Department of Communications

Telecommunications
Regulatory
Service

PERFORMANCE ANALYSIS OF LAND MOBILE PACKET RADIO SYSTEMS

By: J. A. DaSilva, P. Eng.

Government of Canada Gouvernement du Canada

PERFORMANCE ANALYSIS OF LAND MOBILE

PACKET RADIO SYSTEMS

by

J.A. Schwarz daSilva

Consultant in Systems Engineering

Telecommunication Regulatory Service

Department of Communications

February 1981

ABSTRACT

This report is concerned with the performance analysis of packet communications through land mobile radio channels.

The Rayleigh fading character of mobile radio channels is studied through simulation and the statistical distributions of the key parameters of such channels are obtained. Knowledge of such distributions is shown to be instrumental in obtaining expressions for the bit error rate and packet error rate under various fading conditions. In particular, the impact of fading on packet communications is examined by considering the inbound and outbound radio channels of a mobile system operating at 850 MHz. In the former case, the degradation in the throughput of multiple-access schemes is examined. Numerical results indicate that in order to minimize the decrease in throughput, the coverage range of such mobile systems should be limited to about 2 miles which corresponds to the radius of cell sizes for typical cellular systems. For the later case, the solution to the problem of the optimal selection of the packet size is obtained by minimizing the wasted time due to packet overhead and packet retransmissions. Numerous curves are provided showing the influence of the average message length, the bit rate and the signal-to-noise ratio on the optimal packet size.

The problem of multiple-access land mobile radio systems carrying digitized speech and packetized data is then considered. In one of the models examined, 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. In particular, it is shown that spectrum savings can be achieved, not only by interpolating speech packets from a large user population, but also by taking advantage of the fact that the outbound channels need not be accessed on a contention basis. 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 in the form of curves indicate that this delay could be kept within acceptable bounds.

The introduction and implementation of packet speech as well as voice and data integration is now well under way in the context of packet switched computer communication networks. It is hoped that this report will contribute towards the extension of such communications technology to land mobile systems.

TABLE OF CONTENTS

PAGE

LIST OF ILLUSTRATIONS

LIST OF ILLUSTRATIONS (CONTINUED)

LIST OF ILLUSTRATIONS (CONTINUED)

LIST OF TABLES

CHAPTER 1

INTRODUCTION

1.1 Background

Since the early days of wireless communications and certainly as far back as 1906 when the first international radio conference was held, the radio spectrum has been considered a natural resource with some unique characteristics. Indeed, the use of the radio spectrum creates several interference problems which might be regarded as a form of pollution. This particular type of interference or "pollution" is related to the increasing use of the spectrum resource as well as to the lack of an adaptive set of spectrum management techniques. As new and innovative radio spectrum applications are developed within a sometimes obsolete framework of standards and regulations, the radio spectrum becomes progressively "congested".

The congestion problem has economic, regulatory and technical implications. From an economic point of view it is clear that the limits on the use of the radio spectrum are determined by the willingness of the industry to commit the necessary resources for research and development; thereby permitting a more efficient and intensive use of the spectrum. From a regulatory point of view, congestion is determined by the regulator's willingness to develop and enforce an adequate set of standards and equipment specifications that maximize the spectrum efficiency. From a technical point of view, it is clear that the latent communications capacity of the radio spectrum far exceeds any projected demand since innovative communications engineering techniques held the promise to alleviate the congestion in some frequency bands.

The present work is directly relevant to a particular segment of the radio spectrum, namely, the land mobile communication band for which new spectrum efficient concepts are presented.

1.2 Current trends

The concept of a mobile telephone service which would allow a mobile user to establish a connection with a public telephone network has been put to use for the past 40 years with the first experimental land mobile system installed in New York City in 1940. By 1950, the mobile radio industry was providing a regular service to over 24,000 mobile units in the major metropolitan areas. During the next decade, the demand for mobile telephone service as well as dispatch service continued to grow to such an extent that allocated spectrum space became heavily congested. Several methods were proposed for solving the congestion problem. Examples include closer channel spacings, single side band modulation and frequency sharing on a geographic basis. An interim solution which helped to improve the mobile service was the implementation in 1965 of the so-called IMTS (Improved Mobile Telephone Service) offering two-way automatic service and full duplex operation. The limited benefit offered by IMTS was essentially due to the trunking of small groups of channels. At that time the major problem confronting the mobile telephone service was the lack of sufficient spectrum. In 1968, the Federal Communications Commission (FCC) issued a report which stated that the additional frequency space required for the land mobile radio services could be obtained by sharing of the UHF television channels 13 to 20 in the ten largest metropolitan areas. Meanwhile in Canada the possible reallocation of the 406-960 MHz band was being seriously considered since a number of studies had shown that the assigned spectrum for mobile radio service would be exhausted in the early 1980's in major Canadian cities. A clear example of the frequency shortage was given by Toronto's 150 MHz system which experienced more than 65% blocking in the busy hours. Currently both the F.C.C. in the U.S.A. and the D.O.C. in Canada have allocated the 804-890 MHz band to mobile communication systems. It is anticipated that with the opening of this new band, the yearly growth rate for mobile communications will be maintained at 15%, which implies that the 1985 world-wide mobile radio equipment market will reach about 8% of the overall world telecommunications market, estimated at $60 billion.

The mobile radio market is expected, however, to be affected by the rapidly advancing Large Scale Integration (LSI) technology. For example, new generations of automatic mobile telephone units currently in planning will be based on a complete solid state design and will make a substantial use of digital integrated circuit technology. Similarly, custom logic circuits will play a role in digital frequency synthesizers to allow for automatic transmitter identification and thus relieve the mobile user from a number of traditionally manual operations.

Today, the proliferation of computer applications is making it necessary to bring computer resources close to the individual. It is thus extremely desirable to create more flexible and more economic communication techniques. With the decreasing costs of electronic logic and memory circuitry and with the advent of new devices and terminals that generate information in digital form, a growing number of applications have emerged that require the exchange of such information through mobile radio channels. Classical examples of such applications are given by law enforcement systems where it is desirable to access data banks from a vehicle, and rapid transit systems where buses automatically report their location to a central controller. Despite the advances made in the digital communications field, the land-mobile digital revolution is, however, still in its infancy, mainly because of the absence of an appropriate regulatory framework. It is, anticipated, however that in the near future, a phenomenum equivalent in magnitude to the personal minicomputer boom will take place in the mobile digital field. As an indication of the expected demand, it was estimated that some 200 minicomputers were dedicated to personal use in 1974 while recent forecasts indicate that by 1990 over 2 million of such personal minicomputers will be used throughout North-America.

1.3 Solutions to the Spectrum Congestion

In the previous sections we have referred to the growth trends in conventional land-mobile systems and have argued that the convergence of data processing and mobile communications will bring about a saturation of the land-mobile frequency bands. Since the allocation of new frequency bands to these types of communications will only provide an interim relief, there has been a growing concern with respect to the system concepts that can best accommodate this spectrum demand. The following are some of the system concepts that are being actively researched:

- Cellular Architectures
- Spread Spectrum
- Packet Switched Radio for Data Transmission
- Packet Switched Radio for Speech
- Integration of data and voice

1.3.1 Cellular Architectures

The cellular based land-mobile system is certainly one of the most favoured and commonly discussed concepts for increasing the radio spectrum usage efficiency.

As a result of this forecasted congestion and also because of its desire to expand opportunities in mobile communication, Canada announced new spectrum allocations for mobile communications in the 406-960 Mhz band {1}. In particular, the 806-890 MHz band was reallocated from UHF television to mobile communications, without initially specifying the sub-allocations. It is anticipated that the first Canadian cellular systems for analog communications will undergo field tests commencing in 1983.

In 1974, the Federal Communications Commission (FCC) of the U.S.A. government decided to reallocate on a nationwide, primary basis, TV channels 70 to 83 (806 to 890 MHz) to the land-mobile service, with 825 to 845 MHz being earmarked for mobile terminals, and 870 to 890 MHz earmarked for base stations, associated with cellular networks.

In March 1977, the FCC authorized Illinois Bell Telephone to construct and operate a developmental cellular system in the Chicago area and field tests and evaluation of performance are currently underway. During the trial period the Advanced Mobile Phone Service (AMPS) will have a capacity of 2500 mobile customers but after approval for regularized service the system expansion would allow customer growth to more than 100,000.

Another cellular mobile telephone system, called Dynatac is scheduled for installation in the Washington-Baltimore area. This system designed by Motorola will accommodate both portable and mobile terminals { 2 }.

As of January 1980 the Federal Communications Commission released its new cellular mobile radiotelephone "notice of inquiry and notice of proposed rulemaking" expressing the agency's convictions that:

--*"There seems to be little doubt that cellular communications offers the best means for meeting the demands of the mobile communications market through the end of this century," and "...The* *time has arrived for the Commission to establish rules and policies for the commercial operation of cellular mobile radio systems."*

In Japan, the first published studies of high capacity land-mobile radio telephone service had appeared as early as 1967. In 1976, the Japan Radio Technical Council approved use of the 850 MHz band for cellular systems and allocated 50 MHz of spectrum for this use. A cellular network providing commercial mobile telephone service is now in operation in Tokyo and plans have been made to progressively expand it until contrywide coverage is achieved.

As has been discussed, several cellular experimental systems are currently at different stages of design and development. The ultimate objective of these undertakings is to develop a system that can grow and achieve the performance comparable to that of the large scale telephone network. The common design features of these experimental networks are that they will service a variety of mobile users broadly classified as:

(1) dispatch users, and
(2) users requiring telephone service to hand-held and portable telephones.

Structure-wise, these systems are quite similar since they all employ cellular and trunking concepts. There are, however, some dissimilarities and Table 1.1 identifies some of these.

In their earlier stages of development, it can be expected that systems will be designed with "large" cell radius* and as traffic grows, cells will be split and reduced in size, thereby increasing traffic handling capability and spectrum efficiency.

One general rule of thumb (illustrated by the hypothetical example of Table 1.2) that has emerged from the experience to date, is that the traffic handling capacity of a cellular system is approximately proportional to the inverse of the square of the cell radius. There are, however, physical and economical limits to the cell size. A brief survey of the cellular structure concepts is given in Appendix A.

1.3.2 Spread Spectrum

Although spread spectrum techniques can be traced back to Shannon {3} some thirty years ago, the technological difficulties encountered in implementing such systems have led to their slow pace of development. According to Dixon {4} a spread spectrum system is one which meets the following two criteria:

- the transmitted bandwidth is much greater than the bandwidth of the information being sent, and
- some technique (spreading technique) independent from the information being transmitted, but completely known to the receiver, is employed to determine the resulting RF bandwidth.

*A complete description of the Chicago and Tokyo cellular systems can be found in a special issue of the Bell System Technical Journal (Vol. 58, No. 1, January 1979) and the Review of the Electrical Communication Laboratories of Japan (Vol. 25, Nos. 11-12, Nov.-Dec. 1977)

CHARACTERISTICS	UNITED STATES	JAPAN
FREQUENCY BAND (MHz)	825-845 AND 870-890	860-885 AND 915-940
RF CHANNEL SPACING	30 kHz	25 kHz
NUMBER OF CHANNELS	667	1000
CELL RADIUS	2 - 15 km	5 - 10 km
LOCATION OF MOBILE	RANGING	SIGNAL-TO-NOISE-RATIO
CONTROL CHANNELS	10 kbps	300 bps
ERROR CONTROL	FORWARD ERROR CORRECTION AND SPACE DIVERSITY	FREQUENCY DIVERSITY

Table 1.1 SOME CHARACTERISTICS OF CELLULAR SYSTEMS

CHARACTERISTICS	7 CELLS	19 CELLS
SERVICE AREA (km^2)	450	450
CELL RADIUS (km)	≈ 5	≈ 3
CHANNEL WIDTH (kHz)	2x30	2x30
CHANNEL SET SIZE (CHANNELS)	100	100
NUMBER OF CHANNEL SETS	3	3
TOTAL SPECTRUM (MHz)	20	20
BLOCKING PROBABILITY	0.02	0.02
ERLANGS / MOBILE	0.03	0.03
MOBILES / MHz	1143	3103
MOBILES / CHANNEL	68	186
ERLANGS / MHz	34	93
ERLANGS / CHANNEL	2	5.58
MOBILES / MHz / km^2	2.54	6.9

Table I. 2 APPROXIMATE SPECTRUM EFFICIENCIES OF CELLULAR SYSTEMS

Spread spread systems can be classified, according to the type of spreading technique that is used. Essentially, two types of spread spectrum systems exist:

- Direct Sequence or Pseudo Noise Systems
- Frequency Hopping Systems

1.3.2.1 Direct Sequence or Pseudo Noise

In a direct sequence system the spreading is accomplished by modulating (multiplying) the message signal A(t) by a binary pseudo-random sequence B(t) whose symbol rate, also called the chip rate, is much higher than the information bandwidth. In most situations the bandwidth of B(t) is 100 to 1000 times that of A(t) which effectively results in producing the wide band spectrum of B(t). Prior to transmission the resulting signal is usually PSK or MSK modulated which results in a double sided spectrum where the main power is contained within a bandwidth equal to twice the clock frequency of the pseudo-random sequence generator. Denoting by f_B the bandwidth of the base-band signal and by f_c the frequency of the code generator, the spreading factor, G, also called the process gain, is given by:

$$G = \frac{f_c}{f_B} \qquad \text{..... (1.1)}$$

As shown by the block diagram of Figure 1.1, the transmitted signal corrupted with noise, is multiplied by a synchronized code signal and then filtered to match the information bandwidth.

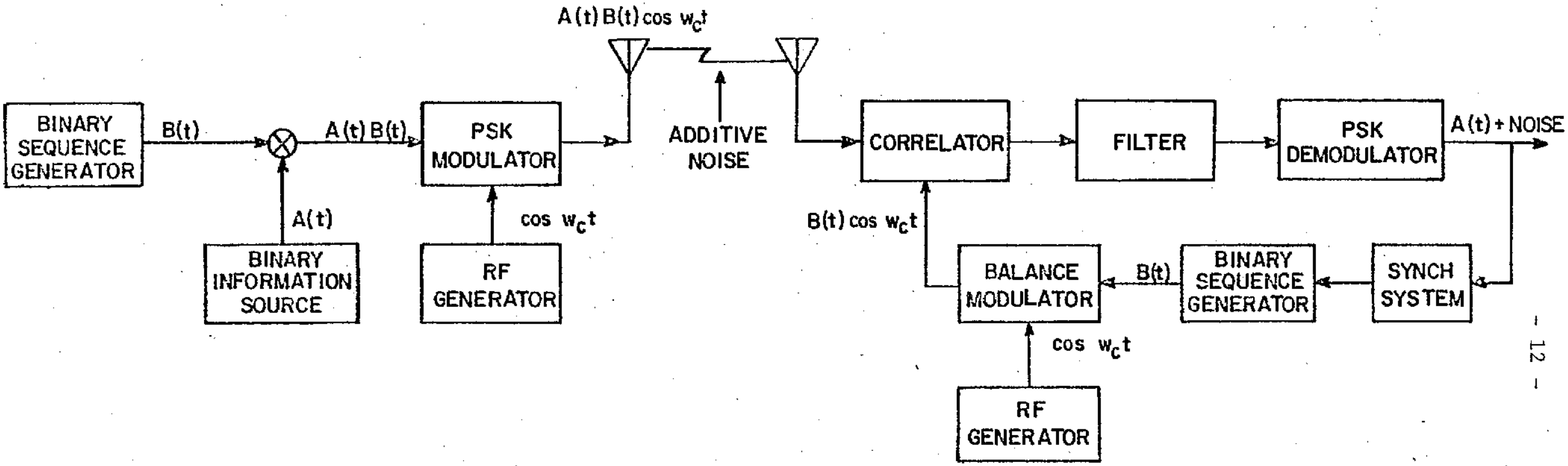


Figure 1.1 DIRECT SEQUENCE TRANSMITTER AND RECEIVER

1.3.2.2 Frequency Hopping

While direct sequence systems operate in the frequency domain since energy is present at all times across the spreading bandwidth, frequency hopping systems operate in the time domain because at any instant in time all the energy is radiated around a single carrier frequency. In the basic frequency hopping system the carrier frequency is pseudo-randomly changed in discrete increments, at hop rates typically varying anywhere from 10 to 1000 hops per sec. This process is equivalent to braking the overall transmission bandwidth in a number of uniform frequency slots, each with a bandwidth comparable to that of the information, with the carrier frequency allowed to take on any of the available center frequency values. In the receiver (see Figure 1.2) the input signal is mixed with a local oscillator signal controlled by a synchronously operated code generator.

1.3.2.3 Spread Spectrum Limitations

One of the major limitations of direct sequence systems arises from the so-called near-far problem. Simply put, this problem arises when the received level of the wanted signal is below the level of the received unwanted signal by an amount equal to or greater than the processing gain. This problem does not occur in frequency hopping systems since different code sequences imply different hopping patterns, with the result that at most times the undesired or jamming signal will be on a different frequency from the desired one. Another problem which is peculiar to the frequency hopping systems is related to the need to develop extremely fast frequency synthesizers.

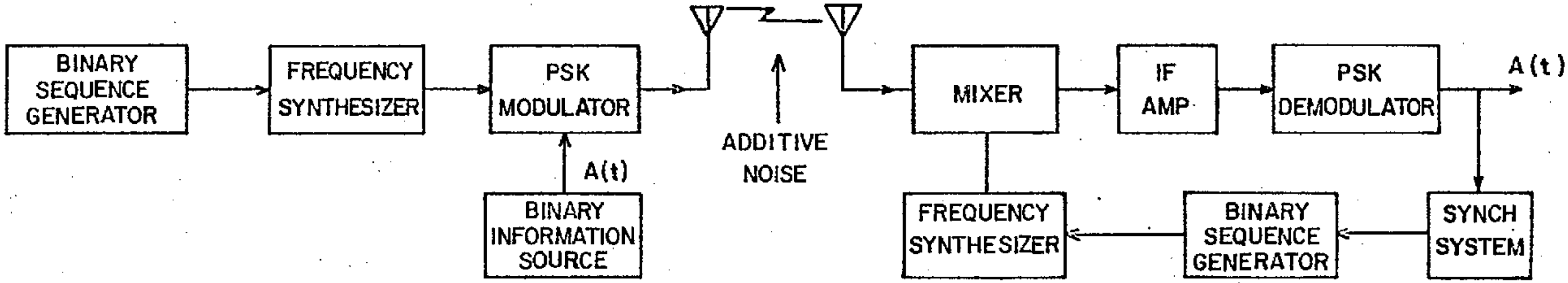


Figure I.2 FREQUENCY HOPPING TRANSMITTER AND RECEIVER

Until recently the most serious problem of any spread spectrum system appeared to be related to the necessity to provide time synchronism to spread spectrum users, since time accuracies of the order of 10-100 nano-seconds are required. However with the increasing trend toward the use of digital processing circuitry, large-scale-integration techniques, charge-transfer devices, surface-acoustic-wave devices and microprocessors, most limitations are being removed, making practical but still expensive many sophisticated designs for spread spectrum.

The primary advantages to be gained by using spread spectrum techniques in land-mobile radio are related to selective addressing, interference rejection and code division multiplexing. The fact that spread spectrum emission have a low power density and are immune to jamming has led Utlaut {5 } and Ormondroyd {6 } among others, to suggest that spread spectrum land-mobile systems could be overlayed on the frequency bands used by television. So far it is not clear if spread spectrum systems are more or less spectrum efficient than cellular based or conventional mobile systems. Cooper {7} claims that spread spectrum will always be superior to conventional techniques so far as the efficiency of bandwidth and energy utilization are concerned. On the other hand, it was shown by Berry and Haakinson {8}, using both analysis and simulation, that a spread spectrum service in interference limited multi-user systems would not be as spectrum efficient as conventional FM land-mobile radio.

1.3.3 Packet Switched Radio for Data Transmission

The single most important factor that determines how well a radio channel is shared among its users is the scheme that is used to gain access to it. The characteristics of various random access schemes that have been proposed for packet radio communication systems are especially attractive because all these schemes suggest an efficient sharing of the spectrum resource.

The first packet radio communication system to be described in the literature was the ALOHA System, implemented at the University of Hawaii {9}. This system provided communication between a central computer and its geographically scattered but fixed terminals. The "ALOHA" concept was extended to include investigations involving mobile terminals {10},{11},{12}. It was also quickly learned that packet transmissions over satellite channels offered many attractive options for the efficient collection and distribution of data over large areas {13}. As will be shown later*, another appealing feature is the simplicity and elegance of packet radio concepts, and their wide range of applicability in such diverse areas as wire line networks {14},{15},{16},{17} and cable television systems {18}.

The fundamental concepts behind resource sharing (allocation) and why it is of such importance in the field of computer communications are summarized by Kleinrock {19}, as follows:

"... Resource allocation is at the root of most of the technical problems we face today in and beyond the information

* See Appendix B

industry. These problems occur in any multi-access system in which the arrival of demands as well as the size of the demands made upon the resources are unpredictable. The resource allocation problem in fact becomes that of resource sharing and one must find a means to effect this sharing among the users in a fashion which produces an acceptable level of performance..."

1.3.3.1 Experimental Packet Radio Networks

The ALOHA System {9} is the first example of an experimental packet radio network. Built to illustrate the feasibility of the packet radio concept, it was a one hop system where all terminals were in line-of-sight and within range of the central station. Packet repeaters were later added to provide coverage beyond the range of the central station.

Another experimental packet radio network {11}, designed to serve as a research facility, was subsequently built in the San Francisco Bay area with the support of the Advanced Research Projects Agency (ARPA) of the United States Department of Defence. Its planned experimental program and features are described by Kahn {12} and include a number of investigations involving spread spectrum modulation and anti-jam protection. This network is still under development and experiments are being carried out {11}.

In parallel with this ground activity the ARPA has been funding since the early 1970s the development of packet satellite technology in order to evaluate its potential for long haul computer communications amongst geographically distributed users. The most prominent example of this technology is the Atlantic Packet Satellite Network, known as SATNET, which has been in operation since 1975. Current plans call for a continued use of SATNET as the primary link between ARPA and a number of research organization in Europe. In addition to its international links, SATNET will form the basis for a U.S. packet satellite network involving a number of earth stations in Massachusetts, California and the Washington area. The emphasis of the planned research is centered around using the packet satellite technology to explore the integration of packet voice, graphics, facsimile and computer to computer traffic. An excellent description of the experimental activities being carried out over SATNET with particular relevance given to the packet speech performance can be found in a recent paper by Chu {21}. A number of other papers {22},{23},{24} address some technical aspects of the SATNET experiment.

In Canada, packet radio techniques have been experimented with since 1978, when the Federal Department of Communications regulated its use by radio amateurs. With the forthcoming launching of an amateur satellite it is expected that a nation-wide packet radio network will be implemented.

In the U.S., GTE Telenet recently announced plans to use local intra-city packet radio communications on a commercial basis.

1.3.4 Packet Switched Radio for Speech

When digitized voice techniques were first introduced in the 1930's, the subject of bandwidth requirements was not directly addressed. The primary motivation for the original work appears to have been the furtherance of fundamental speech research. During World War II, secure voice communication was an important military need, and that further enhanced the relevance of digitized speech communications.

As communications technology developed in the 1950's and 1960's, it became clear that 3 KHz telephone lines could be used to transmit bit streams at speeds as high as 9600 bps by using sophisticated modulation methods. Good quality digitized speech could be securily transmitted using the telephone facilities. However, the high cost of digital signal processing has, until recently, limited the applications of digital speech to military purposes.

Two major factors have revived the research in digital speech and made the digital voice terminal a practical reality, namely; the advances in digital signal processing and the development of sophisticated speech algorithms. While digital signal processing have drastically reduced both the cost and the size of the digital circuits, sophisticated speech algorithms have been developed which led to a continuous improvement of the way in which the speech parameters are described and transmitted with relatively few digits. There are essentially two ways* by which speech can be encoded:

- Waveform reconstruction
- Speech analysis-synthesis

These two techniques are quite distinct in their approach for describing the speech signals, as well as in their bit rate performance and complexity. It is generally agreed to that a bit rate greater than or equal to 16 kbps is characteristic of waveform reconstruction techniques, while speech analysis and synthesis use lower bit rates.

Since the 1940's, the advantages of digital, as opposed to analog transmission of the speech signal have been well documented. Some of

---

* See Appendix C for a brief review of speech encoding techniques.

its advantages can be summarized as follows:

- Digital transmission offers a greater reliability for long distance connections over noisy channels, since digital signals require a lower signal-to-noise ratio to achieve a quality equivalent to that of analog voice transmission.
- Switching and concentration of information are more efficiently realized when the information is in digital form.
- Digital transmission permits channel multiplexing.
- Digital transmission permits speech encryption.
- Due to the recent advances in vocoder design, digitized voice becomes more efficient in terms of bandwidth occupancy*.

In addition to the benefits achievable by transmitting speech in digital form, it is also possible to exploit the ON-OFF characteristics of speech. Indeed as shown by Brady {25}, during a telephone conversation the transmission channel remains idle for about 60 to 75% of the time. It is therefore possible to interpolate a large number of users on a lesser number of voice circuits. The first known speech interpolation system, the analog TASI, proposed by Bullington {26} in 1959, was able to concentrate 72 speech inputs on 36 channels, thereby achieving

---

* Codex among other companies is currently marketing a system that permits up to four simultaneous conversations to be multiplexed onto a single 9600 bps circuit.

an interpolation gain of 2. Since 1959 a number of digital speech interpolation systems, based on flexible methods of managing overload conditions, have been suggested.

Campanella {27} studied a digital version of TASI as well as a scheme known as SPEC (Speech Predictive Encoded Communications) and showed that both methods achieve interpolation gains greater than two. In 1978, Woitowitz {28} reported on the effects of speech statistics on a scheme, similar to SPEC, known as IPM (Instantaneous Priority Multiplexing). This author suggests that IPM is particularly applicable when the number of traffic sources is small. A number of other speech interpolation schemes have been reported {29},{30} and compared by Seitzer {31}. Recently, Soumagne {32} proposed a speech interpolation scheme that can achieve a compression ratio of 4 allowing for instance 128 users to share 32 transmission channels. With the exception of IPM,the quality of digital speech interpolation systems depends to a large extent on being able to detect speech in the presence of noise. One of the simplest methods of speech detection, used by Brady {25} to study the ON-OFF Speech Characteristics, is based upon the fact that speech bursts occur at a higher level than noise. When several consecutive speech samples exceed a given threshold level, the presence of a talkspurt is detected. One of the earlier versions of a voice-activated switch used in the SPEC design {27} used a threshold level which was made to vary according to the noise level of the circuit.

Other properties of speech and noise have led Drago {33} to design and evaluate two dynamic speech detectors whose quality was evaluated by a number of subjective tests. More recently, and in particular since 1975 {34} with the advent of packet speech concepts, new digital speech detection mechanisms have been proposed. The latest scheme, due to Dhadesugoor {35}, is based upon encoding the voice using CVSD (Continuous Variable Slope Delta Modulation) at 16 kbps and detecting a given pattern of bits corresponding to a silence period. Currently, the development of reliable speech detectors and the related speech processing devices for bit rates lower than 16 kbps, are still an area of very active research. However, the latest advances in component technology will allow a greatly expanded sophistication in circuit implementation thereby ensuring a widespread use of digital voice and speech interpolation in the context of land line communication systems.

As mentioned above, Forgie {34} in 1975 appears to have been the first author to evaluate the prospects of transmitting voice in packet form through packet switched networks. In essence packet speech attempts to achieve, over store-and-forward networks, the transmission economies that can be realized with speech interpolation over dedicated links. The main difference between packet speech and speech interpolation lies in the fact that packets transmitted over store-and-forward network will encounter variable delays from origin to destination. Particularly in situations of overload and even under normal operating conditions it is now well established that store-and-forward networks can occasionally introduce excessive delays, which have disruptive effects on a conversation. In addition to excessive delays which might render the conversation psychologically unacceptable, packet speech can potentially be degraded and rendered unintelligible if the packet loss rate exceeds some threshold level which depends on the selected packet size, the speech encoding technique and its corresponding digitization rate. If an appropriate environment for packet speech can be achieved, by careful consideration of the main network issues such as flow control procedures and routing strategies, the implications of packet voice will go far beyond the speech interpolation advantage. The following are some of its main implications:

* Speech transmission across a number of connected networks can be mode compatible with end-to-end network security procedures
* Speech transmission can be achieved at different rates and at different quality levels in accordance with the state of congestion of the network
* Speech messages can be stored and retrieved later
* Packet speech can provide a technology base for integrated data/voice packet switched networks.

It is believed that the successful implementation of packet speech over experimental store-and-forward networks will significantly alter the architecture of future networks and allow for a variety of new services to be offered. Such networks and services will be available to a larger community of end users, including those that have a requirement for mobile or portable communications equipment. Packet speech and speech interpolation over mobile communication channels can significantly contribute towards the solving of the spectrum congestion problem.

1.3.5 Integration of Data and Voice

The motivation for exploring new switching techniques that achieve greater efficiency of transmission resources started in 1970 with the advent of computer-communication networks. In recent years, the potential for increased network flexibility through the integration of voice and data, has been illustrated by a number of authors {36},{37}, {38}. The framework for the search of efficient integration techniques is based upon the following question:

> *Given a limited resource (switch or transmission channel) is it advantageous to: 1) allocate on a fixed basis a fraction of the resource to voice traffic and reserve the remainder of the resource to data traffic, 2) allocate part of the resource on a fixed basis and the remainder on a dynamic basis, 3) allocate the resource on a dynamic basis.*

No general answer to the above question can be given without prior knowledge of the particular environment where voice and data are to be integrated. The three alternative solutions give rise to three switching technologies and variations thereof, with circuit switching corresponding to 1), hybrid switching corresponding to 2) and packet switching corresponding to 3). Since both circuit and packet switching are well known techniques we will limit ourselves to a brief description of hybrid switching.

Essentially there are two variations of hybrid switching; namely, the fixed boundary methodology and the moveable boundary methodology. Both methodologies are based on imposing a frame structure to a high speed digital bit stream. Frames of fixed duration are partitioned either on a fixed basis or on a moveable basis. Port {39} studied the case of a fixed boundary and Coviello {40} proposed a moveable boundary scheme. The main advantage of the moveable boundary over the fixed boundary methodology is due to the fact that packet switched data traffic can utilize the temporarily idle channel capacity assigned to circuit switched voice traffic. Fischer {41} derived exact analytical expressions for both methodologies and presented some typical cases that clearly show the superiority of the moveable boundary solution. The analysis provided by Fischer fails, however, to indicate the overall performance of such an integrated scheme over a network consisting of several switching nodes in tandem. In this case, as indicated by Watanabe {42}, only an approximate analysis and simulation can give some insight to the efficiency of integrated systems. An enhanced hybrid system, using what is known as the Packetized Virtual Circuit concept was subsequently proposed by Forgie {43}. The enhancement is obtained by a TASI like approach that takes advantage of the voice silence gaps to increase the overall system capacity.

More recently Ross {44} advocated what is known as a Flexible Hybrid scheme where frames of 10 to 20 msec are partitioned into two regions, separated by a moveable boundary, with one region dedicated to circuit-switched synchronous traffic and the other dedicated to packet-switched or message-switched traffic. Both voice and interactive data are transmitted as packet-switched data while bulk and burst data information are transmitted as circuit-switched data. By transmitting voice as packet-switched data the TASI interpolation gain is achieved since during speech silence gaps the transmission process is stopped. In addition, this approach allows for high priority voice or data to use the circuit-switched region thereby ensuring low delays.

An interesting approach to the integration of voice and data, which is proposed for satellite channels but can also be applied to mobile channels, is due to Derosa {45}. Voice traffic originating at the various earth stations (mobile terminals) is carried by an uplink TDMA channel and a downlink TDM channel. Data traffic accesses the satellite (base station) through a number of slotted ALOHA FDMA uplinks and reaches the earth stations through the same downlink channel used for voice. Derosa suggests that higher priority could be attributed to the voice traffic which implies that data packets upon reaching the satellite will be discarded if temporarily no idle downlink slot is available. Interestingly enough no attempt is made to mark as idle those downlink slots corresponding to speech silence gaps.

In the context of mobile channels the most recent attempt to the integration of voice and data is due to Marsan {46}. His approach is based on the assumption that for certain applications, the length of voice messages is fairly short (of the order of 6.5 sec), which clearly precludes its application to a conventional land mobile system with voice holding times of the order of 3 minutes. Despite this shortcoming, it is however clear that similar approaches could lead to substantial spectrum efficiencies for land mobile systems.

1.4 Objectives of this report

In the previous sections we have summarized the efforts that have been undertaken so far to arrive at system concepts to resolve the spectrum congestion problem.

Cellular architectures and spread spectrum modulation techniques are fairly well understood at present and are still the subject of extensive research and development.

Packet radio, despite being an already well established technique, has not been considered in a practical way as a technique that can be applied to mobile communication systems. As an indication of this fact, most recent studies assume, that radio channels are noiseless even though it is well known that broadcast channels of the mobile radio type are error prone.

Packet speech and voice/data integration over land mobile channels, as indicated by the preliminary attempts of Marsan {46}, Pan {47} and Ellershaw {48}, is a promising approach which offers practical solutions to the spectrum congestion problem.

Based on the above discussion we can formulate the two main objectives of this study as follows:

* *Development of a better understanding of the key parameters of the land mobile channel and study of their impact on the throughput and efficiency of packet radio random access schemes.*

* *Investigation, through analysis and simulation, of the applicability of packet speech concepts to land mobile channels and study of the performance of integrated voice/data mobile systems.*

The first objective of this report is dealt with in Chapters 2 and 3, where in addition to describing the simulation model of the land mobile channel we obtain, as a function of the signal to threshold ratio, the statistical distributions of the fade durations, level crossings and interfade durations. Knowledge of these distributions will lead us to develop approximate expressions for the bit error rate and packet error under fading conditions. We will also indicate how the knowledge of the packet error rate can be used to derive an expression for the optimal packet size which is shown to be useful in particular for base to mobile communications. In the opposite direction, i.e. for mobile to base communications, we consider the effects of fading on the throughput of packet radio random access schemes. Numerical results are provided which indicate that for environments where the packet radio terminals are mobile, the capacity degradation of packet radio random access schemes can be quite substantial.

The second objective of this report is addressed in Chapter 4. After a review of the relevant performance measures of a packet speech land mobile system we discuss a variation of the CSMA protocol as applied to a mobile system where talkspurts are transmitted as multi-packet messages. We show in particular the potential bandwidth savings that can be achieved by a packet speech system, especially in the downlink channels.

In the same chapter we consider one approach for integrating voice users and data users and present analytical expressions for the delay experienced by data packets, from arrival to successful departure. To corroborate the analytical results we have built a GPDS simulation model that shows that, depending on the data users traffic characteristics, large numbers of data users can share a unique voice channel by taking advantage of the idle channel periods.

1.5 Contributions

The main contributions of this report, presented in Chapters 2, 3 and 4, can be summarized as follows:

* Chapter 2

  - *We obtain the statistical distributions of the three key variables of Rayleigh fading mobile channels namely; the fade durations, the intervals between consecutive level crossings and the non-fade durations.*

* Chapter 3

  - *We derive a simple and accurate expression for the packet error rate on mobile fading channels and use it to 1) show how the maximum throughput of the mobile-to-base random access channels can be degraded, and to 2) evaluate the optimal packet size on the base-to-mobile channels.*

* Chapter 4

  - *We describe and evaluate the performance of two new architectures for land-mobile systems namely; a packet speech based land mobile system and an integrated voice/data land mobile system.*

1.6 Report Organization

The remaining four chapters of this report are organized as follows:

Chapter 2 presents a model of the fading land mobile channel and identifies some of its key parameters, namely the average fade duration and the average level crossing rate. A simulation model is described which obtains the main statistical distributions that concern the data user.

Chapter 3 builds upon the information gathered from the simulation study to obtain simple but approximate expressions for the bit error rate and the packet error rate. We show how these expressions can be used to compute the optimal packet size for the base to mobile link and also to study the throughput degradation that results from fading errors and packet collisions due to the contention nature of random access schemes.

Chapter 4 is concerned with the feasibility of applying packet speech concepts to the land mobile channel, in addition to investigating one possible approach to the integration of voice and data on the same channel.

Chapter 5 contains some concluding remarks and recommendations for further work.

CHAPTER 2

THE LAND MOBILE CHANNEL

2.1 Introduction

As indicated in Chapter 1, the possibility of transmitting digital information over land-mobile fading channels has recently received considerable attention. This is mainly due to the emergence of a growing number of applications that require the exchange of data through such channels. Prominent examples of such trends are given by high capacity mobile systems, computerized dispatch mobile systems and automatic vehicle location systems. All such mobile systems share a common characteristic. Whether it is for digital signalling or for access and control purposes, additional channels are required to convey information in digital form.

The above discussion justifies the importance of assessing the impact of the channel characteristics on the transmitted digital information. After a review of the mechanisms that lead to the designation of land-mobile channel as a Rayleigh fading channel (Section 2.2), we identify two of the most important characteristics of such channels, namely, the level crossing rates and the fade durations. The probability distribution functions of both parameters are then obtained using a computerized fading simulation model. Following a brief description of the basic building blocks of the simulation model (Section 2.3), we examine its validity by comparing simulation results with available theoretical equations. Having established the accuracy of the simulation model, we proceed in Section 2.4 to obtain the distribution functions of the intervals between consecutive level crossings, the fade durations and the non-fade durations, as functions of the relative signal levels.

The results of this chapter will be used in Chapter 3 to derive an expression for the optimal packet size and to study the problem of channel capacity degradation.

2.2 Theoretical Model for the Fading Signal

In estimating both the bit error rate as well as packet error rate of mobile data channels it is essential to consider the main features of the fading signal received by the mobile unit. This section will briefly outline these features.

When measuring the envelope of a transmitted sinusoidal carrier as seen by a mobile unit radio antenna in an urban environment, it is observed that this signal is time varying, although the transmitted envelope is not {49},{50}. It is also observed that when the receiving antenna is standing still, the envelope is almost constant. This strongly implies that the envelope is constant in time, but varies spatially. A model which has been developed previously to capture this spatial variation is described next.

Ossanna {51} was the first to propose a statistical model in terms of a set of vertically polarized horizontally travelling plane interfering waves. Gilbert {52} suggested a more general model which represents the basis of the widely accepted analysis given by Clarke {50} and is the one considered here.

The model can be easily understood by considering Figure 2.1, where N vertically polarized, horizontally travelling plane waves are superimposed. Every wave has an angle (spatial angle) of arrival α_n and a phase shift ϕ_n, which is uniformly distributed throughout 0 to 2π.

Denoting the angular carrier frequency by ω_c, the resulting field point (x_o, y_o) can be written as:

$$E(t) = \sum_{n=1}^{N} E_n(t) \qquad \ldots\ldots (2.1)$$

where

$$E_n(t) = \cos\left[\frac{2\pi}{\lambda}(x_o \cos\alpha_n + y_o \sin\alpha_n) + \phi_n\right] \cdot \cos\omega_c t - \sin\left[\frac{2\pi}{\lambda}(x_o \cos\alpha_n + y_o \sin\alpha_n) + \phi_n\right] \cdot \sin\omega_c t \qquad \ldots\ldots (2.2)$$

and λ is the carrier wavelength.

By giving the point (x_o, y_o) a velocity υ in any direction, the power spectrum of the envelope can be found analytically. Using Eqn. (2.2), we can rewrite Eqn. (2.1) as follows:

$$E(t) = T_c(t) \cos\omega_c t - T_s(t) \sin\omega_c t \qquad \ldots\ldots (2.3)$$

where $T_c(t)$ and $T_s(t)$ are, by the central limit theorem, Gaussian random variables corresponding to the in-phase and quadrature component of E(t).

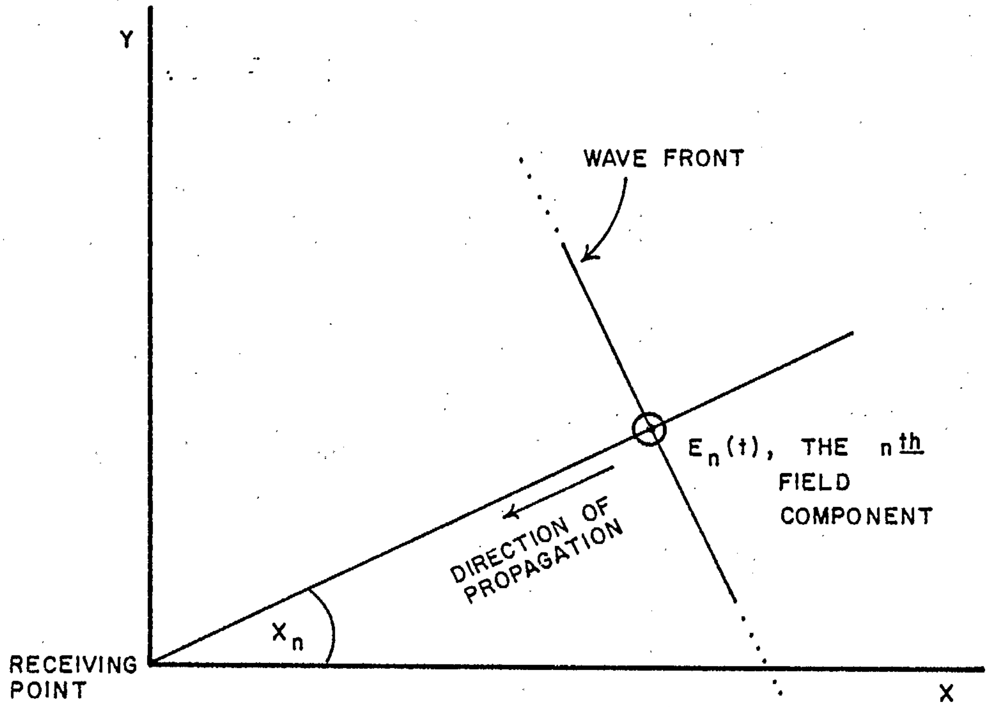


Figure 2.1 A PROPAGATION MODEL FOR URBAN LAND MOBILE RECEPTION

Denoting by V and ϕ respectively the amplitude and the phase of the received signal, we have:

$$V = \left[T_c^{\,2} + T_s^{\,2} \right]^{1/2} \qquad \text{..... (2.4)}$$

and

$$\phi = \tan^{-1} \left[T_s / T_c \right] \qquad \text{..... (2.5)}$$

where $V \geqslant 0$ and $0 \leqslant \phi \leqslant 2\pi$.

For a fixed time, t, the independent random variables T_c and T_s are defined as sums of independent Gaussian random variables with zero mean and variance equal to σ^2. Hence their joint probability density is given by:

$$P(T_c, T_s) = \frac{1}{2\pi\sigma^2} \exp\left(- \frac{T_c^{\,2} + T_s^{\,2}}{2\sigma^2} \right) \qquad \text{..... (2.6)}$$

It can be shown that the joint probability density function of the amplitude and phase random variables is obtained from Eqns. (2.4),(2.5) and (2.6):

$$P(V, \phi) = \frac{V}{2\pi\sigma^2} \exp\left(- \frac{V^2}{2\sigma^2} \right) \qquad \text{..... (2.7)}$$

Integrating (2.7) over ϕ from 0 to 2π gives:

$$P(V) = \begin{cases} \dfrac{V}{\sigma^2} \exp\left(- \dfrac{V^2}{2\sigma^2} \right) & ; \quad 0 \leqslant V < +\infty \\ 0 & ; \quad \text{otherwise} \end{cases} \qquad \text{..... (2.8)}$$

which corresponds to the density function of a Rayleigh distributed variable.

Also, integrating (2.7) over V from 0 to $+\infty$ gives:

$$P(\phi) = \frac{1}{2\pi} \quad ; \quad 0 \leqslant \phi \leqslant 2\pi \qquad \ldots\ldots (2.9)$$

In practice one is more interested in the signal power level, hence Eqn. (2.8) can be rewritten as:

$$P(y) = \begin{cases} \frac{1}{y_o} \exp\left(-\frac{y}{y_o}\right) & ; \quad 0 \leqslant y \leqslant +\infty \\ 0 & ; \quad \text{otherwise} \end{cases} \qquad \ldots\ldots (2.10)$$

where y is the instantaneous power level $\left(= \frac{V^2}{2}\right)$ and y_o is the average power level $(= \sigma^2)$. Note that y is an exponentially distributed variable.

Eqns. (2.8) and (2.9) show that the phase of the received signal is uniformly distributed from 0 to 2π, while the signal amplitude is Rayleigh distributed.

It follows from Eqn. (2.10) that the probability that the signal power, y, exceeds some threshold power level, y_t, is

$$\text{Prob}[y > y_t] = \exp(-y_t/y_o) \qquad \ldots\ldots (2.11)$$

Apart from the statistical distribution of the signal envelope, there are two important parameters of the fading process which can be defined {49}; namely, the average level crossing rate, N, and the average fade duration, τ. N is defined as the rate at which the signal envelope crosses a certain threshold level, y_t, in the positive direction, and τ is defined as the average period of time the signal stays below the threshold level. These two parameters are given by:

$$N = f_d \sqrt{2\pi\rho} \quad \exp(-\rho) \qquad \text{..... (2.12)}$$

and

$$\tau = \frac{\exp(\rho) - 1}{f_d \sqrt{2\pi\rho}} \qquad \text{..... (2.13)}$$

where $\rho = y_t / y_o$

and f_d = maximum Doppler frequency shift $= \frac{\upsilon}{\lambda}$.

Obviously, the product $N\tau$ represents the percentage of time the sign signal stays below the threshold level and it is independent of the Doppler frequency shift, f_d. Combining Eqns. (2.11)-(2.13) we obtain:

$$\text{Prob}\left[y \leqslant y_t \right] = N\tau = 1 - \exp(-\rho) \qquad \text{..... (2.14)}$$

2.3 The Multipath Fading Simulator

In the previous section, we presented expressions for the average fade duration and average level crossing rate. For certain systems this would be the only information required. However, for digital land mobile systems, it is important to obtain the actual distributions of these two parameters, as indicated by Arredondo {53}. To accomplish this task, a computer-generated multipath fading simulator which is similar to the one reported by Smith {54} and is based on Eqn. (2.3) was constructed. Figure 2.2 illustrates the simulator block diagram which consists of two independent Gaussian noise sources, followed by two identical shaping filters and quadrature balanced modulators. The sum of the two quadrature signals is a Rayleigh-distributed RF signal. In the computer simulation, the Gaussian distributed Fourier coefficients of the noise are obtained from a random number generator with each coefficient being weighted by a shaping factor. This spectral shaping factor is selected to match the theoretical spectrum of E(t) when the receiver antenna is a vertical monople. The spectrum of the shaping filter is given {49} as:

$$S(f) = \begin{cases} \dfrac{y_o}{2\pi f_d}\left[1 - \left(\dfrac{f}{f_d}\right)^2\right]^{1/2} & ; \quad f \leq f_d \\ 0 & ; \quad f > f_d \end{cases} \qquad \ldots\ldots (2.15)$$

Briefly, the routine creates a sequence of amplitude elements equally spaced in time to produce a multipath fading signal envelope waveform having Rayleigh Statistics. The Doppler frequency f_d is an input parameter, and the output time sequence is stored in an array X. The amplitude values stored in X are normalized to have a zero decibel mean, i.e., $\bar{X}^2 = 1$. The program generates up to 16000 amplitude values spaced Δt second apart, where Δt is an input parameter and must be adequately determined for each Doppler frequency in order to produce a smooth envelope. The 16000 points are divided into 40 frames each containing 4000 points, and the program is designed to generate any

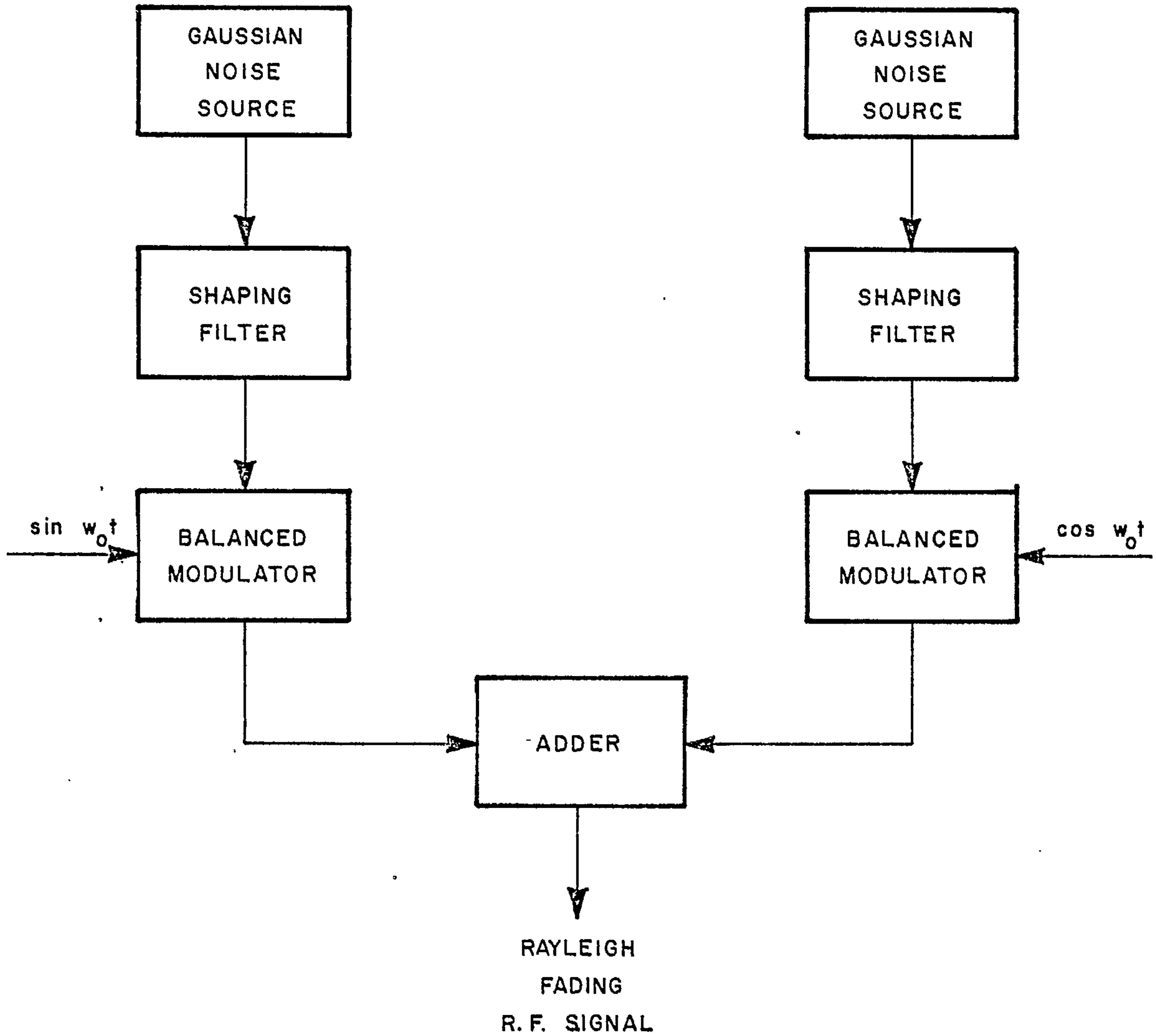


Figure 2.2 FADING SIMULATOR BLOCK DIAGRAM

number of frames up to 40. A sample of the computer-generated envelope is shown in Figure 2.3 for a particular carrier frequency and vehicle speed. As indicated, during a time span of 0.5 seconds, signal fades of more than 30 dB below the average level can occur.

2.4 Fading Statistics

Four additional programs were constructed to compute from the data stored in the array X the different statistics of the fading signal. The first program is used to compute the average number of fades per second (Eqn. 2.12) and the average fade duration (Eqn. 2.13). The program is also used to verify that the generated amplitude envelope is Rayleigh distributed. The statistics of the computer-generated fading signal are compared to the theoretical statistics in Figures 2.4-2.6. Figure 2.4 shows the probability that the signal amplitude level stays below a certain threshold level (the abcissa). The comparison extends over 30 dB level range, and it is apparent that the agreement between the theoretical and simulation curves is excellent. The average level crossing rate and average fade duration as a function of the relative power level are given by Figures 2.5 and 2.6 respectively. The comparisons extend over the same level range with an excellent agreement between the theoretical predictions and the simulation results.

The second program calculates the statistical distribution of the fade durations as a function of the relative threshold to signal level. In Figure 2.7 we display these statistical distributions for three selected values of the relative threshold to signal level; namely, 0,-10 and -15 dB. The horizontal axis represents the fade width normalized

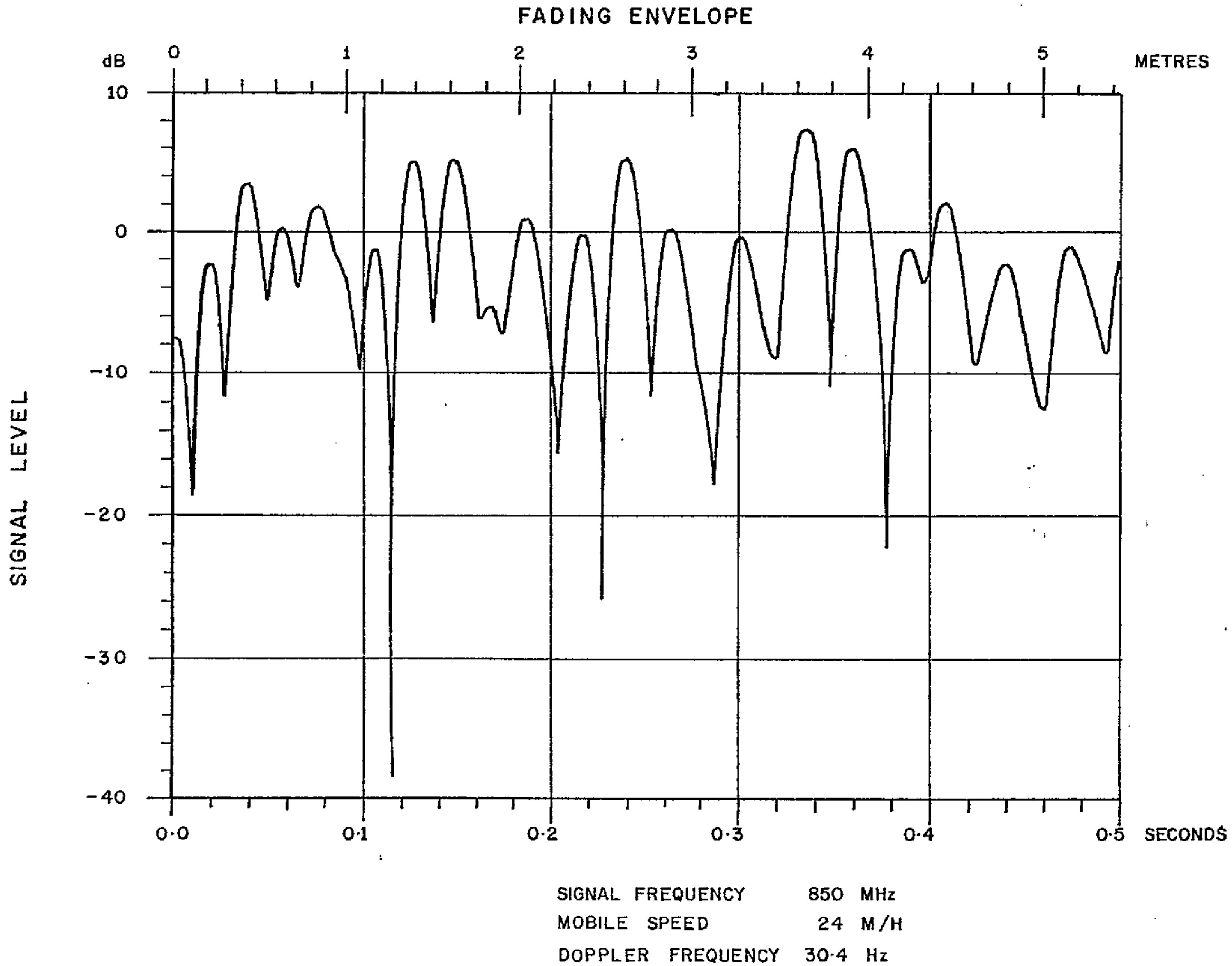


Figure 2.3 SAMPLE OF THE COMPUTER GENERATED ENVELOPES

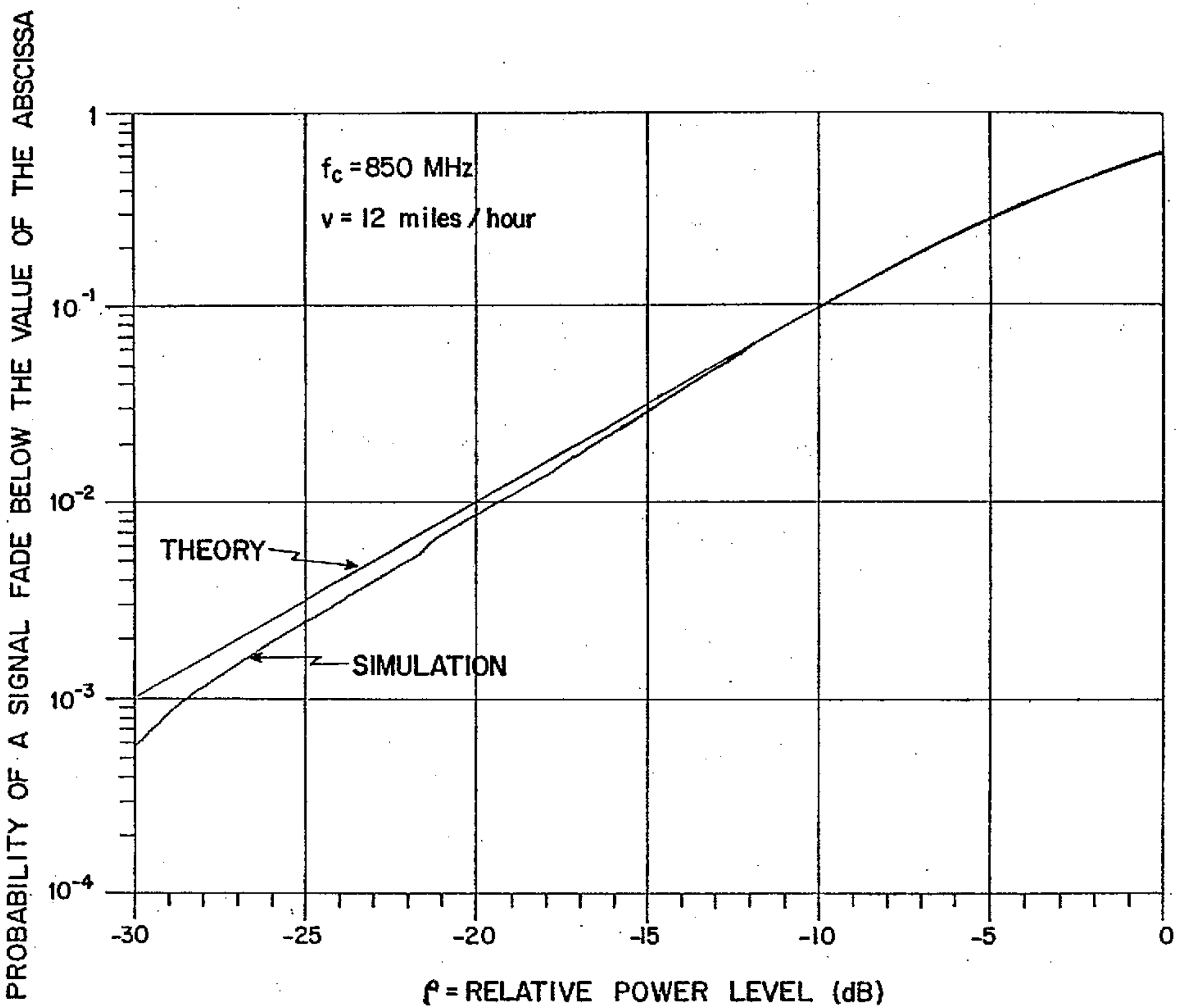


Figure 2.4 THE RAYLEIGH DISTRIBUTION - THEORY AND SIMULATION

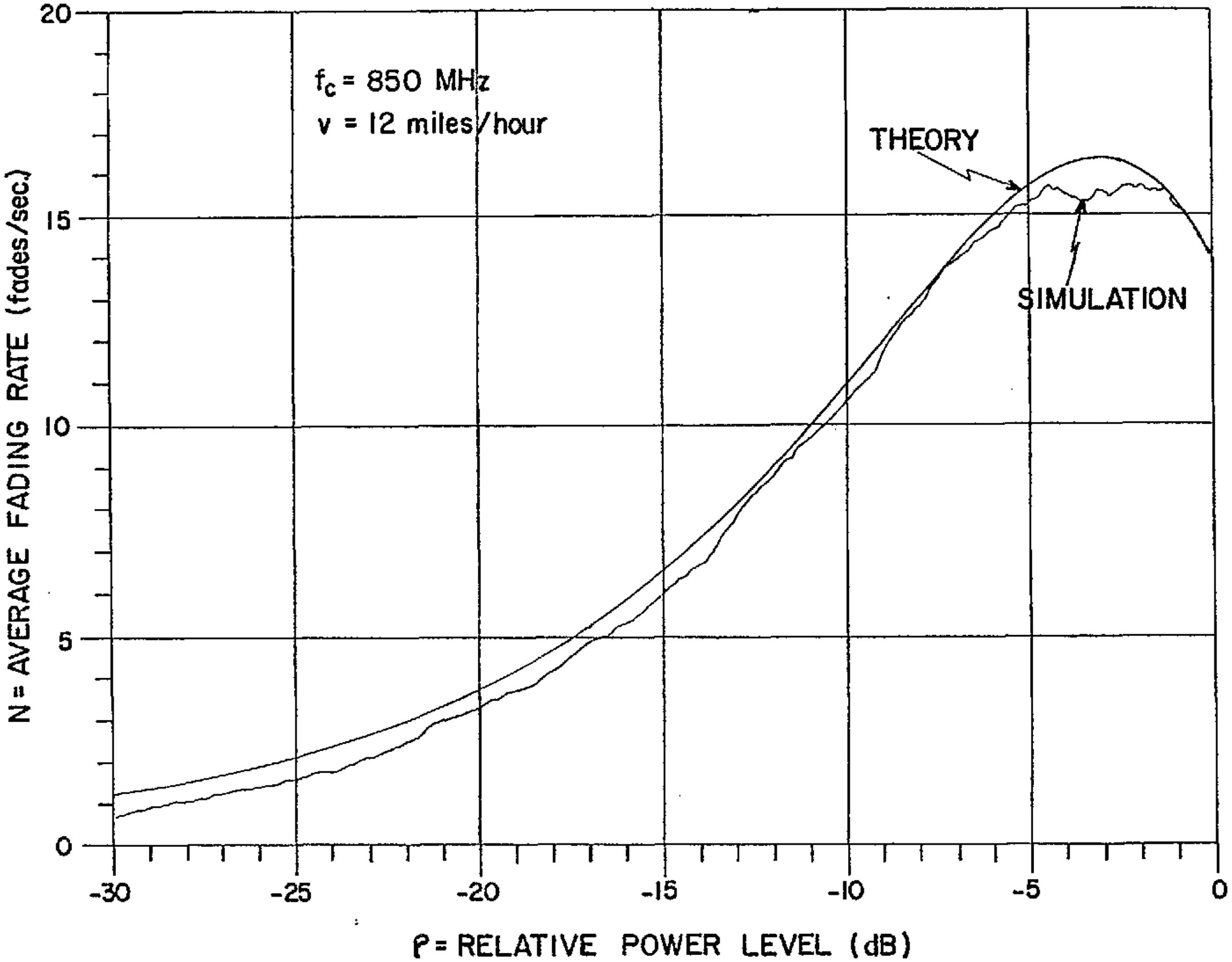


Figure 2.5 AVERAGE FADING RATE - THEORY AND SIMULATION

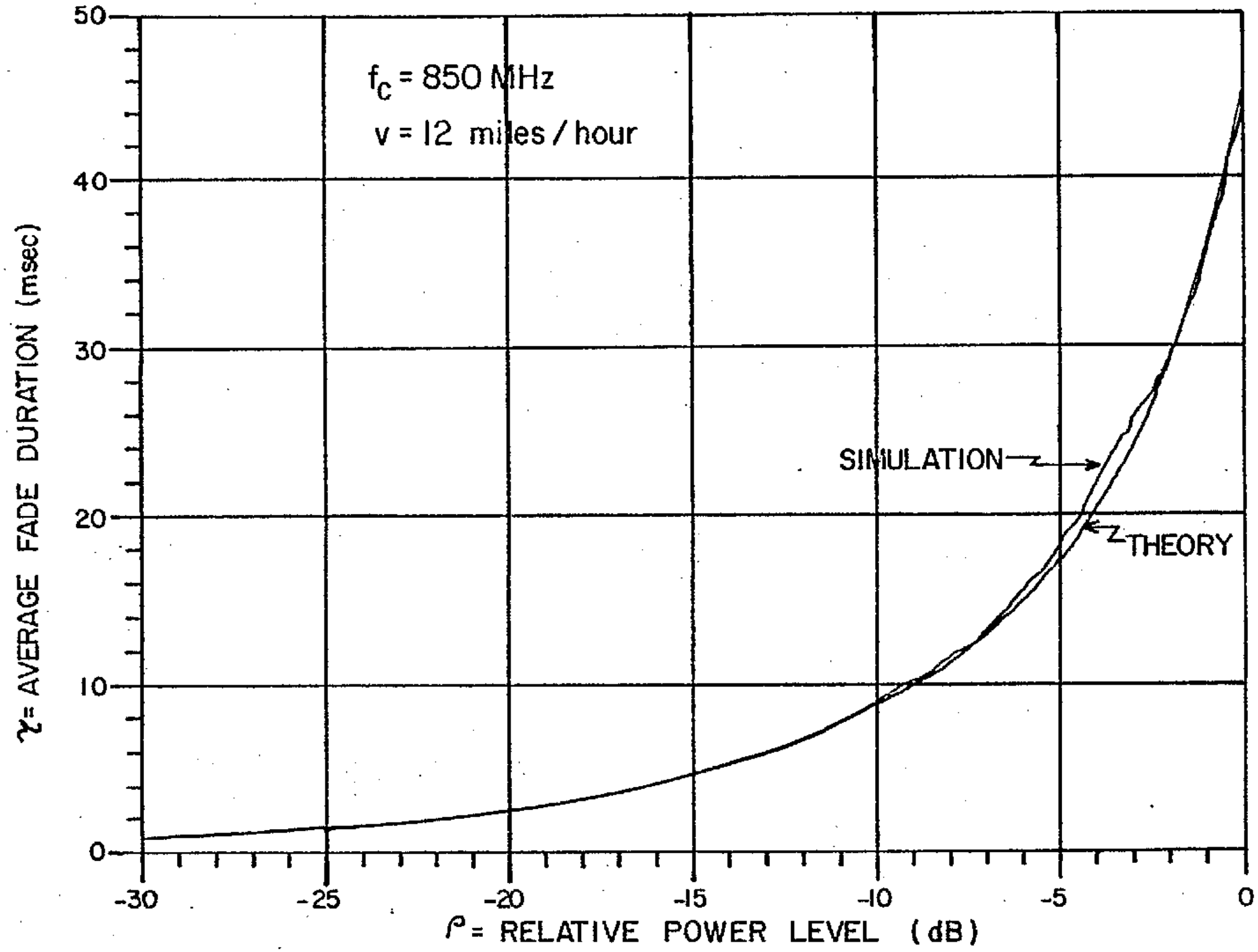


Figure 2.6 THE AVERAGE FADE DURATION - THEORY AND SIMULATION

to the average fade duration and the vertical axis represents the cumulative distribution of the fade width. In the absence of a theoretical model to describe the fade width distribution, we have attempted to compare the simulation results with the exponential distribution. Figure 2.7 indicates that the exponential model may be acceptable for relative threshold to signal levels around 0 dB but is less acceptable for values such as -10 dB or -15 dB.

The third program is used to calculate the distribution of the non-fade intervals. In contrast to the fade width distribution, Figure 2.8 shows that the non-fade distribution agrees with the exponential distribution for values of the relative threshold to signal level below -10 dB which is rather encouraging since in practice the threshold level should be 10 to 30 dB below the average signal level. It is thus concluded that the exponential model can be used to describe accurately the non-fade interval distribution. This property is used to estimate the packet error rate for land-mobile data channels as will be shown in Chapter 3.

The fourth program computes the distribution of the time intervals between consecutive level crossings. In this program the random variable is the interval of time between the start of two consecutive downward level crossings or equivalently, the period of time corresponding to the sum of the fade interval and the non-fade interval. The simulation results shown in Figure 2.9 indicate that the interfade interval and the non-fade interval have nearly similar distributions which can be approximated by an exponential distribution for high values of the signal to threshold ratio.

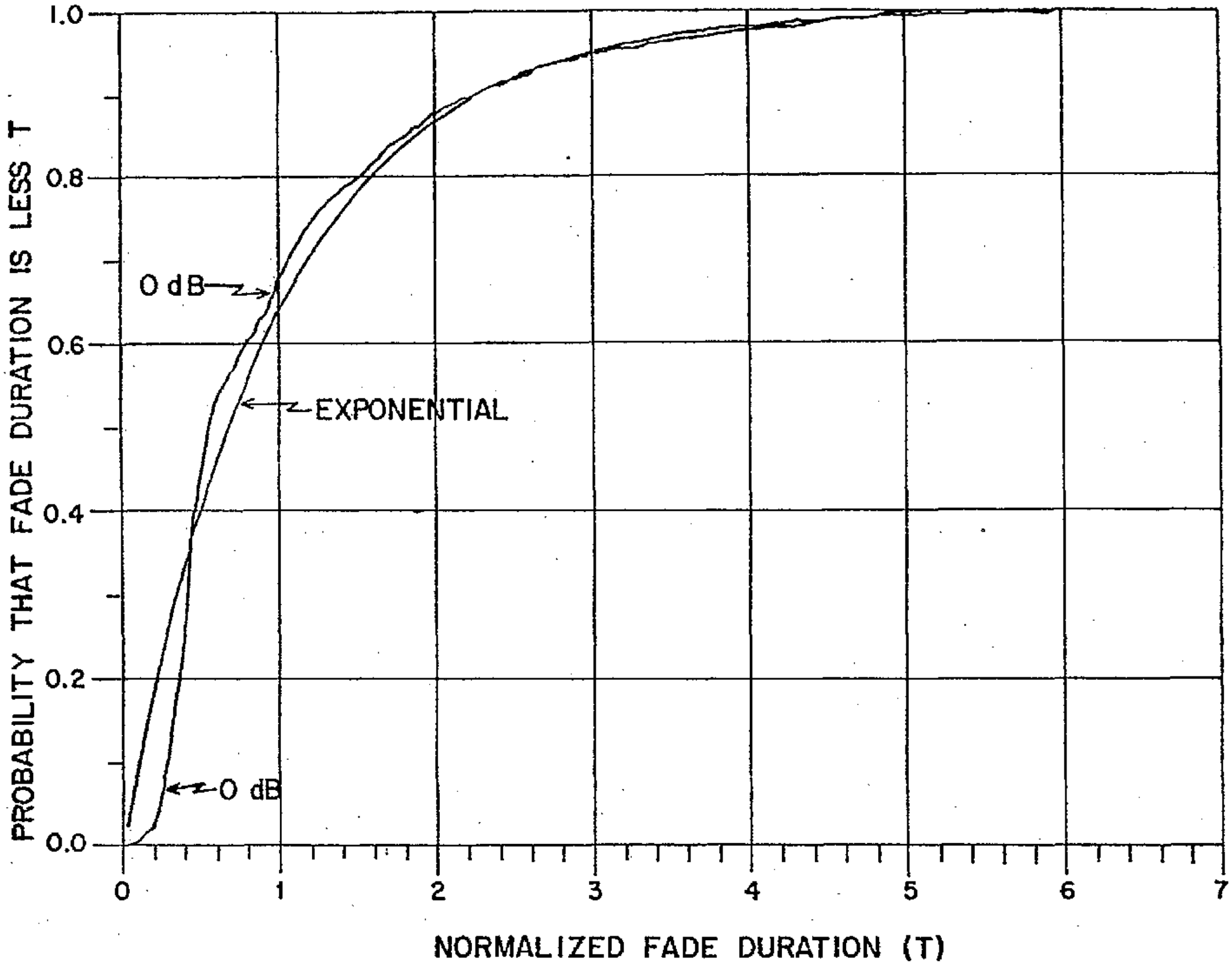


Figure 2.7a THE FADE WIDTH DISTRIBUTION (ρ=0 dB)

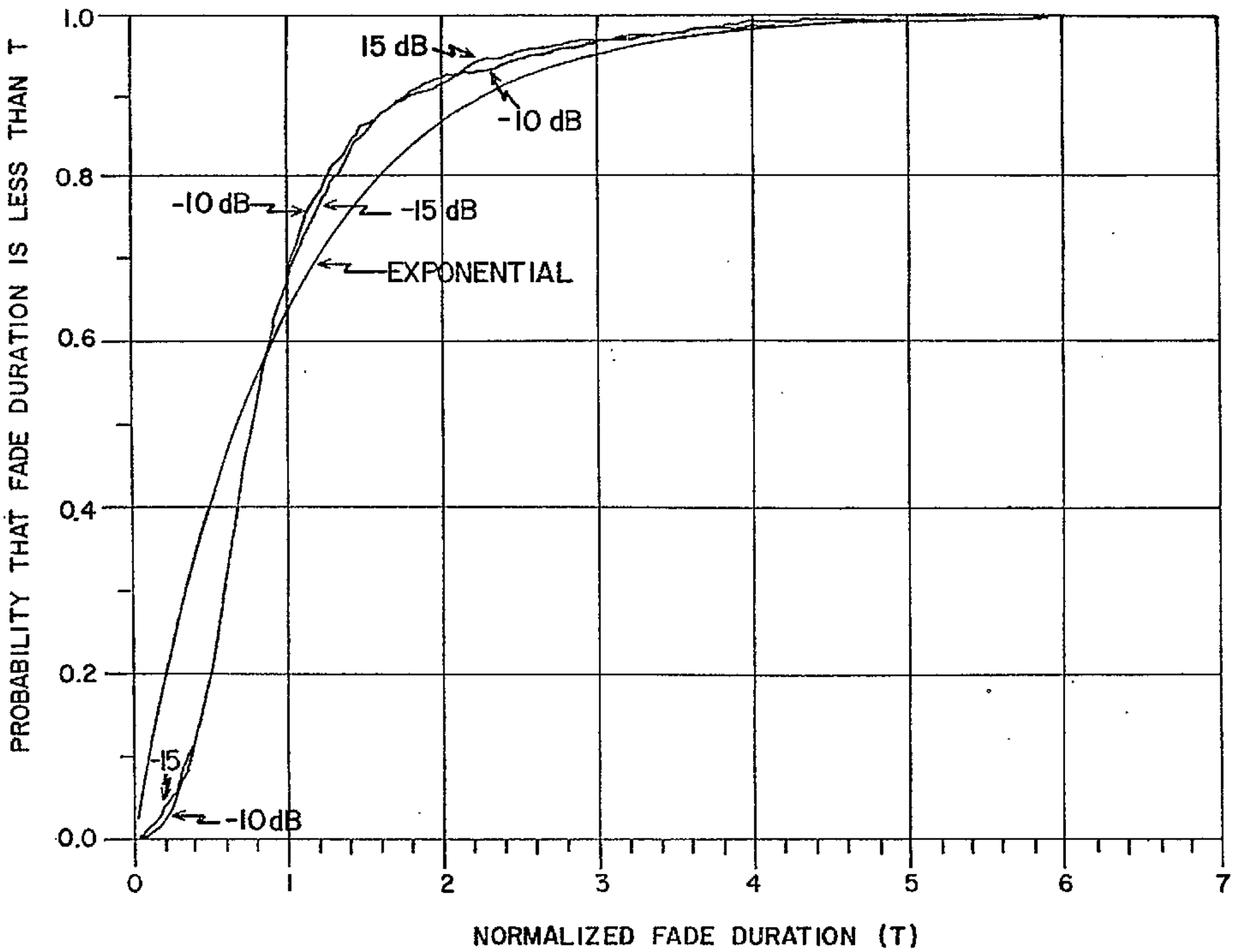


Figure 2.7 b THE FADE WIDTH DISTRIBUTION (ρ=-10,-15 dB)

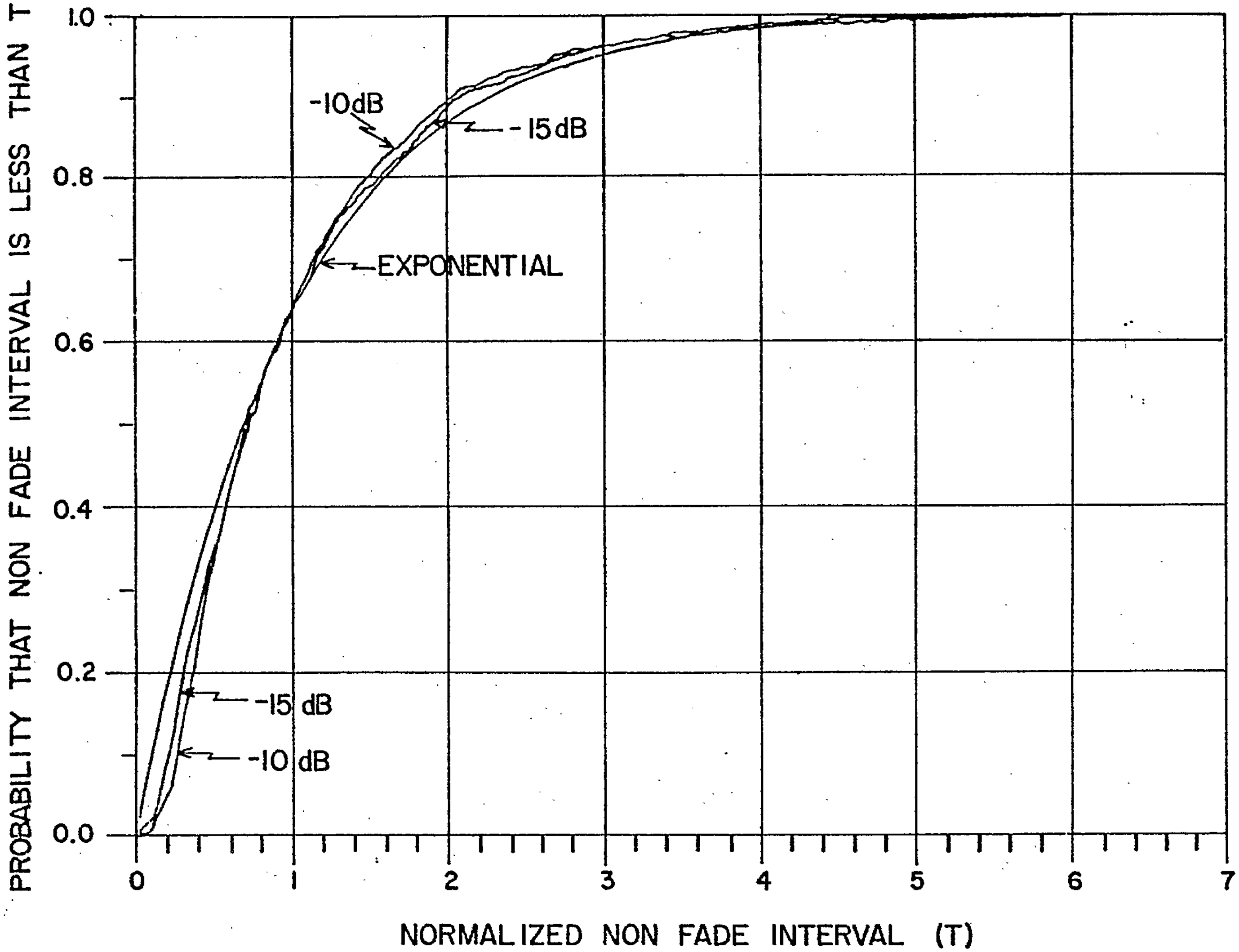


Figure 2.8a DISTRIBUTION OF NON-FADE INTERVALS
(ρ = -10, -15 dB)

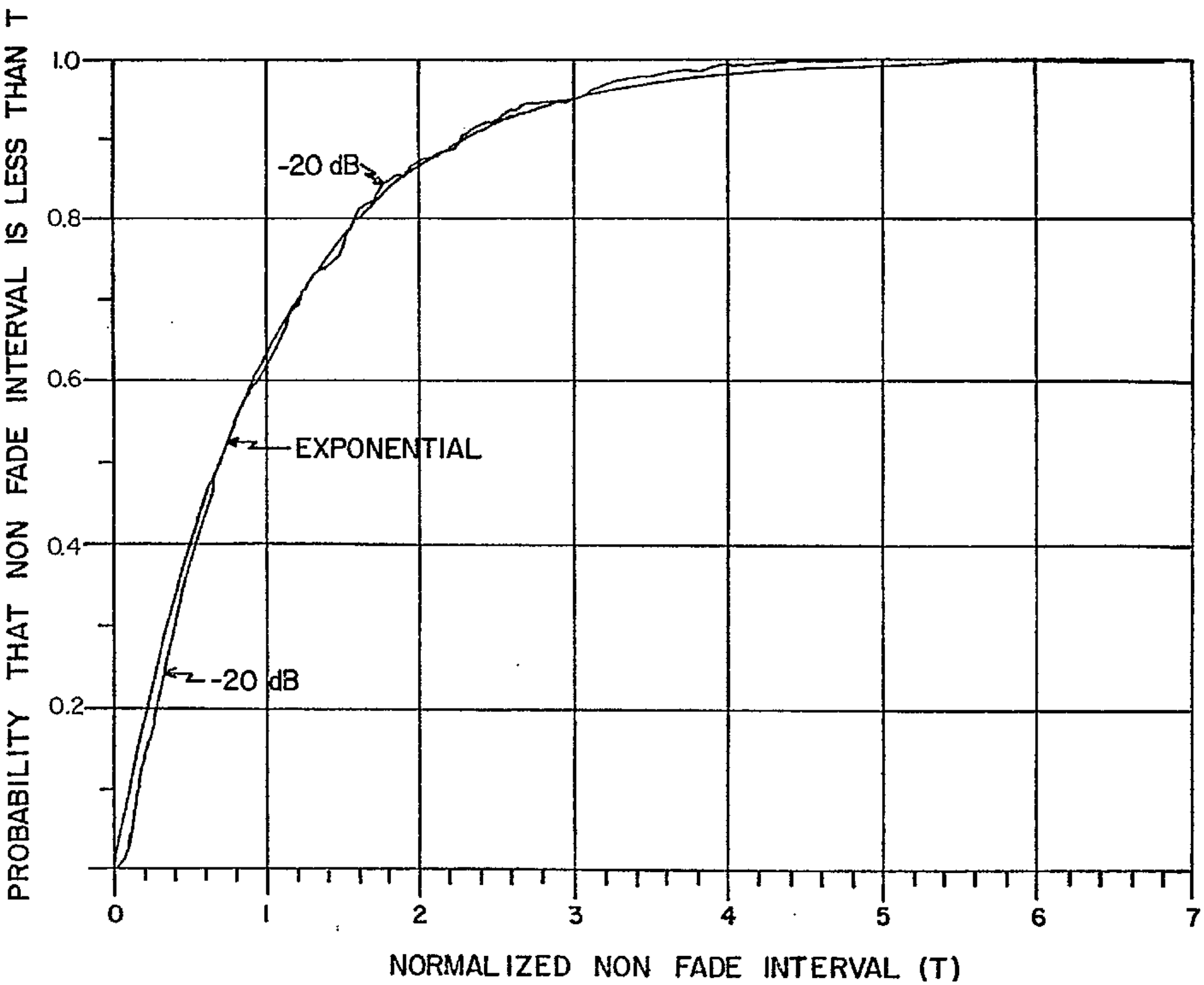


Figure 2.8b DISTRIBUTION OF NON-FADE INTERVALS (ρ = -20 dB)

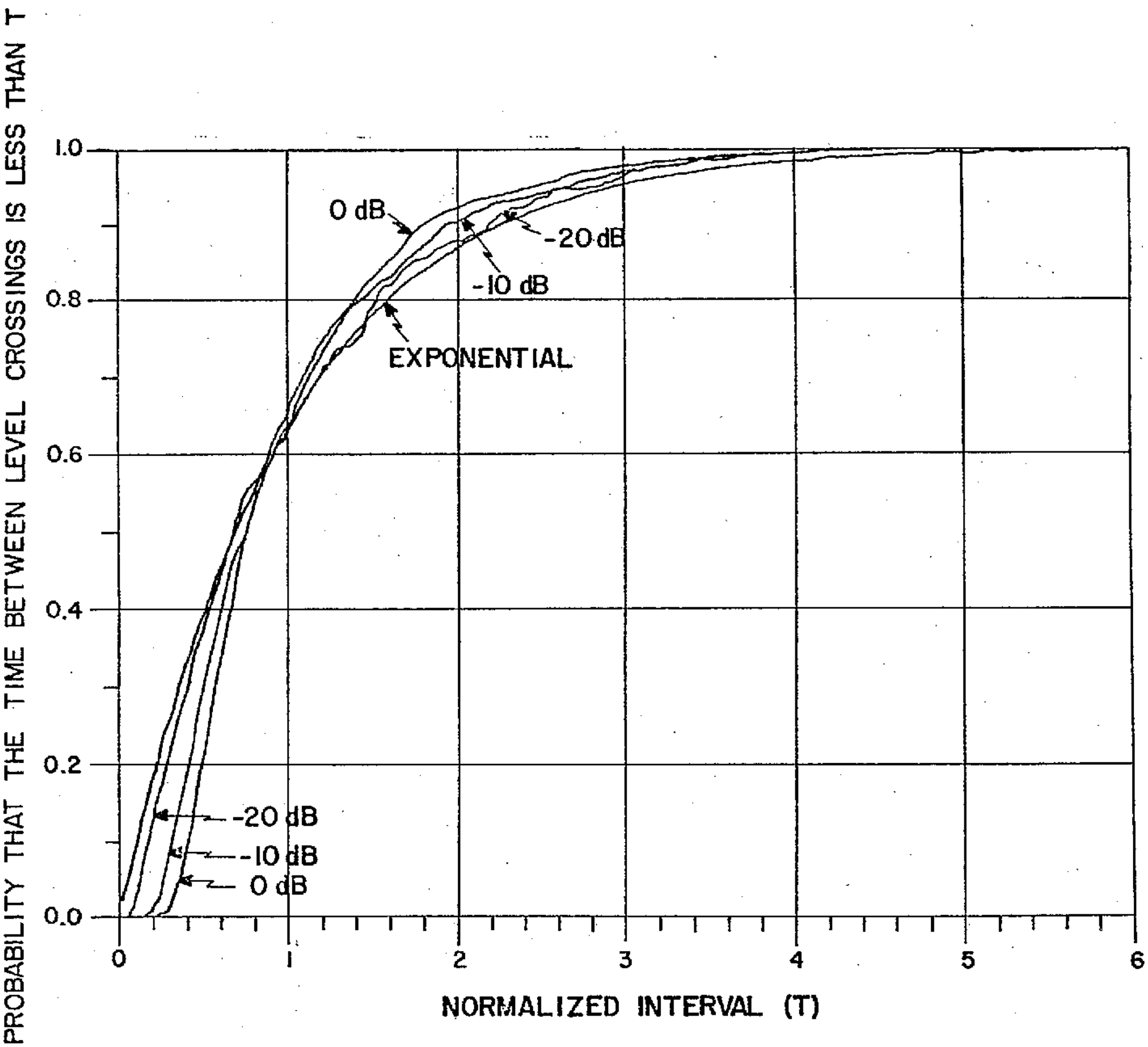


Figure 2.9 DISTRIBUTION OF INTERVALS BETWEEN LEVEL CROSSINGS (ρ = 0, -10, -20 dB)

2.4 Summary of Results

The following is a summary of the results of the Rayleigh fading simulation model developed in this chapter:

* The computer-generated envelope possesses all the characteristics of a Rayleigh fading signal;

* The average fade duration and the average level crossing rates agree well with the theoretical predictions;

* The cumulative distributions of the three variables: fade interval, non-fade interval and interfade interval, vary with the threshold level. While the fade width distribution can only be approximated by an exponential distribution for values of the relative threshold to average ratio close to 0 dB, the other two variables can be assumed to be exponentially distributed whenever the ratio of the average to threshold signal levels is in the range 10 to 30 dB;

* The fact that the non-fade and interfade intervals are nearly exponentially distributed for the practical range of threshold to average levels (<-10 dB) provides an important tool in estimating the packet error rate for land-mobile data channels.

CHAPTER 3

IMPACT OF FADING ON PACKET TRANSMISSIONS

3.1 Introduction

In practise, the transmission of a continuous stream of digital data over communications channel encounters errors due to the presence of random noise, distortion and interference. For many communication channels, (e.g. satellite channel), the thermal noise generated within the receiver is considered to be the major source of error, hence the performance of such channels is usually measured in terms of the signal-to-noise (S/N) ratio, where the noise is often considered to be a White Gaussian Noise, (WGN). Although the WGN assumption is not always valid, it greatly simplifies the analytical calculations of the bit error rate (BER) in a simple closed form expression as a function of the S/N ratio.

The situation is different in land-mobile data channels as signal fading rather than random thermal noise forms the major source of errors. The error characteristics of land-mobile channels differ from that of non-fading channels in essentially three aspects:

- The error rate for land-mobile channel is at least an order of magnitude higher than that of non-fading channels.
- The error rate for non-fading channels usually exhibits a threshold behaviour, i.e., the bit error rate drops to a negligible value when the S/N ratio exceeds a certain value

(usually between 10 and 12 dB for most modulation techniques). Such behaviour does not exist in the case of land-mobile channel, where the bit error rate decreases at constant rate (10 dB/decade for Rayleigh fading channels) with increased values of S/N ratios.

- For non-fading channels bit errors are usually taken to be independent and randomly distributed, while for Rayleigh fading land-mobile channels we obtain bit error distributions which are intimately related to the average fade duration and average level crossing rate. Such channels are referred to as bursty error channels.

This chapter is concerned with the determination of the impact of fading errors on digital transmissions in packet form. More specifically, in Sections 3.2 and 3.3 we describe the approach taken to determine the average bit error rate as well as the packet error rate. The knowledge of the packet error rate will allow us to study in Section 3.4 the problem of the capacity degradation of packet radio channels under fading conditions. In Section 3.5, we present a method for determining the optimal packet length on the base to mobile channel.

3.2 Bit Error Rate

In order to calculate the average bit error rate in land-mobile fading channels, we can establish a certain threshold level, y_t, such that the signal level, at the decision instant, can be greater or smaller than y_t. If the signal level is less than y_t, the probability that the particular bit is in error will typically be 0.5. If the signal level is greater than y_t, the probability of bit error is reduced. Clearly, the absolute value of the threshold level depends upon the modulation technique being used. In the sequel we define the relative threshold level with respect to the average signal level as follows:

$$\rho = \frac{y_t}{y_o} = \frac{\text{The Threshold Level}}{\text{Average Signal Level}} \qquad \ldots\ldots (3.1)$$

Knowing $P_e[1/\rho_i]$, the bit error rate expression for the selected modulation scheme, the average bit error rate $\overline{BER}$ under fading condition is given by:

$$\overline{BER}\,(\rho) = \int_0^\infty P_e[1/\rho_i] \times P[\rho_i/\rho]\, d\rho_i \qquad \ldots\ldots (3.2)$$

where ρ_i is the ratio of the threshold to instantaneous signal level. Using the simulation model described in Chapter 2, the average bit error rate can be computed as follows:

- At the decision instant, we find the instantaneous signal level and the corresponding threshold to signal ratio ρ_i. Then, ρ_i is substituted in the bit error rate equation corresponding to the particular modulation scheme being used, to determine the probability of a bit error.
- The average $\overline{BER}$ as a function of $\bar{\rho}$ can be obtained by repeating the above step for a large number of decision instants.

For a file consisting of 160000 points representing the instantaneous signal level and for a modulation technique based on non-coherent detection (DPSK) we obtain the simulation results given in Figure 3.1. Notice that for values of ρ less than or equal to -10 dB the BER slope is constant at 10 dB/decade. The dotted line in the figure represents the BER curve for non-fading channels which is shown here for comparison purposes. We should note here that the simulation results are relatively insensitive to the Doppler frequency or transmission speed over the practical ranges of these two variables. Finally, it is observed that the simulation results shown are in good agreement with theoretical or experimental results published elsewhere {49}.

An alternative derivation of the average bit error rate can be obtained by a careful consideration of the fading envelope shown as Figure 2.3. For convenience, the fading envelope can be schematically presented as in Figure 3.2, with the three fading variables identified. From Chapter 2 we know that the average fade duration, the average non-fade interval and the average inter-fade interval are given by:

$$\bar{X} = \tau = \frac{\exp(\rho) - 1}{f_d \sqrt{2\pi\rho}} \qquad \text{..... (3.3)}$$

$$\bar{Y} = \bar{Z} - \bar{X} = \frac{1}{N} - \tau \qquad \text{..... (3.4)}$$

$$\bar{Z} = \frac{1}{N} = \{f_d \sqrt{2\pi\rho} \exp(-\rho)\}^{-1} \qquad \text{..... (3.5)}$$

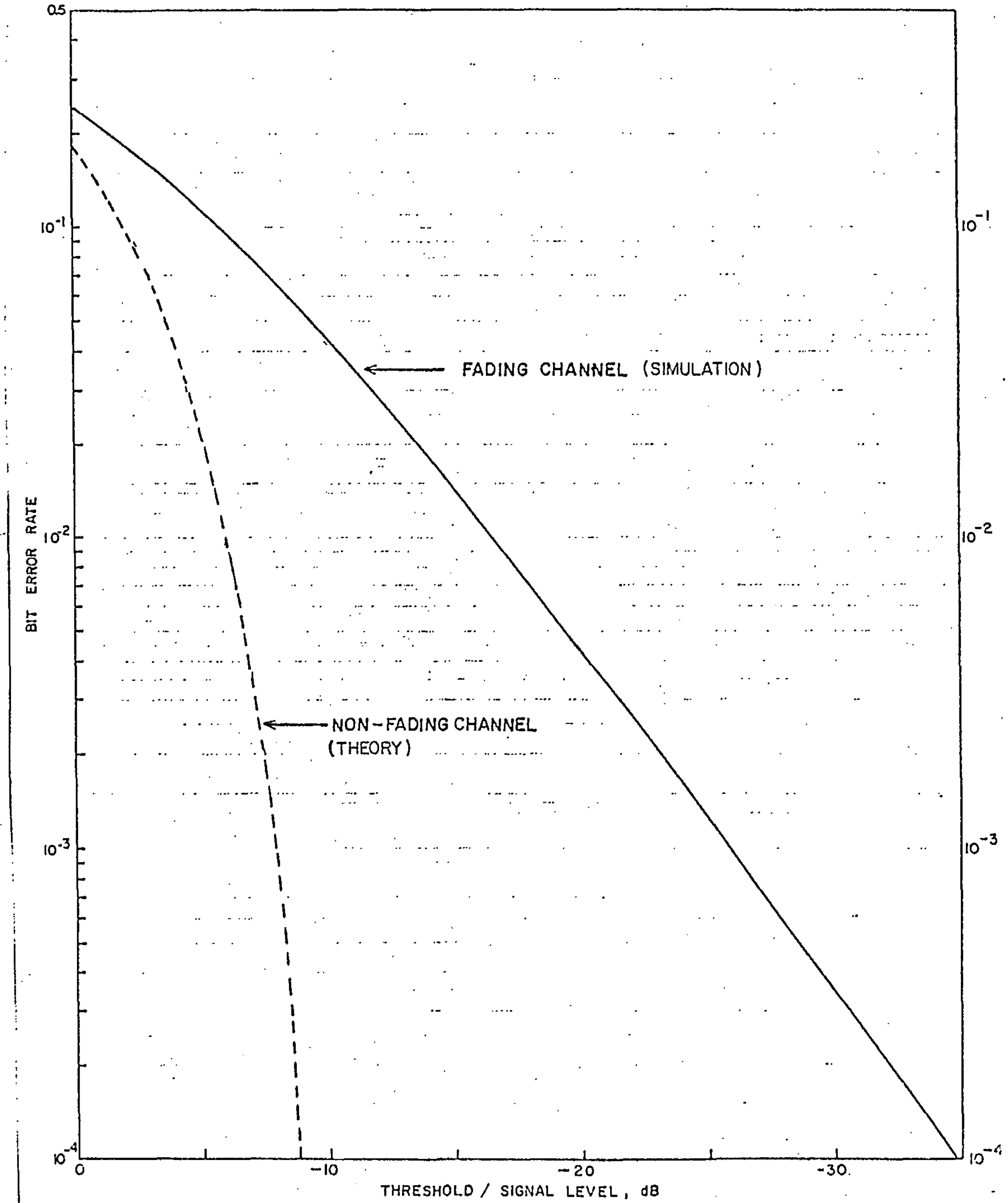
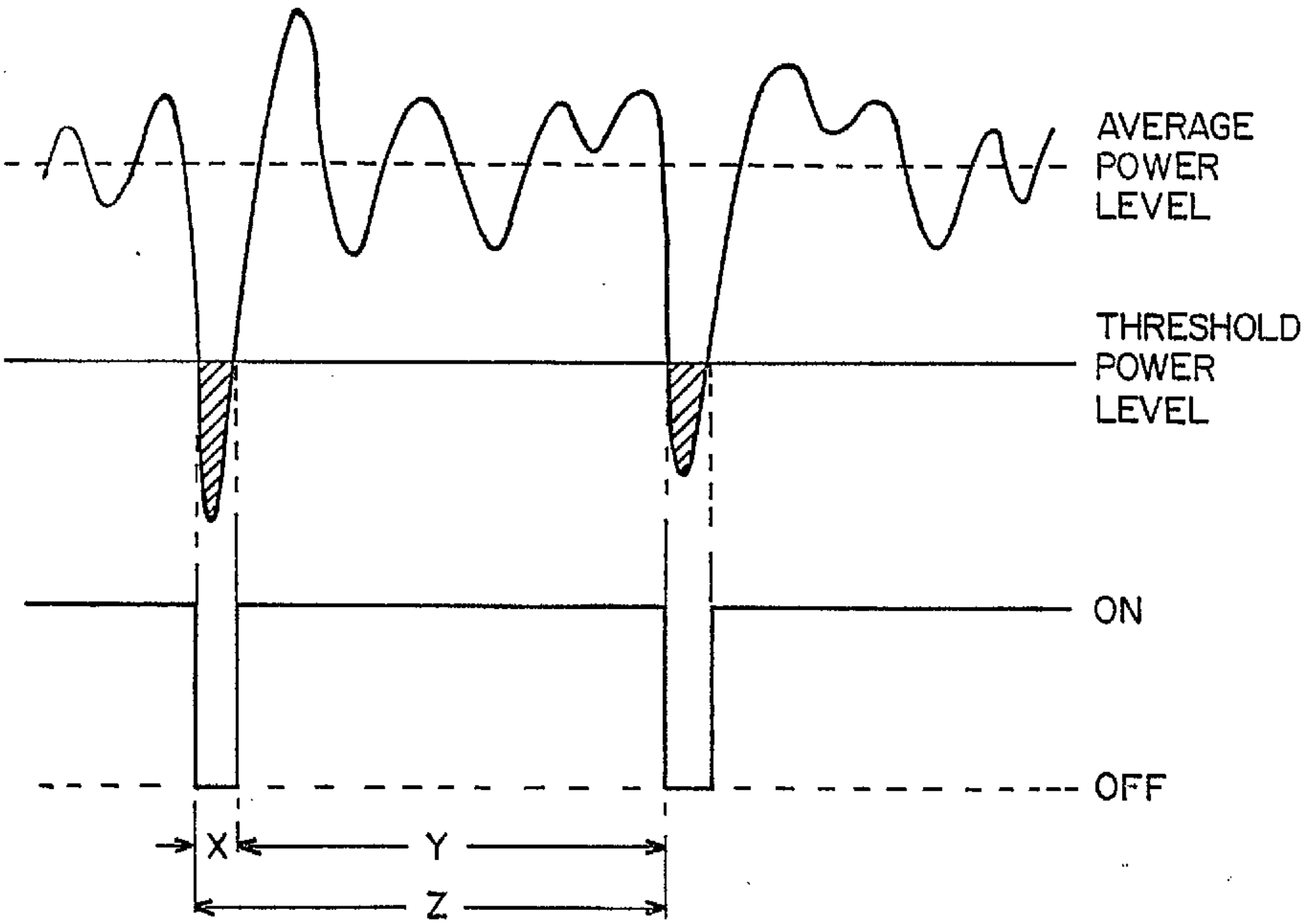


Figure 3.1 BER CHARACTERISTICS FOR RAYLEIGH FADING CHANNELS



X = FADE DURATION

Y = NO-FADE DURATION

Z = INTER-FADE INTERVAL

Figure 3.2 THE ON-OFF PATTERN OF THE FADING ENVELOPE

Selecting DPSK as a modulation technique, the probability of a bit error will be either equal to 0.5 if the bit hits a fade period, or equal to:

$$Pe = (0.5) \exp(-1/\rho) \qquad \ldots\ldots (3.6)$$

if the bit hits a non-fade interval. Hence the average bit error rate $\overline{BER}\ (\rho)$ will be given by:

$$\overline{BER}\ (\rho) = (0.5)\ \frac{\bar{X}}{\bar{Z}} + (Pe)\ \frac{\bar{Y}}{\bar{Z}} \qquad \ldots\ldots (3.7)$$

which can be written as:

$$\overline{BER}\ (\rho) = 0.5 \left\{ 1 - \exp(-\rho) + \exp\left[-(1 + 1/\rho) \right] \right\} \ldots\ldots (3.8)$$

3.3 Packet Error Rate

Equation (3.8) gives an exact expression for the average bit error rate since it is identical to the expression obtained from Eqn. (3.2), however, it cannot be applied whenever the digital information is transmitted in packet form, as is the case in many data communication systems. In such systems each data packet enters the system as a separate entity with a header that contains the identification of the source and the destination. A packet also contains a number check-sum bits which allow the receiver to detect and/or correct the errors incurred during the transmission process. Upon reception of a data packet, the receiver checks the information contained in the packet against possible transmission errors and takes the appropriate action required by the particular communication protocol that is being used.

In such systems, it is important to calculate the Packet Error Rate (PER) which, for our purposes, is defined as the fraction of packets that have at least one detected error. For non-fading channels where noise is the major source of errors and bit errors are randomly distributed, the PER and BER are related through the following equation:

$$PER = 1 - (1 - BER)^L \qquad \text{..... (3.9)}$$

where L is the packet length.

In the case of fading channels no equivalent formula is available for the PER since it is extremely difficult to model the error bursts due to fading. Most workers in this field, such as French {55} and Mabey {56}, have resorted to field trials to try to gain an understanding of the error distributions due to fading. In the following we will derive a simple but approximate closed form expression for the packet error rates in fading channels. To do so we will assume, in agreement with the results given in Chapter 2, that the non-fade intervals are exponentially distributed. Referring again to Figure 3.2, we assume the packet to have been received correctly if it was contained entirely within the non-fade interval, Y. If the packet, on the other hand, overlaps to any extent with a fade slot, it is considered to have at least one error. Based on these assumptions, the PER can be written as:

$$PER = 1 - \frac{\bar{Y}}{\bar{Z}} \; P[Y>T] \qquad \text{..... (3.10)}$$

where T is the packet duration, $\bar{Y}$ and $\bar{Z}$ are the mean values of the non-fade and inter-fade intervals, respectively.

Assuming the non-fade intervals to be exponentially distributed we have:

$$P[Y>T] = \exp(-T/\bar{Y}) \qquad \text{..... (3.11)}$$

Using Eqns. (3.4) and (3.5) we obtain:

$$PER = 1 - \exp\{-(\rho + f_d T\sqrt{2\pi\rho})\} \qquad \text{..... (3.12)}$$

The above equation is plotted in Figure 3.3 for a mobile system operating at 850 MHz, for two packet sizes over a 35 dB range of relative threshold levels. We have further selected a channel transmission rate of 4.8 kbps and a vehicle speed of 48 km/h. The figure clearly shows that the packet error rate can exceed 10^{-2} for the selected set of parameters.

In order to confirm the validity of Eqn. (3.12), which we recall is based on the assumption of exponentiallity for the non-fade intervals, a number of simulation tests were performed using the fading simulator described in Chapter 2. These tests showed that in particular for values of the threshold to average signal ratio in the range -25 to -30 dB, the theoretical predictions gave slightly pessimistic results when compared with the simulation results. The differences are attributed to the lack of accuracy of the simulation model in this particular range, as indicated by Figures 2.4 and 2.5.

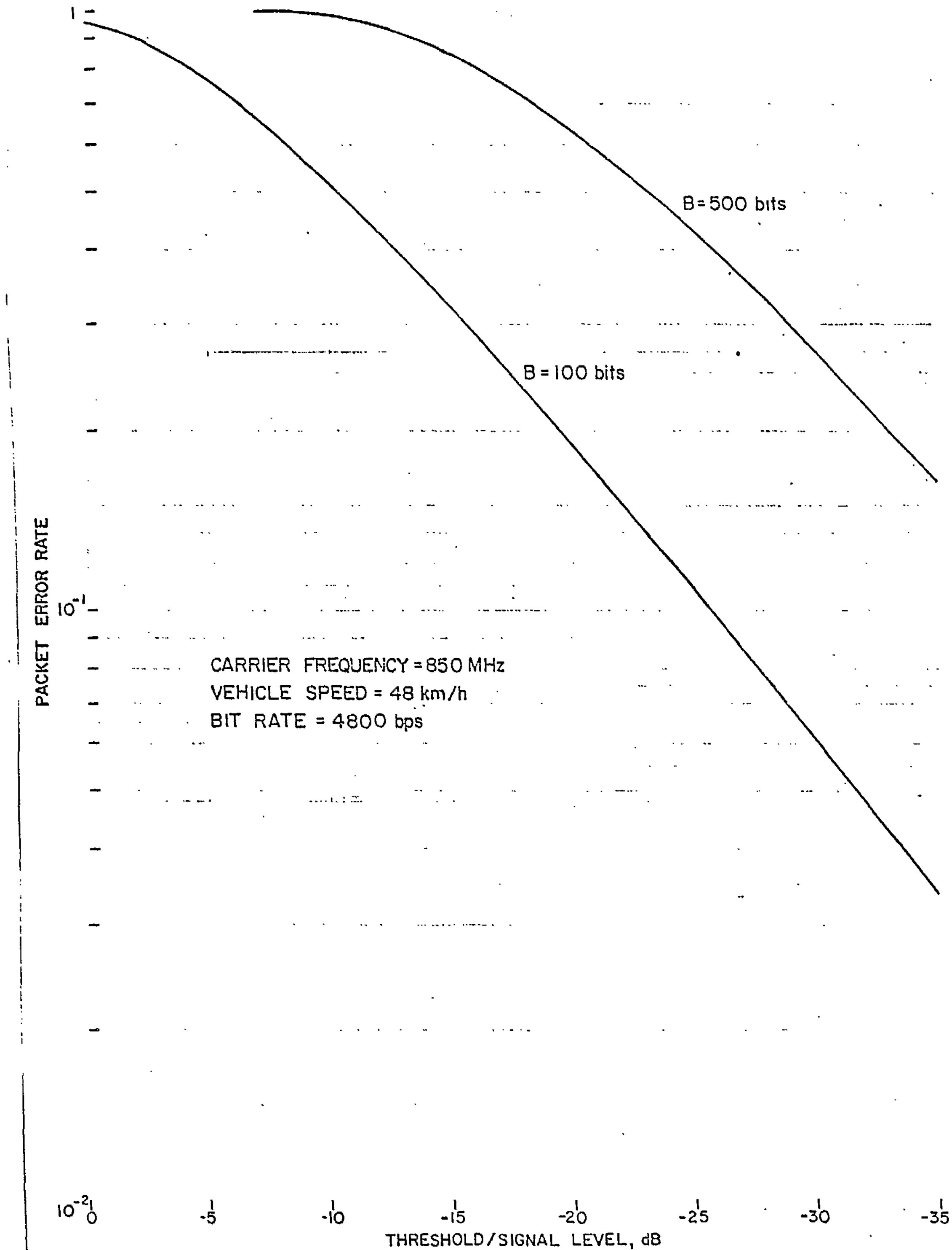


Figure 3.3 PACKET ERROR RATE CHARACTERISTICS OF RAYLEIGH FADING CHANNELS

For all practical purposes the expression given by Eqn. (3 2 although approximate represents a very useful tool for analyzing a evaluating the performance of packet transmissions in the mobile environment. Because of its simplicity, this formula provides e system designer with a quick and relatively accurate way of determ the packet error rate under wide range of system parameters. F instance, the effect of the vehicular speed and the carrier freque can be examined by a simple substitution of the proper values for Doppler frequency f_d in the equation. Similarly, the packet le gt and the speed of transmission can be examined through the proper values of the packet duration.

It should be noted that in the simulation model only the e o by the signal fading were accounted for. Therefore, the simulation results and the PER formula are valid for mobile terminals only stationary terminal, on the other hand, receives a signal with a constant envelope, hence, the errors will be caused basically random thermal noise generated in the receiver front end. In c case, errors will be randomly distributed and the packet error ra given by Eqn. (3.9).

3.4 Capacity Degradation of Packet Radio Fading Channels

The typical assumption used in the study of the throughpu formance of packet radio random access schemes, consists in stati packet collisions represent the only source of packet errors. assumption is to some extent valid provided that the packet radic terminals are stationary. However, packet radio techniques ca applied to an environment where the terminals are mobile as is the case in dispatch land mobile systems. Hence the assumption of a noise-less channel is no longer warranted, since even in the abser of packet collisions, packet errors can occur because of fading. this context, it is relevant to determine how the presence of fad: affects the throughput performance of random access schemes. To c so we consider the model shown in Figure 3.4 where we examine in particular the uplink channel*. We will ignore errors due to ign: noise and assume that the acknowledgement traffic is carried by a channel distinct from the uplink channel (mobiles to base).

In the absence of packet collisions, we have seen in the prev section that the probability of successful packet transmission is given by:

$$P_S = \exp \{ - (\rho + f_d T\sqrt{2\pi\rho}) \quad \ldots.$$

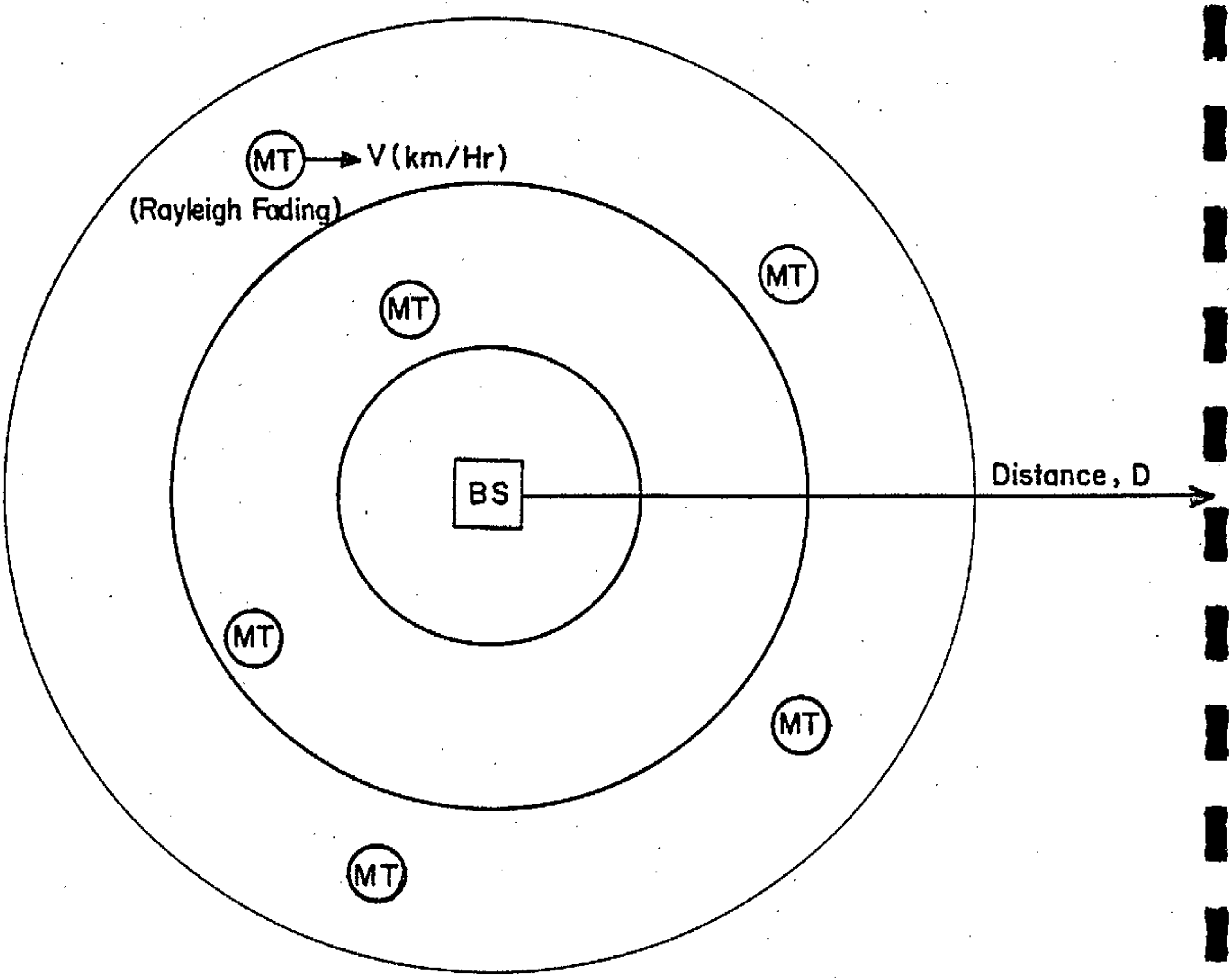
On the other hand, in the absence of fading the relationship l the traffic and the throughput for the classical random access sc! namely; Pure ALOHA, Slotted ALOHA and Non-Persistent CSMA is give:

Pure ALOHA $\qquad S = G \exp(-2G) \quad \ldots.$

Slotted ALOHA $\qquad S = G \exp(-G) \quad \ldots.$

Non-Persistent CSMA $\qquad S = \dfrac{G \exp(-\beta G)}{G(1+2\beta) + \exp(-\beta G)} \quad \ldots.$

* The downlink channel will be examined in Section 3.5.



B.S. = Base Station
M.T. = Mobile Terminal

Down-Link (BS $\rightarrow$ MT) $\Rightarrow$ Fading Errors Only
Up - Link (MT $\rightarrow$ BS) $\Rightarrow$ Fading Errors + Collision

Figure 3.4 CAPACITY DEGRADATION MODEL

where S is the channel input rate (average number of new packets generated per packet transmission interval T), G is the channel traffic rate (average number of new packets plus previously collided packets generated per packet transmission time) and β is the normalized propagation delay (ratio of propagation delay to packet transmission time)

Assuming independence between the fading and collision processes and reinterpreting G as the traffic rate due to new arrivals, collisions and repetitions caused by fading errors, Eqns. (3.14)-(3.16) can be rewritten as follows:

Pure ALOHA $$S = G \exp\{-(2G + \rho + f_d T\sqrt{2\pi\rho})\} \quad \ldots\ldots (3.17)$$

Slotted ALOHA $$S = G \exp\{-(G + \rho + f_d T\sqrt{2\pi\rho})\} \quad \ldots\ldots (3.18)$$

Non-Persistent CSMA $$S = \frac{G \exp\{-(\beta G + \rho + f_d T\sqrt{2\pi\rho})\}}{G(1+2\beta) + \exp(-\beta G)} \quad \ldots\ldots (3.19)$$

Recalling that ρ denotes the ratio between the receiver threshold level and the average received power, we conclude that for a fixed threshold level as the mobiles move away from the base station there will be a degradation in throughput, due to the fact that P_S, as given by Eqn. (3.13), decreases. This phenomenon is shown in Figure 3.5 where we clearly see that, for a carrier frequency of 850 MHz and a vehicle speed of 60 km/h, the maximum value of throughput i.e the channel capacity, is substantially reduced for increasing values of ρ.

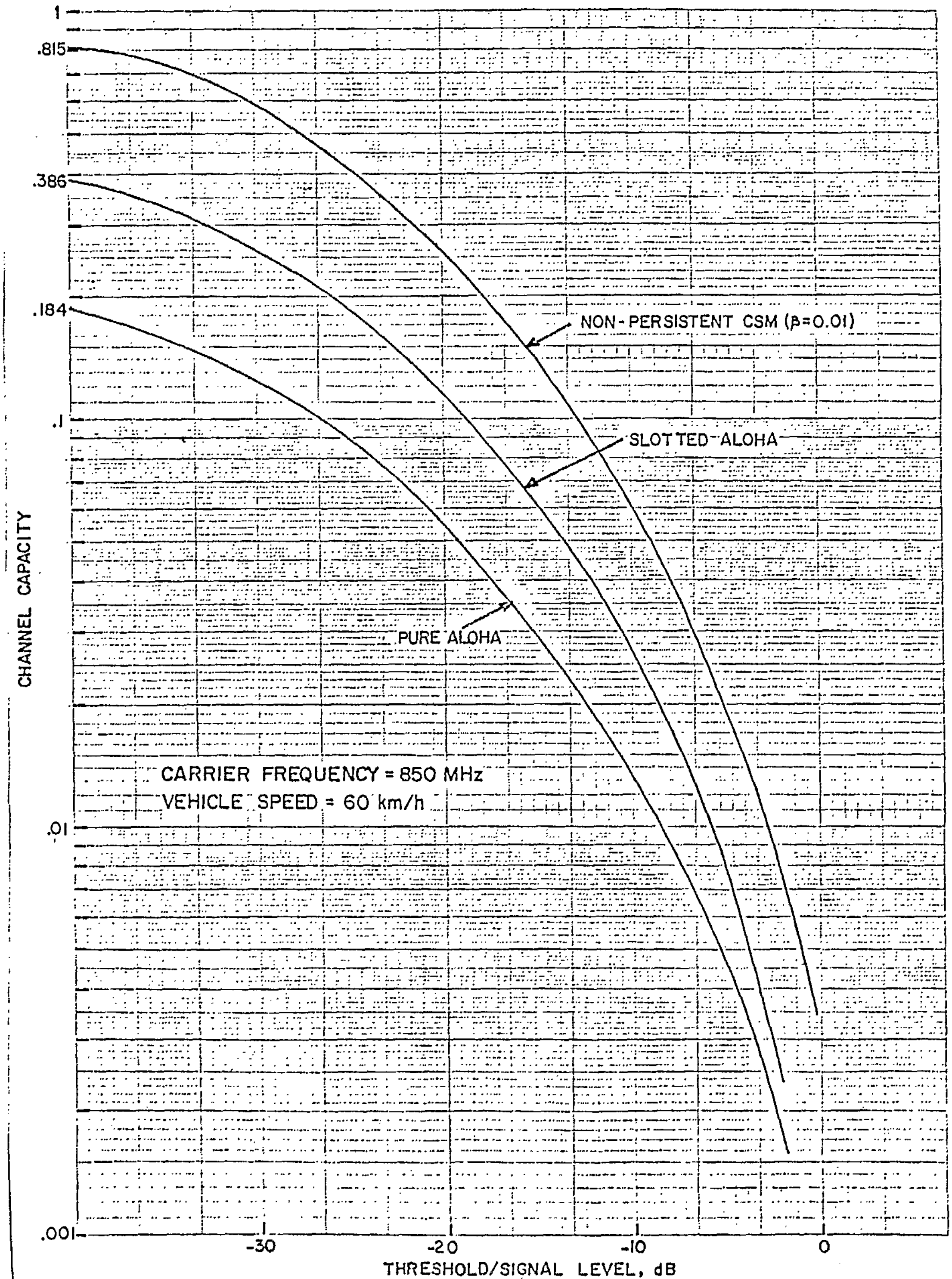


Figure 3.5 THROUGHPUT DEGRADATION AS A RESULT OF FADING

3.4.1 Path Loss Model

For a given set of system parameters it then becomes important to relate the values of ρ to the corresponding distances to the base station. To do so we consider a path loss model proposed by Okumura {57}. According to this author the total path loss attenuation in dB can be approximated by:

$$P_L(\text{dB}) \approx 37 + 20 \log f + 20 \log D + A_{Fs} \qquad \ldots\ldots (3.20)$$

where:

f = frequency in MHz

D = distance in miles

A_{Fs} = median attenuation relative to free space (dB)

Using average effective transmissions heights for the base station and terminal of 45 meters and 3 meters respectively, the median attenuation A_{Fs} can be approximated by:

$$A_{Fs} = 13.3 \log D + 36 \qquad (\text{for } D \leq 14 \text{ miles}) \qquad \ldots\ldots (3.21)$$

Hence at a carrier frequency of 850 MHz the total path loss will be:

$$P_L(\text{dB}) \approx 131.6 + 33.3 \log D \quad (\text{for } D \leq 14 \text{ miles}) \ \ldots\ldots (3.22)$$

If we now assume an effective radiated power of +40 dBm and a received threshold power of -130 dBm we obtain the following relationship between the distance, the total path loss and the ratio of threshold to average received power:

D(mi)	1	2	5	8	10	14
P_L(dB)	131.6	141.6	154.9	161.7	164.9	169.7
ρ(dB)	-38.4	-28.4	-15.1	-8.3	-5.1	-0.3

Comparing these values with the curves of Figure 3.5 it becomes clear that if the average distance between the mobile terminals and the base station exceeds 5 miles the throughput degradation becomes substantial. In the derivation of the above results it was implicitly assumed that all mobile terminals were at the same distance from the base station. If we assume that terminals are uniformly distributed within a circle of radius R, where R represents the coverage range of the base station, we should replace D and ρ by their average values.

For the same set of parameters we obtain the following relationship between the average value of ρ and the average distance to the base station:

$$\bar{\rho}(\text{dB}) = -38.4 + 33.3 \log \bar{D} \qquad \text{..... (3.23)}$$

The uniformity assumption implies that:

$$\bar{D} = \frac{2R}{3} \qquad \text{..... (3.24)}$$

Hence:

$$\bar{\rho}\ (\text{dB}) = -44.26 + 33.3 \log R \qquad \text{..... (3.25)}$$

Taking as an example, the Pure ALOHA random access scheme we obtain the following results:

R(mi)	1	2	5	8	10	14
$\bar{\rho}$(dB)	-44.26	-34.23	-20.98	-14.18	-10.96	-6.09
S_{MAX}	0.184	0.152	0.058	0.026	0.016	0.005

3.5 Optimal Packet Length

As we have already mentioned the problem of designing packet radio systems encompasses a number of system variables such as the network topology, channel configuration, access policy, modulation schemes, routing, error control and flow control procedures.

The error control procedures are of significant importance for packet radio since transmission errors will result from noise, fading and packet overlaps. However, most studies concerning multi-access channels ignore error recovery procedures.

Exceptions include the studies of Schwartz {58}, Binder {59} and Tobagi {60}. Schwartz, in what is probably the earliest attempt to assess the impact of channel errors, compared the performance of FEC and ARQ for a pure ALOHA channel. His analysis is based on the assumption that collisions are the only source of channel errors. Later Binder {59} considered the degradation due to acknowledgement error control traffic on a two channel system; an uplink ALOHA channel from terminals to the central station and a downlink from the central station to the users. More recently Tobagi {60} investigated the effect of acknowledgement traffic on maximum throughput for the slotted ALOHA and CSMA protocols.

On single access channels, a number of authors have examined the performance and efficiency of ARQ and FEC systems, typically under the assumption of random errors. Sastry {61},{62} among others has been concerned with the performance of hybrid error control systems which use both ARQ and FEC, for channels characterized by random and burst errors. Recently Townsley {63} examined a number of variations of the basic ARQ technique, with respect to maximum throughput and average queue lengths. The most recent study of error control systems in multi-access environments, due to Saeki {64}, only assumes random noise errors.

In this section we focus our attention on the send and wait error control procedure, often called stop-and-wait (SW-ARQ), as it is representative of many data communication systems. According to this strategy, the transmitter sends one packet of data at a time and waits for an acknowledgment (ACK) from the receiver before proceeding. This wait, also referred to as the time out, will last at most A time units after which if the transmitter has not received an ACK the same packet is retransmitted. This procedure is repeated until the ACK is received at which point in time another packet will be transmitted.

A mathematical model, based on the work of W.W. Chu {65}, is used to determine the optimal packet size that minimizes the expected waiting time in retransmission and acknowledgment delay and thus maximizes the channel efficiency of mobile radio channels in a fading environment.

The model used assumes that transmission errors are only caused by fading i.e errors due to other sources of noise and interference are not considered. Traffic originates at a fixed station and is destined to a mobile terminal. Thus we only consider outbound traffic. We further assume that the acknowledgment traffic carried by a separate channel arrives at the fixed station reliably and at no cost.

3.5.1 Model Description

Several attempts to describe and quantify errors on mobile radio channels have been made {55},{66},{67}, however, the available results assume that the sending station transmits a continuous bit stream to the mobile terminal. No attempt has been made so far to assess the impact of the error characteristics of mobile channels when the sending station formats the bit stream in packets of data and transmit them to the mobile terminal. Following a model originally proposed by W. Chu {65} we will be concerned with the determination of the optimal packet size that minimizes the time wasted in acknowledgments, retransmission and packet overhead. The assumption is made that messages arriving at the sending station are geometrically distributed with mean $\bar{\ell}$, that is $P_L(\ell) = pq^{\ell-1}$, with $\bar{\ell} = 1/p$ and $p+q=1$. We will consider a stop and wait transmission strategy and assume that error bursts are due to the fading of the received signal below a certain threshold level which is receiver dependent. Random noise errors due to thermal noise or ignition noise are not considered to be significant in the presence of fading and therefore neglected.

If we assume geometrically distributed message lengths with messages partitioned in fixed size packets of B bits per packet the total expected wasted time, $\bar{W}(B)$, to transmit a message as a series of packets is given {65} by:

$$\bar{W}(B) = \left(1-q^B\right)^{-1} \left[A + \left(\frac{P_B}{1-P_B}\right)\left(A + \frac{B+b}{R}\right) + \frac{B+b}{R}\right] - \frac{1}{pR} - \frac{b'}{R} \qquad \ldots\ldots (3.26)$$

where:

P_B = probability that a packet transmitted over the channel will have at least one detected error

A = acknowledgment delay associated with each packet (sec)

B = packet data (bits)

b = packet overhead (bits)

b' = unpacketized message overhead (bits)

R = channel transmission rate (bits/sec).

We shall consider a full-duplex mobile terminal with an acknowledgment message equal in length to the packet overhead, b, bits. Then,

$$A = \frac{b}{B} \qquad \ldots\ldots (3.27)$$

Denoting by T the total packet duration (i.e $T = (B+b)/R$) and equating the unpacketized message overhead with the individual packet overhead (i.e $b' = b$) we can rewrite Eqn. (3.26) as follows:

$$\bar{W}(B) = \left(1-q^{B}\right)^{-1} \left[\frac{b}{R} + \left(\frac{P_B}{1-P_B}\right)\left(\frac{b}{R} + T\right) + T\right] - \frac{1}{pR} - \frac{b}{R} \qquad \ldots\ldots (3.28)$$

We seek the optimal packet size, B_o, such that

$$\bar{W}(B_o) = \min\{\bar{W}(B)\} \qquad \ldots\ldots (3.29)$$

Differentiating $\bar{W}(B)$ with respect to B, denoting the first derivative of P_B by P_B' and equating it to zero gives:

$$\left.\frac{\partial W(B)}{\partial B}\right|_{B=B_o} = X(B_o) - Y(B_o) = 0 \qquad \ldots\ldots (3.30)$$

where

$$X(B) = (q^{-B}-1)\left[\frac{1}{b} + \left(1 + \frac{RT}{b}\right)\left(\frac{P_B'}{1-P_B}\right)\right] \qquad \ldots\ldots (3.31)$$

and

$$Y(B) = -\left(1 + \frac{RT}{b}\right) \cdot \ln(q) \qquad \ldots\ldots (3.32)$$

We now recall, from Section 3.3, that the packet error rate was obtained as:

$$P_B = 1 - \exp\{-(\rho + f_d\, T\sqrt{2\pi\rho})\} \qquad \ldots\ldots (3.33)$$

Hence P_B' , its first derivative with respect to the packet size B will be given by:

$$P_B' = \frac{f_d\sqrt{2\pi\rho}}{R}\ \exp\{-(\rho + f_d\, T\sqrt{2\pi\rho})\} \qquad \ldots\ldots (3.34)$$

Substituting P_B and P_B' in Eqn. (3.31) we obtain:

$$X(B) = (q^{-B}-1)\left[\frac{1}{b} + \left(1 + \frac{B+b}{b}\right) \cdot \frac{f_d\sqrt{2\pi\rho}}{R}\right] \qquad \ldots\ldots (3.35)$$

The optimal packet size, B_o, can be obtained by solving the following equation for the packet size, B_o:

$$(2b+B_o)\ \ln(q) + (q^{-B_o} - 1)\left[1 + \frac{f_d\sqrt{2\pi\rho}}{R}\ (2b+B_o)\right] = 0 \qquad \ldots\ldots (3.36)$$

and the corresponding minimum time $\bar{W}(B_o)$ can be obtained by substituting B_o into Eqn. (3.28).

3.5.2 Numerical Results

The optimal packet size for stop-and-wait transmission strategy is computed by solving Eqn. (3.36) numerically using an iterative method. We terminate the iteration when the improvement on $\bar{W}(B)$ for a new value of B is less than 10^{-4} seconds. In all the results presented here, the carrier frequency, the vehicle speed and the overhead bits were kept constant at 850 MHz, 30 mi/hr and 32 bits, respectively.

Figure 3.6 shows the optimal packet size vs bit rate for different signal-to-noise ratios and average message lengths. It is shown in the figure that the optimal packet length increases for higher signal-to-noise ratio, higher bit rate or longer messages. For example, for average message length of 10,000 bits and for 40 dB signal-to-noise ratio, B_o varies from 240 bits at 1 kbps up to 750 bits at 16 kbps, while the corresponding range for 1000 bits message length is 190 bits to 330 bits. Another important observation is that for relatively short messages, B_o exhibits a minor change for a wide range of bit rates (R>4 kbps). Figure 3.7 illustrates another important observation, that is the optimal packet length is almost independent of the message length (for $\bar{\ell} \gtrsim 4000$ bits) for low rates and/or low signal-to-noise ratios.

Figures 3.8 and 3.9 illustrate the minimum wasted time as a function of bit rate, signal-to-noise ratio and average message length. This minimum wasted time is achieved only when the optimal packet length is chosen. For all other packet lengths the time will be larger as indicated in Figures 3.10 and 3.11.

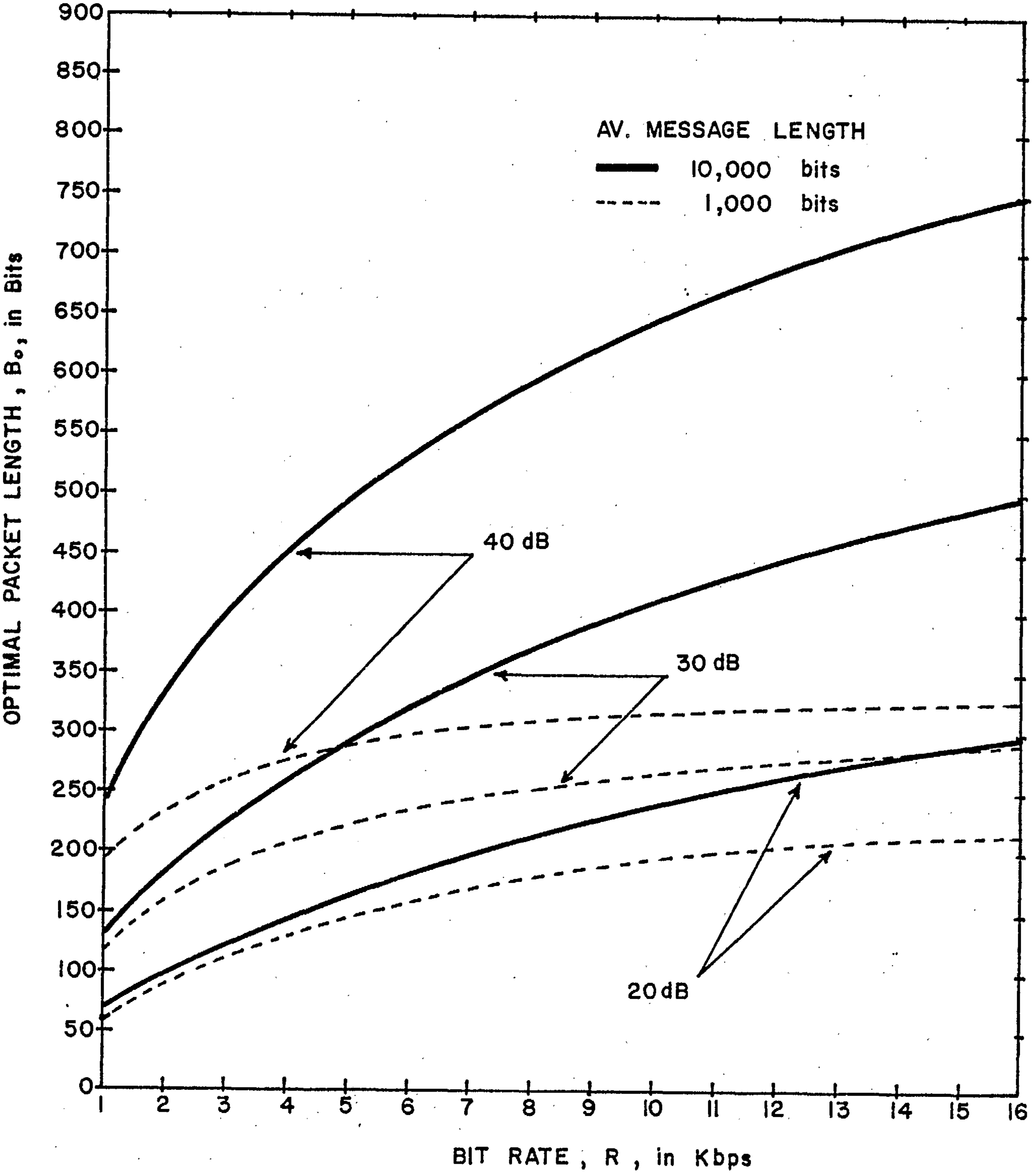


Figure 3.6 OPTIMAL PACKET LENGTH vs BIT RATE FOR THREE SIGNAL TO NOISE RATIOS

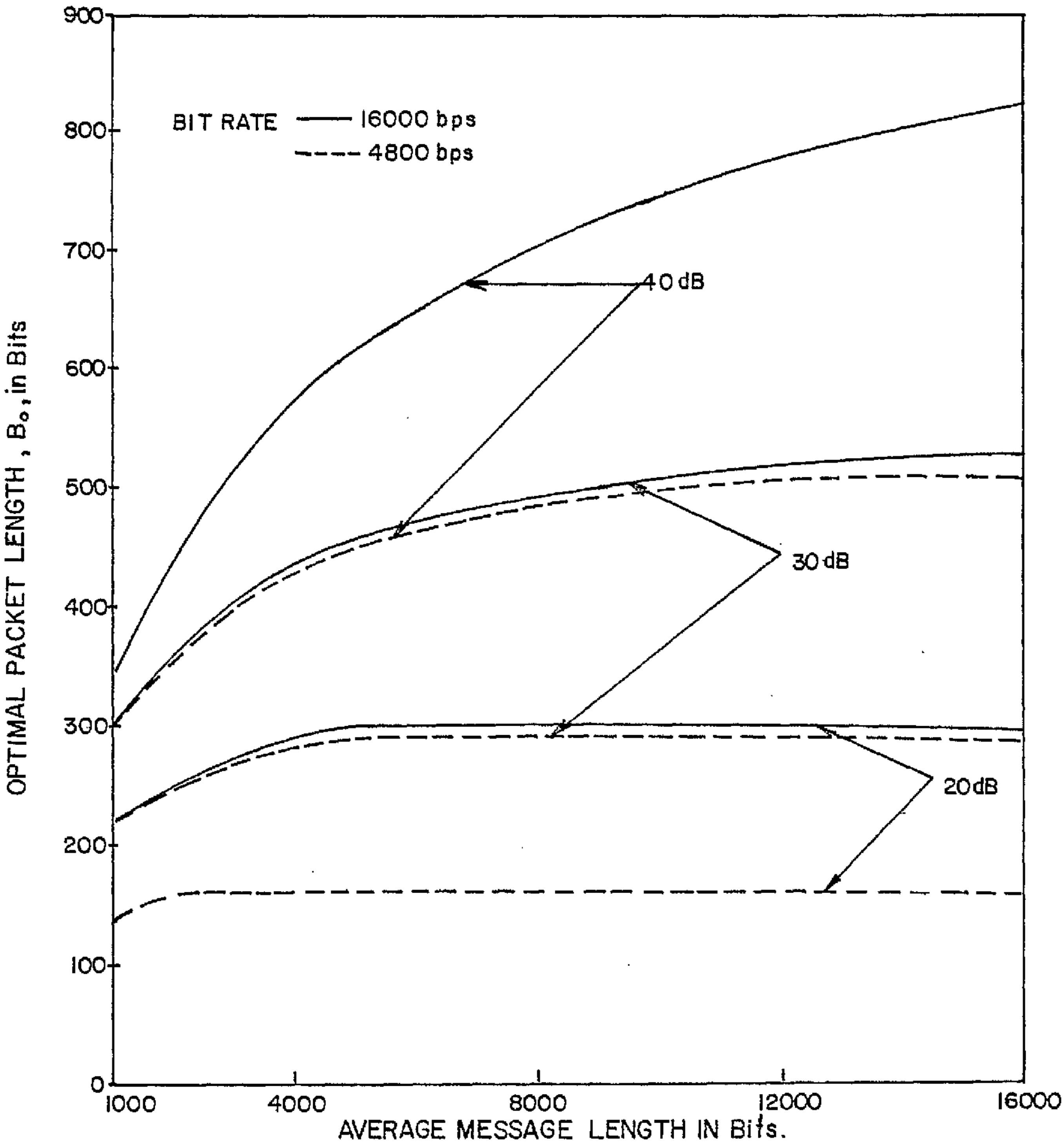


Figure 3.7 OPTIMAL PACKET LENGTH vs AVERAGE MESSAGE LENGTH WITH BIT RATE AND SIGNAL TO NOISE RATIO AS PARAMETERS

The above analysis has provided a means by which the optimal packet length for a fading mobile channel can be evaluated. We have shown the effects of various system parameters such as channel bit rate, relative signal level, and average message length. Other parameters such as carrier frequency were not explicitly considered though they are imbedded in the equations. Curves similar to those given in Figures 3.6-3.11 would be obtained if we were to consider the carrier frequency as a parameter.

3.6 Summary of Results and Discussion

In this chapter we have investigated the impact of fading on the transmission of digital data in packet form. We have obtained a simple but approximate expression for the packet error rate under fading conditions which we have used to study the throughput degradation of packet radio random access schemes and to determine the optimal packet length for base to mobile transmissions.

The numerical results presented clearly indicate that signal level variations resulting from fading can contribute in a substantial manner to lessen the performance of mobile data transmission systems. One possible means of combating the large signal-to-noise ratio fluctuations that accompany signal variations is to use frequency diversity, time diversity or space diversity. Space diversity as indicated by Jakes {49} appears to be one of the most favoured techniques to solve the fading problem. Essentially space diversity is based on the fact that the probability of two or more transmission paths becoming unreliable at the same time, is considerably less than the corresponding probability for an individual path.

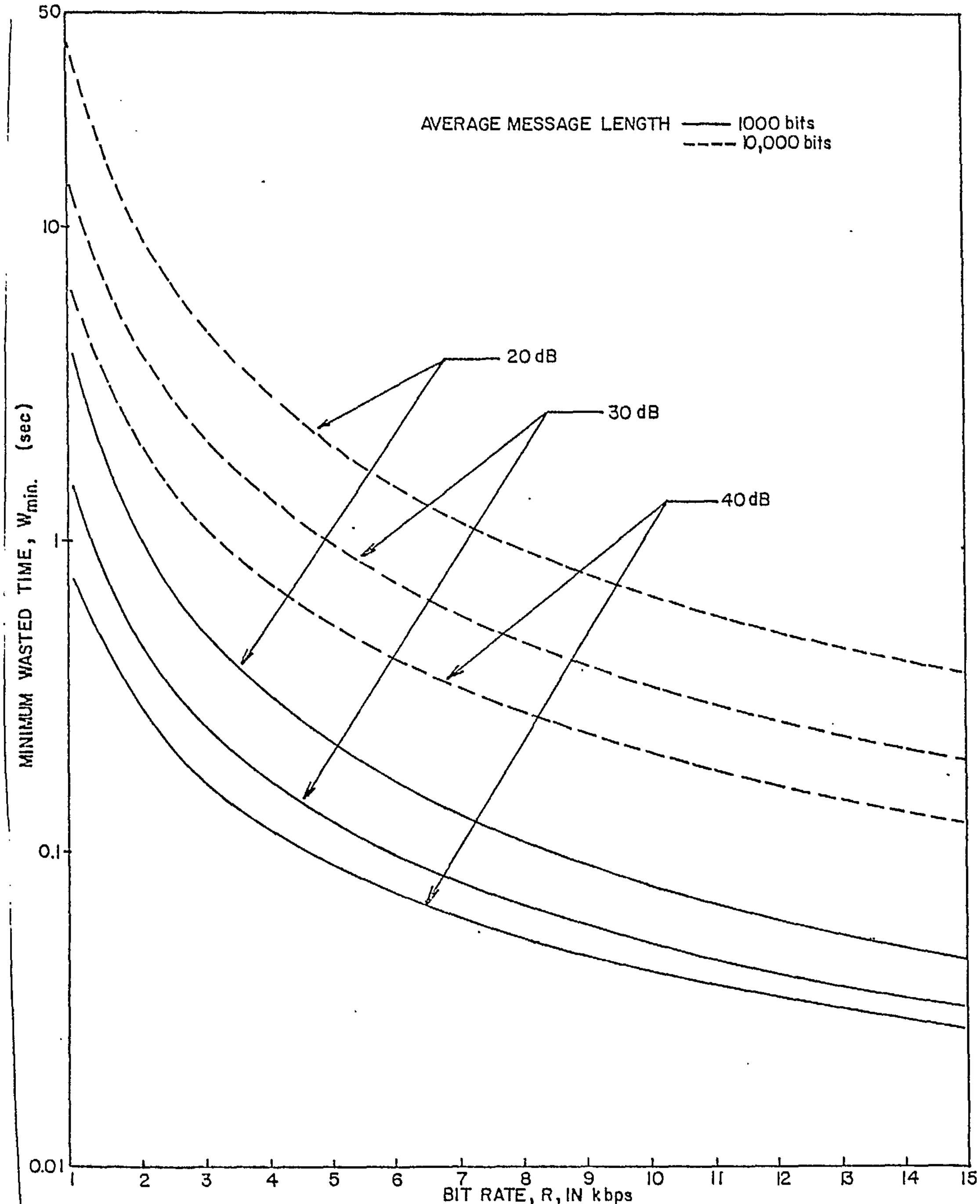


Figure 3.8 MINIMUM WASTED TIME vs BIT RATE WITH AVERAGE MESSAGE LENGTHS AND SIGNAL TO NOISE RATIO AS PARAMETERS

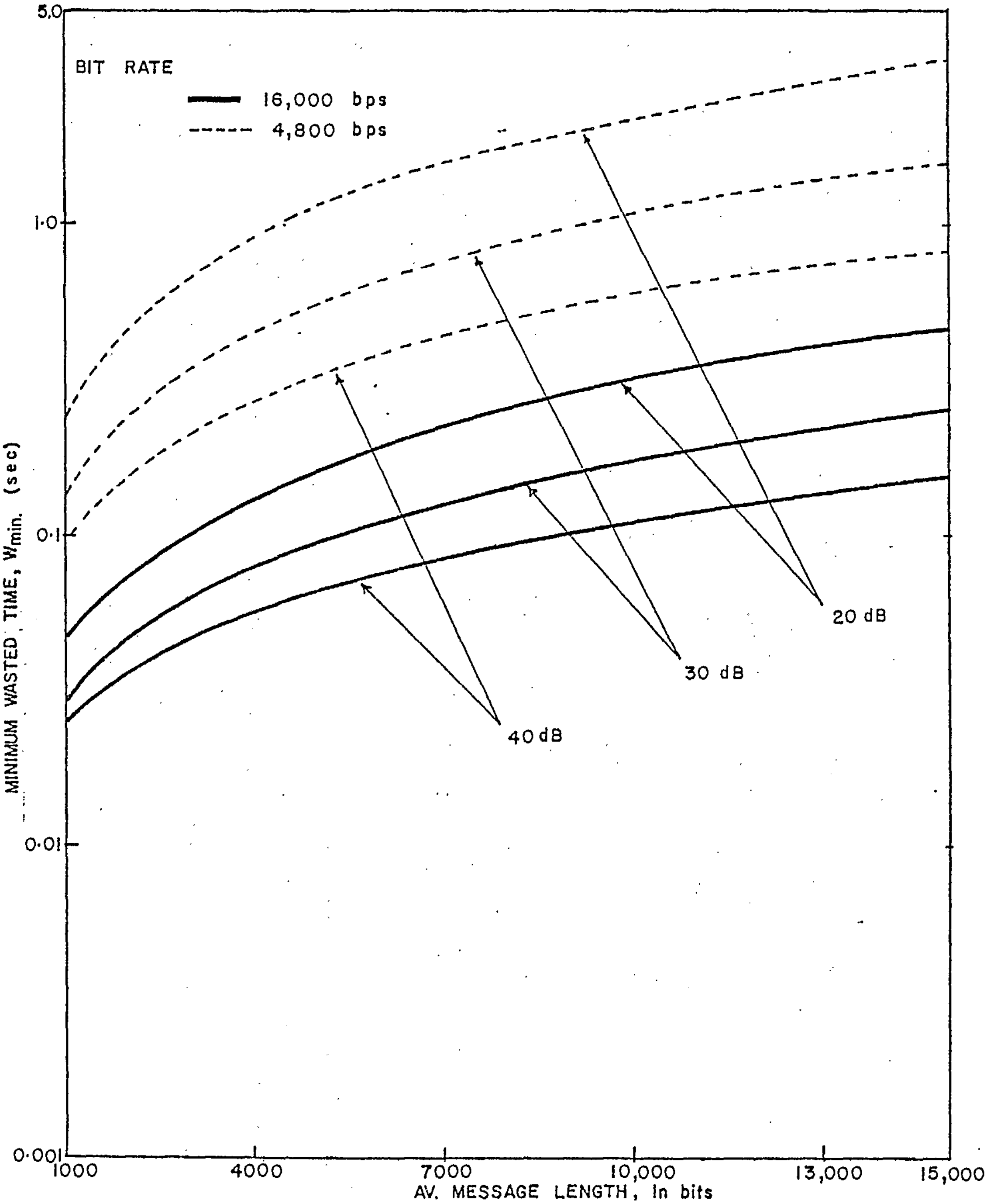


Figure 3.9 MINIMUM WASTED TIME vs AV. MESSAGE LENGTH WITH BIT RATE AND SIGNAL TO NOISE RATIO AS PARAMETERS

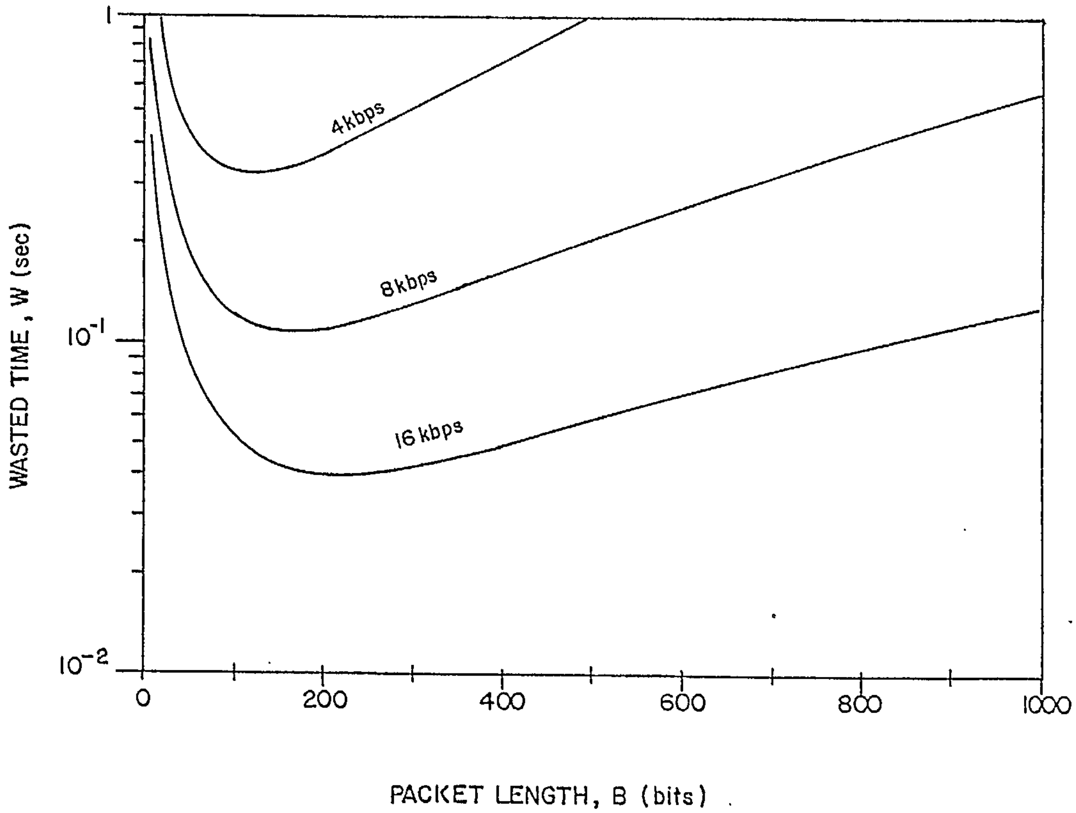


Figure 3.10 WASTED TIME vs PACKET LENGTH
(S/N = 20 dB, $\overline{\ell}$ = 1000 bits.)

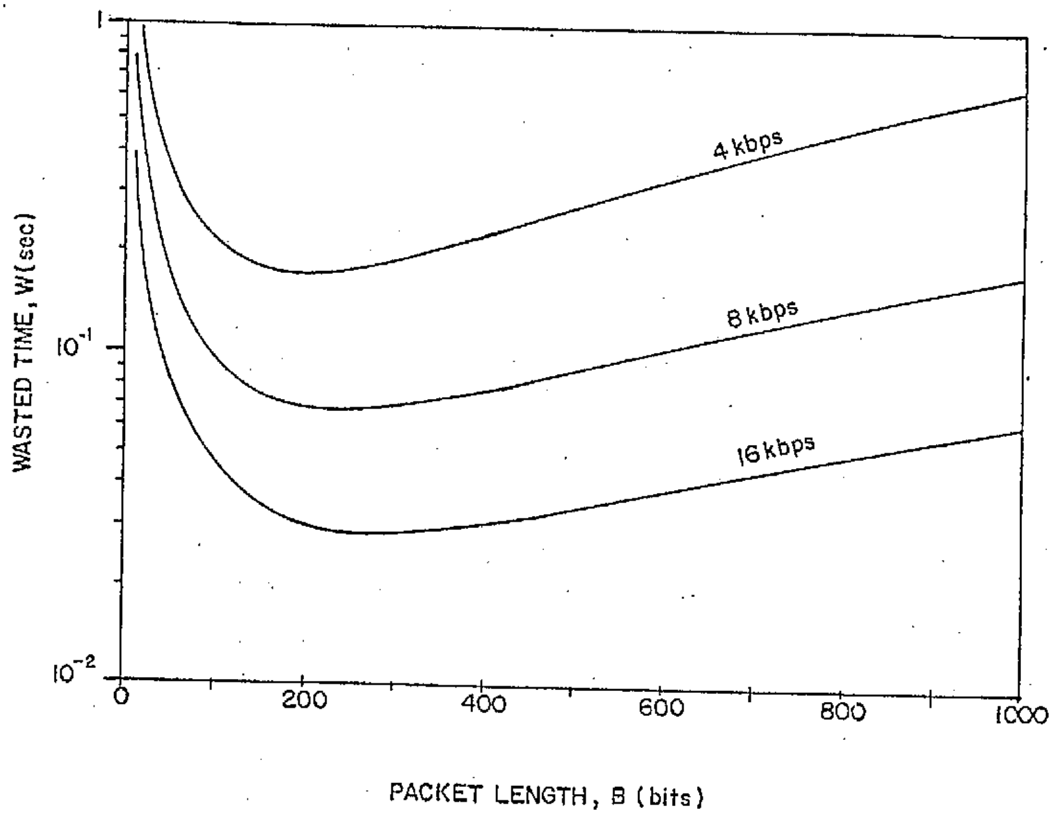
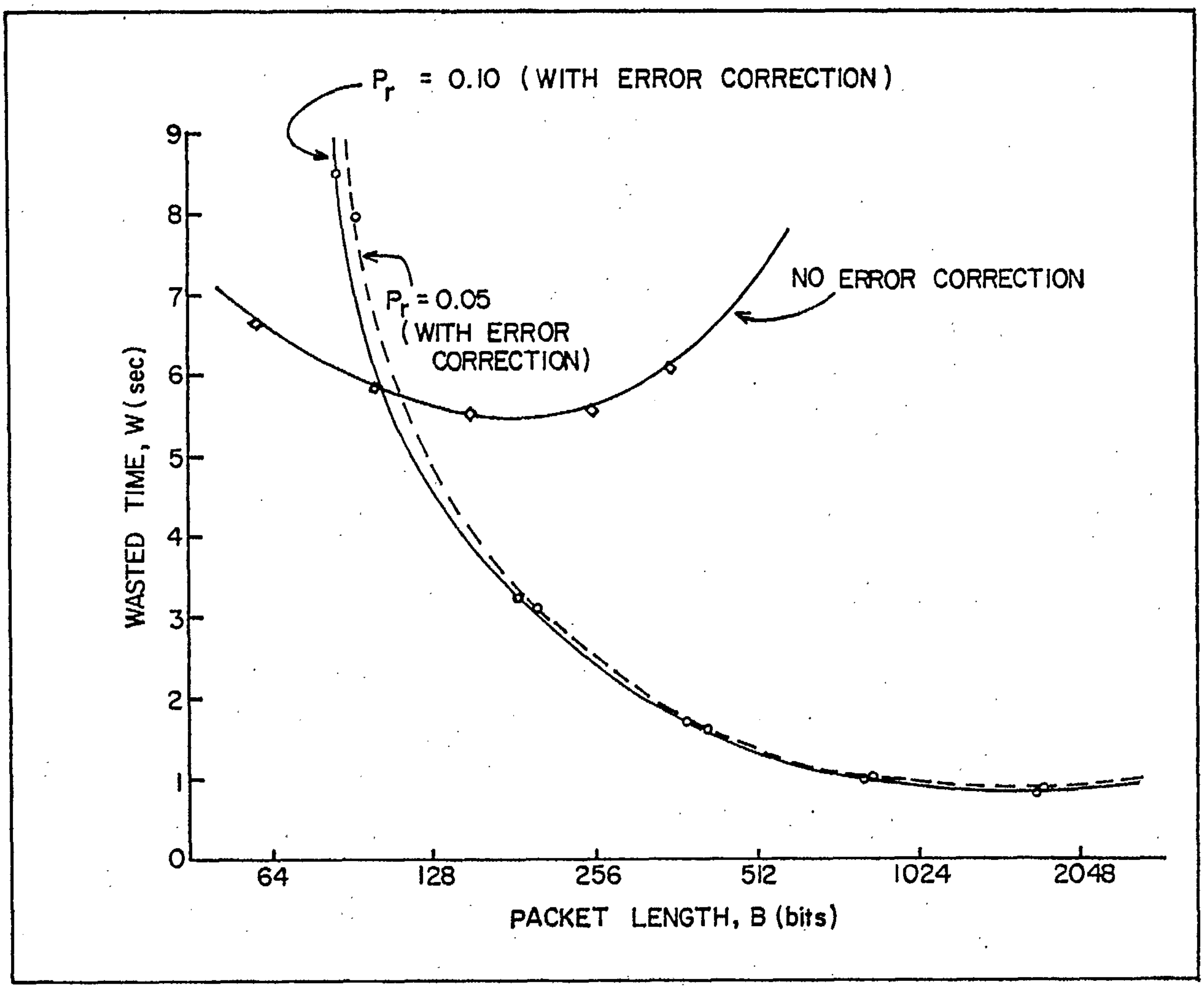


Figure 3.11 WASTED TIME vs PACKET LENGTH
(S/N = 30 dB, $\bar{x}$ = 1000 bits)

Alternatives to diversity can take various forms. Cavers {68} suggests that the transmission of information can be made to vary according to the signal-to-noise ratio at the receiver. Kadokawa {69} has successfully extended the classical Lyncompex concepts to VHF land mobile systems. Mabey {56} suggests that bit interleaving can be applied to distribute the errors randomly. This entails that packets are temporarily stored and the bits are interleaved before transmission. Hence a burst of errors due to fading will tend to place single bit errors in each of a number of packets, thereby transforming a burst error channel into a random error channel.

The gains achieved by diversity techniques and bit interleaving can be further increased by the use of forward error correction which can take various forms. Mabey {56} conducted an evaluation of forward error correction codes, by measuring the error distribution for transmissions at 462 MHz using data rates of 1200 bps and 4800 bps, and concluded that random error correcting codes performs as well if not better than burst error correcting codes. Mahmoud {70} using the fading simulation program discussed in Chapter 2, obtained curves giving the probability of N errors for packets consisting of n bits. Based on these results he further showed how, by using BCH random error correcting codes, the expected waisted wasted times due to packet repetitions and packet overhead, could be decreased. Figure 3.12 illustrates the type of gains that can be achieved by using forward error correction. Clearly at the expense of some additional overhead, the optimal packet length can be considerably increased.



$P_b = 10^{-2}$ = AVERAGE BIT ERROR RATE

R = 4000 bits/sec.

$\bar{\ell}$ = 2000 bits

A = 0.2 sec

b = 30 bits

P_r = PROBABILITY OF RETRANSMISSION

f_d = 25.5 Hz

Figure 3.12 WASTED TIME vs PACKET LENGTH WITH AND WITHOUT ERROR CORRECTION

CHAPTER 4

NEW ARCHITECTURES FOR LAND-MOBILE SYSTEMS

4.1 Introduction

As stated in Chapter 1, the frequency spectrum assigned to land mobile communication systems is gradually becoming congested. Attempts to increase the spectrum efficiency of traditional analog land mobile systems have taken various forms ranging from the provision of some degree of channel assignment automation to channel trunking. In parallel with the development of analog systems, a gradual shift towards mobile digital systems seems to be taking place. The need to provide privacy and security, aided with the developments in speech processing technology has led to the development of mobile systems where digitized voice is encrypted {71}, {72}. These developments have motivated the study of newer digital modulation schemes that attempt to confine the radiated bandwidth of digital voice within the spectral limits of the analog voice channel (25 kHz or 30 kHz). Examples of such studies can be found in de Jager {73} and Hirade {74}. Clearly, such high speed modulation schemes are useful not only for voice based communication systems, but also for data communication systems. Moreover, it is now becoming possible to envisage a mobile set that provides the user with a dual capability of digital voice and data.

This chapter is concerned with the analysis of new architectures for digital mobile systems. In particular, we discuss in Section 4.3 one possible configuration for a digital speech mobile system where fixed length voice packets are transmitted over the communications channel(s) using a variation of the NPCSMA protocol. In Section 4.4, we study a different architecture in which digital speech (or even analog speech) is not packetized. Here it is proposed that the channel idle periods, either within two consecutive calls or within a given voice call, can be used by a population of data users.

Prior to discussing these new system architectures, we will present in the following section a summary of the issues relevant to packetized speech networks.

4.2 Packet Speech Issues

Clearly the most important issue in an interactive man-to-man voice conversation is the overall quality of the speech signal as subjectively perceived by the end users. In conventional land mobile communications the quality of the voice communications as impaired by channel noise is typically "measured" in terms of what is referred to as the Articulation Score which can be related to the signal-to-noise ratio.

The Articulation Score (AS) is a measure of the percentage of words or syllables that are correctly and intelligibly received over a communication channel. In conducting AS tests, the subjects and words are chosen so that only the parameters corrupting the radio channel will influence the differences in intelligibility of the received words. In order to avoid using long and tedious tests, some automated tests and mathematical procedures have been devised that give an indication of the AS in a relatively short time. One such procedure gives rise to a measure of channel quality referred to as the Articulation Index {75}.

In addition to the AS and the AI, other methods can be used to evaluate the voice quality. One such method is the result of extensive field experiments with mobile systems, where the Circuit Merit (CM) or required voice quality is given in terms of the speech-to-noise ratios. In Table 4.1 we show typical values of the speech-to-noise ratios required to achieve a certain circuit merit.

In cases where speech is digitally encoded the problem of assessing voice quality becomes more involved, because of the interdependence between the quality related parameters of the speech signal, the encoding schemes and the noise environment (additive and multiplicative noise). So far, as mentioned by Jayant {76}, the use of the signal-to-noise as a speech quality performance measure has proved to be inadequate. Hence most researchers in the field use various kinds of subjective tests.

Melnick {77} provides curves that show how word intelligibility is related to the channel bit error rate (random errors only), for various encoding rates using variable slope delta modulation. His intelligibility test, known as the Modified Rhyme Test (MRT), consists in asking the participants to select from six possible word choices, the one that is nearest to the transmitted word. As seen in the curve reproduced as Figure 4.1 a word intelligibility of about 80% is satisfied at all encoding rates provided that bit error rate does not exceed 7%. It should be emphasized however that for many applications a low word intelligibility score is sufficient to ensure a perfect conversation intelligibility. This being the case one can conclude that there is no need to encode speech at rates higher than 7.2 kbps provided, the bit error rate does not exceed 10%.

CIRCUIT MERIT	DEFINITION	SPEECH TO NOISE RATIO dB NOMINAL VALUE	RANGE
CM5	PERFECTLY READABLE, NEGLIGIBLE NOISE	- -	ABOVE 30
CM4	PERFECTLY READABLE BUT WITH NOTICEABLE NOISE	22	16 TO 30
CM3	READABLE WITH ONLY OCCASIONAL REPETITION (COMMERCIAL)	12	9 TO 16
CM2	READABLE WITH DIFFICULTY REQUIRES FREQUENT REPETITION (NONCOMMERCIAL)	7	5 TO 9
CM 1	UNUSABLE, PRESENCE OF SPEECH BARELY DISCERNIBLE	- -	BELOW 5

Table 4.1 TYPICAL SPEECH TO NOISE RATIOS FOR INDICATED CIRCUIT MERITS IN NOISE LIMITED SYSTEMS

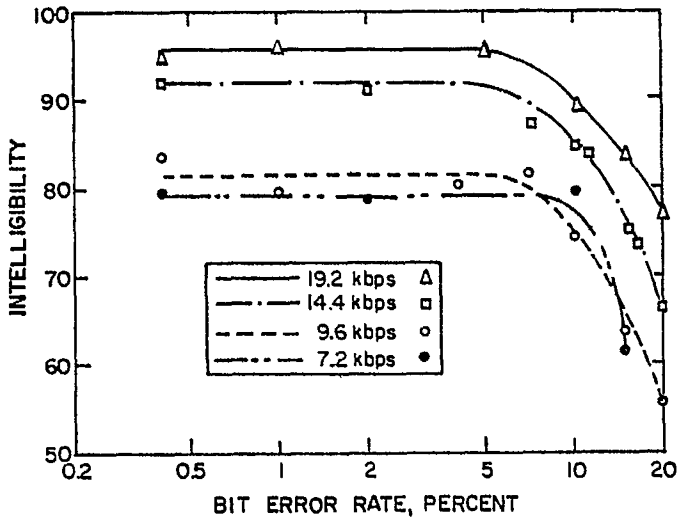


Figure 4.1 INTELLIGIBILITY PERFORMANCE OF VARIABLE SLOPE DELTA MODULATION

Recently, Dhadesugoor et al {35} reported on an extensive set of experiments designed to assess the quality of continuous and packetized speech under various conditions. In particular the performance of the Song Mode Voice Digital Adaptive Delta Modulation (SVADM) was compared to the Harris and Motorola CVSD. It was found that SVADM outperforms the other two techniques and has word intelligibilities of 99% at 16 kbps and 90% at 9.6 kbps in the absence of channel errors. In the presence of randomly distributed channel errors the voice ceased to be intelligible at 9.6 kbps when the error rate exceed 10^{-2}.

In packet transmission of speech where all silent intervals within and between sentences are discarded the perceptual quality of voice will still depend on the encoding technique, the digitization rate and the distribution of channel errors. However protocol features and other network impairments can affect, not so much the quality of the reconstructed speech, but rather its psychological acceptance. In general the important issues in packet speech networks are related to:

* Packet end-to-end delay
* Speech continuity
* Background noise
* Packet length
* Packet loss rates

4.2.1 Packet end-to-end delay

The first element of delay in the overall end-to-end delay is the so-called packetization delay or time required to format a packet. If for instance the packet size is fixed at 800 bits and speech is being digitized using CVSD at 16 kbps, it takes exactly 800/16000=0.05 sec to format a packet. Following the packetization phase, a header is attached to the packet at which point in time the transmission process can begin. Depending on the specific channel access protocol being used the packet might be immediately transmitted or might join the terminal buffer thereby encountering a queueing delay.

Clearly, both the buffer size and the buffer management scheme must be tailored to the access protocol and channel activity in order to ensure that queueing delays are kept within acceptable bounds. After some queueing delay the packet is eventually transmitted at a rate which exceeds the digitization rate. Assuming again a packet size of 800 bits a packet header of 16 bits and a transmission rate of 40 kbps it will take 816/40000=0.0204 sec to transmit the packet.

On its way to the destination node, the packet will be delayed at each intermediate node by varying amounts which depend upon the state of congestion of the network. In the case of store-and-forward networks, the selection of the routing strategy that minimizes the queueing delays of voice packets represents a crucial design problem. In land mobile networks the routing problem is more straightforward since from source to destination the packet will typically traverse one (base station) or three (satellite receiver, central station, satellite transmitter) intermediate nodes.

Upon arrival to the receiver's buffer the depacketization phase is initiated and a hopefully identical replica of the original voice signal is delivered to the receiver. Clearly the depacketization time will be equal to the packetization time. In addition to the time required to depacketize the incoming speech it might be desirable in some circumstances to introduce an artificial element of delay in order to preserve speech continuity.

The overall end-to-end delay which is given by the summation of the above partial delays, must clearly be kept within acceptable psychological bounds. Some empirical tests conducted at the Bell Laboratories indicate that overall delays of one second or more were considered to be unacceptable. Also as indicated by the experiments being carried out over SATNET {21}, one-way delays between any pair of hosts, in excess of one second did not contribute significantly to the degradation of the voice conversation. However as shown by Klemmer {78} a "commercially acceptable" overall delay should not exceed 200 msec, consequently such value should represent a design objective for land mobile radio systems. Such design objective should not preclude delays to temporarily exceed such value, provided of course that such delays are rare and far apart.

4.2.2 Speech Continuity

Depending on the variability of end-to-end delays and the amount of speech contained in each packet, interruptions in reconstructed speech due to packet losses, silence gaps and packet late arrivals, can render conversations unintelligible. Studies by Flanagan {79} indicate that listeners can normally bridge the gaps between two consecutive speech packets. In addition some latitude exists when silent intervals are recreated by the receiver, since lengthening or shortening within sentence silences by up to ±50% are perceptually acceptable. Clearly such latitude could help alleviate the problem of buffer design.

4.2.3 Background Noise

In general the amount of background noise present in the environment of the speaker influences the performance of the voice encoding algorithm. Furthermore in the context of packet speech where silence gaps are removed, it becomes extremely difficult in the presence of noise, to detect the presence or absence of a talkspurt which entails that high speech interpolation gains might not be achieved. Clearly then, care must be exercised in selecting the most appropriate noise-cancelling microphones or audio pre-processing circuits. To the author's knowledge no studies have been published on the types of background noise present in land-mobile environments.

4.2.4 Packet Length

The selection of the optimal packet length is a very complicated problem because of the large number of interrelated parameters and variables that are functions of the packet size. Minoli {80} derived an expression for the optimal packet length as a function of the transmission rate, speech digitization rate and packet overhead. In his context, optimal means the packet length that minimizes the overall end-to-end delay. The approach of Minoli {80} neglects to take advantage of the fact that the human ear cannot perceive speech losses of up to 30 msec. By selecting such a packet length an increase in overall end-to-noise delay can be balanced against the packet loss rate, which implies that at the receiver end a packet could be discarded to maintain the overall delay below the acceptable threshold, without impairing the voice quality.

4.2.5 Packet Loss Rates

In a packet switched network voice packets can be "lost" for several reasons, including misrouting, excessive network delay, collision with other packets and loss of packet header. Whenever a packet is lost, a gap referred to as a "glitch", will be introduced in the reconstructed speech. To ensure speech continuity such gaps could be filled in by replaying the last arrived packet or alternatively could be left as is if the glitch duration does not exceed the perceptual bridging interval. An important issue to be resolved whenever packets are lost concerns the error detection and correction protocols. Voice packets as opposed to data packets have more stringent delay requirements, which implies that ARQ techniques should not preferably be used for voice packets since they might not lead to an acceptable performance. However in order to minimize the probability of packet loss it might be desirable to use a FEC protocol particularly on that part of the voice packet corresponding to the header. Clearly if no error correction is attempted and the packet header is corrupted by noise, the entire packet would be lost. As shown by Coviello {81} the amount of packet overhead needed to identify each voice packet could be kept under 32 bits, since for N active calls only $\log_2 N$ bits

As we mentioned above the effect of packet losses on the overall speech quality is intimately related to the packet length. Huggins {82} has shown that on one hand the intelligibility decreases to about 10% as the packet size approaches 250 msec, and on the other hand the intelligibility increases to about 80% as the packet size approaches 19 msec. In particular he suggests that packets should contain between 10 and 50 msec of speech. The same author concluded that speech losses as high as 50% can be sustained with marginal degradation if such losses correspond to speech segments of about 19 msec. However it should be clear that the tolerable packet loss rate is dependent upon the redundancy of the digitized voice bit stream. The higher the encoding rate the higher the acceptable loss rate. Recently, Dhadesugoor {35} compared the performance of CVSD and SVADM at two encoding rates namely 16 kbps and 9.6 kbps. Using four packet lengths (2048,1024,512,256) he showed that speech was intelligible at loss rates of 10% for any packet size and any encoding rate.

4.3 Packet Speech for Mobile Radio

One of the earliest attempts to evaluate the traffic carrying capacity of a packet speech mobile radio system is due to Ellershaw {48} who studied two system architectures in particular. His first architecture corresponds to a time division multiplexed system where transmission slots are allocated to speech packets of different users, by the central base station. Hence the system is under the control of the base station whose slot allocation is based upon the correct reception of service requests transmitted to it over a reservation channel accessed on a slotted ALOHA basis. No attempt is made to take advantage of the silence gaps characteristic of voice conversation, hence the system behaves as a circuit switched system where the signalling channel corresponds to the reservation channel. The second architecture suggested by Ellershaw is essentially a variation of the first one, the major difference being that speech talkspurts are formatted into packets of fixed length with requests for channel time transmitted on a contention basis over a signalling channel. By not transmitting silence gaps, he shows that this architecture can support twice as many traffic sources. One of the major weaknesses of this approach consists in assuming that channel bit rates of up to 1 Megabit are possible in a land-mobile environment. However it is well known that only spread spectrum based systems can attain such high bit rates. Indeed frequency selective fades impose a maximum bit rate of the order of 100 kbps for digital modulation schemes in the 850 Mhz frequency band.

In this section we consider an interpolative packet speech land-mobile system where speech packets are not allocated specific transmission time slots but rather gain access to the available FDM channels on a contention basis. Hence there is no need to reserve a channel for signalling purposes.

4.3.1 Network Model

Consider a conventional land mobile system where a large number of speech sources want to exchange voice communication through a finite number of radio channels available at a given base station. If we assume an Erlang B traffic model, the grade of service or blocking probability is given by:

$$P_B = \frac{\rho^M/M!}{\sum_{n=0}^{M} \rho^n/n!} \qquad \text{..... (4.1)}$$

where ρ is the traffic offered to the group of M channels by the population of users. The total traffic carried by these channels is therefore:

$$\rho_c = \rho\left[1-P_B\right] \qquad \text{..... (4.2)}$$

The number of users that can be supported is easily obtained by dividing ρ_c by the traffic load per voice source. If as an example we consider 100 channels, a blocking probability of 0.02 and a load per voice source of 0.02 erlangs, we conclude that approximately 49 users per channel could be supported.

Clearly the same channels can be shared, more efficiently, by (1) taking advantage of the statistical characteristics of voice (2) digitizing the talkspurts and encoding them into packets of fixed size and (3) transmitting these packets at high speed.

As shown in Figure 4.2 we assume that speech packets can use any one of the M available uplink channels. Packets that successfully arrive at the base station will be processed and transmitted over any of the L available downlink channels. We further assume that the channel spacing is 30 kHz and we ignore packet transmission errors due to

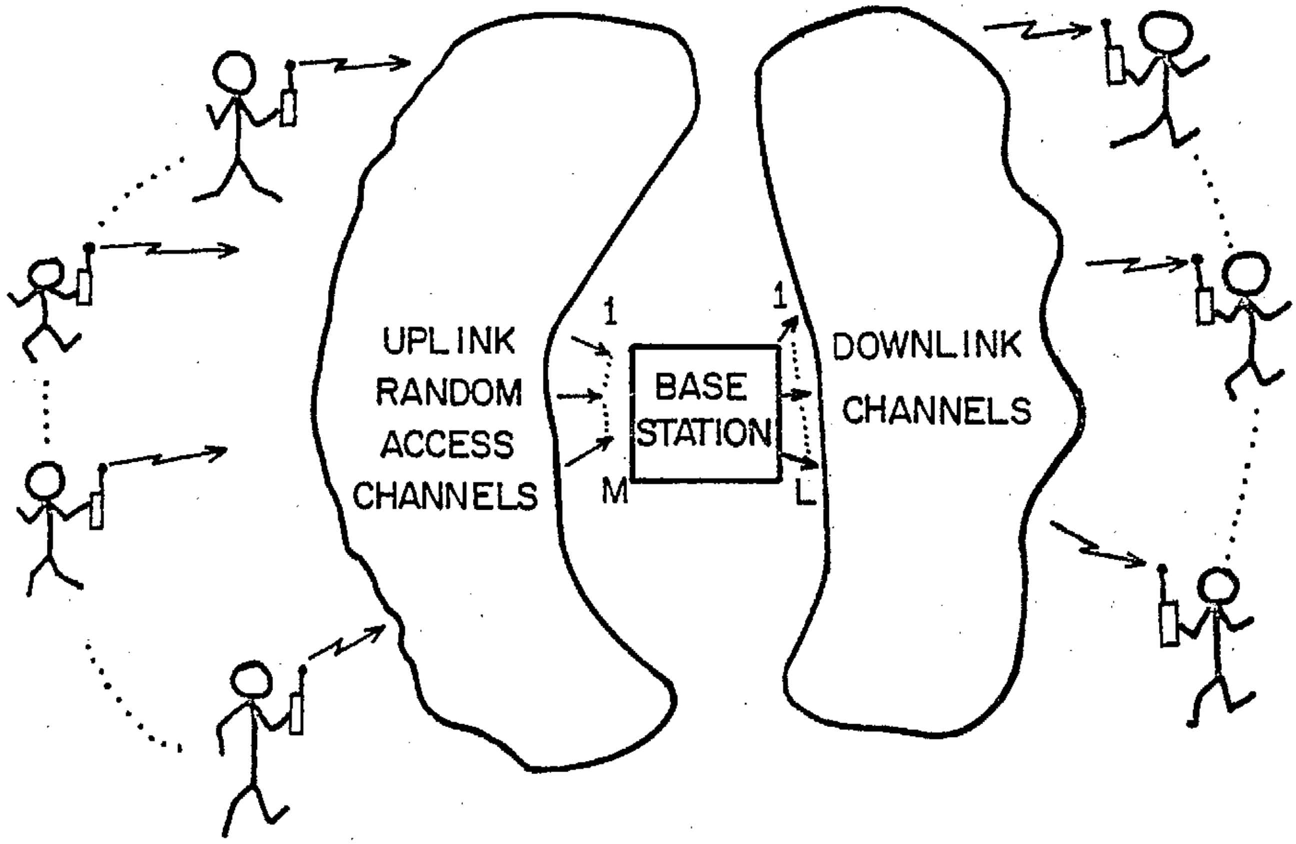


Figure 4.2 NETWORK MODEL FOR PACKET SPEECH

fading, random noise or impulsive noise. The reason why we disregard fading errors is outlined in Chapter 3 where we showed that the capacity degradation of packet radio fading channels can be minimized by ensuring a high average to threshold power ratio. With the specific system parameters discussed in Chapter 3 we concluded that the base station coverage range should be kept under 2 miles. Moreover we assume a space diversity system where we use selection diversity at the base station and switched diversity at the mobile units. To alleviate packet synchronization problems we assume that a Manchester bit encoding format is adopted.

4.3.2 Random Access Model

In Figure 4.3 we show the process of speech detection and packetization that takes place for every talkspurt corresponding to a segment of a conversation between two mobile users. Clearly a very large number of speech sources could share the channel if it were possible to perfectly utilize the pauses in the conversations. It is also clear from the same figure that in such an ideal case the number of speech sources that could share the channel would be a function of the talkspurt length distribution, the pause length distribution, the encoding rate (R), the packet size (B), the overhead (b) and the transmission speed (C).

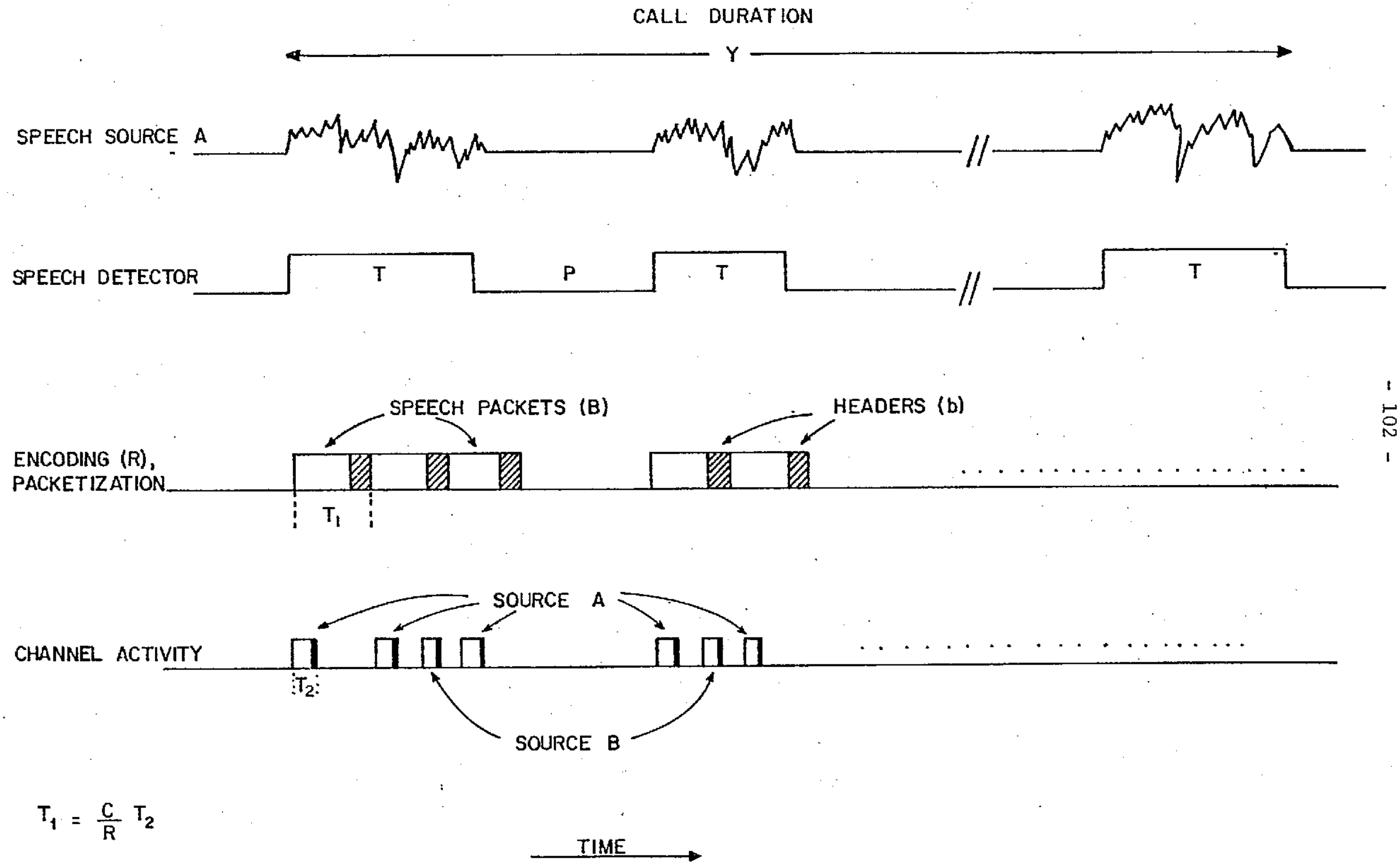


Figure 4.3 PACKET SPEECH MODEL

In practice it is however, virtually impossible to perfectly schedule the activity of the channel without resorting to a centralized control mechanism. A possible alternative is to use a form of distributed control, which simply means that a packet from a given user could be prevented from being transmitted if the channel happened to be busy. If however at some point in time the channel is idle, it is quite possible that no other terminal is in the process of transmitting a voice packet. As described in Appendix B a number of multiple access protocols can be implemented in the context of a mobile channel.

One of these protocols known as the Non-Persistent Carrier Sense Multiple Access (NPCSMA) scheme operates as follows:

- If a terminal has a packet ready for transmission it senses the channel and if the channel is sensed idle, the packet is transmitted. It can however, collide with some other packet during a time window, which is related to the propagation delay between terminals.

- If the channel is sensed busy the terminal does not persist in sending the packet and simply reschedules the transmission of the packet according to some random delay distribution. At this new point in time, the channel is sensed again and the same procedure is repeated.

- If a terminal learns that a packet collided with some other packet, it reattempts a retransmission according to the above procedure.

Defining by S the average number of packets generated per packet transmission time and by G the average number of new and rescheduled packets per packet transmission time it can be shown {83} that S and G are related by:

$$S = \frac{G\ e^{-\delta G}}{G(1+2\delta) + e^{-\delta G}} \qquad \text{..... (4.3)}$$

where δ, the normalized propagation delay is given by:

$$\delta = \frac{\tau}{T_p} \qquad \text{..... (4.4)}$$

where τ is the one-way propagation delay between any two terminals and T_p is the packet transmission time. In Figure 4.4 we depict the relationship between S and G for various values of δ.

It can also be shown {83} that when a packet is ready for transmission, the probability "θ" that the channel is busy is given by:

$$\theta = \frac{G(1+\delta) - 1 + e^{-\delta G}}{G(1+2\delta) + e^{-\delta G}} \qquad \text{..... (4.5)}$$

The relationship between θ and G is displayed in Figure 4.5, for two values of the normalized propagation delay.

One of the characteristics of the NPCSMA random access scheme is that not all arrivals result in actual transmissions. Some packets will be blocked because of channel unavailability and only those that find the channel idle will actually be transmitted. Following the notation introduced in {83} we will denote by H the actual rate of transmitted packets. Hence we can write:

$$H = G(1-\theta) \qquad \text{..... (4.6)}$$

It is customary to define the probability of successful transmission by the ratio S/G. However, since G includes packets that were not transmitted but were merely blocked prior to transmission, we believe it is more appropriate to define the probability of successful transmission by the ratio S/H. From Eqns. (4.3),(4.5),(4.6) we obtain:

$$\xi = \frac{S}{H} = \frac{e^{-\delta G}}{1+\delta G} \qquad \text{..... (4.7)}$$

Using two values of the normalized propagation delay, we illustrate in Figure 4.6 the relationship between ξ, H and G.

For the case of voice packets we suggest a modification of the NPCSMA protocol, namely we do not attempt to retransmit a packet that has collided. This implies that, in the above equations, we should interpret G as the rate of new and rescheduled packets, H as the rate of transmitted packets and S as the rate of successfully transmitted packets.

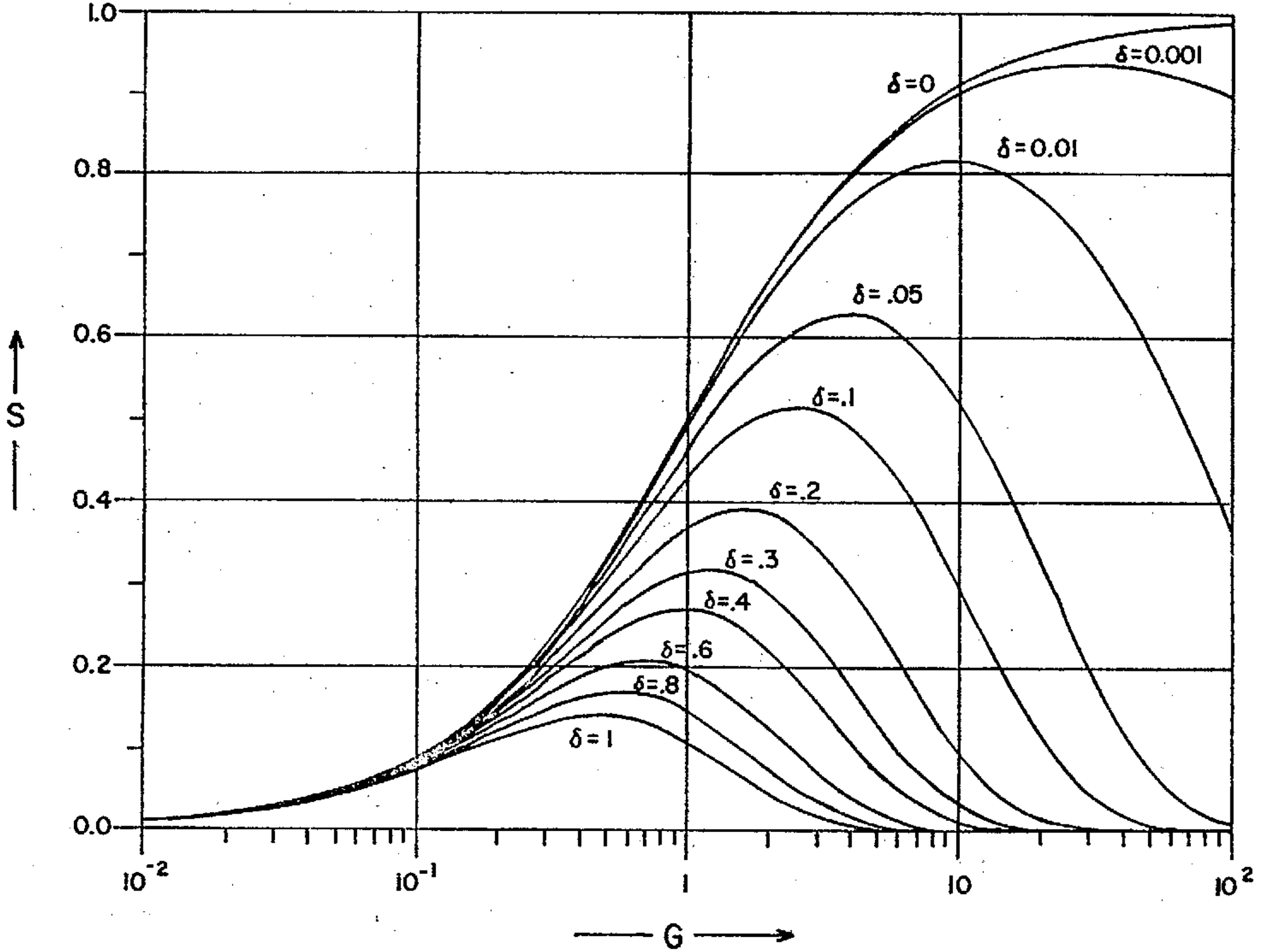


Figure 4.4 TRAFFIC-THROUHPUT CHARACTERISTICS OF NP CSMA

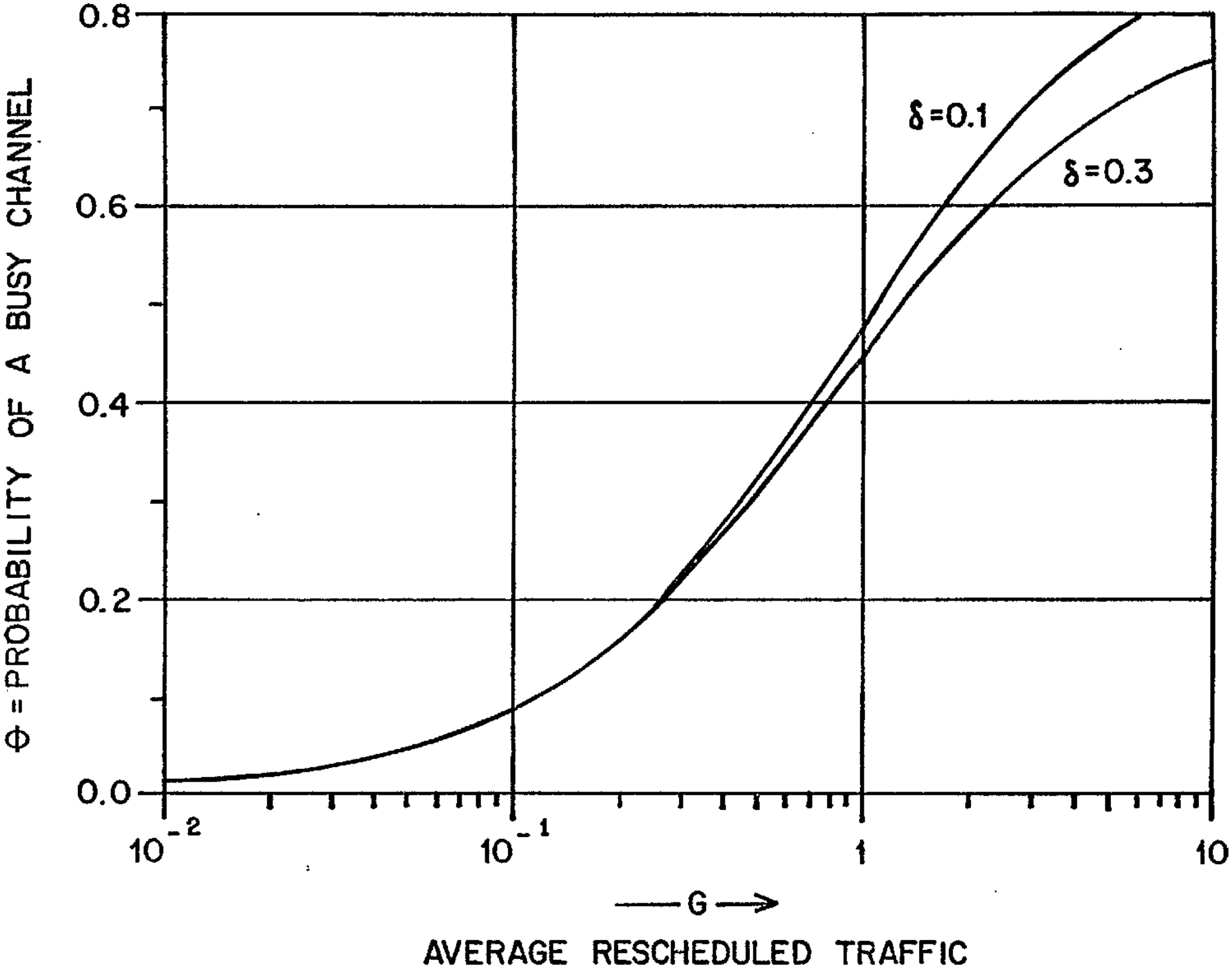


Figure 4.5 PROBABILITY OF A BUSY CHANNEL vs AVERAGE RESCHEDULED TRAFFIC

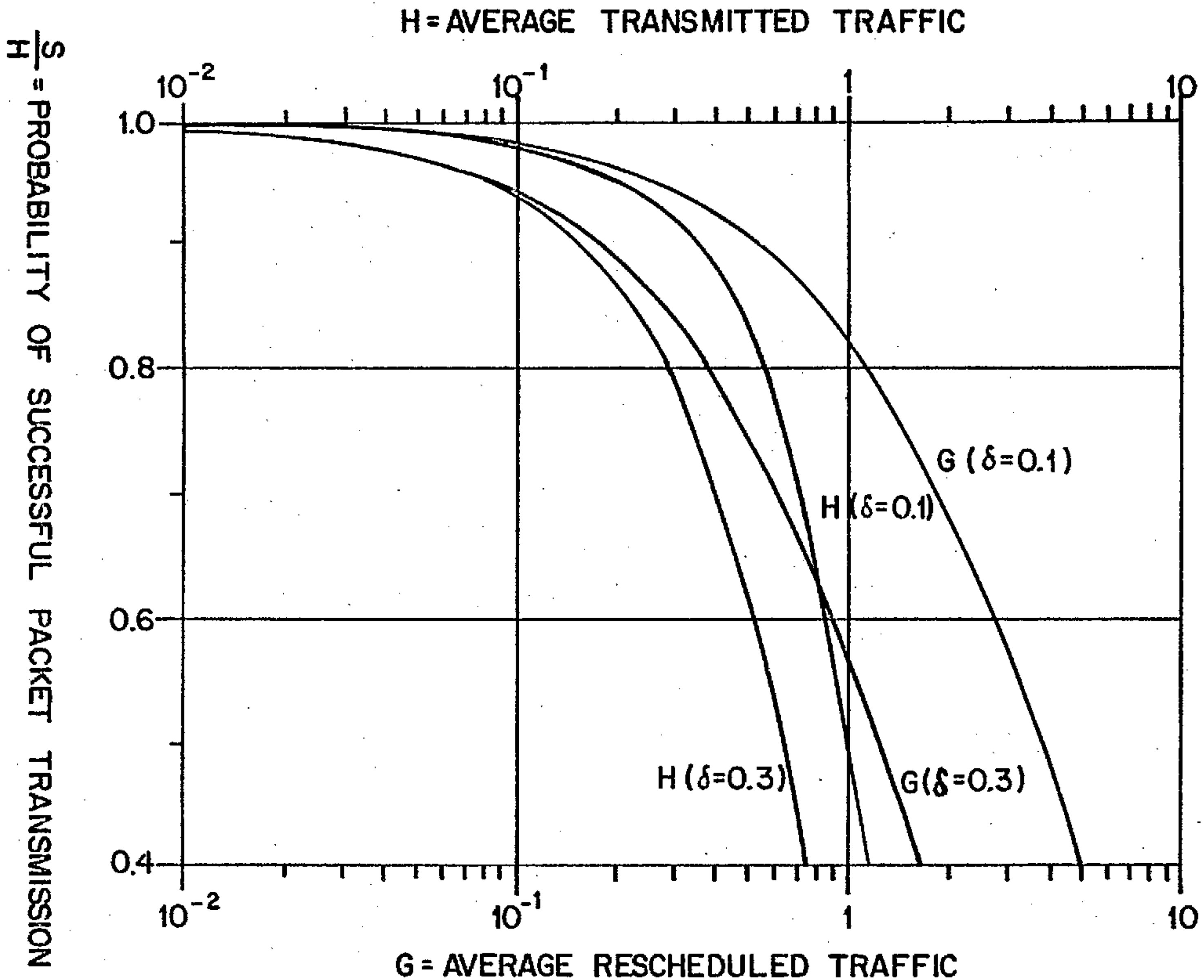


Figure 4.6 SUCCESSFUL PACKET TRANSMISSION vs G AND H

4.3.3 Speech Model

In the previous section we have given expressions for the probability of sensing a busy channel and for the probability of successful packet transmission, in terms of H the normalized offered channel traffic. We must now relate H to the calling patterns of the voice sources as well as to their talkspurt statistical properties.

Consider a sequence of talkspurts and pauses characteristic of the speech pattern of a normal user and assume that the talkspurts as well as the pauses are exponentially distributed {84} with means $\bar{T}$ and $\bar{P}$. Hence the probability density functions for the random variables T and P are given by:

$$f_T(t) = \frac{1}{\bar{T}} \exp\left(-\frac{t}{\bar{T}}\right) \qquad \ldots\ldots (4.8)$$

and

$$f_P(t) = \frac{1}{\bar{P}} \exp\left(-\frac{t}{\bar{P}}\right) \qquad \ldots\ldots (4.9)$$

We assume that the digitally encoded talkspurts are broken up into n packets of length B. If the voice digitization rate is denoted by R, we have:

$$P\left[RT \leq nB\right] = 1 - \exp\left(-\frac{nB}{R\bar{T}}\right) \qquad \ldots\ldots (4.10)$$

Hence, the probability that exactly "n" packets will be needed is:

$$P\left[(n-1)B < RT \leq nB\right] = \exp\left(-\frac{(n-1)B}{RT}\right) - \exp\left(-\frac{nB}{RT}\right) \qquad \text{..... (4.11)}$$

Thus the mean number of packets per digitally encoded talkspurt is obtained as:

$$\bar{n} = \{ 1 - \exp(-\frac{B}{R\bar{T}}) \}^{-1} \approx \frac{R\bar{T}}{B} \qquad \text{..... (4.12)}$$

It is well known that the speech source activity ratio, a, can be defined as the ratio of the average talkspurt duration to the sum of the average talkspurt duration and average pause duration, as follows:

$$a = \frac{\bar{T}}{\bar{T} + \bar{P}} \qquad \text{..... (4.13)}$$

The source activity ratio, typically of the order of 0.4, can also be interpreted as the probability that an active speech source is issuing a talkspurt at some random time.

Since each packet generated during a talkspurt must be identified by a header of size b the time required to transmit a packet is given by:

$$T_p = \frac{B+b}{C} \qquad \text{..... (4.14)}$$

where C denotes the channel transmission rate.

Finally denoting by ρ_i the offered load (in erlangs) per voice source during the busy hour, we obtain the following expression for the normalized offered traffic per source:

$$h = \rho_i \, a \, \frac{R}{C} \cdot \frac{B+b}{B} \qquad \ldots\ldots (4.15)$$

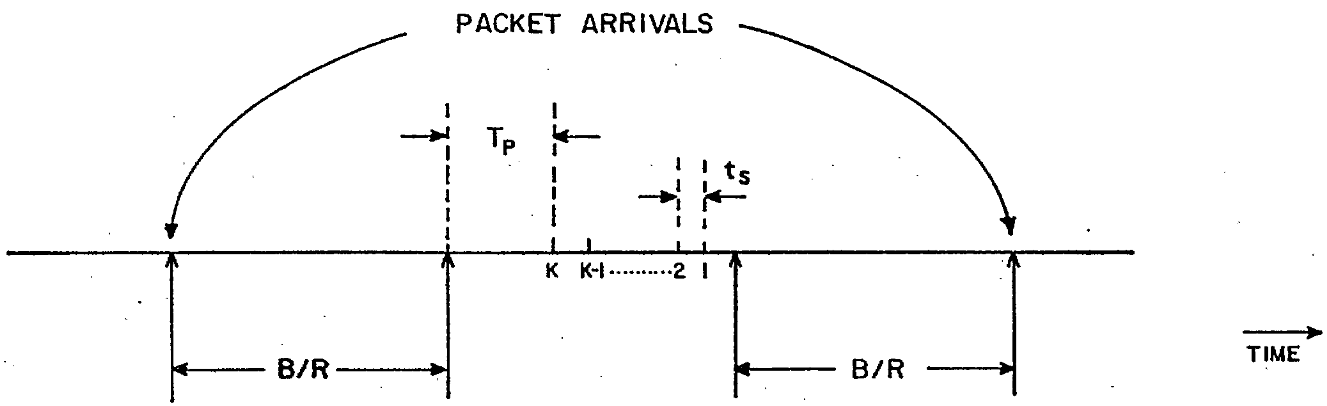
Hence the total traffic offered to each of the M channels by a population of N voice sources is given by:

$$H = \rho_i \, a \, \frac{R}{C} \cdot \frac{B+b}{B} \cdot \frac{N}{M} \qquad \ldots\ldots (4.16)$$

4.3.4 Traffic Model

Consider Figure 4.7 where we see that during a talkspurt, packets "arrive" in a very regular manner. Indeed if each packet contains B bits, there will be a packet arrival every B/R units of time. This inter-packet arrival time corresponds to the so-called packetization delay, that is the time required to form a packet of B bits.

Upon arrival of the first packet of a talkspurt to the buffer of the radio terminal, the process of selecting a channel for transmission, is initiated. Suppose that one of the M available channels is selected. After a period of time t_s, the logic unit within the transceiver will decide whether or not the channel is busy. If that particular channel is found to be idle, a header is attached to the packet currently in the buffer and a packet size B+b is transmitted. If the channel is sensed busy, another channel is selected at random among the available M channels and the process is repeated. Since the terminal has a buffer capable of containing a single packet and since



t_s = SENSING PERIOD

T_P = PACKET TRANSMISSION TIME

B/R = TEMPORAL PACKET LENGTH

Figure 4.7 TRAFFIC MODEL FOR SPEECH PACKET ARRIVALS the interarrival time of two consecutive packets is equal to $\frac{B}{R}$, a packet currently in the buffer that does not find an idle channel within this period of time is discarded.

More specifically if we divide the period of time $(B/R - T_p)$ into k slots each equal to the sensing time t_s we will discard a packet if after k sensing points an idle channel was not found.

Since, θ, the probability that a given channel is busy, is the same for all channels, the probability α of being forced to discard the packet is given by:

$$\alpha = \theta^k \qquad k=1,2,.... \qquad ..... (4.17)$$

Naturally the probability $(1-\alpha)$ of being able to transmit the packet within the allowable time period is given by:

$$(1-\alpha) = \sum_{i=1}^{i=k} \theta^{i-1}(1-\theta) = 1-\theta^k \qquad ..... (4.18)$$

An important question can now be raised concerning the number of discarded and collided packets. Since we do not propose that collided packets be retransmitted, we would like to determine the fraction of lost packets (discarded and collided). A given packet within a talk-spurt can be discarded with probability α, and a transmitted packet can collide with probability $1-\xi$, hence the fraction of lost packets in the uplink channel is given by:

$$\emptyset_u = \alpha + (1-\alpha)\ (1-\xi) \qquad ..... (4.19)$$

To determine the fraction of lost packets on the downlink channels we assume that packets arrive at the base station via M channels. Denoting by S the probability of a successful arrival over any one of these channels, if there are L downlink channels (L<M) the fraction of lost packets, $\emptyset_d$, is obtained from:

$$\emptyset_d = \frac{\binom{M}{L}(S)^L}{\sum_{i=0}^{L}\binom{M}{i}(S)^i} \qquad \text{..... (4.20)}$$

Figures 4.8 and 4.9 illustrate the relationship between $\emptyset_d$ and S for M=50 and M=100. Clearly by selecting the appropriate number of downlink channels any constraint on $\emptyset_d$ can be met.

To obtain Eqn. (4.20) we have assumed that no buffering is provided at the base station. This is a reasonable assumption since it is well known that large delays in packet voice transmission will be intolerable. However if there is a need to further decrease $\emptyset_d$, this can be achieved by increasing the overall end-to-end delay i.e by providing buffering at the base station. It should also be emphasized that so far we have delt with what we could call "incestuous" traffic. Indeed we have assumed that communications take place between mobile terminals, and have excluded communications coming in from or addressed to land terminals.

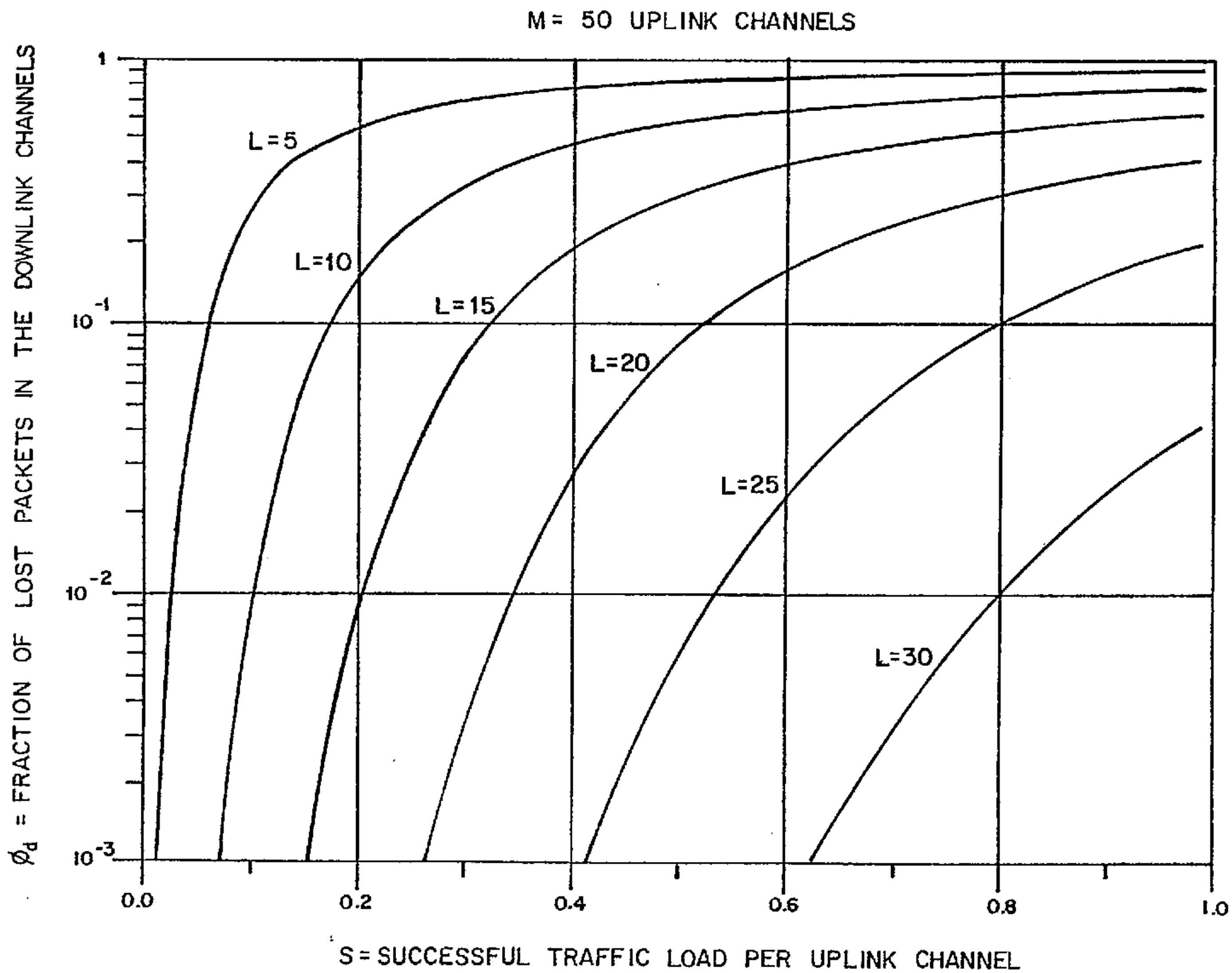


Figure 4.8 FRACTION OF DOWNLINK LOST PACKETS vs AVERAGE SUCCESSFUL TRAFFIC (M=50)

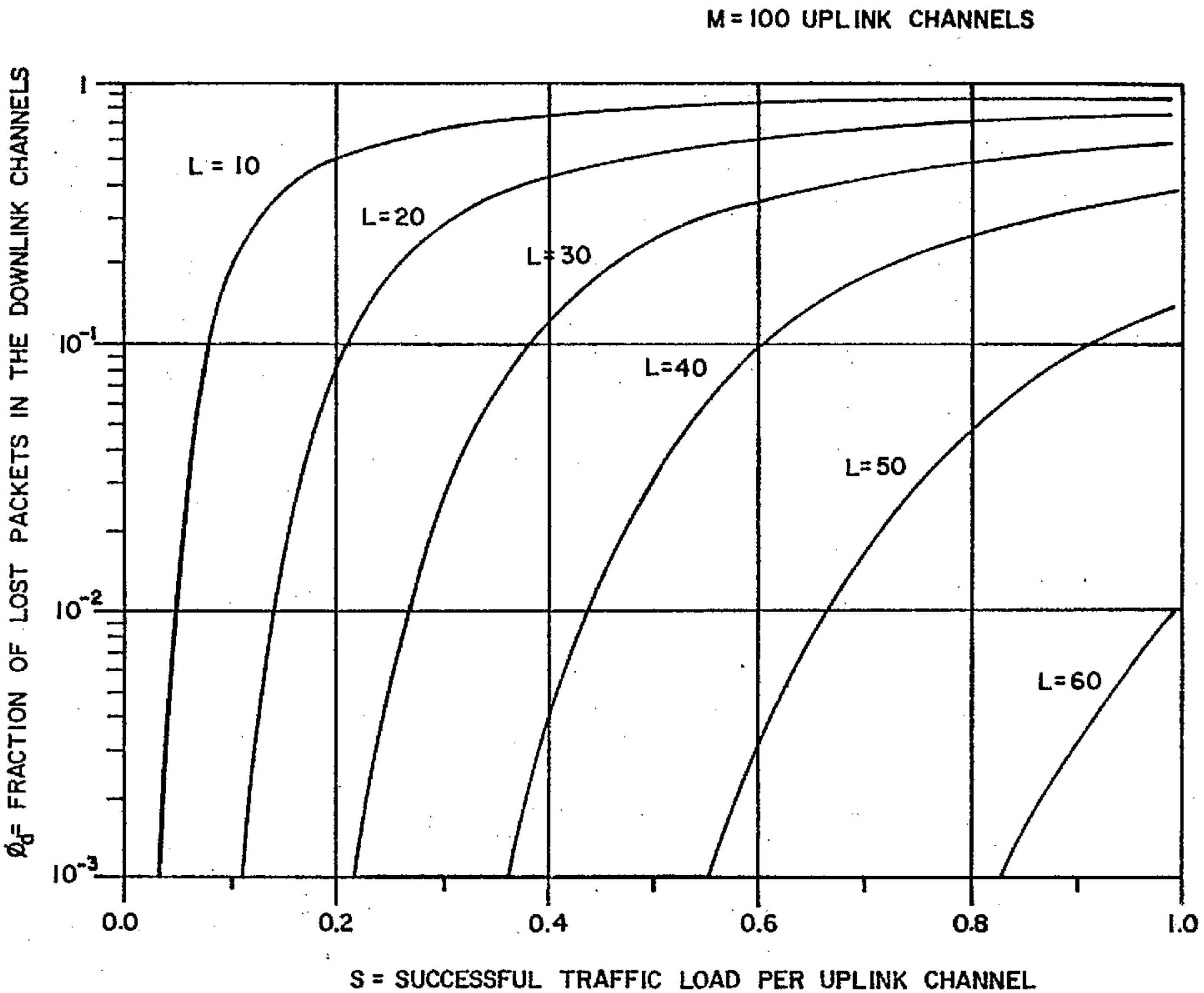


Figure 4.9 FRACTION OF DOWNLINK LOST PACKETS vs. AVERAGE SUCCESSFUL TRAFFIC (M=100)

4.3.5 End-to-End Delay

From the system description given so far it is clear that we have chosen to minimize the end-to-end delay at the expense of an increased packet loss rate. For those packets that were successfully transmitted from origin to destination through the base station we can easily derive an upper-bound for the maximum end-to-end delay as follows:

$$D = \frac{3B}{R} \qquad \ldots\ldots (4.21)$$

where the first B/R is the packetization delay, the second B/R is the sum of the maximum pretransmission delay and the transmission time and the third B/R is the depacketization delay. Note that we have ignored in the above expression any processing delays that take place at the base station.

4.3.6 Discussion and Numerical Results

Before applying the previous equations to a specific set of system parameters we summarize some of the factors that can play a role in selecting such parameters. Measurements carried out on the characteristics of speech during conversations have shown that human speech is bursty in nature. Brady {25}, among others, has confirmed that the actual channel utilization during a one-way conversation is only about 40%. He has further shown that the exponential distribution fits reasonably well the distribution of talkspurt lengths with a mean value of about 1.3 sec. On the other hand, results {34},{85} obtained to date on the transmission of packetized speech in the ARPANET indicate that to maintain a high quality speech it is necessary to ensure that:

- a nearly synchronous voice output is generated by the receiver
- end-to-end network delays do not exceed 300 msec.

Moreover, packets of lengths varying from 10 to 50 msec of speech intelligibility can be lost without seriously affecting the voice quality output and degradation begins to be observed when the fraction of lost packets exceeds a certain level, which is a function of the redundancy of the speech signal. According to some of the published data, the tolerable fraction of lost packets varies from 0.5% at low digitization rates to probably 50% at high digitization rates.

Denoting by ϕ the total fraction of the lost packets we have:

$$\phi = \phi_u + \phi_d \leq \begin{cases} 0.0005 & \text{for } R \approx 2.4 \text{ kbps} \\ 0.2 & \text{for } R \approx 16 \text{ kbps} \\ 0.5 & \text{for } R \approx 32 \text{ kbps} \end{cases} \qquad \ldots\ldots (4.22)$$

From Eqn. (4.16) it can be clearly seen that, H, the normalized offered traffic is highly dependent on the channel transmission rate C. It is also clear that both the probability "α" of discarding a packet and the probability, $1-\xi$, of losing a packet, obtainable from Figures 4.5 and 4.6, are highly dependent on the values of, G, and H. Hence by increasing the channel transmission rate we are clearly increasing the system efficiency measured in terms of the fraction of lost packets. There is however a practical upper-bound on the transmission rate that can be derived from a 30 KHz land mobile channel. Indeed a number of modulation schemes that have been devised recently {73}, suggest that transmission speeds of the order of 40 kbps, can be achieved over a 30 KHz channel.

Another parameter that can influence the system performance is the packet overhead. For the purposes of our analysis we will assume that an abbreviated header of 16 bits {81} is all that is required to properly address the packets, while providing header error correction.

Finally the last parameter of crucial importance is the time t_s required to sense a channel. As we have mentioned above, $(B/R - T_p)$ the period of time during which channels can be sensed is divided into a number of sensing slots of length t_s. Hence the maximum number of sensing slots is given by:

$$K = \frac{B/R - T_p}{t_s} \qquad \ldots\ldots (4.23)$$

which, for all practical purposes is a very large number. Indeed from Table 4.2, if we assume a channel speed of 40 kbps and a sensing time of 0.03 msec, we see that in the worst case varies from about 53 (R=32 kbps and T_p=8.4 msec) to about 320 (R=32 kbps and T_p=40.4 msec). This immediately implies that for values of θ below 0.5, the probability α of discarding a packet can be ignored.

The values contained in Table 4.2 which were obtained for two temporal packet lengths of 10 msec and 50 msec (with ρ_i=0.02 and a=0.4) indicate, as expected, that for a given channel speed (40 kbps), as

	B/R = 10 msec				B/R = 50 msec			
R(kbps)	B(bits)	B+b(bits)	T_p (msec)	h($\times 10^{-3}$)	B(bits)	B+b(bits)	T_p (msec)	h($\times 10^{-3}$)
2.4	24	40	1	0.8	120	136	3.4	0.54
4.8	48	54	1.35	1.08	240	256	6.4	1.02
16	160	176	4.4	3.52	800	816	20.4	3.26
32	320	336	8.4	6.72	1600	1616	40.4	6.46

Table 4.2 MODEL PARAMETERS FOR PACKET SPEECH

the voice digitization rate is increased, the average normalized offered traffic per voice source is also increased. This suggests that in order to increase the maximum number of voice sources that can be supported by the system, the voice digitization rate should be kept as low as possible. However, as indicated above the tolerable fraction of lost packets is a function of the speech redundancy which is quite low for low digitization rates. Since the tolerable lost packet level decreases faster than the normalized offered traffic per voice source, there is little advantage in decreasing the voice digitization rate beyond a certain value. It will be apparent that rather on the contrary, the voice digitization rate should be kept above 16 kbps. If we consider a single radio channel supporting an amount of traffic H given by Eqn. (4.16) and assume that $\emptyset_d$, the fraction of packets lost on the downlink channel is negligible, the total fraction of lost packets when $\alpha \to o$, will be:

$$\emptyset = \emptyset_u = \alpha + (1-\alpha)\ (1-\xi) \approx 1-\xi \qquad \ldots\ldots (4.24)$$

Now from Figure 4.6 we see that, for $\delta = 0.1$, H is related to ξ by:

$$H \leq \begin{cases} 1 & \text{for} \quad \xi \geq 0.5 \\ 0.55 & \text{for} \quad \xi \geq 0.8 \\ 0.03 & \text{for} \quad \xi \geq 0.995 \end{cases} \qquad \ldots\ldots (4.25)$$

which implies from (4.22) and the data of Table 4.2, that the maximum number of voice sources that can be supported is given by:

$$N \leq \begin{cases} \frac{1}{h} = 148 & \text{for } R = 32 \text{ kbps} \\ \frac{0.55}{h} = 156 & \text{for } R = 16 \text{ kbps} \\ \frac{0.03}{h} = 37 & \text{for } R = 2.4 \text{ kbps} \end{cases} \qquad \ldots\ldots (4.26)$$

Note that to derive the above numbers we have assumed a packet duration of 10 msec. For packet durations of 50 msec there is a slight increase in the number of users. However, from a delay point of view it is preferable as indicated by Eqn. (4.21) to keep the packets as short as possible. Note also that in practice δ can be smaller than 0.1. If for instance we take a one-way propagation delay of 54 μsec and a packet transmission time of 1 msec we obtain $\delta = 0.054$. This implies in particular, that for R = 2.4 kbps the maximum number of voice sources that can be supported exceeds the number given by Eqn. (4.26).

In the case where we have M uplink channels, if we assume that the traffic is evenly distributed among these channels, the total number of voice sources that can be supported, is obtained by multiplying the results of Eqn. (4.26) by M. Since we want to minimize the fraction of lost packets on the downlink channels, it is essential that for a given value of S we select the appropriate number L of downlink channels. As an example assume that the digitization rate R is equal to 16 kbps and that we can tolerate a total fraction of lost packets of the order of 20%. From Figure 4.6 (with δ=0.1) we find that H should be less than 0.55 hence the average successful traffic will be S=0.44. Using Eqn. (4.20) with M=50, we see from Figure 4.8 that in order to keep ϕ_d below 0.001, the number L of required downlink channels is of the order of 27. We can then achieve a spectrum saving of the order of 46% which for 30 KHz channels represents about 0.69 MHz. Additional savings in spectrum can be obtained by allowing ϕ_d to increase while keeping ϕ below 20%.

Various system configurations were simulated to determine the accuracy of ϕ_d the theoretical fraction of lost packets on the downlink channels. In Figure 4.10 we see that the agreement between the theoretical and simulation results obtained for 4 uplink channels and 2 downlink channels, is excellent over a wide range of traffic loads.

4.4 The Integration of Voice and Data

Prior to describing our own model for an integrated voice and data mobile system, we will briefly summarize the only two models that have been advocated so far. The earliest model due to Pan {47} suggests that voice and data can share a number of FDMA or TDMA channels on a contention basis. No attempt is made to packetize speech or data, however silence gaps in a voice conversation are not transmitted. Contrary to a circuit switched system where voice calls can be blocked in Pan's model calls are not blocked. However because of the contention character of the access scheme, initial segments of talkspurts can be locked out. Assuming a 40% activity ratio, this author showed that for a maximum talkspurt lockout of

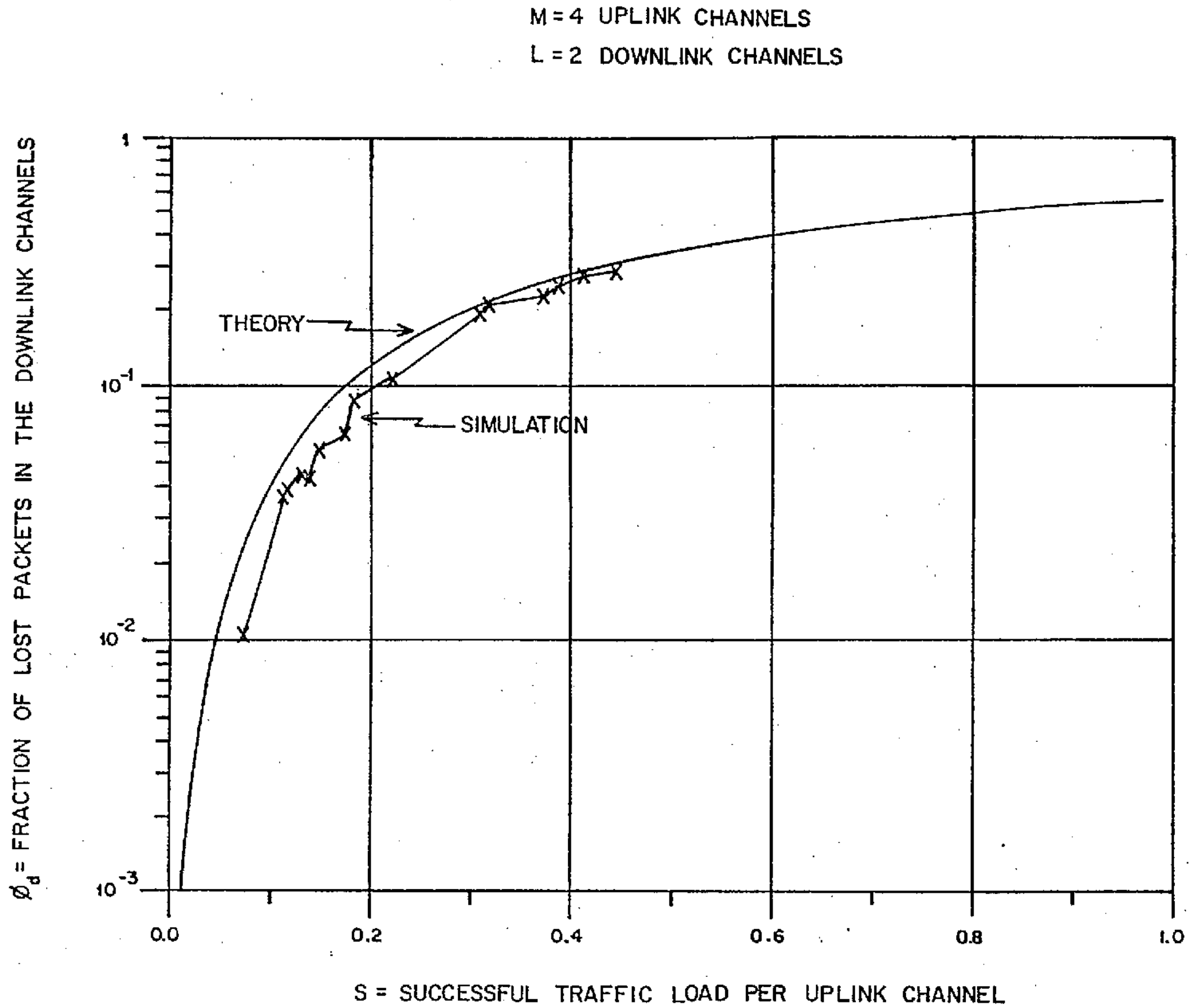


Figure 4.10 FRACTION OF DOWNLINK LOST PACKETS (SIMULATION RESULTS FOR M=4, L=2)

30 msec, the number of channels required to carry the offered traffic could be halved in comparison with a conventional circuit switched mobile system working at a 5% level of blocking.

The only other attempt to study the integration of voice and data over mobile channels is due to Marsan {46}. He considers a system where exponentially distributed voice messages (average 6.5 sec) have absolute priority over the fixed size data packets (400 bits). Even though the transmission of voice messages is not centrally controlled, interference between voice messages is disregarded since it is assumed that human operators take care not to interfere with each other. If however voice messages collide, the assumption is made that their speech content would remain intelligible. For the data users Marsan proposes two random access schemes namely pure ALOHA and NPCSMA. In the later case, data terminals attempt to determine if the channel is busy (with voice or data). Data packets are assumed to be lost after a collision and hence are retransmitted after a randomly distributed delay. Among the curves provided, two show the influence of the transmission speed on the number of packet reschedulings or on the average packet throughput. One of the interesting assumptions of Marsan's model is that voice messages are transmitted using an analog modulation while data is digitally modulated.

The approach we take to develop our model differs from Pan's and Marsan's approach both in terms of the model assumptions as well as in the overall perspective and model performance measures.

4.4.1 Network Model

As in the model described in Section 4.3.1 we consider a conventional land mobile system where a finite number, M, of speech traffic sources exchange communications through a given number, C, of radio channels. Denoting by α the average number of voice calls per traffic source during the busy hour and by $1/\mu$ the average call duration and assuming a "blocked calls lost" discipline, the fraction of time, T_c, during which the channels are busy* is given by:

$$T_c = \frac{\binom{M}{C} \rho_i^{\,C}}{\sum_{j=o}^{C} \binom{M}{j} \rho_i^{\,j}} \qquad \text{..... (4.27)}$$

where $\rho_i = \alpha/\mu$ denotes the offered load per voice source.

Again, following Brady {84}, we consider each voice call to be formed of a sequence of talkspurts and silence gaps. Thus, the probability, ϕ, that, on any given channel, there is a talkspurt in progress is given by:

$$\phi = T_c \cdot \frac{\bar{T}}{\bar{T}+\bar{P}} \qquad \text{..... (4.28)}$$

where $\bar{T}$ and $\bar{P}$ denote respectively the average talkspurt duration and the average silence duration.

---

* If we consider an infinite population of traffic sources Eqn. (4.27) should be replaced by the classical Erlang B equation.

A question is now raised regarding the feasibility of transmitting data packets during the silence periods, both within a given call and between two consecutive calls without disturbing in any way the flow of voice calls. One possible scheme that can be envisaged consists in preventing the transmission of data packets during talkspurts by using a sub-audible busy tone within the voice channel. In addition, it is necessary to ensure that those voice sources that have been granted access to the voice channels, activate their transmitters only during a talkspurt period. While contention amongst voice sources can be avoided by using a blocked calls lost mechanism, a distributed access scheme is for all practical purposes sufficient to ensure an efficient utilization of the available time windows on any given set of frequency multiplexed channels. In essence, what we are advocating is a form of hybrid switching with voice calls served on a circuit switching basis and data packets served on a packet switching basis.

Because of the real time nature of voice communications, any arriving talkspurt will be immediately transmitted; thereby pre-empting any data packet that might be in progress. By properly adjusting the configuration of both voice and data transmitters, and by carefully selecting the data packet length and channel transmission rate, it is possible to keep the potential damages to the talkspurts within reasonable limits. It should be noted that we are not advocating that the same mobile transmitter be used alternatively as a voice or data terminal. We are only concerned with the integration of voice and data at the radio frequency channel level.

Prior to describing the protocol to be used by the data packet terminals we will first state the major assumptions of the proposed system's model.

- The arrival processes of voice calls and data packet arrivals are taken to be Poisson processes with different mean arrival rates
- Talkspurt and pause lengths are exponentially distributed with different mean lengths
- Data packet lengths or equivalently data packet transmission times are constant
- A unique radio frequency channel is shared by voice and data traffic sources.

A number of distributed random access mechanisms have been proposed in recent years (see Appendix B) with the objective of handling in a cost-effective fashion the ever increasing need to exchange communications in packet form. Some of these schemes are more complex and more efficient than others, particularly in situations corresponding to those found in land mobile environments. However, for the sake of simplicity we will restrict our attention to a slightly modified version of the so-called Pure ALOHA random access scheme.

According to this protocol a packet generated by a data terminal joins the terminal's buffer until reaching the head of the buffer which we refer to as the transmit buffer. It is assumed that all terminals are notified of the channel busy status (talkspurt in progress) by a busy tone transmitted by the base station through which all voice and data traffic is routed. Hence when a packet enters the transmit buffer, the transmitter will sense the status of the channel. If the channel is currently busy with a talkspurt in progress the packet will wait until the end of the talkspurt at which time it is scheduled for transmission after a time duration selected from a random distribution as a multiple of the packet transmission time. It is assumed that this duration of time will be taken from a uniformly distributed scheduling function which varies between 0 and K. Prior to packet transmission, the sending mobile will sense the channel status and only if the channel is silent will the packet be transmitted. The fact that the channel was found to be idle does not by itself guarantee its correct transmission since the probability exists that the packet might collide totally or partially with a packet from another terminal. Partial packet overlaps can occur since a packet arriving to the transmit buffer during a silence period will be transmitted without delay.

Additionally, we will assume that a packet collision is detected by the sending terminal exactly after a period of time corresponding to a packet transmission time. Having detected a collision the packet is rescheduled for transmission after a time duration selected from the uniformly distributed scheduling function mentioned above.

4.4.2 Analytical Model

The protocol described above is to some extent similar to the Pure ALOHA random access scheme as far as the random access to the channel during the silent time windows is concerned. To account for the fact that the channel is not available to data users on a full time basis, we introduce in the classical traffic-throughput relationship of Pure ALOHA, a factor $\emptyset$ as follows:

$$S = G(1-\emptyset)\, e^{-2G(1-\emptyset)} \qquad \ldots\ldots (4.29)$$

where S and G represent respectively the average number of offered packets per packet transmission and the average number of new and rescheduled packets per packet transmission time. The factor $\emptyset$ given by Eqn. (4.28) represents the probability that the channel is busy with a talkspurt.

The interpretation of Eqn. (4.29) is quite straightforward for when $\emptyset$ approaches one the throughput S vanishes to zero while when $\emptyset=0$ we obtain the classical Pure-ALOHA traffic-throughput equation. Note however that even if there were not intercall gaps, there would always exist the intra-call gaps. Hence the maximal value of $\emptyset$ would be of the order of 0.4.

In Figure 4.11, we depict the relationship between S and G for various values of $\emptyset$ and it is quite clearly seen that the maximum value of S is always 0.184. The corresponding value of G will however vary between G=0.5 ($\emptyset$=0) and G=0.83 ($\emptyset$=0.4). As the channel becomes busier with voice calls the probability of successful packet

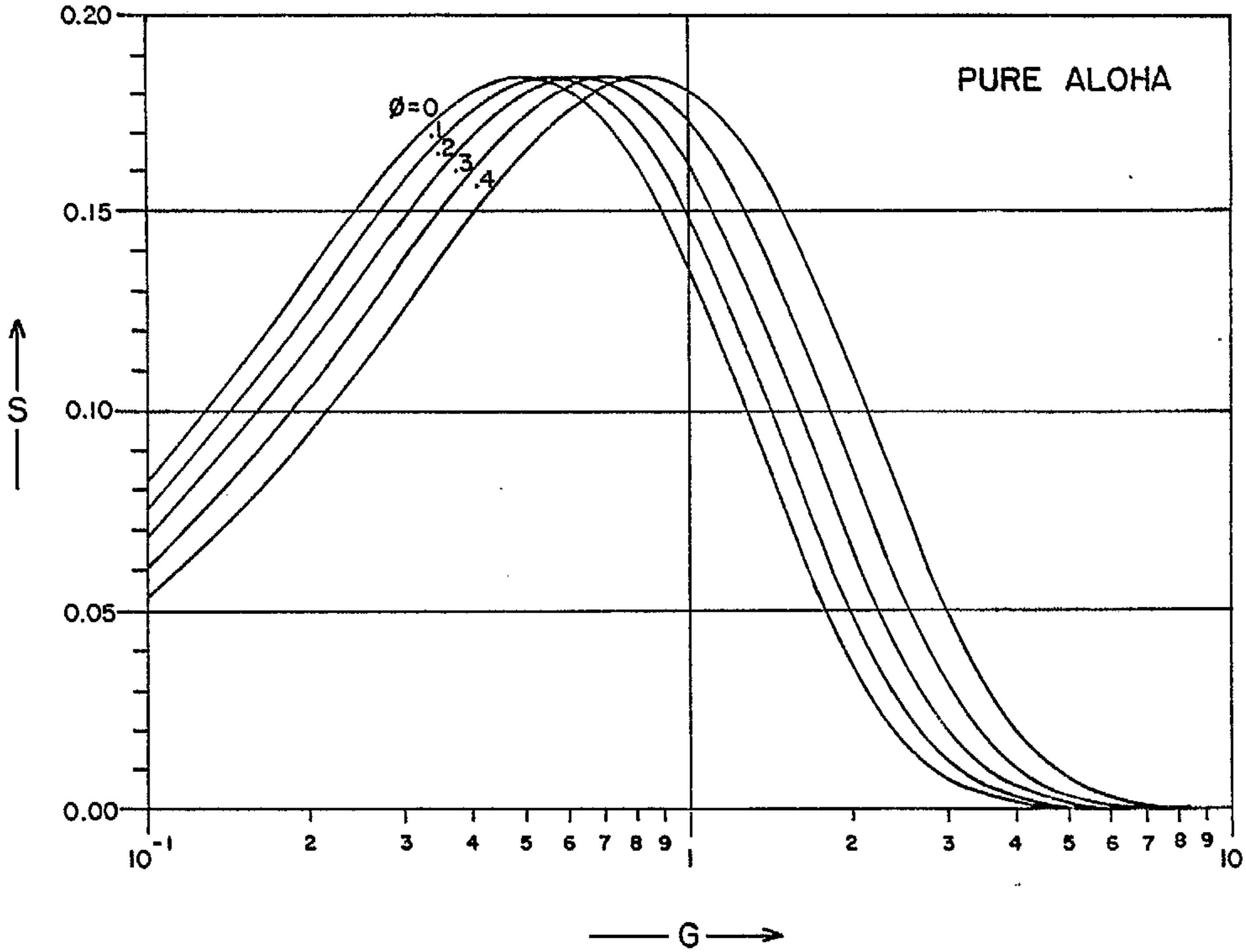


Figure 4.11 TRAFFIC THROUGHPUT CHARACTERISTICS OF BUSY-TONE PURE ALOHA WITH Ø AS A PARAMETER

transmission, given by S/G, decreases. The same conclusion can be reached by noting that the average number of transmissions per packet, given by G/S, increases. This has an impact on the average delay experienced by a packet from its arrival until its successful transmission.

Hence, the fact that voice traffic has precedence over the data traffic will introduce an additional delay component in the overall average packet delay from the time the packet is accepted in the transmitter buffer pool to the time the packet is successfully transmitted. Consequently, in the following analysis we seek to determine the total average packet delay as well as the average buffer occupancy as a function of the statistical characteristics of voice and the average random retransmission-rescheduling delay. We will furthermore define an upper-bound for the average random retransmission delay since the larger its value the more "unstable" will the channel become.

4.4.3 Total Average Packet Delay

By approximating the packet generation process at each data terminal with a Poisson process and modelling the resulting queueing system as an M/G/1 queue, the overall total average packet delay is given by:

$$T_D = \bar{T}_s + \frac{\lambda(\bar{T}_s^2 + \sigma_{T_s}^2)}{2(1 - \lambda\bar{T}_s)} \qquad \ldots\ldots (4.30)$$

where:

$$\bar{T}_s = \text{average packet service time}$$

$$\sigma^2_{T_s} = \text{variance of the packet service time}$$

$$\lambda = \text{average packet arrival rate}$$

We are now faced with the difficult task of determing $\bar{T}_s$ and $\sigma^2_{T_s}$ as a function of the uniformly distributed random retransmission delays and the exponentially distributed talkspurt lengths. In doing so it is helpful to consider the diagram of Figure 4.12.

Having reached the head of the buffer pool, the packet enters the transmit buffer and proceeds to sense the status of the channel. After a random amount of time W during which the channel was sensed and found to be busy a number of times, the packet is transmitted for the first time. If it collides the process is reinitiated until after a number of collisions the packet successfully departs.

Denoting by X_i the random variable representing the elapsed time between the i^{th} and i^{th}+1 failed attempt for transmission and assuming the variables X_i (i=1,....n) to be independent and identically distributed with mean and variance given by:

$$\eta = E(X_i)$$

$$\sigma^2 = \sigma^2_{X_i}$$

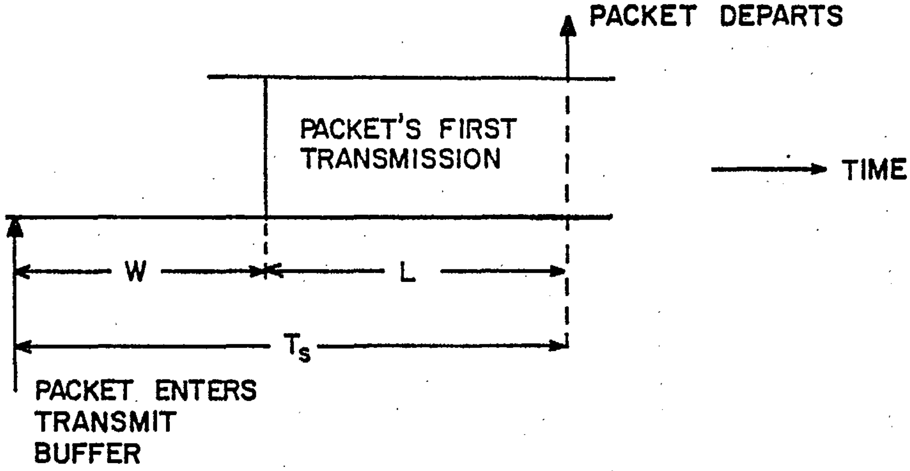


Figure 4.12 MODEL FOR THE PACKET SERVICE TIME

we obtain the following relationships:

$$\eta = \bar{T} + \bar{K} \qquad \text{..... (4.31)}$$

$$\sigma^2 = \bar{T}^2 + \frac{(K+1)(2K+1)}{6} - \bar{K}^2 \qquad \text{..... (4.32)}$$

where:

$\bar{T}$ = average talkspurt length

$\bar{K}$ = average rescheduling delay

K = maximum value of the rescheduling delay

Note that we have made use of the fact that talkspurts are exponentially distributed while the rescheduling delays are uniformly distributed.

Clearly the random variable W is related to the variables X_i by the following random sum:

$$W = \sum_{i=1}^{N} X_i \qquad \text{..... (4.33)}$$

where the variable N representing the number of unsuccessful transmission attempts has the following distribution:

$$P[N=i] = \emptyset^{i} (1-\emptyset) \qquad i=0,\ldots.\infty \qquad \text{..... (4.34)}$$

where $\emptyset$ denotes the probability that the channel is busy.

To obtain the expected value and the variance of W we make use of the following relationships:

$$\bar{W} = \eta \, E(N) \qquad \text{..... (4.35)}$$

$$\sigma_W^2 = \eta^2 \, \sigma_N^2 + E(N) \quad \sigma^2 \qquad \text{..... (4.36)}$$

Since N is geometrically distributed we have:

$$E(N) = \frac{\emptyset}{1-\emptyset} \qquad \text{..... (4.37)}$$

$$\sigma_N^2 = \frac{\emptyset}{(1-\emptyset)^2} \qquad \text{..... (4.38)}$$

Hence Eqns. (4.35) and (4.36) can be rewritten as follows:

$$\bar{W} = \frac{\eta \, \emptyset}{1-\emptyset} \qquad \text{..... (4.39)}$$

$$\sigma_W^2 = \frac{\emptyset}{(1-\emptyset)^2} \left[\eta^2 + \sigma^2 \, (1-\emptyset) \right] \qquad \text{..... (4.40)}$$

which specify the random variable W in terms of its two most important parameters namely the mean and the variance.

We must now conduct a similar analysis to derive the corresponding parameters of the random variable L representing (see Figure 4.12) the time between the first packet transmission and its successful departure. Before doing so consider the diagram shown in Figure 4.13.

Following the i^{th} transmission and after one time unit corresponding to the packet transmission time, the packet collision is detected. Consequently, the packet is rescheduled for transmission by a random amount of time drawn from a uniform distribution, at which point in time the channel sensing process is reinitiated. The sensing process is stopped by the $i^{th}+1$ transmission of the same packet. Assuming independence between the two random variables Y and W we obtain:

$$\bar{Z} = \frac{\eta \emptyset}{1-\emptyset} + 1 + \bar{K} \qquad .....(4.41)$$

$$\sigma_Z^2 = \frac{\emptyset}{(1-\emptyset)^2}\left[\eta^2 + \sigma^2 \ (1-\emptyset)\right] + \frac{(K+1)(2K+1)}{6} - \bar{K}^2 \qquad .... (4.42)$$

Since the random variable Z represents the elapsed time between two consecutive transmissions, the random variable L (see Figure 4.12) will be given by:

$$L = \sum_{j=1}^{H} Z_j \qquad ....j=1,....H \qquad ..... (4.43)$$

where H the random variable denoting the number of transmissions prior to the successful transmission, is distributed as follows:

$$P\left[H=i\right] = P_c^{\ i} \ (1-P_c) \qquad i=0,....\infty \qquad ..... (4.44)$$

where P_c denotes the probability of packet collision during the voice gaps.

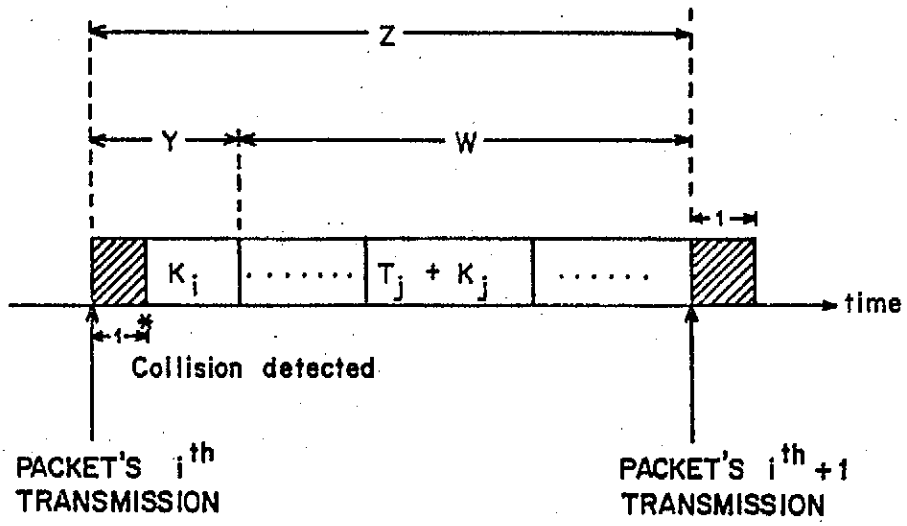


Figure 4.13 MODEL FOR THE TIME BETWEEN TWO CONSECUTIVE PACKET TRANSMISSIONS

It is known {86} that P_c is a function of the data traffic rates as well as the average retransmission delay, however we will assume that $\bar{K}$ is large enough so that P_c approaches its limiting value. Hence we can write:

$$\bar{L} = \bar{Z} \frac{P_c}{1-P_c} \qquad \text{..... (4.45)}$$

$$\sigma_L^2 = \frac{P_c}{(1-P_c)^2} \left[\bar{Z}^2 + \sigma_Z^2 (1-P_c) \right] \qquad \text{..... (4.46)}$$

Referring again to Figure 4.12 and assuming that the random variables W and L are independent we finally obtain the average and the variance of the service time as follows:

$$\bar{T}_s = \bar{W} + \bar{L} \qquad \text{..... (4.47)}$$

$$\sigma_{T_s}^2 = \sigma_W^2 + \sigma_L^2 \qquad \text{..... (4.48)}$$

which implies that:

$$\bar{T}_s = \frac{\eta\, \emptyset}{1-\emptyset} + \frac{P_c}{1-P_c} \cdot \bar{Z} \qquad \text{..... (4.49)}$$

$$\sigma_{T_s}^2 = \frac{\emptyset}{(1-\emptyset)^2} \left[\eta^2 + \sigma^2 (1-\emptyset) \right] + \frac{P_c}{(1-P_c)^2} \left[\bar{Z}^2 + \sigma_Z^2 (1-P_c) \right] \qquad \text{... (4.50)}$$

The results in conjunction with knowledge of the average packet arrival rate are all that is needed to determine by use of Eqn. (4.30), the total average packet delay $\bar{T}_D$. As is well known, by applying Little's result, we can also determine the average buffer occupancy $\bar{n}$ as follows:

$$\bar{n} = \lambda \bar{T}_D \qquad \ldots\ldots (4.51)$$

4.4.3.1 Upper-bound for K

Usually in problems of random access communications a lower-bound for the maximum value of the random retransmission delay K is specified below which the channel is unstable in the sense that the number of collisions is so high that the real packet throughput vanishes to zero. A recent study {87} has however clearly shown that there is a need to define an upper-bound for K beyond which the packet delays become extremely large or what is equivalent there is a high probability of buffer overflow. In the context of this paper we can also define an upper-bound for K and to do so we simply select the value of $\bar{K}$ such that:

$$\lambda \bar{T}_s < 1 \qquad \ldots\ldots (4.52)$$

From Eqn. (4.49) we obtain:

$$\bar{K} < \frac{(1-P_c)/\lambda - \left[\bar{T}\emptyset/(1-\emptyset) + P_c \right]}{\emptyset/(1-\emptyset) + P_c} \qquad \ldots\ldots (4.53)$$

4.4.4 Modified Protocol

A modification of the protocol upon which the above equations were derived can be suggested where by a packet upon finding the channel busy with a talkspurt is immediately rescheduled by a random amount of time K drawn from a uniform distribution. This modification implies that the parameter $\bar{T}$ will not be present in the above equations. Following the same arguments the equations required to determine the total average packet delay and the average buffer occupancy are given by:

$$\bar{L} = \frac{\bar{K}}{1-\emptyset}\left(\emptyset + \frac{P_c}{1-P_c}\right) + \frac{P_c}{1-P_c} \qquad \ldots\ldots (4.54)$$

$$\sigma_L^2 = \frac{\sigma_Z^2}{1-P_c} + \frac{P_c}{(1-P_c)^2}\,\bar{Z}^2 \qquad \ldots\ldots (4.55)$$

where:

$$\bar{Z} = 1 + \bar{K}/(1-\emptyset) \qquad \ldots\ldots (4.56)$$

$$\sigma_Z^2 = \frac{\emptyset}{(1-\emptyset)^2}\left[\bar{K}^2 + \sigma^2\,(1-\emptyset)\right] \qquad \ldots\ldots (4.57)$$

$$\sigma^2 = (K+1)(2K+1)/6 - \bar{K}^2 \qquad \ldots\ldots (4.58)$$

From Eqn. (4.53) we can also determine the new upper-bound for $\bar{K}$ which will be:

$$\bar{K} < \frac{\left[1-P_c(1+\lambda)\right](1-\emptyset)}{\lambda\left[\emptyset+P_c(1-\emptyset)\right]} \qquad \ldots\ldots (4.59)$$

4.4.5 Discussion and Numerical Results

The analysis given above should be considered as an approximate analysis since the assumptions pertaining to the independence between random variables and the Poisson nature of the packet arrival process have yet to be justified. However, the simulation results of Figures 4.14 to 4.17, indicate that the approximate analytical model can be considered a valid basis for system's design. For illustration purposes we have also shown in the same set of figures some simulation points corresponding to the slotted ALOHA random access scheme.

Since for most cases of interest there is virtually no difference (except for small values of $\bar{K}$) between the protocol described in Section 4.4.1 and what we called the modified protocol we have chosen to present the analytical and simulation results for the modified protocol.

Also among all the possible options available concerning the characteristics of the voice traffic, we have selected to use an average talkspurt length of 1.3 sec and average pause length of 1.8 sec {84}. Using these values we obtain for the various simulation runs a probability of busy channel of the order of 0.258 which corresponds to about 20 voice users at one call every hour and 3 minutes per call.

For all runs we have used a packet transmission time of 100 msec and have expressed all values in terms of multiples of the packet transmission time.

Figure 4.14 displays the classical S-G curves for both the pure and slotted ALOHA cases as well as some simulation points corresponding to a finite population of data terminals. It is clearly seen that the limiting case given by the theoretical curves will only be attained for a very large number of data sources as well as for a very large value of the average retransmission delay which would lead to an infinite average packet delay. As expected, the agreement between analysis and simulation is excellent for low traffic values. Ignoring the delay factor one can conclude that a single integrated voice-data radio channel can theoretically support about 2000 data sources (slotted ALOHA), each generating one 100 msec long packet every 10 minutes. However, since delays cannot be ignored in any practical system, it is interesting to establish an upper-bound for what could be a reasonable average packet delay, say 2 sec, and determine how many data sources could the system support. Such trade-offs are displayed in Figure 4.15 where it is shown that up to 600 data sources (pure ALOHA), each generating one packet every 20 minutes, can be supported provided that $\bar{K}$ (the average value of the retransmission delay) is kept below 15. As shown by the simulation curve corresponding to 600 users, the value of $\bar{K}$ needed to ensure channel stability should not be decreased below 10. We note that the proposed analytical model cannot be used for predicting the lower bound for $\bar{K}$.

Figure 4.15 also shows some simulation points corresponding to pure and slotted ALOHA with 300 sources each generating one packet every 20 minutes. As expected for such low traffic cases there is virtually no difference between the two random access schemes.

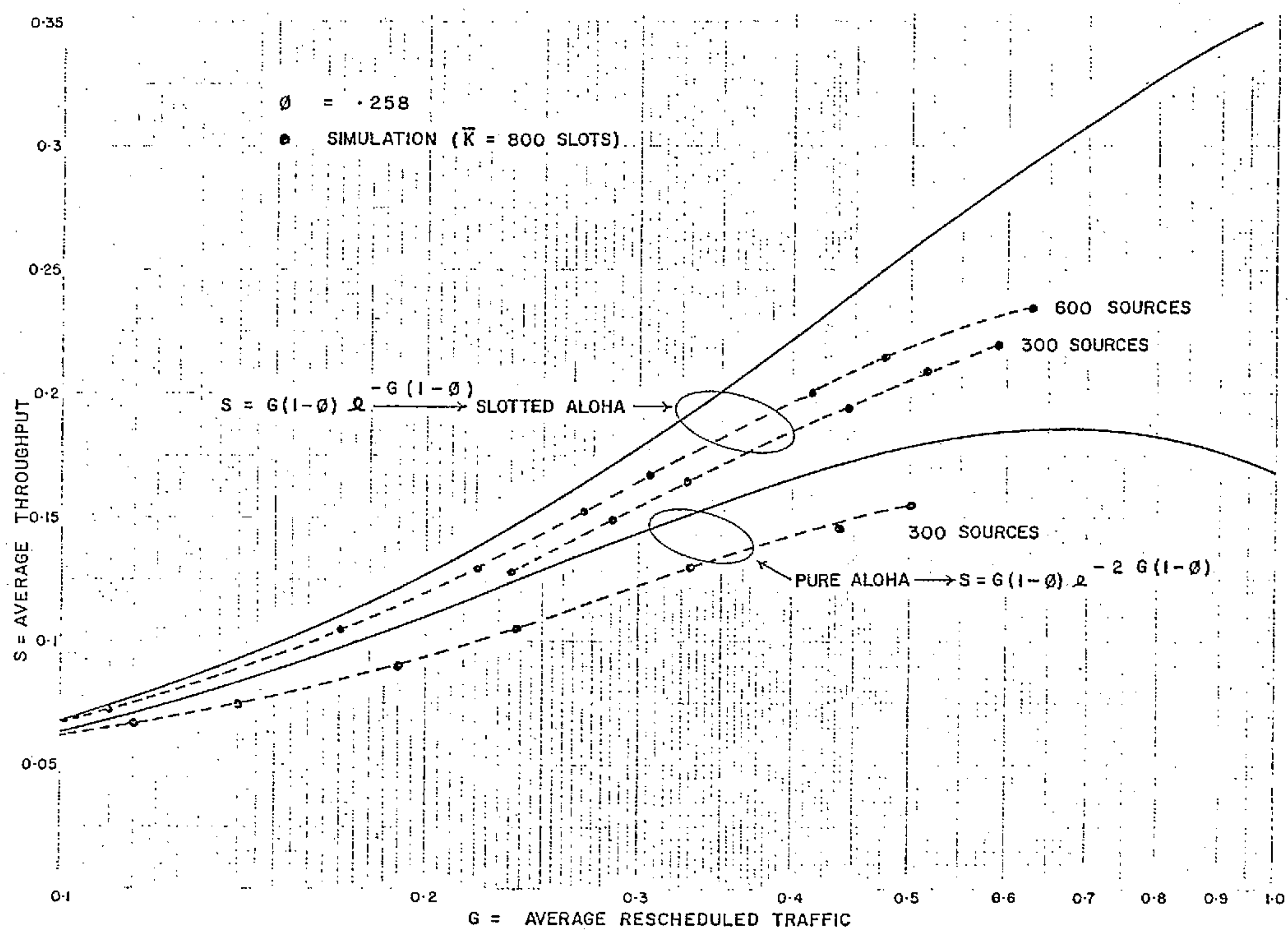


Figure 4.14 TRAFFIC - THROUGHPUT FOR INTEGRATED VOICE/DATA

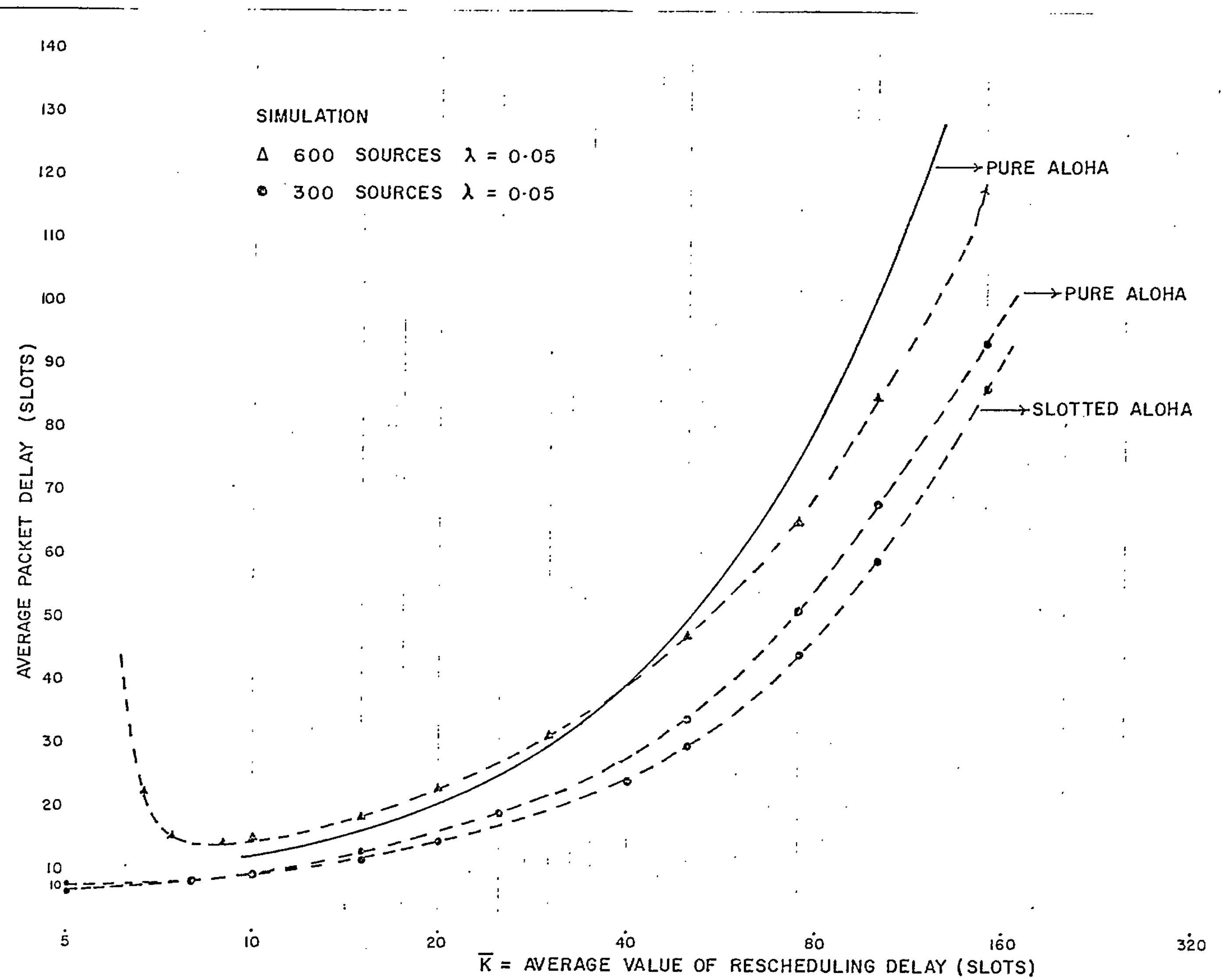


Figure 4.15 INFLUENCE OF $\overline{K}$ ON THE AVERAGE PACKET DELAY (LOW TRAFFIC)

To illustrate the behaviour of the proposed system in cases of higher traffic loads we display in Figures 4.16 and 4.17 the average packet delay and average service time as a function of $\bar{K}$. Again the agreement between the theory and simulation is excellent for a wide range of $\bar{K}$. We also note that the slotted ALOHA scheme offers a great improvement over the pure ALOHA both in terms of channel stability and delay. Indeed for the pure ALOHA method the lower-bound on $\bar{K}$ is approximately 50 which gives an average packet delay close to 10 sec, while for slotted ALOHA the minimum achievable average packet delay is about 1 sec for $\bar{K}=5$.

In Figure 4.18 we show how the average packet delay varies with the number of traffic sources for a constant channel input rate. We see for instance that for Slotted ALOHA, $\bar{K}=25$ and channel input rate of 0.2 two "unstable" regions are clearly identified. For a small number of large users the average packet delay is dominated by the average packet queueing time which in term is dependent upon the value of $\bar{K}$. Too large a value of $\bar{K}$ will make the channel unstable. On the other hand for a large number of small users an originally stable channel becomes unstable with the average packet delay becoming dominated by the average packet service time. In this case to small a value of $\bar{K}$ will make the channel unstable. By reducing the channel input rate we see that for the same value of $\bar{K}$ the average packet delay remains essentially constant for the given range of traffic sources.

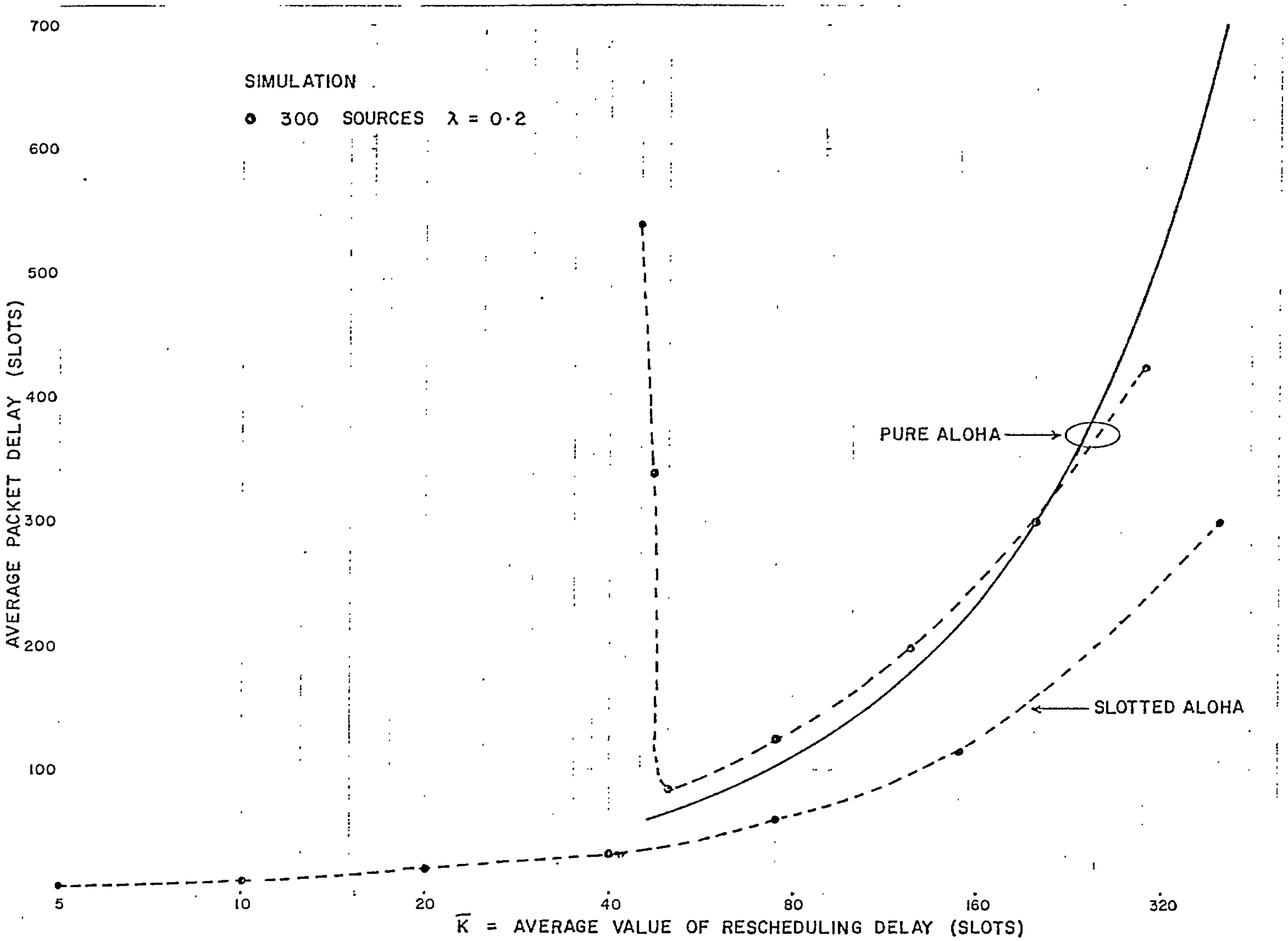


Figure 4.16 INFLUENCE OF $\bar{K}$ ON THE AVERAGE PACKET DELAY (HIGH TRAFFIC)

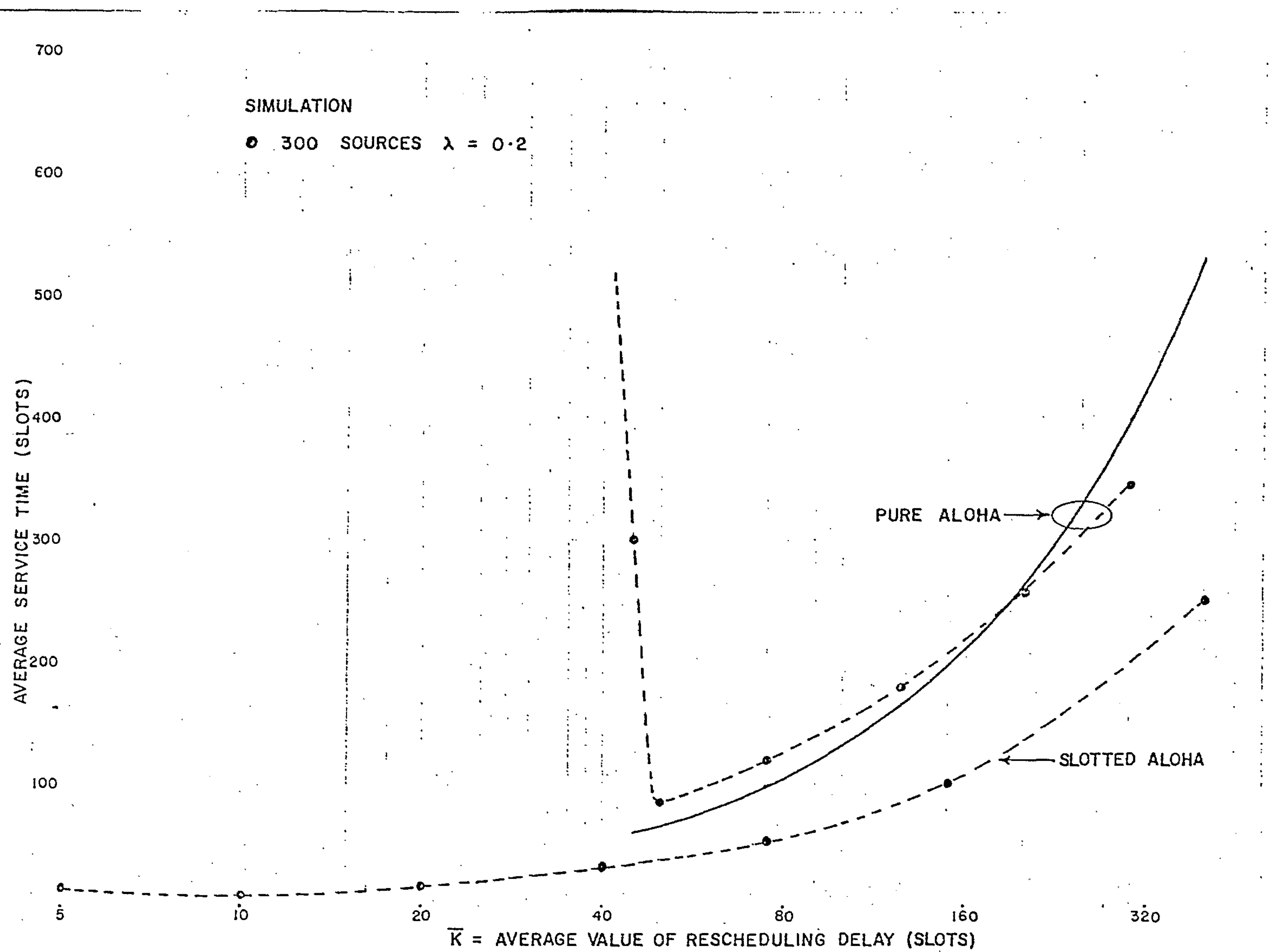


Figure 4.17 INFLUENCE OF $\overline{K}$ ON THE AVERAGE SERVICE TIME (HIGH TRAFFIC)

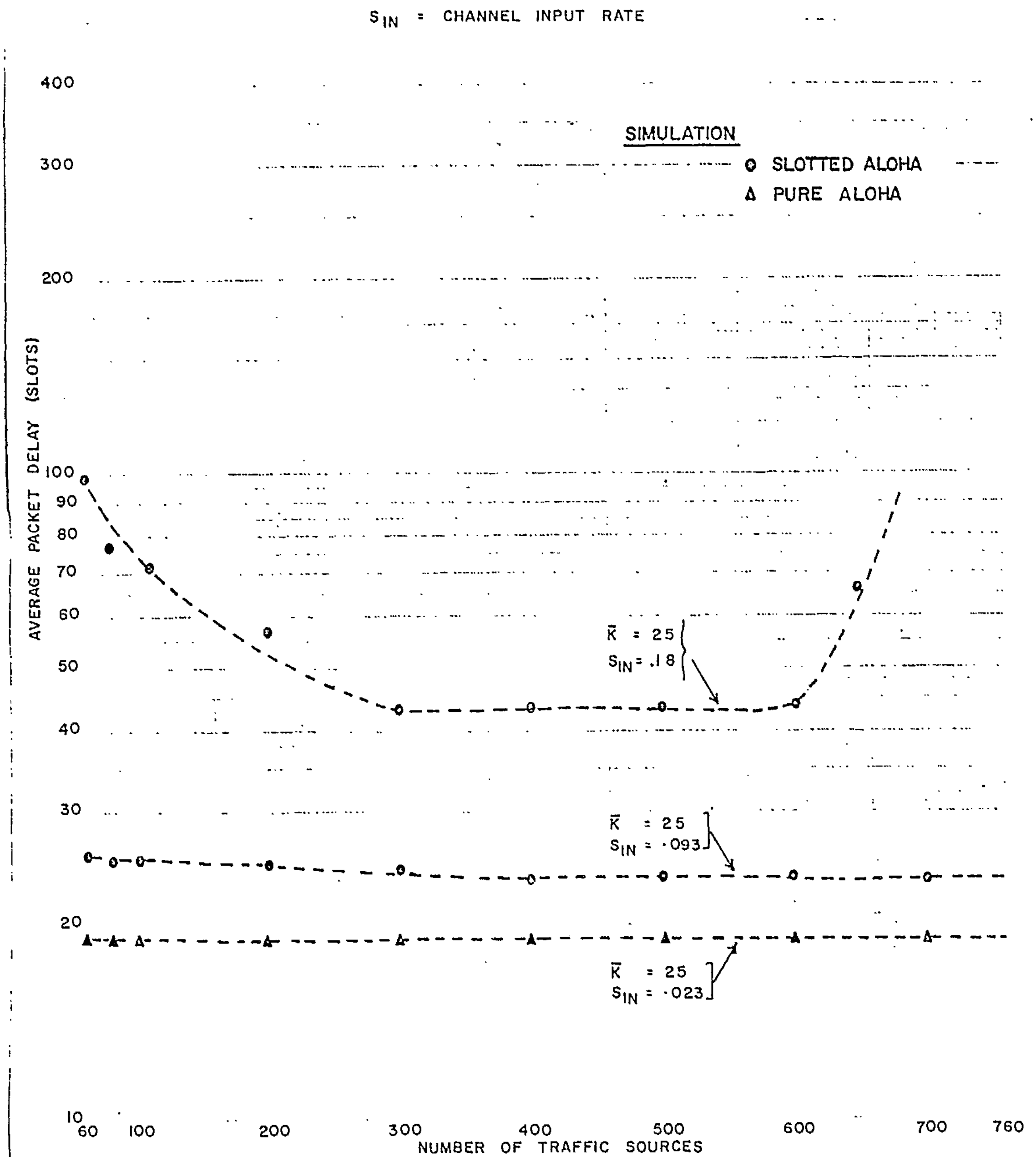


Figure 4. 18 AVERAGE PACKET DELAY vs NUMBER OF TRAFFIC SOURCES

4.5 Summary of Results

We have examined in this chapter two new system configuration that can be advantageously used in the context of land-mobile communication systems. The packet speech model presupposes that voice is digitally encoded and transmitted using an efficient digital modulation technique such as TFM. However, in the case of speech and data integration there is no need to digitally encode the talkspurts. This last model assumes the availability of an efficient and economic speech detector. As discussed elsewhere in this dissertation, several techniques can be used for speech detection. From a technological point of view the simplest means to detect the presence of speech requires that the voice signal be digitally encoded. In this case, silence detection is accomplished by examining specific bit patterns corresponding to the silence segments of a voice conversation. In Figures 4.19 and 4.20 we show the configurations of the integrated mobile terminal and the integrated base station. As indicated in these figures both the voice and data signals share the same modulator. When voice is transmitted by a mobile station, there is a need to incorporate in the transmitted signal a sub-audible tone which is used by the base station to inform all terminals of the presence of voice. A mobile unit wishing to transmit data packets is prevented from doing so whenever a busy tone is present on the channel.

In summary, based on the numerical results it appears that the concept of packet speech can be advantageously applied to land-mobile channels. In particular, it was shown that for a given number of uplink channels, the maximum number of voice sources that can be

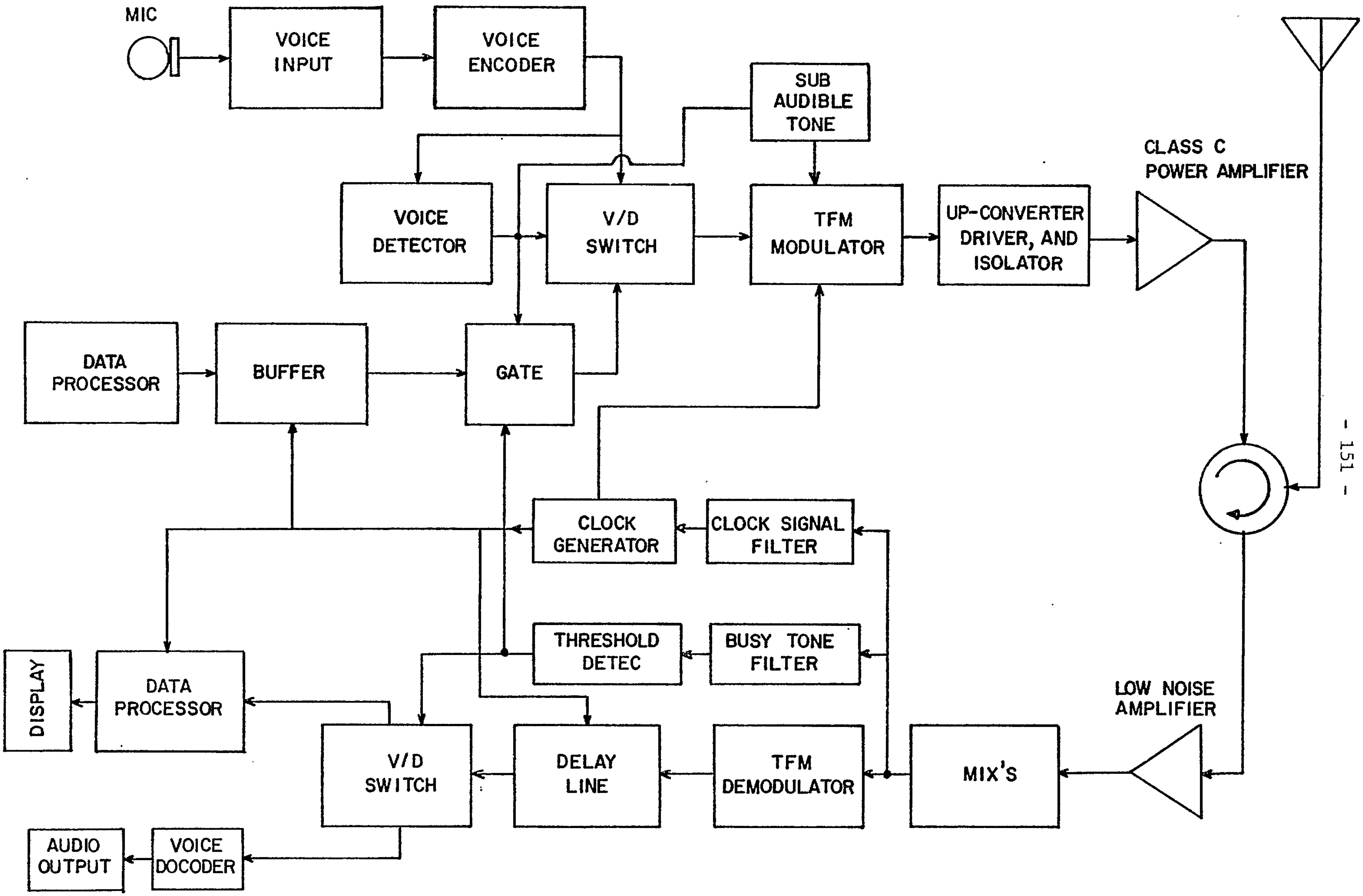


Figure 4.19 INTEGRATED MOBILE TERMINAL

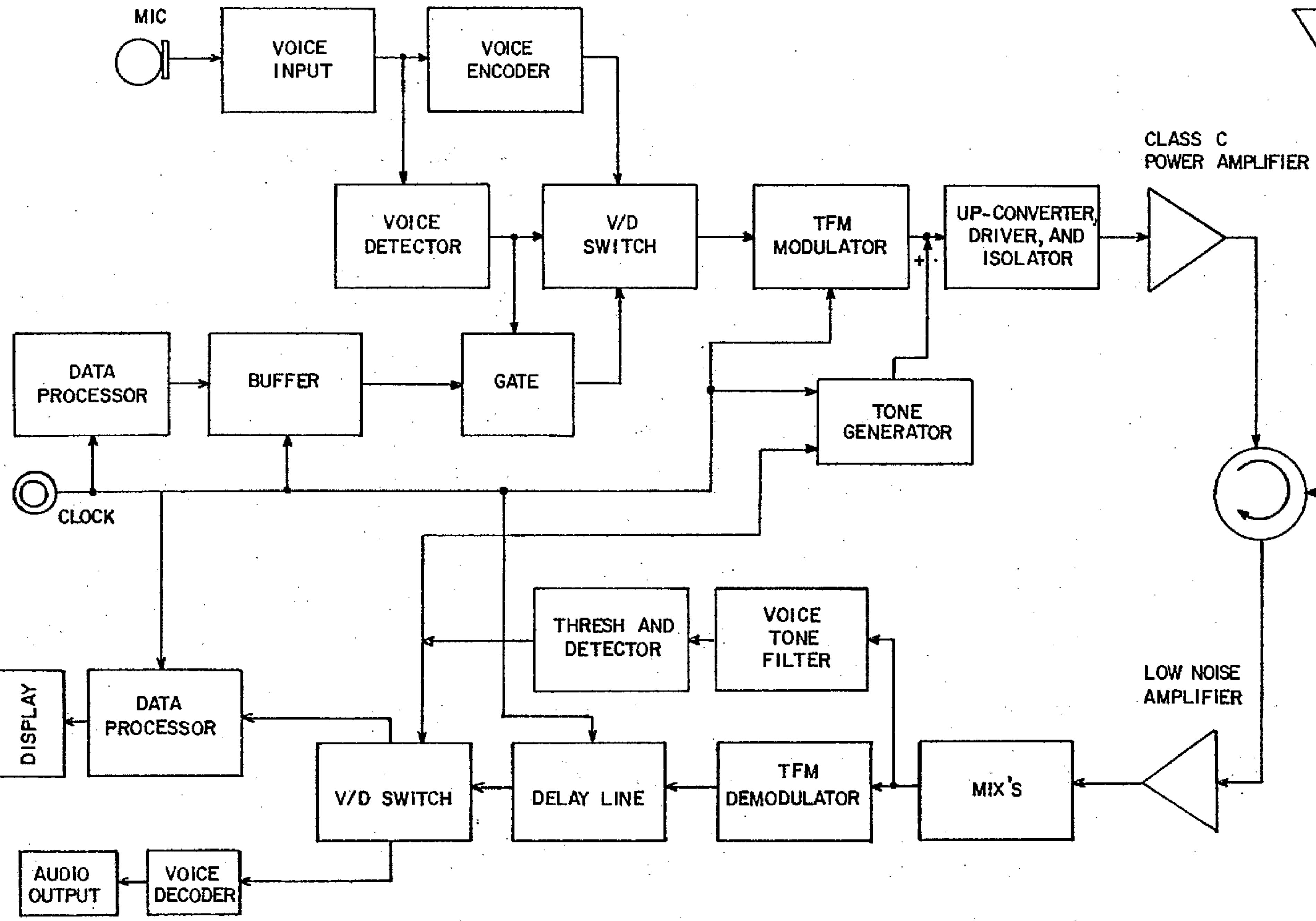


Figure 4.20 INTEGRATED BASE STATION

supported will, under some assumptions, attain a value of ≈ 150 M, which is a three fold increase over what can be achieved with conventional analog land-mobile channels. It was also shown that in a packet speech system the amount of spectrum required for the downlink channels represents about 50% of the bandwidth required in a comparable analog land-mobile system. This is because conventional land-mobile systems require the allocation of the same bandwidth in both directions.

We have also presented a mathematical analysis of a land mobile communications channel where data packets are transmitted during the pauses of voice conversations. The numerical results provided indicate that under some constraints, the integration of circuit switched voice users and packet switched data users on the same channel is a viable alternative to the current frequency spectrum shortage. These numerical results were confirmed by extensive simulation runs which conclusively show that a single radio channel is capable of supporting up to 600 bursty data traffic sources with average packet delays of the order of 2 sec. A careful examination must be applied, however, to such optimistic figures. In any operational system of this type, small fluctuations in the voice traffic characteristics can lead to substantial packet delays. While a given average packet delay might be acceptable for a specific application, such systems should be designed to ensure for instance that 95% of all packets encounter delays of less than 2 sec. Consequently, this leads to a decrease in the number of data traffic sources accessing the system. This problem can be overcome by employing higher transmission speeds.

CHAPTER 5

CONCLUSIONS AND SUGGESTIONS FOR FURTHER WORK

5.1 Conclusions

The work reported in this report attempted to achieve two objectives:

* *Development of a better understanding of the key parameters of the land mobile channel and study of their impact on the throughput and efficiency of packet radio random access schemes.*

* *Investigation, through analysis and simulation, of the applicability of packet speech concepts to land mobile channels and study of the performance of integrated voice/data mobile systems.*

The first objective was achieved by conducting an analysis of the key parameters of the land mobile channel and assessed their impact on the throughput and efficiency of multiple-access schemes. Using a classical model of the propagation environment of land mobile systems, a computerized fading simulator was constructed and its validity was examined by comparing it with available theoretical results. In order to gain further insight into the performance of digital transmissions under fading conditions, we obtained the statistical distributions of the fade duration, the non-fade duration and the interfade duration

In the second specific fading problem that we studied, we were concerned with finding the packet size that minimizes the time wasted in retransmissions. An ARQ protocol was used and the variation of how the optimal packet size varied as a function of the bit rate, the signal-to-noise ratio and the average message length was examined.

The second objective of this report was met by conducting a thorough study of the applicability of packet speech and voice/data integration to mobile channels. In the packet speech area, we developed a feasible system configuration and showed that by taking advantage of the idle channel periods, a large number of voice sources could be multiplexed onto a number of uplink FDMA channels. Our analysis of this problem was further extended to show that substantial bandwidth savings could also be obtained on the downlink channels. Using appropriate system parameters, we showed that the traffic carrying capacity of packet speech mobile systems is increased by a factor of three over what can be achieved using conventional mobile systems.

The possibility of exploiting the presence of silence gaps in a two-way conversation to transmit data in packet form was also examined. A simple but accurate model of the integration of voice and data, showed that a large number of data sources could indeed be interpolated in the mobile radio channel. Because of the limited time availability of the voice channel, data packets have to be retransmitted more often, which implies that the integration of voice and data will result in a penalty of higher average packet delay. Extensive simulations confirmed the accuracy of the theoretical model developed for delay analysis. The average delay between two consecutive retransmissions was shown to influence significantly both the traffic carrying capacity of such integrated systems and the total expected packet delay. An upper-bound for the average rescheduling delay was also obtained.

5.2 Suggestions for Further Work

The results of the work reported herein represent important contributions towards the objective of achieving an understanding of the issues and problems in digital integrated voice and data mobile radio systems. The following is a summary of the areas requiring further research:

* Investigation of the characteristics of mobile radio channels in the presence of fading, shadowing and impulsive noise by means of an experimental as well as a computerized fading simulator.

* Determination of the statistical distributions of the main variables of the fading channel, for mobile systems using diversity techniques, with a view to obtaining analytical expressions for the packet error rate.

* Study of the throughput degradation of packet radio channels in the presence of shadow fading and impulsive noise.

* Determination of the optimal packet size for fading channels in the presence of impulsive noise. Besides a thorough investigation of error control techniques, particular emphasis should be given to the synchronization problem and its impact on the optimal packet size.

* Theoretical analysis of the feasibility and cost effective implementability of alternative multiple-access protocols for packet speech and voice/data mobile systems.

* Investigation of the spectrum efficiencies of cellular structures for multiple-access radio fading channels carrying speech and data packets.

In the context of packet speech and voice/data integration, a number of experimental research areas are open for further research. These can be summarized as follows:

* Construction and testing of a laboratory model of a mobile channel simulator that could be used to assess the impact of fading, channel interference and impulsive noise on the transmission of speech packets.

* Subjective testing of the quality and intelligibility of voice communications over such channels. This study entails in particular, a thorough examination of cost-effective voice encoding schemes and digital modulation techniques. Particular emphasis should be given to the possibility of using forward error correction in speech packets and its impact on voice quality. To meet the security requirements in certain applications, the influence of encryption techniques on voice quality should be assessed.

* The speech interpolation efficiency of a packet speech network depends to a large extent on the reliable detection of the talkspurt and silence gaps. This requires the availability of an efficient speech detector that can accurately detect the presence of speech in a noisy environment. Several speech detectors should be implemented and tested in the presence of simulated background noise characteristic of land mobile systems.

* In a packet speech radio network, one of the crucial design parameters is the time required to sense and switch the carrier to a free channel. It is desirable to keep such time short compared to the time required to transmit a packet. Experimental studies are required to design and implement a frequency synthesizer that satisfies these requirements.

* Coherent FM signals and coherent reception provide a compact RF spectrum and low error rate, hence they are highly desirable in digital radio networks. However coherent reception of a digital packet entails that a fast synchronization scheme is implemented whereby the carrier and clock references are accurately and rapidly acquired. Studies of this particular aspect of packet radio will rely on novel hardware circuit designs.

APPENDIX A

CELLULAR STRUCTURES

A.1 Introduction

A conventional land-mobile system, illustrated in Figure A.1 uses a number of fixed transmitter stations, called base stations, which are typically located well above the ground level. Current regulations on effective radiated power limit the coverage area of a given base station operating in the VHF band to about 3000 km^2 with a radius of operation of about 30 km.

In practice the coverage area is determined from the distance at which the signal-to-noise ratio would drop below a certain specified level. For example, in the case of two mobile systems using the same frequency it can be easily shown that the required separation between the two base stations must be of the order of 100 km. This spacing is of course power dependent and could be reduced if the transmitting power were to be reduced.

Very simply put the cellular system concept is based on this fact, namely that the same frequency (channel) can be reused within a given area by decreasing the radiated power of the corresponding base stations. It is however important to realize that the traffic carrying capacity of a channel, defined at some acceptable blocking probability would remain the same within the coverage area. The payoff in improved overall network traffic handling comes from this multiple reuse of the same channel.

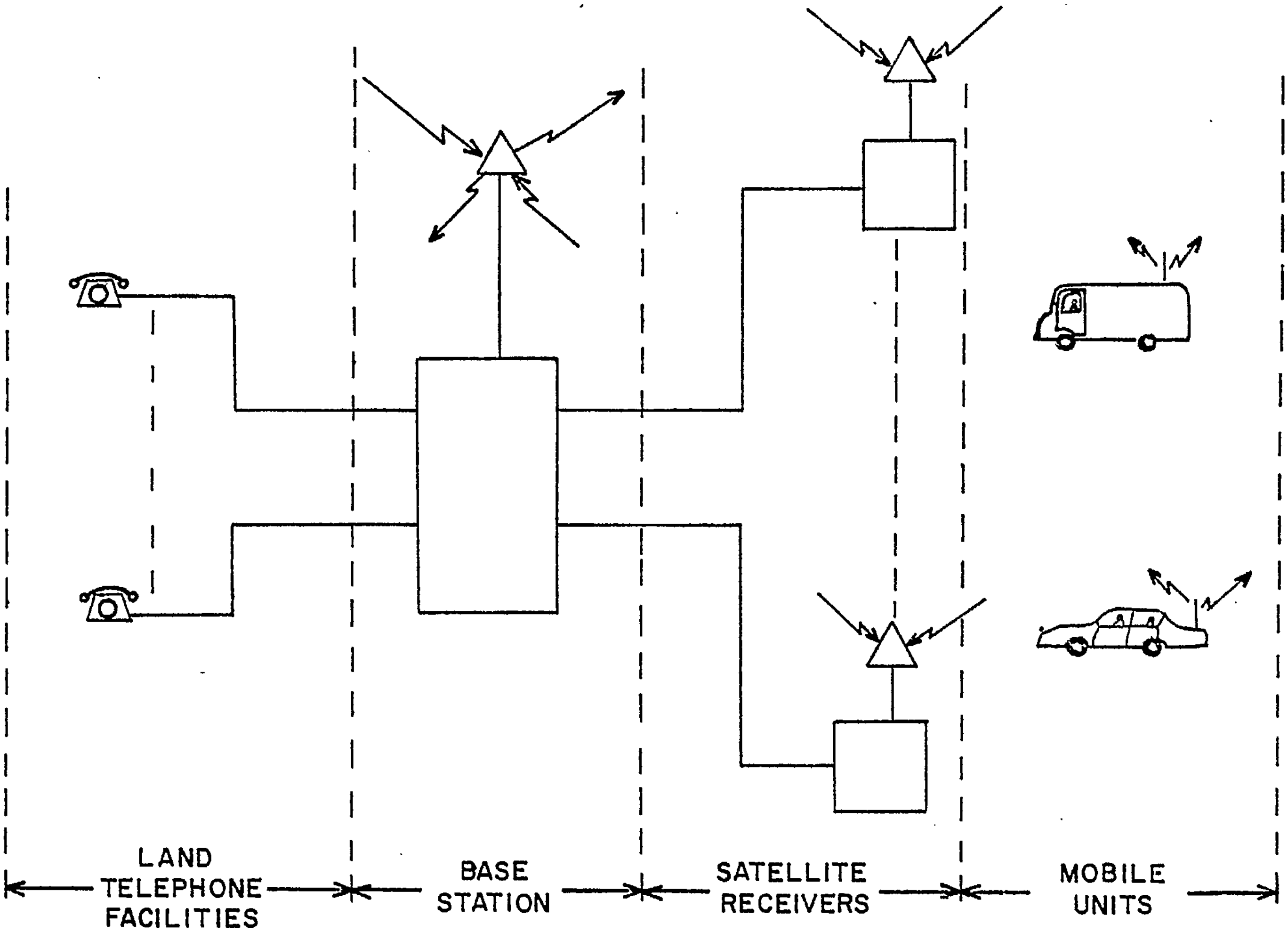


Figure A.1 GENERAL MOBILE RADIO NETWORK

In this appendix we will review the concepts attached to the so-called cellular structures and describe the various techniques that have been proposed to assign channels or frequencies to the various cells within a cellular system. The main reason why we have chosen to review some of these concepts, stems from the fact that cellular structures are a natural architecture for packet radio systems on mobile data communication systems in general. It is the author's opinion that the merging of the packet radio and the cellular technology, will definitely contribute towards solving future spectrum congestion problems.

A.2 The Cellular Structure

The boundaries of each cell are dependent upon many factors such as terrain, antenna heights and the distance to adjacent cell sites. Only three geometric shapes can be used to subdivide a given metropolitan area into many non-overlapping cells (see Figure A.2). However, for analysis and modelling purposes, cells are assumed to be hexagonal in shape, since the hexagon best approximates the RF pattern given by an omnidirectional antenna. Some planned systems will be using 60° or 120° directional antennas in addition to omnidirectional antennas. For those cases where directional antennas are in use, triangular cells could be used to approximate the antenna radiated pattern.

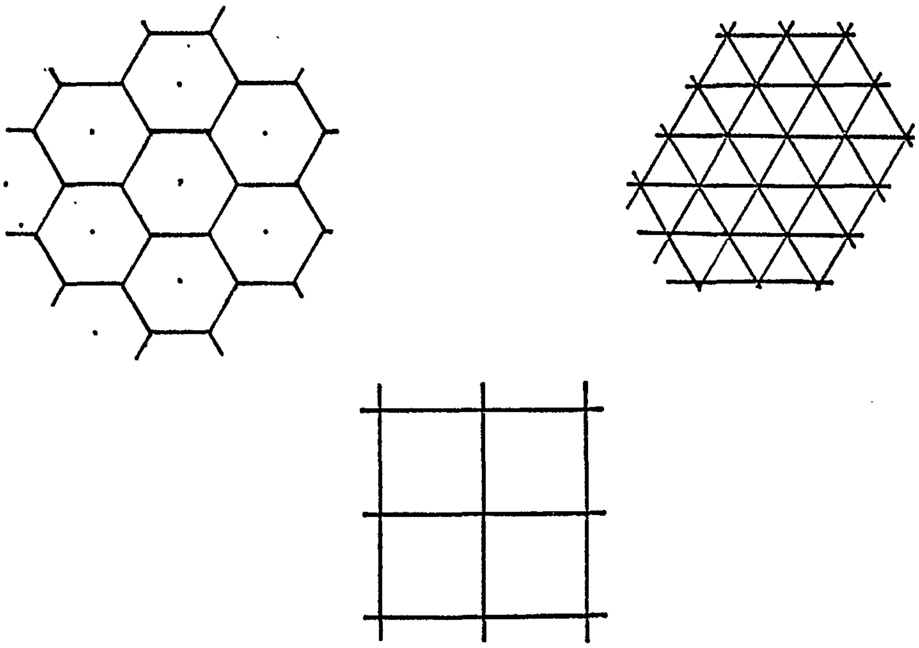


Figure A.2 GEOMETRIC SHAPES FOR CELLULAR SYSTEMS

In comparing cellular network arrangements, two parameters are used. These are:

- the so-called reuse ratio D/R which is the ratio of the distance between co-channel base stations transmitters to the cell radius, and
- C the number of distinctly unique sets of frequencies.

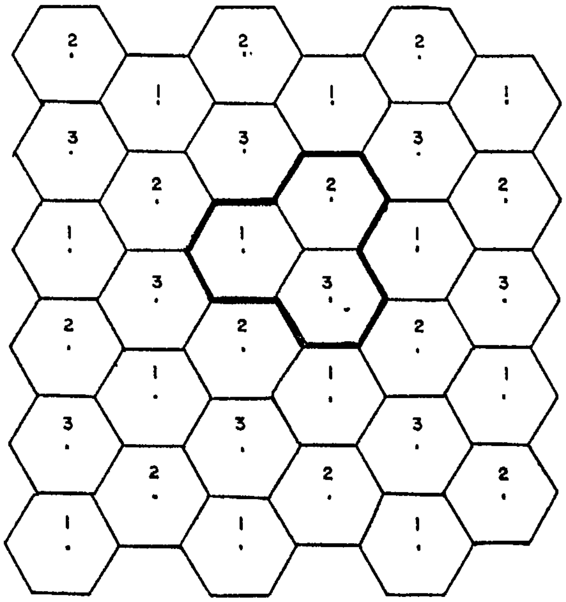
For hexagonal cells, it can be shown that these are related as follows:

$$\frac{D}{R} = \sqrt{3C}$$

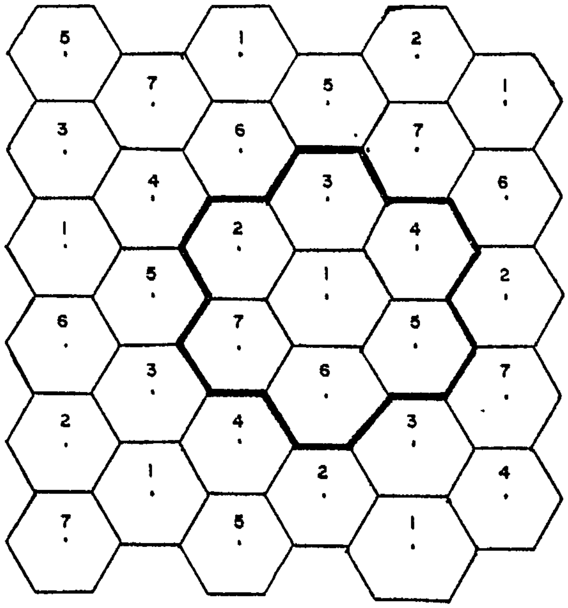
where C can take only the selected values:

$$C = 3,4,7,9,12,13....$$

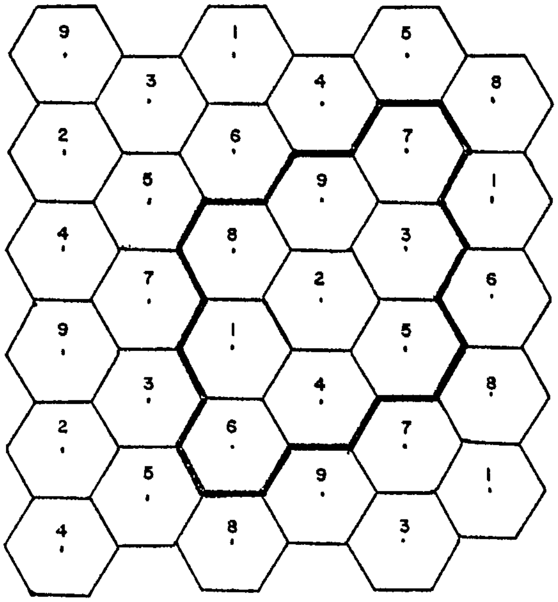
Figure A.3 shows the four cellular structures that are obtained by letting C = 3,7,9 and 12. Each cell has a set of distinct frequencies that will not interfere with frequencies assigned to adjacent cells. The frequencies (channels) within each cell are used for communications between the base station (which is connected to the telephone network) and the mobile terminals. Also shown in dark lines in Figure A.3 are arrangements of the different sets of frequencies among the cells. By repeating this pattern, the same set C or frequencies can be reused to cover a given area.



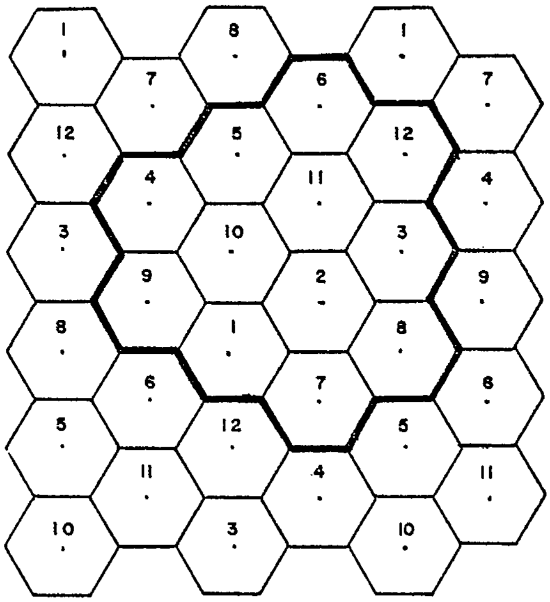
•CELL-SITE LOCATION
C = 3 , DIR=3



•CELL-SITE LOCATION
C = 7 , DIR=4.6



•CELL-SITE LOCATION
C = 9 , DIR= 5.2



•CELL-SITE LOCATION
C=12 , DIR = 6

C = no. of Channel Sets
R = "Radius" of cell
D = Distance between co-channels

Figure A.3 CELLULAR STRUCTURES FOR C= 3, 7, 9, 12

The actual values of D and R are functions of blocking probability, the number of available frequencies and traffic characteristics. However, it turns out from analysis that D/R, the minimum co-channel reuse ratio, is a function of the signal-to-noise and carrier to interference ratios. The minimum co-channel reuse distance D brings about a cluster of cells known as an Interference Cell Group. As indicated in Figure A.4 for a given D and R there is a minimum number of channel sets required to provide service to the system.

The traffic handling capability of a given cellular network can be greatly increased by a technique called splitting. This involves the addition of new cells midway between existing cells reducing the transmitter power and, hence, the coverage areas, while retaining the same co-channel reuse ratio. This is the rationale behind the "remote control of transmitter power" feature that is being built into the Motorola system.

In order to manage a cellular network a number of control channels must also be provided. These control channels

- assign the specific pair of frequencies (duplex channel) whenever a mobile terminal originates a call
- locate the given mobile terminal that is being called (called Paging)
- reassign frequencies as required to maintain communications as mobile terminals move from cell to cell (called Hand-off).

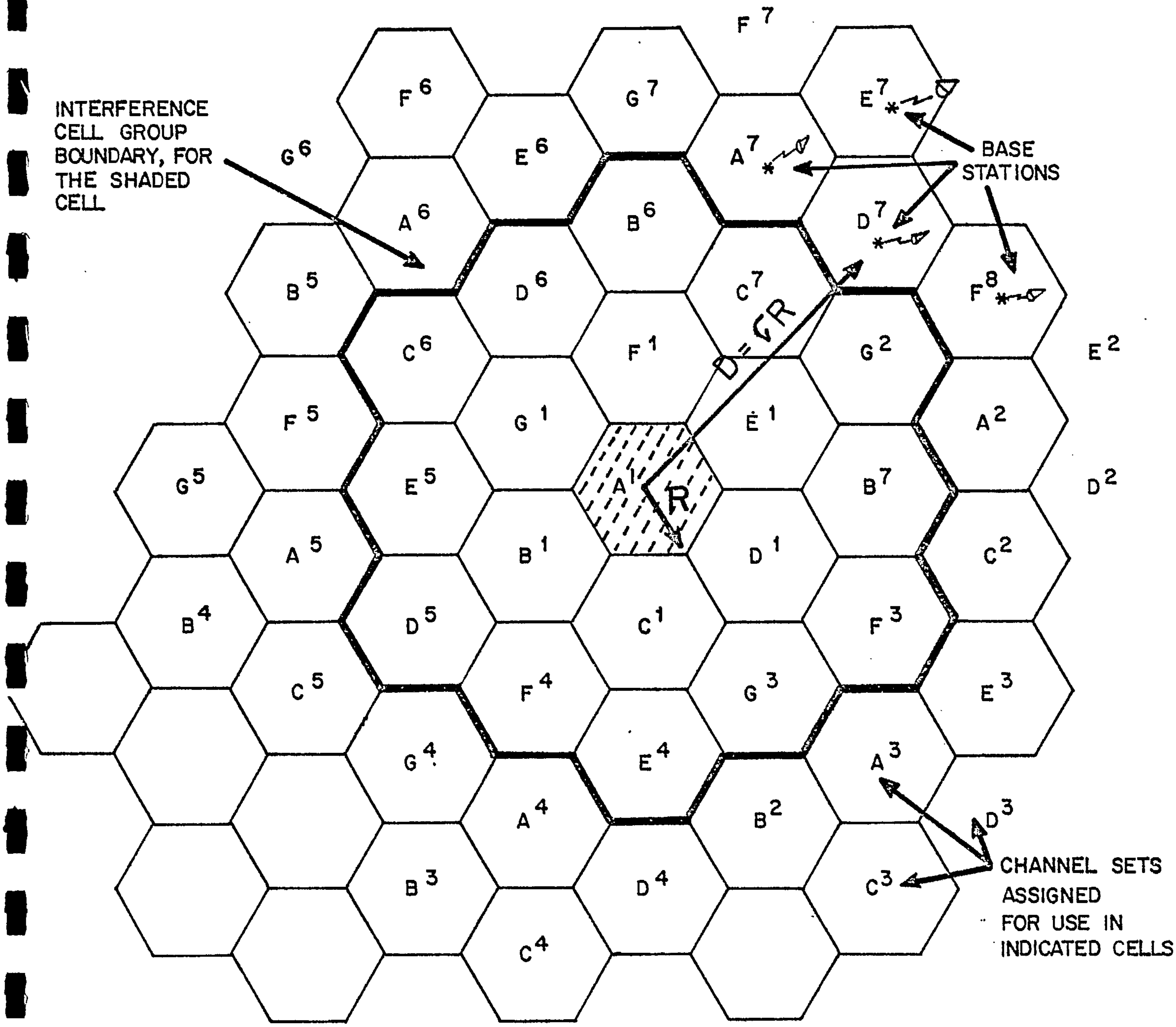


Figure A.4 CELLULAR SYSTEM LAYOUT

Control channels are therefore known as either paging or access channels, and both types operate in a digital mode. The operation is as follows.

When a particular mobile terminal wishes to place a call, it sends a request to do so on the uplink access channel. This request is decoded by the base station which then makes available a duplex channel, and the mobile terminal is so notified via the downlink access channel. In the case of a land originated call, the exact cell location of the desired mobile station is unknown. A paging (locating) signal, must therefore be broadcast by base stations in all cells over the downlink paging channel*. All mobile terminals have a distinc distinct address and automatically "tune in" (listen) on the paging chan channel when not in use. Thus, the mobile terminal being sought is readily located and responds over the uplink paging channel. This digital response signal is detected by the closest base station in the cell in which the mobile is located and the channel assignment procedure mentioned above is repeated.

Since adjacent cells do not use the same radio channels, as an active mobile unit crosses a cell boundary a new radio channel must be quickly assigned. Several schemes have been proposed to overcome the hand-off problem. One scheme requires that all active mobile terminals

*In actual practice, in order to avoid interference between transmitters located in each of the cells, a set of different and distinct paging channels is required. The mobile terminals must then scan all of these paging channels when not in use. To reduce the total number of required paging channels, the assignment of these channels is made on the basis of paging areas comprising groups of cells.

continually transmit their identity over the paging channel so that the closest base station can ensure that a channel is always assigned. Another scheme, which reduces the uplink paging channel traffic, requires that mobile terminals transmit their identity only when there is a change in identity of the base station that they are communicating with.

A.3 Channel Assignment Schemes

Central to the design of a cellular network is the choice of the channel assignment scheme. This choice involves specifying the set of channels and how they should be assigned at each given point in time so as to minimize the probability of a call being blocked or in other words to maximize the traffic carrying capacity of a system for a given acceptable grade of service and channel reuse ratio. Channel assignment schemes have been actively investigated {88},{89},{90} and fall into three classes:

- static assignment schemes
- dynamic assignment schemes
- hybrid assignement schemes

Static Assignment Schemes

As the name implies, these involve a time invariant allocation of specified sets of frequencies to cells. As discussed above, the required number of sets of frequencies is a function of the co-channel reuse ratio. It is easy to see that the main drawback of static schemes is that a call originated by a mobile terminal will be blocked if all channels within its cell are currently in use even though neighboring cells might have idle channels.

The probability that a call is blocked* in a cell having c channels can immediately be calculated using the Erlang-B formula:

$$B(c,\rho) = \frac{\rho^c/c!}{\sum_{n=o}^{c} \rho^n/n!}$$

where "ρ" is the traffic offered to this group of c channels. The total traffic carried by a given cell is therefore

$$\rho^1 = \rho[1 - B(c,\rho)]$$

The total traffic carried by a cellular system with L cells employing a static allocation scheme and M channel sets is readily obtained from the above equations.

Dynamic Assignment Schemes

Static assignment schemes are efficient as long as the traffic being generated by the terminals is uniformly distributed across the system. In actual practice, this will not be true, since the traffic being generated in any cell will be a function of the density of mobiles in that cell and this varies with the geographical location of the cell.

* assuming that the traffic being generated is Poisson.

In order to overcome the problems arising from the location dependent traffic patterns, dynamic channel assignment schemes have been proposed to manage and reallocate the frequencies in cells on the basis of the stochastic fluctuations occurring in the traffic.

All of these dynamic management schemes work on the basis that currently unused frequencies can be assigned to any of the cells provided that the actual total number of channels, etc. is not violated. Within this framework, a variety of strategies {91},{92},{93},{94}, have been studied, and each includes criteria for determining which channels of the available should be assigned.

Closed form analytical models of dynamic assignment schemes have proven elusive and difficult. Simulation is still our only tool for studying the performance of these schemes. In this regard, one interesting result obtained by simulation is that "random channel selection for re-assignment" strategy will always lead to traffic handling results comparable to those of any of the more exotic dynamic assignment schemes that have been studies so far.

Conceptually, the problem of computing the blocking probability in a given cell under dynamic assignment involves determining all system states that satisfy the co-channel reuse distance constant. From a computational point of view, this becomes an intractable problem as the number of system states increases approximately as a factorial function of the number of cells in the network.

As has happened so many times in the past, when combinatorial problems has presented computational difficulties, Graph Theory can be applied. In particular, Sengoku {95} gives a graph theoretical method for finding the system blocking probability.

This method first involves determining the probability P(i), that a given channel is used in "i" cells simultaneously, as a function of the system offered load ρ from the following result:

$$P(i) = \frac{(\rho/L)^i \, \nu_i}{\sum_{j=o}^{k} (\rho/L)^j \, \nu_j}$$

where each ν_i is the number of i-cliques of the graph obtained from the given cellular system by connecting those pairs of nodes (cells) which are not adjacent to each other, and k is the largest number of cells in which a given channel can be simultaneously used without causing interference.

Once P(i) has been determined, the average amount of traffic a_c that can be carried by the network can be computed as:

$$a_c = \sum_{i=1}^{k} iP(i)$$

and the average system blocking probability would therefore be:

$$P_B = 1 - \frac{a_c}{\rho}$$

Sengoku {95} also proposed a dynamic channel assignment algorithm based on the method of cliques. The principle behind this algorithm is that it attempts to minimize the system blocking probability by selecting for assignment a channel which minimizes the decrease of cliques. This, however, involves extensive computation and it appears that the algorithm could not be implementable in a practical cellular system. There is, however, little doubt that it generates good channel assignment schemes and, as such, could be used as a basis against which other heuristic schemes are evaluated.

The research to date indicates that for low blocking probabilities, dynamic schemes would give better traffic handling performance than static schemes. However, for very high blocking probabilities, the opposite is true. The best compromise seems to be a combination of these schemes, and this brings us to what are known as hybrid assignment schemes.

Hybrid Assignment Schemes

These assignment schemes attempt to take advantage of the best features of static and dynamic schemes. They all operate by assigning some channels for dynamic assignment to cope with the statistical fluctuations of traffic.

The problem then involves determining the split of available channels. A number of specific cases have been studied. Cox {89} for example, studied the case of 10 channels available for assignement and found the optimum division to be 8 static channels and 2 dynamic channels.

In a recent paper Kahwa {90} simulated several hybrid assignment schemes under an Erlang B service discipline and provided some guidelines as to how channels should be split. Briefly, his results indicate that hybrid schemes are always superior to static ones if traffic load peaks up exceeding average load by as much as 50% are present in any of the cells. More recently Sin {96} undertook a simulation of a cellular system employing a hybrid channel assignment scheme with an Erlang C service discipline. Figure A.5 gives a comparison between the Erlang B and Erlang C service disciplines. For light loadings, systems with a high number of dynamic channels give the best performance. As the load is increased the number of dynamic channels should be reduced until all channels are assigned on a static basis.

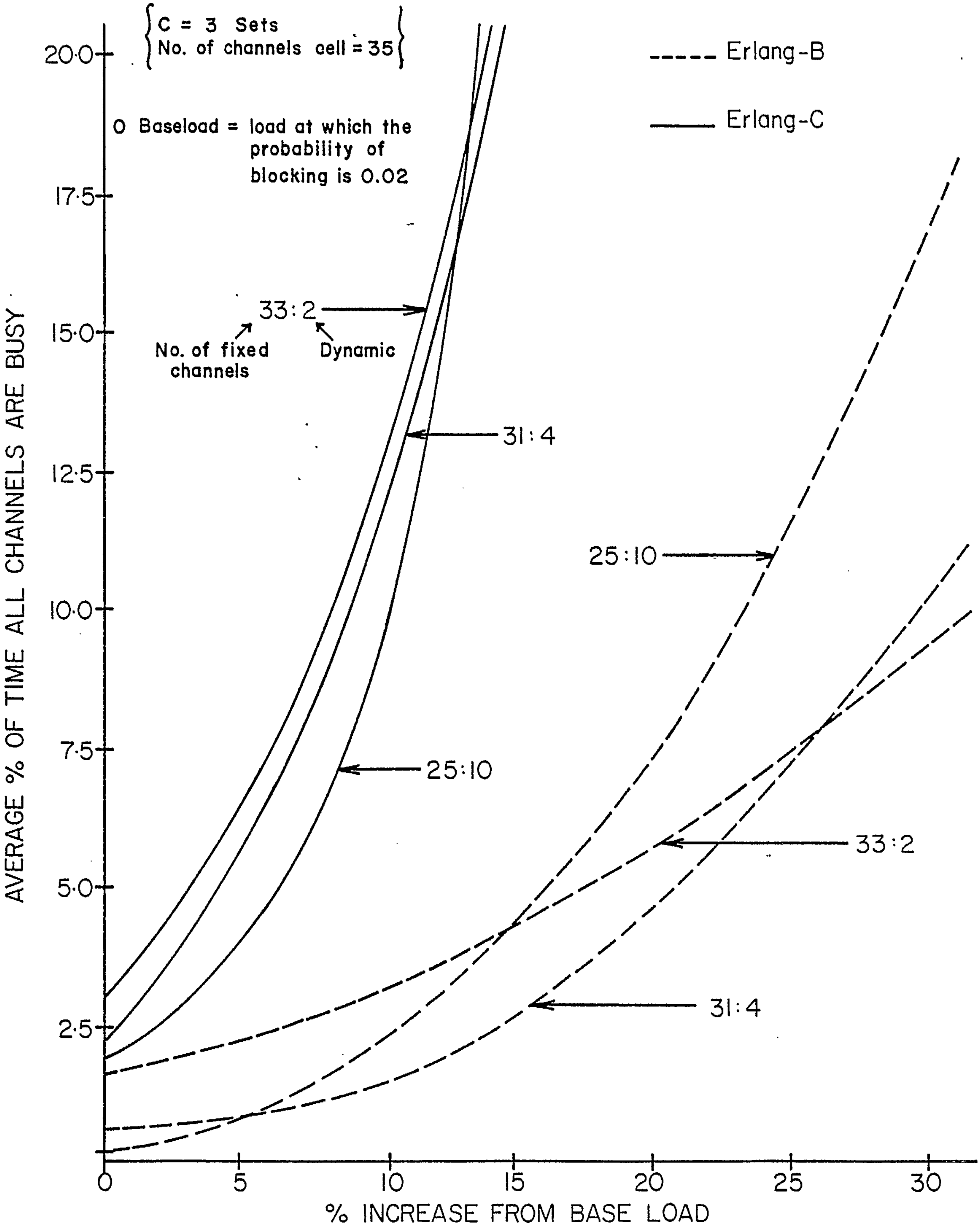


Figure A.5 PERFORMANCE OF CELLULAR SYSTEMS UNDER HYBRID ASSIGNMENT SCHEMES

APPENDIX B

PACKET RADIO RANDOM ACCESS SCHEMES

B.1 Introduction

When one considers an environment where a large population of low traffic users wishes to communicate with each other either directly or through a central station, an immediate question can be raised regarding the method by which the communication channel or channels will be allocated. In the context of radio communication channels, three main channel allocation methodologies namely; fixed, dynamic and random, can be investigated. FDMA and TDMA are examples of fixed allocation schemes, while ATDMA would correspond to a dynamic or demand based allocation method. Schemes such as FDMA or TDMA, which can typically be implemented in a distributed control environment, have been shown to be wasteful of bandwidth. The third main class of allocation schemes, namely random access, has on the contrary been shown to be extremely spectrum efficient.

In this appendix we will review the classical random access schemes and briefly comment on the various reservation protocols that have been proposed in the literature. Some of the problems and relevant issues that can be raised in the context of packet radio networks, will also be given.

B.2 THE PURE ALOHA RANDOM ACCESS SYSTEM

The pure ALOHA scheme {97},{98}, appears to have been the first random access scheme specifically designed to optimally share a radio communication channel among a large number of data terminals. In this scheme terminals transmit at random data packets (of fixed length) over a high speed radio channel, without consideration of the transmissions that are being made by other terminals using the same channel. In those cases where packet transmissions overlap (collide), no acknowledgement is received at the transmitting terminal and the transmission is repeated after a period of time of random duration. This retransmission process is repeated until a successful transmission and acknowledgement occur or until the process is terminated by the terminal.

In the pure ALOHA scheme, only packets received without error are acknowledged. To prevent collisions between packets and their acknowledgements the downlink radio channel from the central station to the terminals, is not the same radio channel that is used for uplink communication.

Finally, the literature has concentrated on packet overlaps as the only source of packet errors. However, in a land-mobile environment errors in packet transmission are due to ignition noise, fading and shadowing.

Capacity of the Pure ALOHA Channel

In order to describe this result assume that the total traffic offered to a radio channel by a large population of terminals, consists of newly generated packets as well as previously collided packets.

Define the channel input rate S, as the average number of new packets generated per packet transmission interval, T, and the channel traffic rate G as the average number of new and retransmitted packets per packet transmission interval. Then under steady state conditions S, becomes equal to the channel throughput rate. The expression for the maximum achievable value of S, was first shown by Abramson {99} to be:

$$S = Ge^{-2G} \qquad \text{..... (B.1)}$$

A plot of this equation is shown as Figure B.1. In particular, as shown in this figure, maximum packet throughput is achieved at G = 1/2 and is equal to 1/2e ≃ 0.184 of the total capacity of the channel. The ratio S/G can also be interpreted as the probability of a successful packet transmission while (G/S - 1) is the average number of retransmissions required before a packet is successfully received.

Delay

If we define the expected packet delay D as the average time from when a packet is generated until it is successfully received the following expression is obtained for a pure ALOHA channel.

$$D = \left[\frac{G}{S} - 1\right] R + 1 + \beta \qquad \text{..... (B.2)}$$

where R is the average delay between two consecutive packet transmissions.

Assuming that acknowledgement packets are always correctly received, and denoting by α the transmission time of the acknowledgement packet (measured in units of packet transmission times) we obtain:

$$R = 1 + 2\beta + \delta + \alpha \qquad \text{.....(B.3)}$$

where β and δ are respectively the one-way propagation delay, and the average value of some specific retransmission time probability distribution. Both β and δ are expressed in units of packet transmission time.

It should be kept in mind that the above equation does not include the time required to error check packets or the time taken to generate acknowledgement packets. Furthermore it is easy to see that the normalized propagation delay β will typically be a very small fraction of the packet transmission time.

Intuitively it should also be clear that, δ, should not be chosen too large since this will produce a delay that is also large. If, on the other hand, δ is too small, interference (collisions) increase and so does the channel traffic G, and the channel degrades. Fortunately, it has been shown {86} that for any particular value of S, an optimal δ exists that will minimize the delay per packet for any given retransmission delay probability distribution function.

Remarks

Despite the relatively low channel utilization that it achieves (of the order of 18% of capacity) the pure ALOHA random access scheme is attractive because of its inherent simplicity and low implementation cost. However, the major limitation of pure ALOHA (as well as slotted

ALOHA channels) is their instability as traffic increases. In this regard, Lam {86} has shown that as a result of the stochastic fluctuation in channel input, channel saturation can occur causing the channel to become congested with collisions and retransmitted packets. Under these conditions, the real traffic actually passing through the system falls off dramatically. For example in Figure B.1, the channel throughput vanishes to zero whenever the traffic offered to the channel exceeds the value of 0.5 of its capacity.

In order to deal with this instability condition various dynamic control procedures have been proposed {100} {101}. Essentially, these procedures minimize the occurrence of instability by requiring that each terminal take appropriate action (such as, cease transmission) to prevent channel saturation as the unstable region is being approached.

The above results for the pure ALOHA scheme apply only in those cases where each message is transmitted as a fixed size packet(s). In cases where messages are made-up of variable length packets, analysis techniques based on message switching are used. For example, Ferguson {102},{103} has developed a model for analyzing the traffic handling of radio networks where messages are variable length packets. His analysis shows that the throughput of a packet radio network with fixed packet lengths is greater than the throughput of a network employing variable length packets.

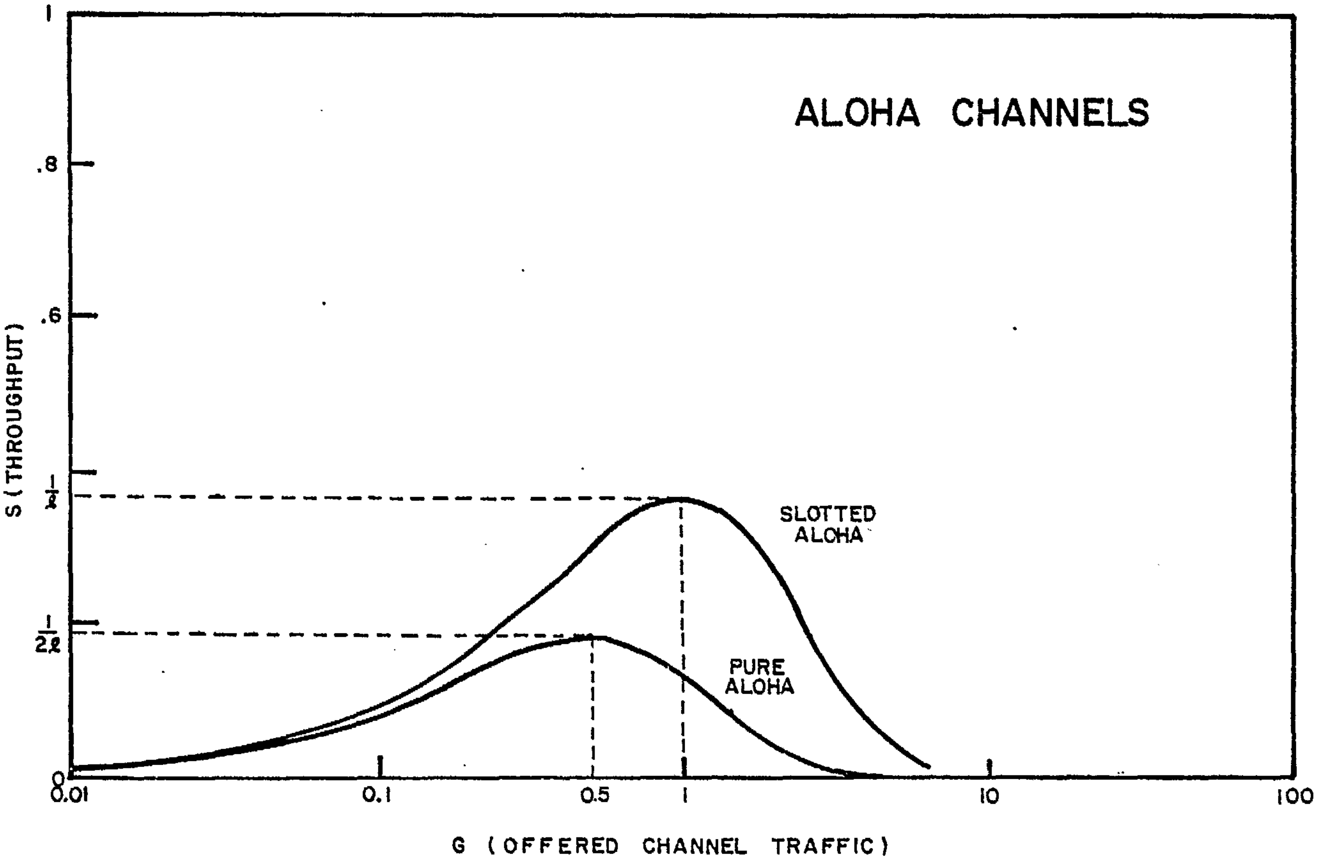


Figure B.1 PERFORMANCE OF ALOHA CHANNELS

B.3 THE SLOTTED ALOHA RANDOM ACCESS SCHEME

This scheme is directed at reducing the period of time (called) vulnerable period) during which packets can collide. To achieve this reduction, Roberts {104} suggested that terminals should be synchronized in some sense. This synchronization is necessary to produce what is called slotting of the time axis. These slots or segments of time axis are equal to the transmission time of a packet of constant length. It is then easy to see that if slotting can be achieved on an instantaneous basis for each packet then, the period of time for which a packet would be vulnerable would only be ½ of that of the pure ALOHA case. This is because all terminals would be required to transmit their packets at the beginning of a slot, and therefore, only total packet overlaps can occur. Apart from this modification, the operation of the slotted ALOHA is identical to that of the pure ALOHA and is therefore not repeated here.

Capacity of the Slotted ALOHA Channel

Since collisions between packets in the channel can only occur during a period of time equal to one slot time, the probability of collision is reduced and as shown by Roberts {104} the throughput equation for the channel becomes:

$$S = Ge^{-G} \qquad \text{..... (B.4)}$$

Again, as shown in Figure B.1, the maximum channel throughput using the slotted ALOHA scheme is achieved at G = 1 and is equal to 1/e=0.368 of the channel capacity. This, as expected, is twice the throughput that the pure ALOHA random access scheme would achieve on the same channel.

Delay

Defining the average delay D incurred per packet (measured in units of packet transmission time) as a function of the packet transmission time, the average retransmission delay and the propagation delay β, we have*:

$$D = \left(\frac{G}{S} - 1\right) \left(\beta + \frac{k+1}{2}\right) + 1 + \beta \qquad \ldots\ldots \text{(B.5)}$$

where k represents the maximum retransmission delay expressed as an integer number of k slots or packet transmission times.

Lam {86} proved that for each value of S, an optimum value of k could be selected which would minimize the packet delay.

Remarks

The throughput of slotted ALOHA is twice that of pure ALOHA. However, when the traffic being offered to the channel is close to the theoretical maximum throughput, the channel saturates rapidly and the throughput of actual message packets falls off dramatically. One serious limitation of the slotted ALOHA scheme is the requirement that all users be synchronized, and that the slotting of packets takes place "instantaneously". This is a non-trivial implementation problem.

---

*Note δ (random retransmission delay) $= \frac{k+1}{2}$ for the case of the uniform distribution for random retransmission time.

B.4 CARRIER SENSE MULTIPLE ACCESS SCHEMES (CSMA)

In the previously described schemes, the major factor affecting throughput was the collision of packets that were being transmitted by various terminals unaware of the presence of each other.

To reduce the impact of this limitation, a number of authors {83 },{105},{106} suggested that terminals should first "sense the carrier" (listen to the channel) prior to transmitting. This scheme, which is called the CSMA scheme, assumes that the propagation delay between two terminals is a small fraction of the total time taken to transmit a packet. When this is not the case, the information upon which the terminal bases its decision whether or not to transmit could lead to incorrect decisions.

A terminals "sensing the channel" can employ any of three variations of the CSMA scheme. These are 1-persistent CSMA, p-persistent CSMA, and the non-persistent CSMA. Each of these CSMA schemes differ in the action that a terminal that wishes to transmit takes after sensing the channel. However, in all three cases, the retransmission delay that follows an unsuccessful packet transmission is selected from some retransmission delay distribution. Finally, it is possible to construct slotted as well as unslotted versions of these three CSMA schemes.

The 1-Persistent CSMA Scheme

This scheme operates as follows:

Whenever a terminal has a packet ready for transmission, the channel is "sensed" and if it is idle (no other packet in it), then the terminal transmits its packet. If the channel is sensed busy, then the terminal keeps sensing the channel until it determines that it is idle and then transmits its packet.

The p-Persistent CSMA Scheme

This scheme operates as follows:
Whenever a terminal has a packet ready for transmission the channel is "sensed" and if it is idle, the terminal transmits its packet with probability p, or delays by an amount equal to the propagation time with probability (1-p). If a delay has occurred, the terminal again senses the channel and repeats the above procedure. If the channel is sensed busy, the terminal reschedules its "sensing for transmission" sequence in accordance with the retransmission delay distribution, and operates as described above.

The Non-Persistent CSMA Scheme

This scheme operates as follows:
When a terminal has a packet ready for transmission it senses the channel and if the channel is sensed idle, the packet is transmitted. If the channel is sensed busy the terminal does not persist in sending the packet and simply schedules the transmission of the packet in accordance with the retransmission delay distribution. At this new point in time, the channel is sensed and the same procedure is repeated.

Channel Capacity

A complete discussion of the throughput - traffic characteristics of the three versions of CSMA is given in {83}. One important point to keep in mind is that while the throughput achieved by both the pure ALOHA and slotted ALOHA access is independent of the propagation delay, that this is not so for the CSMA schemes. For example, in the case of non-persistent CSMA Tobagi {83} showed that the throughput S was:

$$S = \frac{Ge^{-\beta G}}{G(1 + 2\beta) + e^{-\beta G}} \qquad \text{.....} \quad \text{(B.6)}$$

The graph of this equation is shown in Figure B.2. For a value of β close to one, that is when the propagation delay is comparable to the packet transmission time, it is readily seen that both pure ALOHA and slotted ALOHA are superior to CSMA.

Delay

For any of the three CSMA schemes, the average packet delay, D, is again a function of both the channel throughput S and the average re-transmission delay δ. For each value of S a minimum delay can be achieved by choosing an optimal value of δ. For the non-persistent protocol the average delay neglecting the propagation delay as given in {83} is:

$$D = \left(\frac{G}{S} - 1\right)\left(1 + \alpha + \delta\right) + 1 \qquad \text{.....} \quad \text{(B.7)}$$

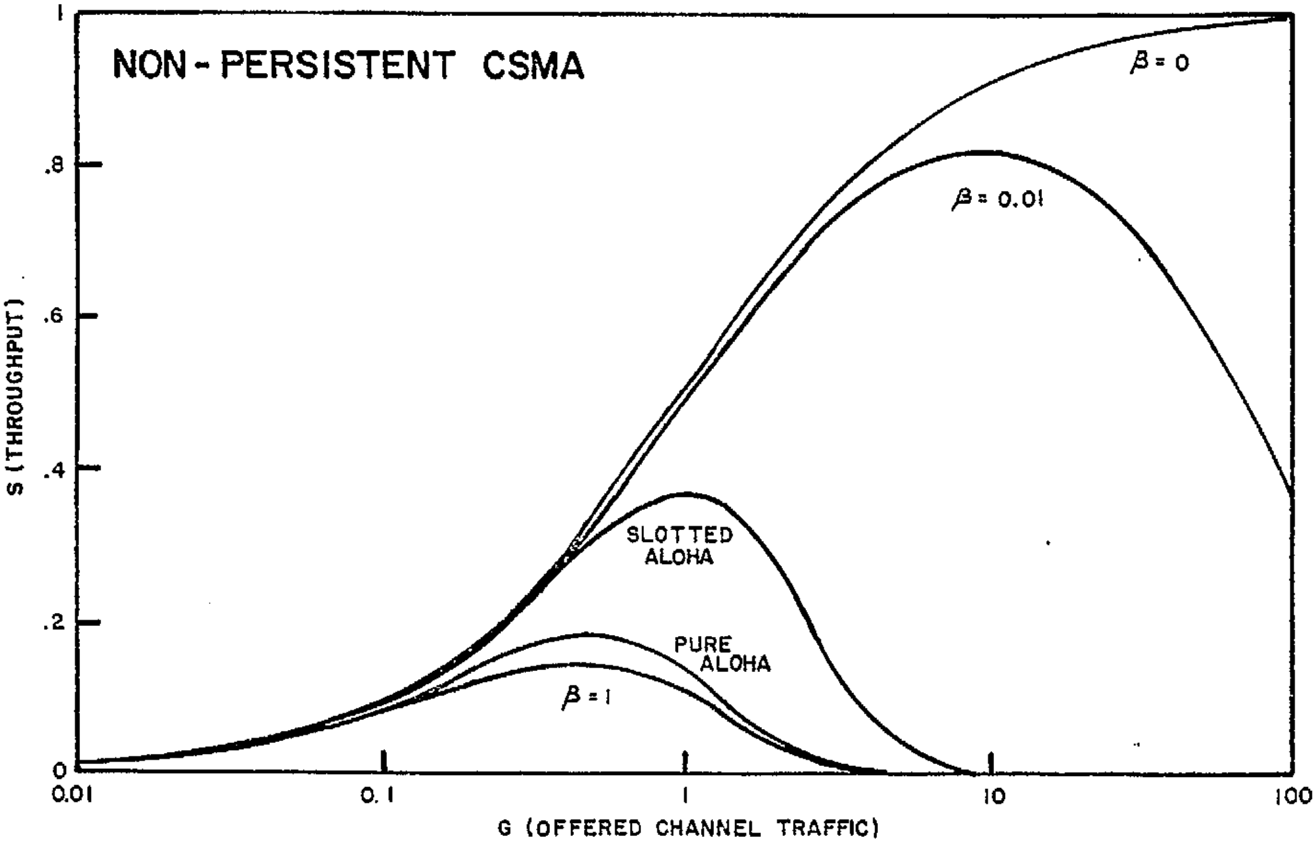


Figure B.2 PERFORMANCE OF NON-PERSISTENT CSMA

Remarks.

The various CSMA schemes do theoretically yield improved channel utilization when compared with the pure and slotted ALOHA ones. However, it must be remembered unless the propagation delay β is neglected the throughput vanishes for a large value of the channel traffic (i.e. as $G \rightarrow 1$). Also for finite values of β, the channel will drift into saturation and instability and dynamic control procedures must be invoked. In addition, the appealing simplicity of implementation of pure ALOHA is lost. Of these CSMA schemes only the non-persistent, non slotted CSMA appears simple to implement.

B.5 "RESERVATION" RANDOM ACCESS SCHEMES

A number of random access schemes have been proposed by various authors to satisfy in some optimal manner the particular environment in which they were designed to operate. Most of these schemes fall under a general class of so-called "reservation" schemes. In essence they work on the basis that a significant portion of the traffic is composed of multi-packet messages. Hence, rather than making an access request for each packet, one request is made per message. The central station maintains a queue of requests and informs the terminals of their position in the queue. Alternatively it is possible to envisage a distributed global queue where terminals would decide when to transmit. This entails however that reservation packets are correctly received by all terminals. Some of the various types of reservation schemes that are known will now be briefly described.

Reservation I

A description of one of the earliest reservation random access schemes is due to Roberts {107}. In this scheme, the channel is time divided into two sub-channels of which one is used, under the slotted ALOHA mode, for reservation packets while the other is operated in a dedicated mode for data packets. Hence, collisions can only occur on the request channel. Furthermore, no control by a central station is required, since it is assumed that all terminals can "hear" the successful requests for data slots. A similar reservation scheme proposed by Szpankowski {108} suggests that both the reservation and message channel could be divided into a number of slots.

Reservation II

In 1976 Tobagi {109} proposed a similar technique that required that the channel be frequency divided into two subchannels - one for control information and one for data. This technique known under the name split Channel Reservation Multiple Access (SCRMA) suggests that the control of the data channel instead of being distributed should be centralized. Tobagi further suggested that the control channel could be accessed using pure ALOHA, slotted ALOHA or CSMA.

Reservation III

Somewhat earlier, Crowther {110} suggested a reservation technique, known as R-ALOHA, which is based on a slotted ALOHA channel, in which a number of slots are grouped to form a frame. A terminal is allowed to transmit a packet during any one of the unoccupied frame slots.

A terminal "own rights" to the slot for a period of time until its message is successfully transmitted. However, no terminal is permanently assigned any of the frame slots, and must compete with other terminals for them. The performance of this multiple access scheme was subsquently analyzed by Lam {111} both through analysis and simulation.

Reservation IV

A variation of the above random access scheme, suggested by Binder {112}is based on a permanently assigned TDMA structure, i.e. each terminal is permanently assigned a slot in each frame. However, any other terminal using the channel can use the slot of a particular terminal if this terminal is idle. In essence, Binder's approach consists of a dynamic time-assignment system which retains the stability and allocation fairness of a fixed TDMA structure.

Reservation V

This scheme takes advantage of the fact that in slotted ALOHA channels the throughput can exceed 1/e if the terminals have different packet transmission rates. Based on this, Sastry {113} proposed a reservation technique that automatically allocates a number of transmission slots, say n, to a large user, whenever such user has managed to capture an available slot. Then, because of the broadcast nature of the nature of the channel, all terminals know that the large user has just transmitted a successful packet and thus refrain from transmitting in the next n slots.

Reservation VI

Another reservation technique known as Multi-Level Multi-Access (MLMA) has recently been proposed by Rothauser {114}. Terminals which are ready to send, enter their requests into a request slot which is just long enough to allow the identification of those currently active requestors. By using a "one-out-of N" code where N is the number of terminals, simultaneous requests for transmission space can be properly recognized.

Reservation VII

A reservation protocol designed to handle both packetized data and voice traffic, known as CPODA (Contention-based Priority Oriented Demand Assignment) was proposed by Jacobs {23} in the context of SATNET experiments. Similarly to Robert's scheme {107} the channel time is partitioned into a reservation subframe and an information subframe. Within the reservation subframe a number of slots are accessed on a contention basis by reservation packets. Channel time can also be reserved by piggybacking reservations in the header of scheduled message transmissions. While an explicit reservation is made for each data packet generated by bursty traffic sources a unique reservation is sent for stream type traffic characteristic of speech or facsimile. CPODA measurements and simulation results were reported by Chu {21}.

Reservation VIII

To cope with the long waisted time delays characteristic of satellite channels Balagangadhar {115} has recently advocated a new reservation based time division multiple access scheme, known as RMA. As in Binder's scheme the time axis is divided into frames, however in each frame there are a number of reservation minislots (one minislot per terminal), followed by preassigned time slots (one slot per terminal) and a variable number of reservation access slots. At the beginning of the frame all terminals transmit their reservation requests on their dedicated minislots. After a round trip delay the vector of reservation demands is received by all terminals who then sequentially transmit on their assigned access slots.

Reservation IX

The SRUC (Split Reservation Upon Collision) technique, proposed by Borgonovo {116} is a hybrid contention-reservation scheme in the sense that a reservation scheme is invoked when contention occurs. Frames are divided into a number of slots with clusters of terminals being assigned one slot in every frame. Every slot within a frame is further partitioned into a data subslot and a signalling subslot. Terminals access the data subslots according to a specified access protocol which relies on a channel control procedure. An evaluation of the performance of SRUC is conducted and a comparison with other access techniques for both ground systems and satellite based systems, shows that SRUC has some advantages in particular for high channel utilizations. From an implementation point of view the SRUC technique appears to be quite simple.

B.6 OTHER RANDOM ACCESS SCHEMES

We will now briefly discuss some representative random access schemes that do not belong to the class of "reservation" techniques.

Pure ALOHA Variation

A modification of the pure ALOHA access scheme proposed by Tasaka {117} is claimed to achieve a higher throughput than pure ALOHA. Such improvement is obtained by using signal processing techniques whenever two or more packets overlap, to recover the "least destroyed" packet. The author believes that these results are of theoretical interest only at this time.

Schemes for Low Number of Users

When the number of terminals (possibly buffered), that compete for a given radio channel, is very low (≤ 20), Scholl {118} showed that a number of collision free schemes were actually more effective than CSMA in terms of their delay-throughput characteristics particularly in the case of heavy traffic. For channels with about 50 buffered user terminals a technique known as Mini-Slotted Alternating Priorities (MSAP) was shown to provide an almost perfect scheduling. The MSAP technique presents the additional advantage of not requiring the control from a central station.

An Optimal Adaptive Scheme

This scheme proposed by Yemini {119} is the first attempt to solve the problem of distributed control of a slotted channel that is shared by a number of terminals. The scheme gives better throughput performance than either the optimally controlled slotted ALOHA (for light loads), or the TDMA for heavy loads. It would be difficult and costly to implement and remains of theoretical interest only at this time.

Group Random Access

It may sometimes be desirable to dedicate in time certain portions of the channel to a specified group of network terminals selected on the basis of priorities or traffic characteristics. The Group Random Access (GRA) technique studied by Rubin {120} provides a specific group of terminals with a periodic sequence of channel access periods during which the group uses a random access scheme. Other groups of terminals share the remaining time available on the channel using similar or different random access schemes.

Generalization of CSMA

Recently, Hansen {121} suggested a generalization of the various carrier sense access techniques proposed by Tobagi {83}. The main difference between the two schemes is that Hansen proposes a frame of sensing slots with groups of users being assigned a specific sensing slot on every frame. After listening to its own sensing slot the user follows the non-persistent CSMA protocol. By a careful selection of M the number of users who are assigned to each sensing slot, Hansen suggests that substantial improvements in delay-throughput performance can be obtained. A further extension {122} of the above scheme, reported by the same author, includes the random assignment of channel users to sensing slots. No improvement over the fixed assignment of sensing slots can be obtained when M is optimized.

Framed ALOHA

A variation of the slotted ALOHA concept, known as framed ALOHA has been proposed by Okada {123}, in particular for satellite channels. A slotted frame containing L slots is available to a finite number of users, each user being allowed to send at most one packet per frame. While the average throughput of the framed ALOHA remains unchanged with regards to slotted ALOHA, the variance of the throughput is reduced by a factor L where L is the number of slots per frame.

The Broadcast Recognizing Access Method

This decentralized random access protocol, applicable to radio channels or coaxial cable buses is based on the ability to recognize the identity of the terminal associated with a successful transmission, and relies on the use of a scheduling function which defers the packet transmission epochs by at least the channel propagation delay. Four variants of the basic protocol are discussed by Chlamtac {124}.

Multiple Access Tree Protocols

These tree protocols, originally formulated by Capetanakis {125}, {126} were followed recently by a new access protocol that according to Meubus {127} substantially improves the original Capetanakis' scheme. In essence tree protocols are based upon assigning the traffic sources to the leaves of a free. Whenever collisions occur the tree is searched and the conflicting traffic sources are progressively partitioned so as to reduce the probability of collision. Ultimately all traffic sources are partitioned into separate sets, each set containing at most one active terminal at which point in time collision free transmissions can take place. Both finite and infinite user populations have been considered by Capetanakis, who shows that in the later case the maximum average throughput does not exceed 0.43.

Schemes for Processing Satellites

DeRosa {128} and Ng {129} were among the first authors to consider the benefits that could be gained by a processing satellite as opposed to the current satellites which can be viewed essentially as transponders.

Indeed in a random access technique such slotted ALOHA there is no need for the satellite to transpond back to the earth stations, collisions and unused slots. The bandwidth savings obtained, by an intelligent satellite, on the downlink channel, could be given to the uplink channel. In 1978 DeRosa studied a packet satellite network with n uplink FDMA channels accessed on a slotted ALOHA basis, one downlink TDM channel and no buffering at the satellite. Later on the same author extended his analysis to the case where the satellite is provided with a small buffer. Eaves {130} at about the same time extended the original results of DeRosa to include multiple TDM downlinks. He further showed how to optimize the selection of the number of downlink and uplink channels for a given bandwidth constraint. A further generalization of the DeRosa model was recently obtained by Molle {131} who refers to these channel configurations as concentrated ALOHA satellite links. He showed in particular that at the expense of increased satellite processing the average packet delays could be reduced. The above schemes are readily applicable to mobile communication systems since processing and buffering at the base station level presents no technical or economical difficulties.

B.7 PACKET RADIO NETWORKS

In the previous section, a number of random access schemes and associated delay - throughput characteristics were discussed. In the design of packet radio networks consisting of a large number of terminals, several repeaters and one or more central stations as shown conceptually in Figure B.3, the random access scheme is only one of the system design considerations. Other considerations of equal importance affect the overall system's capacity, the tolerable end-to-end delay, as well as the system's cost and complexity. The choice of routing, network topology, channel bandwidths and modulation schemes are but some of these important design considerations. Thus, the design of a packet radio network involves a number of mutually dependent considerations. Statements of optimality must therefore be carefully specified in terms of these aspects. In most instances, only a small number of considerations can simultaneously be taken into account. Some of these are discussed below:

The Routing Problem

Three approaches for routing messages in multi-hop packet radio networks have been proposed by Gitman et al {132}. These are called:

- Broadcast routing
- Hierarchical routing
- Directed Broadcast routing

In packet radio networks, as distinct from point-to-point networks (where the routing algorithm must determine the outgoing link for packets) the packet repeaters must decide whether or not to accept an incoming packet. Such a decision is based on an algorithm whose purpose is to regulate the traffic by preventing looping and cycling of packets. This decision process leads to the concept of a broadcast routing scheme.

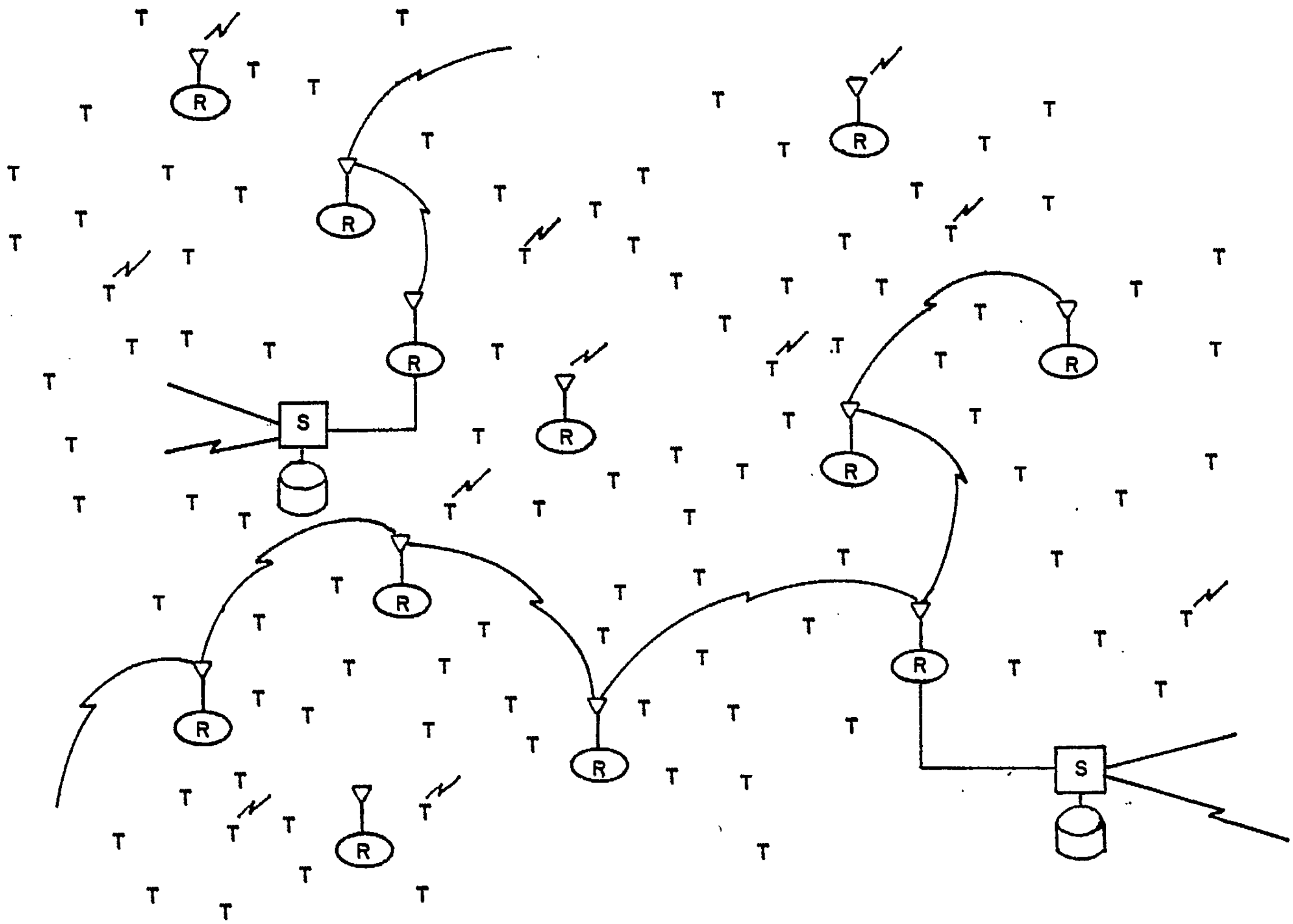


Figure B.3 MODEL OF A PACKET RADIO NETWORK

However, this scheme broadcasts packets and this can have severe consequences as a large number of duplicate packets can be generated thereby waste network resources to handle them.

To overcome this problem, hierarchical routing strategy information in the form of a label must be assigned to each repeater so that the "route of a message" forms a hierarchical tree structure. A shortest path route between any pair of terminals can then be specified thereby eliminating the generation of duplicate copies of packets.

A scheme called "the directed broadcast routing scheme" takes advantage of the best properties of the two previous schemes, in that it uses shortest path routing while at the same time provides for the generation of extra copies of packets along the path as required.

The Acknowledgement Problem

In packet switched networks at least two types of acknowledgement schemes are required to ensure that packets are not "lost" within the network. These acknowledgement schemes are the Hop-by-Hop acknowledgement that is sent whenever a packet is successfully received by the next node on the path; and the End-to-End acknowledgement that is transmitted whenever the packet finally reaches its destination. A comparison of the traffic handling performance of these two schemes was carried out by Gitman {133} who showed that the total average source-destination packet delay with a Hop-by-Hop scheme is always smaller than the corresponding delay with an End-to-End scheme. In particular, the performance of the Hop-by-Hop scheme is significantly better for multihop networks in which the probability of successful transmission on a per hop basis becomes small.

The Reliability Problem

An evaluation of the reliability (probability that message sent will be received) of different routing schemes and topological designs, in the presence of repeater failures was undertaken by Ball et al {134}. Their study, which is based on three reliability measures, showed that the reliability of a multihop packet radio network employing repeaters is extremely sensitive to the routing scheme used and the network topology configuration.

The Channel Configuration Problem

There are two distinct aspects to the channel configuration problem. The first involves the determining of whether or not there should be two distinct channels, one for each direction of transmission or whether a common channel should be dynamically shared between the two directions of transmission. The investigation of this problem was undertaken by Gitman et al {135} who showed that for one-hop networks, the common channel performance is consistantly better than the performance of the split channel, when the ratio of traffic to and from the station is not known or varies.

The other aspect of the other channel configuration problem was studied by the same author {136}, when he considered the possibility of using directional antennas by repeaters and stations for a single station two-hop network. The results obtained assume that separate channels from station to terminals and from terminals to stations are used and that the slotted ALOHA random access scheme is employed. The main results can be summarized as follows: namely that directional antennas can significantly increase the system capacity under some circumstances. This paper also gives some specific design guidelines that will maximize the system's capacity as a function of the repeater interference level. However, the models did not take into account the packet delay in passing through the network.

The throughput-delay performance of two-hop star connected configurations was investigated by Tobagi {137}. This analysis was however limited to the inbound traffic, i.e. traffic originated at the terminals and destined to the station. Later the same author {138} refined his analysis to consider two-hop fully connected configurations and evaluated its throughput-delay performance in terms of the network topology (number of repeaters and network connectivity) as well as the repeater's transmission protocol.

The Network Capacity Problem

The typical assumption used in the analysis of packet radio networks, is that all terminals are within range of each other as well as within range of the central station. Such networks are considered to be single hop networks as opposed to multi-hop networks where a packet must be relayed by several repeaters before reaching its destination. Kleinrock {139} considered range restricted networks consisting of randomly located nodes and showed that it was possible to obtain a capacity that was proportional to the square root of the number of network nodes. For an optimal average node degree of 6 the network throughput is 0.0976 $\sqrt{n}$ as opposed to 1/e for a fully connected slotted ALOHA network. This study was further extended by Silvester [140] who considered the problem of determining what traffic patterns would maximize the throughput.

APPENDIX C

SPEECH CODING AND INTERPOLATION

C.1 Introduction

In this appendix, we shall first review the characteristics of speech waveforms from two viewpoints. In particular we will consider the statistical characteristics of the ON-OFF pattern of the speech (or talkspurt-silence pattern). Then, we shall consider the time and frequency characteristics of the speech waveform.

This review will be followed by a brief summary of voice encoding techniques which for our purposes will be classified into wide-band and narrow-band techniques.

We will close this appendix with a description of two digital speech interpolation techniques namely TASI and SPEC.

C.2 The On-OFF pattern of the speech

The information presented in this section is obtained mainly from the work conducted by Brady {25},{84},{141}, in the late 1960's. His preliminary work led him to suggest a model for generating ON-OFF speech patterns representative of two-way telephone conversations. Before presenting any results, we shall first indicate which assumptions and constraints have been used in developing the model:

- The statistical data obtained and consequently the resultant statistical distributions are highly influenced by the choice of the speech detector and the threshold level. The speech detector is described in {84}, and the threshold level is taken as a parameter with three values -45dBm (most sensitive), -40dBm and -35dBm.

- All talkspurts of length smaller than 15 msec are rejected and all the silence intervals not exceeding 200 msec are filled in. That was necessary to avoid false detection due to noise and momentary stop consonants or other minor breaks.
- The model as shown in Figure C.1 consists of 6 states with the state transitions governed by 12 different parameters. We should note, however, that these parameters (α's and β's) do not represent state transition probabilities but rather rates of transition. This implies that by multiplying them by dt (5 msec) we obtain transition probabilities.
- The system is observed every dt (5 msec) to determine the present state.

The α's and β's parameters are given in {25} for 16 conversations. If we neglect, for the moment the 15 and 200 msec modifications mentioned above, then the system becomes a Markov chain with all of the statistical variables having exponential distributions. The percentage of time the system stays in a certain state can be obtained by solving the transition matrix of the Markov chain. A computer program was developed to calculate the time percentage corresponding to each state and the results are given in Table C.1.

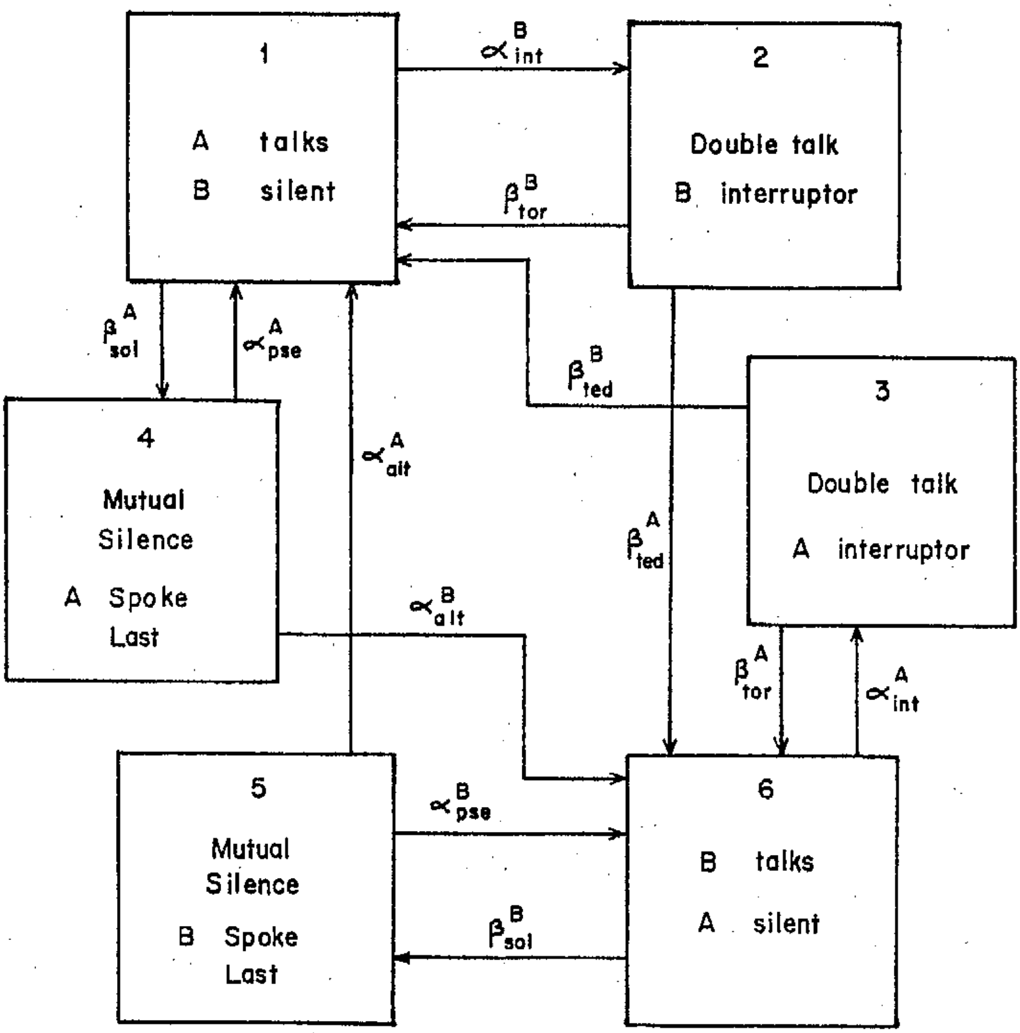


Figure C.1 THE SIX-STATE MARKOVIAN MODEL FOR TWO-WAY CONVERSATIONS

On the average, each of the speaker (A or B) talks for approximately 40% of the time, and falls silent for approximately 60% of the time. In reference {141} the average duration of the talkspurts and silence intervals was calculated based on a slightly different model and the results are given in Table C.2.

Recently Minoli {80} introduced some simplifying assumptions in order to obtain a 2 state Markov chain model which yields more tractable analytical formulations of packet voice communication problems.

There appears to be no wide acceptance for the exponential distribution function of the talkspurts and pauses. The reason seems to be related to the introduction of the 15 msec (throwaway) and 200 msec (fill-in) factors which to some extent ruin the Markovian structure of the model. Brady in particular has argued that the exponential distribution can be accepted as an approximation for the talkspurt duration, but it is entirely unacceptable for the silence intervals. Minoli {80} considered packetized talkspurts which led him to replace a two state Markov process by a two state Markov chain thereby obtaining a geometrical distribution for both the talkspurt length and the silence length.

To summarize, we can conclude the following:

* In a conversation, each party speaks for approximately 40% of the time and remains silent for 60% of the time.

* The average duration of a talkspurt is approximately 1.8 second and the silence interval is 1.8 sec.

STATE	TIME %	STATE	TIME %
1	33.1	4	12.7
2	3.7	5	11.7
3	4.6	6	34.2

Table C.1 FRACTION OF TIME SPENT IN EACH OF THE SIX-STATE SPEECH MARKOVIAN MODEL

EVENT	-45 dBm (sec)	-40 dBm (sec)	-35 dBm (sec)
TALKSPURT	1.37	1.2	0.98
PAUSE	1.8	1.85	1.76

Table C.2 AVERAGE TALKSPURT AND PAUSE DURATION

* The talkspurt length distribution can be assumed to be exponential, while only analytical convenience would justify the use of an exponential distribution for the silent intervals.

C.3 Speech Analysis/Synthesis

As schematically illustrated in Figure C.2, the human sounds are roughly divided into two classes; voiced sounds (such as p,e,m....) and unvoiced sounds (such as s,sh,f,....). Sustained voiced sounds contain a relatively large amount of periodicity, and can be characterized by a fundamental frequency (the pitch) and number of harmonics while the unvoiced sounds are noiselike and cannot easily be parameterized. Two additional parameters used to describe the speech are the volume (which may vary by about 40 dB dynamic range {142}), and the instantaneous spectral envelope of the voiced sound.

The mechanism for voice production, as shown in Figure C.2 is not as simple as it looks for the following reasons:

- Real speech is often not totally voiced or unvoiced but a mixture of both (as in v,z,th,....).
- Due to the highly non-stationary characteristics of speech it is experimentally difficult to distinguish between different classes of speech sounds.

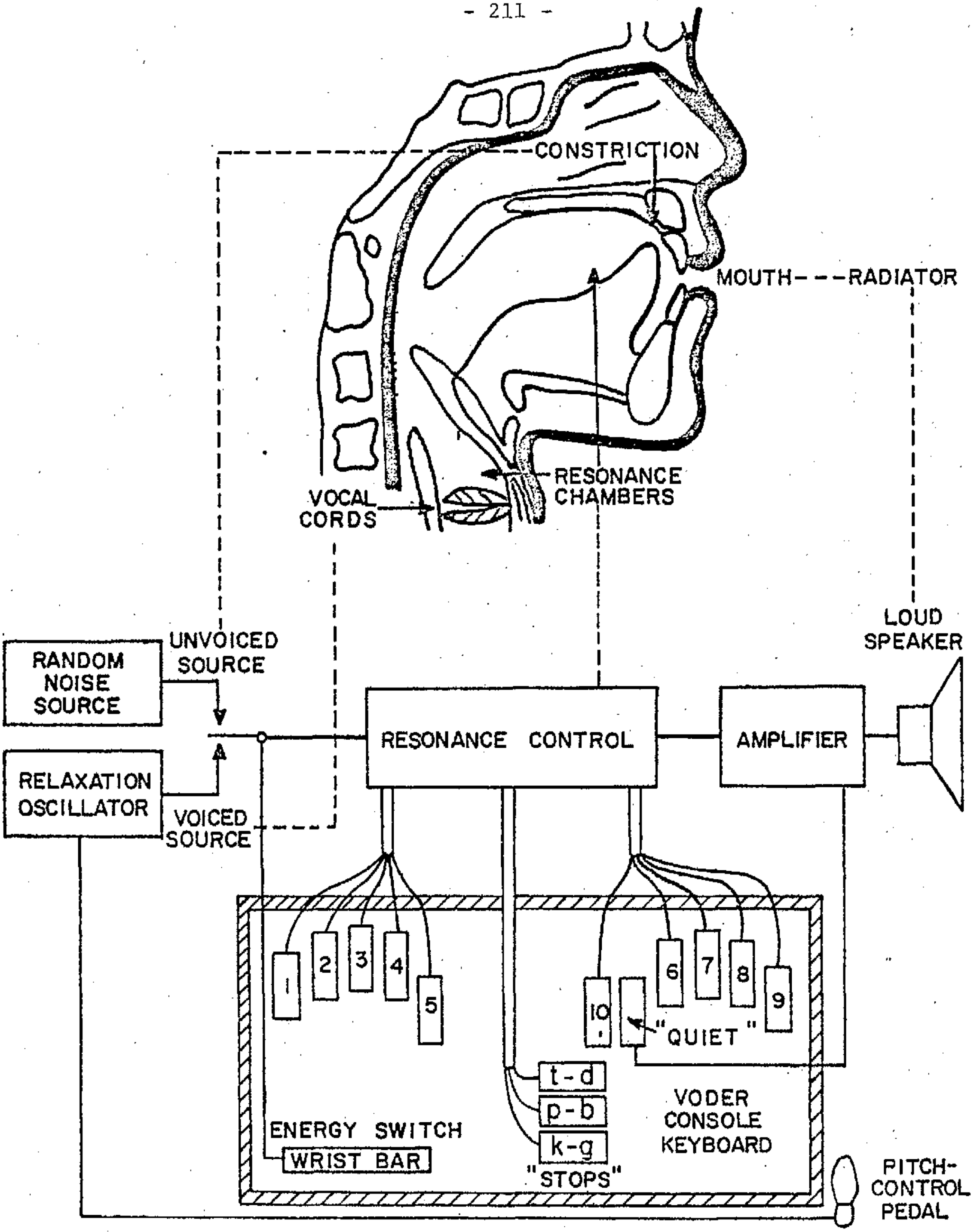


Figure C.2 A MODEL FOR HUMAN SOUND

- During the voiced sound, the fundamental frequency can vary from 50 to 600 Hz.

- Sudden changes in the sound pattern such as the transition from vowel to nasal result in sudden volume and pitch changes in the speech signals.

Further discussion of the speech analysis/synthesis process is beyond the scope of this dissertation. We would like however to stress that voice can only be modelled as a non-stationary process with a high degree of redundancy and high dynamic range. From a practical point of view the redundancy and the range have important implications. The redundancy can be used to reduce the time or bandwidth required to transmit speech while the large dynamic range imposes constraints on the design of speech detectors.

C.4 Voice Coding and Decoding Techniques

As we have previously indicated the speech signal can be regarded as a source with information rates varying between 1200 bps and 64 kbps. At this point, one might ask why is it necessary to consider high rates of transmission since speech-processing algorithms permit the transmission at low data rates such as 2400 bps. The reason stems from the fact that speech processed at a higher rate will be less vulnerable to the environmental degradations and is therefore more robust. It is in general desirable to encode speech at high data rates however network design economics dictates that this goal be compromised to combat traffic congestion.

Voice coding techniques are roughly divided into two categories; namely, wide-band coding (or waveform reconstruction) and narrow-band coding (or speech analysis - synthesis).

In the remaining of this section we shall describe the basic variations of each of the coding categories. We shall first deal with wide-band coding followed by narrow-band coding and a brief discussion of the different trade-offs involved in evaluating the different coding schemes.

C.4.1 Wide-Band Coding {76}

Wide-band coding is a straightforward description of the acoustic time waveform by means of discrete-time, discrete-amplitude representations. Within this category there are basically two classes of coding namely, Pulse Code Modulation (PCM) and Delta Modulation (DM), with many variations such as: Differential Pulse Code Modulation (DPCM), Adaptive Pulse Code Modulation (APCM), Continuous Variable Slope Delta Modulation (CVSD), etc.

Pulse Code Modulation (PCM)

In PCM the analog, band limited, signal m(t) is sampled at a rate of at least 2W Hz, where W is the highest frequency contained in m(t) (usually 4 KHz). The amplitude of each signal sample is, then quantized into one of 2^B levels (typical value of B is 8). The encoder responds to each quantized level by generating a unique 8 bit pattern. This means that the data rate at the output of the encoder is 2WB (=64 kbps). On the receive side, the first operation is to quantize the received signal (the quantization here is a simple 1-0 decision). The purpose of the front end quantization is to separate the signal from the noise which has been added during the transmission along the channel. The quantized signal is then fed to a decoder in order to restore the quantization level and finally to a low pass filter to smooth the output waveform.

The system we have just described is called Linear PCM, since the quantization level is linearly proportional to the input amplitude. Two important modifications have been proposed to reduce the number of quantization levels namely; the Logarithmic quantizer and the A-Law companding (which is an approximation of the Logarithmic quantizer). The advantage of such quantizer is that, without increasing the total number of quantization levels, one can allow large end steps in the quantizer to take care of possible excursions of the speech signal into the (relatively infrequent) large amplitude ranges. It should be noted that the use of a non-uniform quantizer is equivalent to the presentation of a compressed signal to a uniform quantizer, and a subsequent expansion of the output.

Adaptive PCM (APCM)

The first variation of the standard PCM system allows the quantizer step size to vary according to the speech pattern. The idea is to work with a basic quantizer that is very simple (uniform, if necessary), but to modify its step size (for every new input sample) by a factor depending on the knowledge of which quantizer slots were occupied by previous samples. The main advantage of the APCM over the standard PCM is to keep the step size near its optimum value.

Differential PCM (DPCM)

Speech that is sampled at the Nyquist rate (8 KHz) exhibits a very significant correlation between successive samples. One consequence of this correlation is that the variance of the samples difference is smaller than the variance of the speech signal itself. This observation is used to reduce the bit rate of the transmitted signal from 64 kbps (PCM) down to 32-48 kbps. Another important advantage of DPCM over PCM is the better signal-to-quantization error ratio (SNR), since SNR is proportional to the variance of the signal at the input of the quantizer.

Adaptive Differential PCM (ADPCM)

The use of adaptive quantization in ADPCM has the same motivation as the use of an adaptive quantizer in APCM. The adaptive predictor is employed to maximize the signal-to-noise ratio by calculating the optimum predictor coefficients. A simple way to do this is to store a finite section of speech, calculate the auto-correlation function for this section, and then determine an optimum predictor vector. The predictor is periodically updated at suitable time intervals and the predictor coefficient is transmitted to the receiver .

Delta Modulation (DM)

The exploration of signal correlation in DPCM suggests the further possibility of oversampling a signal to increase the adjacent sample correlation, and thus to permit the use of a simple quantization strategy. Delta Modulation (DM), the 1-bit version of DPCM, is precisely such a scheme. The band-limited input signal is sampled at a rate much higher than the Nyquist frequency and each sample is compared with the previous sample. A logic "1" ("0") is transmitted if there is an increase (decrease) in the sample value. On the receive side a staircase approximation of the original speech waveform is constructed using the received data stream.

DM can be affected by two types of quantization errors namely; slope overload distortion and granular noise. Slope overload occurs when the step size Δ is too small to follow a segment of the input waveform while granularily, refers to a situation where the staircase function cannot track a flat segment of the input function.

Adaptive DM (ADM)

As in PCM, it is possible to improve the dynamic range of the DM coder by increasing the step size during a steep segment of the input and decreasing it during a flat segment. Observed samples at the quantizer output are subjected to an adaptation algorithm that calculates the best estimated step size. Among the several algorithms have been proposed to accomplish this objective, the most successful one is the Continuous Variable Slope Delta modulation (CVSD).

Continuous Variable Slope Delta Modulation (CVSD)

CVSD uses the fact that syllabic companding of the speech represents a very useful adaptation strategy to achieve a smooth adaptation in time. An outstanding characteristic of CVSD is its ability, with fairly simple circuitry, to transmit intelligible voice at relatively low data rates. While companded PCM, for telephone quality transmission, requires a data rate of about 64 kbps, CVSD produces a signal of equivalent quality at 32 kbps.

Comparison of Techniques

In comparing the performance of different coding techniques, it must be emphasized that there is no single criterion to assess the performance of coding techniques, primarily because:

- Coder performance varies with the sampling rate and quantizing level.
- Signal-to-quantization error is not the only source of error in a communication channel.
- Quantization errors have different spectra for different coding techniques.
- Human listener react differently to different kinds of errors.

For these and many other reasons there has essentially been no objective measure of the speech quality. The only exception is due to Crochiere et al {143} who formulated various objective measures of performance of speech waveform coders. These authors stress however that extensive experiments are still required to relate subjective and objective measurements.

C.4.2 Narrow-Band Coding

The wide-band coding techniques discussed above were all based on a waveform reconstruction approach. Such approach requires sampling the speech waveform at relatively high rate and transmission of a code that represents either the sample amplitude or the difference in amplitude between two successive samples. The narrow-band coding techniques, on the other hand, are based on entirely different approach. Here, the design is based on the speech analysis/synthesis characteristics without any attempt to preserve the exact shape of the original speech. Such an approach is not a new one, in fact, it is dated back to the 1930's. However, the new advances in digital technology have made such an approach an attractive one at present time.

In the following, we shall review two basic techniques for implementing narrow-band coding namely, channel vocoder and linear predictive vocoder.

Channel Vocoder

The channel vocoder was the first analysis/synthesis device to be designed and implemented back in 1929. To this day, the channel vocoder operational principles remain unchanged.

A bank of band-pass filters is used to partition the speech spectrum into contiguous frequency subchannels. Each subchannel's signal component is then rectified, integrated, and low-pass filtered in order to obtain an estimate of the spectral envelope within that subchannel. Separate circuitry is employed to determine the pitch frequency, the voiced/unvoiced status and the gain. Each parameter is then digitally encoded and a composite digital signal is formed and transmitted to the receiving vocoder.

The synthesizer portion of the receiving vocoder decodes the composite bit stream, breaking it into its constituent signals, which are then used to create a replica of the original voice. Either a pulsed-periodic input at the pitch frequency (for voiced sound), or random noise (for unvoiced sound) is used to excite a filter bank whose output is then weighted by the spectral envelope signals.

Besides the pitch estimation and voiced/unvoiced decisions, the speech quality is parametric in the number of subchannels and the processing time windows. A short time window ($\approx$10 msec) and a large number of subchannels (>16) will result in high quality speech and also high bit rates (4.8 to 9.6 kbps), while longer time windows ($\approx$20 msec) and less number of subchannels ($\approx$16) will result in less quality speech and reduced bit rates ($\approx$2.4 kbps).

Linear Predictive Vocoder (LPC)

Linear predictive coding (LPC) offers the capability for obtaining the majority of the speech parameters directly from the time waveform. The development of the LPC vocoder concept is based on the principle that a reasonable prediction of a sample of the speech wave can be obtained as a linear weighted sum of previously measured speech samples.

The difference between the true value of the speech signal and its predicted value defines an error signal that can be minimized in some sense over a selected time window.

At present data rates as low as 2.4 kbps have been achieved with reasonable quality speech. According to Kang {144} further processing of the LPC data to identify formants and transmit this condensed information, can reduce the rate to 600 bps.

Comparison of Techniques

The comparison between the above two narrow-band coding techniques is based on speech quality, robustness and the bit rate with robustness being closely related to the bit rate. Clearly the higher the bit rate, the higher the redundancy of the speech signal which implies that a higher resistance to errors results. On the other hand, the speech quality depends basically on two sets of parameters. The first set is shared by all analysis/synthesis techniques, and includes the voiced/unvoiced decision and the pitch frequency. A high frequency of errors in these decisions, will result in a serious degradation of the quality and intelligibility of the speech. The second set of parameters varies from one technique to the other, and includes the number of subchannels in channel vocoders or the accuracy of the predictor coefficients for LPC vocoders.

C.4.3 Remarks

In the two previous sections we have discussed the basic features of the main wide-band and narrow-band coding techniques. In the following remarks we will summarize most of these features and present some information regarding the tolerable error rate for each technique.

We have pointed out several times that a high bit rate is always desirable for reliable speech transmission, but that economic network design dictates that this goal be compromised. Clearly as the bit rate increases, the cost of vocoder hardware decreases while the transmission cost (power, bandwidth, equalizers, filters,...) increases with minimum cost being achieved at a bit rate which depends on the type of channel being used. It has been argued that the high cost of low bit rate vocoder is a result of a marketing problem rather than technological problem. Should a large scale market be opened for such devices, their cost would decrease dramatically to offset the artificial cost advantage of wide-band coding techniques. Cost is however not the only reason why narrow-band techniques have historically found limited use. The other reason is related to the fact that a voice communications system conveys more information than merely language. Inflections and emotion in a voice clearly go beyond the synthesized speech intelligibility of narrow-band techniques.

The range of bit rates required to provide intelligible speech for different speech encoding techniques is given in Table C.3.

The bit error rate that can be tolerated in received speech signal depends on the coding technique as well as the bit rate. When the bit rate is very high (such as the case for PCM), an error rate of up to 50% can be tolerated without losing the speech intelligibility. For lesser bit rates the restriction on the error probability becomes tighter. It has however been shown that CVSD can tolerate an error rate of about 10% at approximately 16 kbps. The restrictions become very severe for the narrow-band vocoders. According to Gold {42}, LPC vocoders may tolerate only 1% error. whereas 5% error causes significant and most likely unacceptable degradation in both intelligibility and quality.

DIGITIZATION TECHNIQUE	BIT RATE kbps	DIGITIZATION METHOD
LINEAR PCM................	90-110	WAVEFORM RECONSTRUCTION
LOG PCM..................	48-64	
DPSK....................	32-48	
CVSD....................	16-32	
LPC.....................	2.4 - 9.6	ANALYSIS - SYNTHESIS
CHANNEL VOCODER.......	2.4 - 4.8	
CEPSTRUM VOCODER.......	2.4 - 4.8	

Table C.3 DIGITIZATION RATES

Two particular techniques are becoming widely accepted namely; CVSD (wide-band) and LPC (narrow-band). CVSD offeres a reasonable compromise between the high bit rate PCM, and the narrow-band vocoders, which explains why it is being used whenever the channel is noisy and the bandwidth saving is critical. LPC, on the other hand, is attractive since it provides better speech quality than its channel vocoder counterpart at the same bit rate (2.4 kbps).

C.5 Digital Speech Interpolation

In our discussion of the ON-OFF speech characteristics we have established that the speech signal occupies the channel about 40% of the time, hence it is theoretically possible to interpolate two speakers on a single channel. Such interpolation presumes, that we are capable of perfectly scheduling the conversations of speakers. It is however clear that such an assumption is never realized in practice on a single channel system. On the other hand, in a large system with a large number of active speakers, the talkspurts and silences become more evenly distributed which makes the interpolation process feasible. We refer to the ratio between the number of active users and the number of available channels as the interpolation gain which reaches its theoretical limit (all the silence gaps are filled) when the number of channels is very large (infinite number of channels).

When the speech signal is in a digital form, the interpolation process becomes more flexible because of the greater simplicity in storing and delaying the digital information, and the added flexibility promised by the marriage between the packet switched techniques and the Digital Speech Interpolation (DSI) techniques.

Two different methods of DSI will now be discussed namely a digitized version of Time-Assigned Speech Interpolation (TASI) and Speech Predictive Encoded Communication (SPEC). The operation of each method is briefly reviewed and their performance assessed in terms of the interpolation gain.

TASI

TASI is a technique in which the idle time between calls and the conversation silence gaps are used to accommodate additional calls. Provided that there is a sufficiently large number of channels, most of the idle time on the transmission link can be filled, giving an enhancement in transmission capacity greater than two.

TASI exploits the low speech activity by assigning a transmission channel only when speech is present. In this process, N conversations are carried on M transmission channels where M is less than N and the ratio N/M is referred to as the TASI gain. In practice an accurate measure of the TASI gain, should take into account the channel capacity needed for any assignment information. Talkspurts which begin at times when all channels are busy, are "frozen-out" which implies that either glitches occur or that they wait on a first-come first-served basis for available channels. In the later case in order not to affect the smoothness of the conversation, talkspurts are delayed provided that the waiting time is no longer than the talkspurt duration.

At each TASI terminal the presence of speech is sensed by a speech detector which initiates a request for a transmission channel. Through a channel assignment process an idle channel is assigned to the incoming channel in response to the request. During the time required to make the channel assignment and to connect the listener and talker the speech can be clipped. This phenomenon is called a connect clip and is to be distinguished from the freeze-out clip mentioned previously.

In addition to freeze-out and connect clipping, some time is required to accomplish speech detection and a detector clip can occur. Since it is possible to virtually eliminate detector clipping by an appropriate voice detector design, detector clipping can for all practical purposes be ignored.

If we consider an M-channel TASI system as described above, it can be shown {145} that the cut-out fraction $\emptyset$, or ratio of the cumulative duration of all speech losses to the cumulative duration of all talkspurts is given by:

$$\emptyset = \frac{1}{N\sigma} \sum_{k=M+1}^{N} (k-M) \binom{N}{k} \sigma^{k} (1-\sigma)^{N-k} \qquad \text{..... (C.1)}$$

where N is the number of terminals and σ is the probability that a certain source is issuing a talkspurt at some random time.

A principal factor governing TASI performance from the subjective point of view is the probability of occurrence of voice spurt clips of 50 ms or more. A 2 percent probability of occurrence of clips with durations equal to or greater than 50 ms is typically used as a threshold of acceptability {27}.

Denoting by α the probability of channel activity, the probability that the number of simultaneous talkers on N incoming channels will equal or exceed the number of channels, M, is given by:

$$B_{M,N,\alpha} = \sum_{k=M}^{N} \binom{N}{k} \alpha^k \quad (1-\alpha)^{N-k} \qquad \ldots\ldots \text{(C.2)}$$

Assuming that talkspurts are exponentially distributed with mean $\bar{\bar{T}}$, the probability that a talkspurt is frozen out for more than t units of time, is given by $B_{M,N,\psi}$ where:

$$\psi = \alpha\, e^{-t/\bar{\bar{T}}} \qquad \ldots\ldots \text{(C.3)}$$

Taking t>50 msec and $\bar{\bar{T}}$=1.5 sec the probability of occurrence of clips as a function of the number of transmission channels incoming channels, can be calculated {27}. Results show that the interpolation advantage for TASI system reaches a value of 2.2 for 240 incoming channels and exceeds 2 for all cases in which the number of available transmission channels exceeds 38.

SPEC

SPEC differs significantly from digital TASI in the concept, the implementation scheme and the quality aspects.

The operation principles of a SPEC system can be explained as follows:

- N active speech sources are connected to a central control unit where they are sampled regularly.
- N samples derived during each sample period are compared with the samples previously sent to the receiver and stored in a memory at the transmitter.
- For a given channel if the old and new samples differ by an amount equal to or less than some given number of quantizing steps, the channel is referred to as "predictable", otherwise it is called "unpredictable".
- The unpredictable samples are then transmitted in one frame, which includes N control bits.
- A control bit corresponding to a given channel is set to "1" if the frame contains a sample for that particular channel, and to a "0" if it does not.
- At the receiver the unpredictable samples received in the SPEC frame replace previously stored samples in the receiver's N channel memory.
- The most recent outgoing frame of information thus contains new samples on the unpredictable channels and repetitions of the old samples on the predictable ones.

When the new sample differs from the old one by a number of quantizing levels greater than a certain value called the "Aperture", a, the new sample will be transmitted, otherwise the old sample will be repeated at the receiving end.

If we assume that there are N incoming channels and M transmission slots, where $M < N$, the following two extreme cases can be considered:

- If the aperture, a, is set too high such that for a certain frame all new samples are considered to be predictable, all of the M transmission slots will be empty in that frame.
- If the aperture, a, is set too low such that for a certain transmission frame all N new samples are considered to be unpredictable, N transmission slots are needed to transmit the N samples.

These two extreme cases indicate the need to make the aperture size, a, adaptive to the activity observed over the N incoming channels. When the channel activity is predicted high, the aperture should also be set high such that only M out of N channels are considered unpredictable. When the activity decreases, there will be more slots available for transmission and the value of "a" should also be decreased to fill in these empty slots.

The value of "a" is directly related to the speech quality hence it is desirable to keep the value of "a" as low as possible without overloading the transmission slots.

The performance of SPEC has been studied by Campanella {27} who for the case of satellite channels illustrates the interpolation a advantage of the SPEC and TASI systems as a function of the number of transmission channels. Figure C.3 extracted from {27} indicates a slight advantage of SPEC over TASI, expecially when the number of transmission channels is small. The figure also indicates that the two systems achieve their maximum interpolation gains when the number of channels exceeds 120.

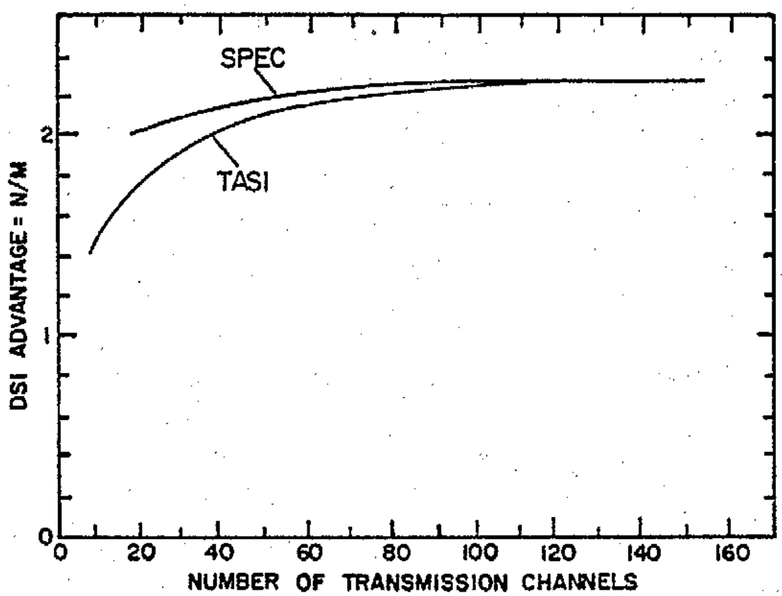


Figure C.3 DSI ADVANTAGE FOR TASI AND SPEC

BIBLIOGRAPHY

{1} D.O.C. "Spectrum Allocation Policy in the 406 to 960 MHz Frequency Band", Information Services, Department of Communications, February 1979.

{2} M. Stern and V. Graziano, "A new portable and mobile radio-telephone system goes to Washington (and Baltimore)" Telephony, July 18, 1977.

{3} C. Shannon, "A Mathematical Theory of Communication", Bell System Technical Journal, Vol. 27, pp 623-656, 1948.

{4} R. Dixon, "Spread Spectrum Systems", John Wiley and Sons Ltd., New York, 1975.

{5} W. Utlaut, "Spread Spectrum - Principles and Possible Application to Spectrum Utilization and Allocation", IEEE Com. Soc. Mag., pp 21-30, September 1978.

{6} R. Ormondroyd, "Spread Spectrum Communication in the Land Mobile Service" Proc. of Communications 78, pp 156-162, 1978.

{7} G. Cooper and R. Nettleton, "A Spread Spectrum Technique for High Capacity Mobile Communications", Proc. of the 27th Vehicular Technology Conf., March 1977.

{8} L. Berry and E. Haakinson, "Spectrum Efficiency for Multiple Independent Spread Spectrum Land Mobile Radio Systems", NTIA Report 78-11, November 1978.

{9} N. Abramson, "The ALOHA System - Another Alternative for Computer Communications", ALOHA System Technical Report B70-1, U. of Hawaii, April 1970.

{10} R.E. Kahn, "The Organization of Computer Resources into a Packet Radio Network", National Computer Conference, 44:177-186, June 1975.

{11} R.C. Kunzelman, "Status Report on the ARPA Packet Radio Experimental Network" COMPCON 78, February 78.

{12} R.E. Kahn, S.A. Cronmeyer, J. Burchfield and R.C. Kunzelman, "Advances in Packet Radio Technology", Proceedings of the IEEE, Vol. 66, No. 11, pp 1468-1498, November 1978.

{13} N. Abramson, "Packet Switching with Satellites", National Computer Conference, 42:695-702, June 4-8, 1973.

{14} R.M. Metcalfe and D.R. Boggs, "Ethernet: Distributed Packet Switching for local Computer Networks", Communications of the ACM, Vol. 19, No. 7, pp 395-404, July 1976.

{15} A. Mok, S. Ward, "Distributed Broadcast Channel Access", Computer Networks, Vol. 3, No. 5, pp 327-335, November 1979.

{16} O. Spaniol, "Modelling of Local Computer Networks", Computer Networks, Vol. 3, No. 5, pp 315-326, November 1979.

{17} D. Clark et al, "An Introduction to Local Area Networks", Proc. of the IEEE, Vol. 66, No. 11, November 1978.

{18} I.T. Frisch, "Experiments on Random Access Packet Data Transmissions on Coaxial Cable Video Transmission Systems", IEEE Trans. Communications, Vol. COM-25, No. 10, pp 1199-1203, October 1977.

{19} L. Kleinrock, "Resource Allocation in Computer Systems and Computer-Communication Networks", IFIP Conference Proceedings, pp 11-18, 1974.

{20} N.Y. Times, pp 1-2, December 15, 1901.

{21} W.W. Chu et al, "Experimental Results on the Packet Satellite Network", Proceedings of the NTC 79, pp 45.4.1 - 45.4.12, Nov. 1979.

{22} R. Kahn, "The Introduction of Packet Satellite Communications", Proceedings of the NTC 79, pp 45.1.1 - 45.1.6, Nov. 1979.

{23} I. Jacobs et al, "Packet Satellite Network Design Issues", Proceedings of the NTC 79, Nov. 1979.

{24} P. Kirstein et al, "SATNET Applications Activities", Proceedings of the NTC 79, Nov. 1979.

{25} P.T. Brady, "A Technique for Investigating ON-OFF Patterns of Speech", Bell System Technical Journal, Vol. 44, No. 1, pp 1-22, January 1965.

{26} K. Bullington and J.M. Fraser, "Engineering Aspects of TASI", Bell System Technical Journal, Vol. 38, pp 353-364, March 1959.

{27} S.J. Campanella, "Digital Speech Interpolation", COMSAT Technical Revue, Vol. 6, No. 1, pp 127-158, Spring 1976.

{28} R. Woitowitz, "The Effects of Speech Statistics in an Instantaneous Priority Speech Interpolation System", Proceedings of the ICC, pp 50.1.1 - 50.1.5, June 1978.

{29} E. Lyghounis, I. Poretti, G. Monti, "Speech Interpolation in Digital Transmission Systems", IEEE Trans. on Communications, Vol. COM-22, pp 1179-1189, Sept. 1974.

{30} A. Dorgelo, H. van der Veer, "Variable Length Coding Increasing Traffic Capacity in PCM Transmission Systems", Zurich Seminar on Digital Communications, pp C6.1-C6.6, March 1972.

{31} D. Seitzer, H. Gerhäuser, G. Langenbucher, "A Comparitive Study of High Quality Digital Speech Interpolation Systems", Proc. of the ICC, pp 50.2.1 - 50.2.5, June 1978.

{32} J. Soumagne, "A High Quality Digital Speech Interpolation System", Proc. of the ICC, pp 37.6.1 - 37.6.6, June 1979.

{33} P. Drago, A. Molinari, F. Vagliani, "Digital Dynamic Speech Detectors", IEEE Trans. on Communications, Vol. COM-26, No. 1, pp 140-145, January 1978.

{34} J.W. Forgie, "Speech Transmission in Packet-Switched Store and Forward Networks", National Computer Conference, pp 137-142, 1975.

{35} V. Dhadesugoor et al,"Delta Modulators in Packet Voice Networks", IEEE Trans. on Communications, Vol. COM-28, No. 1, pp 33-51, January 1980.

{36} M.J. Ross, A. Tabbot, J.W. Waite, "Design Approaches and Performance Criteria for Integrated Voice/Data Switching", Proc. IEEE, 65(9), pp 1283-1295, September 1977.

{37} J.W. Forgie, A.G. Nemeth, "An Efficient Packetized Voice/Data Network Using Statistical Flow Control", International Conference on Communications, pp 38.2-44 - 38.2-53, Chicago, Illinois, 12-15 June 1977.

{38} C.J. Weinstein, M.L. Malpass, M.J. Fischer, "Data Traffic Performance of an Integrated Circuit and Packet Switched Multiplex Structure", International Conference on Communications, pp 24.3-1 - 24.3-5, Boston, Mass., June 1979.

{39} E. Port et al, "A Network Architecture for the Integration of Circuit and Packet Switching", Proc. of the Third ICCC, pp 505-514, August 1976.

{40} G. Coviello, P. Vena, "Integration of Circuit/Packet Switching by a SENET", Proc. of NTC 75, pp 42.12 - 42.17, Dec. 1975.

{41} M.J. Fischer, T.C. Harris, "A Model for Evaluating the Performance of an Integrated Circuit and Packet-Switched Multiplex Structure", IEEE Trans. on Communications, COM-24, No. 2, February 1976.

{42} M. Watanabe et al, "Queueing Analysis of Integrated Switched Networks", Proc. of the Int. Conf. on Comp. Commun., Kyoto, Japan, pp 181-186, Sept. 1978.

{43} J. Forgie, A. Nameth, "An Efficient Packetized Voice/Data Network using Statistical Flow Control", Proc. of ICC 77, pp 38.2-44 - 38.2-48, June 1977.

{44} M. Ross, A. Tabbot, J. Waite, "Design Approaches and Performance Criteria for Integrated Voice/Data Switching", Proc. of the IEEE, Vol. 65, No. 9, pp 1283-1295, Sept. 1977.

{45} J. Derosa, L. Ozarow, "Packet Switching in a Processing Satellite", Proc. of the IEEE, Vol. 66, No. 1, pp 100-102, January 1978.

{46} M. Marsan, M. Pent, "Mixed Voice and Data Multiple Access Protocols for Mobile Radio Systems", Zurich Seminar on Digital Communications, pp C10.1 - C10.6, March 1980.

{47} A. Pan, "Integrating Voice and Data Traffic in a Broadcast Network Using Random Access Scheme", International Conference on Computer Communications, pp 551-556, Kyoto, Japan, Sept. 1978.

{48} J. Ellershaw, "A Radiotelephone System Using Packet Radio", Proc. of the IREE (Aust), pp 16-24, March 1979.

{49} W.C. Jakes et al., "Microwave Mobile Communications", New York: Wiley, 1974.

{50} R.H. Clarke, "A statistical theory of mobile radio reception", Bell Syst. Tech. J., Vol. 47, pp 957-1000, July 1968.

{51} J.F. Ossanna, Jr., "A model for mobile radio fading due to building reflections: Theoretical and experimental waveform power spectra", Bell Syst. Tech. J., Vol. 43, pp 2935-2971, Nov. 1964.

{52} E.N. Gilbert, "Energy reception for mobile radio", Bell Syst. Tech. J., Vol. 44, pp 1779-1803, Oct. 1965.

{53} G.A. Arredondo, J.C. Feggeler and J.I. Smith, "Voice and Data Transmission", Bell System Tech. Journal, Vol. 58, No. 1, pp 97-122, January 1979.

{54} J.I. Smith, "A Computer Generated Multipath Fading Simulation for Mobile Radio", IEEE Trans. on Vehicular Technology, Vol. VT-24, No. 3, pp 39-40, August 1975.

{55} R. French, "The Mobile Radio Data Channel". Zurich Seminar on Digital Communications, pp D.1.1 - D.1.9, March 1980.

{56} P. Mabey, "Mobile Radio Data Transmission - Coding for Error Control", IEEE Trans. on Veh. Tech., Vol. VT-27, No. 3, pp 99-109, August 1978.

{57} Y. Okumura et al, "Field Strength and its Variability in VHF and UHF Land Mobile Radio Services", Rev. Elect. Comm. Lab. Vol. 16, Sept.-Oct. 1968.

{58} J. Schwartz, M. Muntner, "Multiple-Access Communications for Computer Networks" in Computer Communication Networks edited by N. Abramson and F. Kuo, Prentice Hall, New Jersey, 1973.

{59} R. Binder, L. Castonguay, "Acknowledgment Schemes in an ALOHA Channel", Packet Radio Temporary Note 44, ARPA Inc., April 1973.

{60} F. Tobagi, L. Kleinrock, "The Effect of Acknowledgment Traffic on the Capacity of Packet-Switched Radio Channels", IEEE Trans. on Communications, Vol. COM-26, No. 6, pp 815-826, June 1978.

{61} A.R.K. Sastry, "Performance of Hybrid Error Control Schemes on Satellite Channels", IEEE Trans. Communication, Vol. COM-22, pp 689-694, July 1975.

{62} A.R.K. Sastry and L.N. Kanal, "Hybrid Error Control Using Re-transmission and Generalized Burst-Trapping Codes", IEEE Trans. Communication, Vol. COM-22, pp 385-393, April 1976.

{63} D. Townsley, "Error Detection and Retransmission Schemes in Computer Communication Networks", Proc. of COMPCON 78, pp 12-18, September 1978.

{64} B. Saeki, "Error Control Systems for Multi-Access Communication Channels", Ph.D Dissertation, UCLA, System Science Dept., June 1979.

{65} W.W. Chu, "Optimal Message Block Size for Computer Communications with Error Detection and Retransmission Strategies", IEEE Trans. on Communications, Vol. COM-22, No. 10, pp 1516-1525, Oct. 1974.

{66} R.C. French, "Error Rate Predictions and Measurements in the Mobile Radio Data Channel", IEEE Trans. on Vehicular Tech., Vol. VT-27, No. 3, August 1978.

{67} R.F. Linfield, "Radio Channel Capacity Limitations", U.S. Dept. of Commerce, Office of Telecommunications, Boulder, Colo., Report # OTR 77-132, Nov. 1977.

{68} J.K. Cavers, "Buffer Control for Transmission of Blocked Data over Fading Channels", IEEE Trans. on Communications, Vol. COM-25, No. 5, pp 496-502, May 1977.

{69} Kadokawa et al, "Application of the Lincompex System to the Land Mobile Service in the VHF band", Internal Report, Radio Research Laboratories (Japan), 1975.

{70} S. Mahmoud, "Analysis and Design of Mobile Channels in Packet Mode", Carleton University, Final Report to DOC, March 1980.

{71} R. Bailey, "A Digitally Encoded Voice Encryption System", Proc. of the Vehic. Tech. Conf., pp 8-11, 1977.

{72} Sachs/Freeman, "A Study of Digital Uses and Standards for Land Mobile Channels", A report to the Federal Communications Commission, December 1976.

{73} F. deJager, C.B. Dekker, "Tamed Frequency Modulation, A Novel Method to Achieve Spectrum Economy in Digital Transmission", IEEE Trans. Comm., Vol. COM-26, No. 5, pp 534-542, May 1978.

{74} K. Hirade, K. Murota, "A Study of Modulation for Digital Mobile Telephony", Proc. of the Vehic. Tech. Conf., pp 13-19, June 1979.

{75} CCIR Reports 525, 526, 1974.

{76} N. Jayant, "Digital Coding of Speech Waveforms: PCM, DPCM, and DM Quantizers", Proc. of the IEEE, Vol. 62, No. 5, pp 611-632, May 1974.

{77} M. Melnick, "Intelligibility Performance of a Variable Slope Delta Modulation", Proc. of the ICC 73, pp 46.5 - 46.7, June 1973.

{78} E. Klemmer, "Subjective Evaluation of Transmission Delay in Telephone Conversations", Bell System Tech. Journal, Vol. 46, No. 6, pp 1141-1147, July-Aug. 1967.

{79} J. Flanagan, "Digital Techniques for Speech Communications: Some New Dimensions", Proc. of the Int. Conf. on Cybernetics and Society, pp 46-50, 1977.

{80} D. Minoli, "Issues in Packet Voice Communications", Proc. of the IEE, Vol. 126, No. 8, pp 729-740, August 1979.

{81} G. Coviello, "Comparative Discussion of Circuit vs Packet-Switched Voice", IEEE Trans. on Comm., Vol. COM-27, No. 8, pp 1153-1160, August 1979.

{82} A Huggins, "Effect of Lost Packets on Speech Intelligibility", ARPA, NSC Note 78, February 1976.

{83} F. Tobagi, "Random Access Techniques for Data Transmission over Packet Switched Radio Networks", Ph.D Dissertation, U.C.L.A., Dept. of Computer Science, December 1974.

{84} P.T. Brady, "A Statistical Analysis of ON-OFF Patterns in 16 Conversations", Bell System Technical Journal, 7(1), pp 73-91, January 1968.

{85} S. Casner, E. Mader, E. Cole, "Some Initial Measurements of ARPANET Packet Voice Transmission", Proc. of the NTC, pp 12.2.1 - 12.2.5, December 1978.

{86} S.S. Lam, "Packet-Switching in a Multi-Access Broadcast Channel with Application to Satellite Communications in a Computer Network", Univ. of California, Los Angeles, Dept. of Computer Science, Ph.D Dissertation, March 1974.

{87} S. Kamal, S.A. Mahmoud, "A Study of User's Buffer Variation in Random Access Satellite Channels", IEEE Trans. on Communications, Vol. COM-27, No. 6, pp 857-868, June 1979.

{88} L.C. Anderson, "A Simulation Study of Some Dynamic Channel Assignment Algorithms in a High Capacity Mobile Telecommunications System", IEEE Trans. on Vehicular Technology, Vol. VT-22, pp 210-217, November 1973.

{89} D.C. Cox and C.O. Reudink, "Increasing Channel Occupancy in Large Scale Mobile Radio Systems: Dynamic Channel Reassignment", IEEE Trans. on Vehicular Technology, Vol. VT-22, pp 218-222, November 1973.

{90} T.J. Kahwa and N.D. Georganas, "A Hybrid Channel Assignment Scheme in Large Scale Cellular-Structured Mobile Communication Systems", IEEE Trans. on Communications, Vol. COM-26, No. 4, pp 432-438, April 1978.

{91} D.C. Cox and D.O. Reudink, "Dynamic Channel Assignment in High Capacity Mobile Communication Systems", Bell System Technical Journal, Vol. 50, pp 1833-1857, July-August 1971.

{92} D.C. Cox and D.O. Reudink, "Dynamic Channel Assignment in Two-Dimensional Large Scale Mobile Radio Systems", Bell System Technical Journal, Vol. 51, pp 1611-1629, September 1972.

{93} J.S. Engel and M.M. Peritsky, "Statiscally-Optimum Dynamic Server Assignment in Systems with Interfering Servers", IEEE Trans. on Vehicular Technology, Vol. VT-22, pp 203-209, November 1973.

{94} L.C. Anderson, "A Simulation Study of Some Dynamic Channel Assignment Algorithms in a High Capacity Mobile Telecommunications System", IEEE Trans. on Vehicular Technology, Vol. VT-22, pp 210-217, November 1973.

{95} M. Sengoku, K. Itoh and T. Matsumoto, "A Dynamic Frequency Assignment Algorithm in Mobile Radio Communication Systems", The Trans. of the IECE of Japan, Vol. E-61, No.7, pp 527-533, July 1978.

{96} J. Sin, N. Georganas, "A Hybrid Channel Assignment Scheme for Cellular, Land Mobile Radio Systems with ERLANG-C Service", Proc. of the NTC, pp 9.4.1 - 9.4.5, 1979.

{97} N. Abramson, "The ALOHA System", Computer-Communication Networks, N. Abramson and F.F. Kuo, Editors, Prentice-Hall Publishing Company, pp 501-518, 1973.

{98} N. Abramson and F.F. Franklin, "Some Advances in Radio Communications for Computers", Seventh Annual IEEE Computer Society International Conference, COMPCON '73 Digest of Papers, pp 57-60, February 1973.

{99} N. Abramson, "The Throughput of Packet Broadcasting Channels", IEEE Trans. on Communications, Vol. COM-25, No. 1, pp 117-128, January 1977.

{100} L. Kleinrock and S.S. Lam, "Dynamic Control Schemes for a Packet Switched Multi-Access Broadcast Channel", National Computer Conference, pp 143-153, May 1975.

{101} A.B. Carleial and M.E. Hellman, "Bistable Behaviour of ALOHA-Type Systems", IEEE Trans. on Communications, Vol. COM-23, No. 4, pp 401-409, April 1975.

{102} M.J. Ferguson, "A Study of Unslotted ALOHA with Arbitraty Message Lengths", Fourth Data Communications Symposium, October 1975.

{103} M.J. Ferguson, "An Approximate Analysis of Delay for Fixed and Variable Length Packets in an Unslotted ALOHA Channel", University of Hawaii Report TR B76-6, 1976.

{104} L.G. Roberts, "ALOHA Packet System with and without Slots and Capture", Stanford Research Institute, ASS Note 8, NIC DOC 11290, June 1972.

{105} L. Kleinrock and F. Tobagi, "Packet Switching in Radio Channels Part I: Carrier Sense Multiple-Access Modes and Their Throughput-Delay Characteristics", IEEE Trans. on Communications, Vol. COM-23, No. 12, pp 1400-1416, December 1975.

{106} F. Tobagi and L. Kleinrock, "Packet Switching in Radio Channels Part II: The Hidden Terminal Problem in Carrier-Sense Multiple Access and the Busy Tone Solution", IEEE Trans. on Communications, Vol. COM-23, No. 12, pp 1417-1433, Dec. 1975.

{107} L.G. Roberts, "Dynamic Allocation of Satellite Capacity Through Packet Reservation", National Computer Conference, Vol. 42, pp 711-716, June 1973.

{108} N. Szpankowski, K. Ono, Y. Urano, "Simulation and Analysis of Satellite Packet Switching Networks", Proc. of ICCC, Kyoto, Japan, pp 609-615, September 1978.

{109} F. Tobagi and L. Kleinrock, "Packet Switching in Radio Channels: Part III - Polling and (Dynamic) Split-Channel Reservation Multiple Access", IEEE Trans. on Communications, Vol. COM-24, No. 8, pp 832-845, August 1976.

{110} W. Crowther, R. Rettberg, D. Walden, S. Ornstein and F. Heart, "A System for Broadcast Communication: Reservation ALOHA", Sixth Hawaii International Conference on Systems Sciences, January 1973.

{111} S. Lam., "An Analysis of the R-ALOHA Protocol for Satellite Packet Switching", Proc. of the ICC, pp 27.3.1 - 27.3.5, June 1978.

{112} R. Binder, "A Dynamic Packet Switching System for Satellite Broadcast Channels", International Conference on Communications, June 1975.

{113} A.R.K. Sastry, "A Simple Reservation Scheme to Improve Excess Capacity of Slotted ALOHA Channels", International Conference on Communications, June 1977.

{114} E.H. Rothauser and D. Wild, "MLMA - A Collision-Free Multi-Access Method", IFIP Conference Proceedings, pp 431-436, 1977.

{115} M. Balagangadhar, R. Pickholtz, "Analysis of a Reservation Multiple Access Technique for Data Transmission via Satellites", IEEE Trans. on Communications, Vol. COM-27, No. 10, pp 1467-1475, October 1979.

{116} F. Borgonovo, L. Pratta, "SRUC: A Technique for Packet Transmission on Multiple Access Channels", Int. Conf. on Computer Communications, pp 601-607, Kyoto, Japan, September 1978.

{117} S. Tasaka and Y. Yasuda, "A Packet Radio Communication System with a Random Access Scheme", Electronics and Communications in Japan, Vol. 59-A, No. 8, pp 26-34, 1976.

{118} M. Scholl and L. Kleinrock, "Packet Switching in Radio Channels: New Conflict-Free Multiple Access Schemes for a small number of data users", International Conference on Communications, pp 22.1-105 to 22.1-111, 1977.

{119} Y. Yemini and L. Kleinrock, "An Optimal Adaptive Scheme for Multiple Access Broadcast Communications", International Conference on Communications pp 7.2.1 to 7.2.5, 1978.

{120} I. Rubin, "Group Random-Access Disciplines for Multi-Access Broadcast Channels", IEEE Trans. on Information Theory, Vol. IT-24, No. 5, pp 578-592, September 1978.

{121} L.W. Hansen and M. Schwartz, "An Assigned-Slot Listen-Before Transmission Protocol for a Multi-Access Data Channel", International Conference on Communications, Vol. 2, pp 22.2-112 to 22.2-116, 1977.

{122} L.W. Hansen and M. Schwartz, "An Assigned-Slot Listen-Before Transmission Protocol. Dynamic Channel Control Procedures and Random Slot Assignment", International Conference on Communications, pp 7.1.1 to 7.1.6, 1978.

{123} H. Okada et al, "Analysis of Framed ALOHA Channel in Satellite Packet Switching Networks", Proc. of the Int. Conf. on Computer Communications, Kyoto, Japan, pp 617-622, September 1978.

{124} I. Chlamtac, W. Franta, K. Levin, "BRAM: The Broadcast Recognizing Access Method", IEEE Trans. on Communications, Vol. COM-27, No. 8, pp 1183-1189, August 1979.

{125} J. Capetanakis, "Tree Algorithms for Packet Broadcast Channels", IEEE Trans. on Infor. Theory, Vol. IT-25, No. 5, pp 505-515, September 1979.

* {126} J. Capetanakis, "Generalized TDMA: The Multi-Accessing Tree Protocol", IEEE Trans. on Communications, Vol. COM-27, No. 10, pp 1476-1484, October 1979.

{127} C. Meubus, M. Kaplan, "Protocols for Multi-Access Packet Satellite Communications", Proc. of NTC, pp 11.4-1 - 11.4-7, 1979.

{128} J. DeRosa, L. Ozarow, L. Weiner, "Efficient Packet Satellite Communications", IEEE Trans. on Communications, Vol. COM-27, No. 10, pp 1416-1422, October 1979.

{129} S. Ng, J. Mark, "A Multiaccess Model for Packet Switching with a Satellite having some Processing Capability", IEEE Trans. on Communications, Vol. COM-25, pp 128-135, January 1977.

{130} R. Eaves, "ALOHA/TDM Systems with Multiple Downlink Capacity", IEEE Trans. on Communications, Vol. COM-27, No. 3, pp 537-541, March 1979.

{131} M. Molle, L. Kleinrock, "Analysis of Concentrated ALOHA Satellite Links", Sixth Data Com. Symp., pp 87-95, Pacific Grove, November 1979.

{132} I. Gitman, R.M. VanSlike and H. Frank, "Routing in Packet-Switching Broadcast Radio Networks", IEEE Trans. on Communications, Vol. COM-24, pp 926-930, August 1976.

{133} I. Gitman, "Comparison of Hop-by-Hop and End-to-End Acknowledgement Schemes in Computer Communication Networks", IEEE Trans. on Communications, pp 1258-1262, November 1976.

{134} M. Ball, R.M. VanSlike, I. Gitman and H. Frank, "Reliability of Packet-Switching Broadcast Radio Networks", IEEE Trans. on Circuits and Systems, pp 806-813, December 1976.

{135} I. Gitman, R.M. VanSlike and H. Frank, "On Splitting Random Access Broadcast Communication Channels", Seventh Hawaii International Conference on System Sciences, January 1974.

{136} I. Gitman, "On the Capacity of Slotted ALOHA Networks and some Design Problems", IEEE Trans. on Communications, pp 305-317, March 1975.

{137} F. Tobagi, "Performance Analysis of Packet Radio Communication Systems", Proc. of the National Telecommunications Conference, pp 12:6-1 to 12:6-7, December 1977.

{138} F. Tobagi, "Analysis of Slotted ALOHA in a Centralized Two Hop Packet Radio Network", COMPCON 78, pp 19-26, Feb. 1978.

{139} L. Kleinrock, J. Silvester, "Optimum Transmission Radii for Packet Radio Networks or Why Six is a Magic Number", Proc. of the NTC, pp 4.3.1 - 4.3.5, December 1978.

{140} J. Silvester, L. Kleinrock, "On the Capacity of ALOHA Packet Radio Networks for Local Traffic Matrices", Proc. of the 12th Hawaii Int. Conf. on System Sciences, pp 231-237, January 1979.

{141} P. Brady, "A Model for Generating ON-OFF Patterns in Two-Way Conversations", Bell Syst. Techn. Journal, Vol. 48, pp 2445-2472, 1979.

{142} B. Gold, "Digital Speech Networks", Proc. of the IEEE, Vol. 65, No. 12, pp 1636-1658, December 1977.

{143} R. Crochiere et al, "A Study of Objective Measures for Speech Waveform Coders", Zurich Seminar on Digital Communications, pp H1.1 - H1.7, March 1978.

{144} G. Kang, D. Coulter, "600 bit-per-second Voice Digitizer", Naval Research Laboratory Report 8043, November 1976.

{145} C. Weinstein, "Fractional Speech Loss and Talker Activity Model for TASI and Packet Switched Speech", IEEE Trans. on Communications, Vol. COM-26, No. 8, pp 1253-1257, August 1978.