Government of Canada

AN OVERVIEW OF PACKET RADIO COMMUNICATIONS

BY

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Director General
Telecommunication Regulatory Service
MINISTRY OF COMMUNICATIONS
CANADA

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The author owes much to his colleagues John DaSilva, G. Van der Maas, H. Hafez and Joe Kahwa for the many stimulating discussions over the years and for their help in preparing various aspects of this manuscript.

The topics covered herein have been discussed by the author with graduate students in his courses in the faculty of systems engineering and computing sciences at Carleton University, Ottawa, over the last several years.

SUMMARY

This presentation reviews some of the more recent results of interest to the author in the field of packet radio communications.

In particular, Section 2 presents a brief summary and analysis of the results that have been obtained for various packet radio random access schemes and concludes with a discussion of some considerations/ results for designing packet radio networks.

Section 3 surveys and compares the analog land mobile cellular systems from the point of view of channel assignment schemes and traffic handling capabilities.

Section 4 presents a sampling of some recent research results and active investigation in the area of cellular packet radio networks for voice and data applications. In particular, some unpublished results are presented in the problems of optimal packet size and capacity degradation of packet radio channels due to fading. A brief treatment of the interference to UHF television from mobile transmissions is also presented.

Section 5 concludes with a brief review of the recent changes to the regulations in Canada that give Amateur Radio Operators privileges to experiment with packet radio schemes for man/man, man/machine and machine/machine resource sharing.

1. INTRODUCTION

The radio spectrum, while theoretically a non-depletable resource, if improperly shared, tends to take on all the attributes of a finite depletable resource at least as far as the life of humans is concerned. In particular, once made, spectrum allocations are very difficult and time consuming to change even though it can be clearly demonstrated that some new radio service would be more economical and meet a greater need. Over the last ten years in particular, there has been an increasing tendency to employ more scientific considerations to questions associated with the optimum allocation and use of the radio spectrum.

In this regard, packet radio schemes based on cellular concepts are emerging as the ones that hold the greatest promise for effectively sharing the spectrum allocations for mobile voice and data communications.

On the technology side, advances in microprocessors and memory fabrication are leading to the production of increasingly "intelligent" terminals and computer peripherals for mobile and portable communications using packet radio techniques to optimize the spectrum requirements.

The term, 'packet radio', was originally coined by Dr. Abramson at the University of Hawaii who demonstrated its practicality by constructing the "ALOHA" computer network.(1) Since 1970, many authors (see bibliography starting on page 71) have carried out extensive research in this field.

Packet radio concepts for transmitting voice over mobile communication systems are not yet well developed or widely understood. There are, however, many results applicable to packet voice transmissions on land based computer communication networks and digital voice over mobile radio systems. The author is convinced that packet radio is the technique that will realize highly secure economic/integrated data/voice communication networks. Furthermore, packet voice, unlike classical digital voice, does lead to spectrum utilization that is more efficient than that achieved by analog voice.

More exotic and complex modulation techniques can, at significant costs, offer some advantages to users and at the same time improve overall spectrum use. The spread spectrum modulation technique produces a signal that has a transmission bandwidth several orders of magnitude greater than the information bandwidth of the signal being transmitted. Spread spectrum techniques are not discussed further in this presentation. For a readable discussion of spread spectrum concepts in the packet radio field, see Kahn (4).

This presentation is limited to considering random access schemes and cellular and packet radio systems.

Central to the analysis of packet radio cellular systems is the concept of optimal packet size and we will discuss this problem. The propagation behaviour of mobile radio signals is well known and documented. However, the implications of fading and shadowing in the selection of the optimal packet size and channel capacity degradation for packet radio communication, are quite critical and have been largely ignored. Some new and, what we believe are important, results in this area are presented.

Of importance to designers of high capacity cellular systems as well as spectrum managers in North America is the possible interference that could be caused by UHF mobile terminals to UHF television receivers. Some recently published results are incomplete and misleading. In order to clarify these, a new probabilistic model of the potential interference caused by cellular packet radio systems to UHF television transmissions is discussed.

Canadian Amateur Radio operators, as a result of recent changes to the regulations, have embarked on packet radio experimentation involving man/man, man/machine and machine/machine communications. These developments are also briefly reviewed.

2. PACKET RADIO RANDOM ACCESS SCHEMES

In large measure, 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 satisfy the first law of optimum resource sharing*.

The first packet radio communication system to be described in the literature was the ALOHA one implemented at the University of Hawaii (1). This system provided communication between a central computer and its geographically scattered but fixed terminals. The "ALOHA" concept was almost immediately, with the support of the Advanced Research Projects Agency (ARPA), extended to include investigations involving mobile terminals (2), (3), (4). We were also soon to learn that packet transmissions over satellite channels offered many attractive options for the efficient collection and distribution of data over large areas (5). As we shall see, 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 (6), (7), (8), (9) and cable television systems (10).

The fundamental concepts behind resource sharing (allocation) and why it is of such importance in the field of computer communications are nicely summarized by Kleinrock (11), (13), as follows:

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

*"Let whoever needs it use it whenever it is not being used by another" - 1st Law

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..."

The interference/resource sharing problem is not new, as some 70 years earlier, in 1901 (12), the New York times reported in frustration:

"So far as is known, there is no means of preventing successfully the interference of wireless signals and until they become automatically selective it would seem that only one station on each side of New York Bay, would engage in the business. Even during the recent yacht races the wireless telegraph signals were in utter confusion until peace was patched up enabling each party in rivalry to send messages for a few minutes at a time".

The problem expressed in the above quotation was that engineers at the time simply did not know how to share(allocate) a radio channel on either a time, frequency or random access basis.

THE PURE ALOHA RANDOM ACCESS SYSTEM

The pure ALOHA scheme (1), (14), 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 down link radio channel from the central station to the terminals, is not the same radio channel that is used for up link 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. We will discuss some of these later.

Capacity of the Pure ALOHA Channel

In order to describe this result assume that the total traffic offered to a radio channel of bandwidth W that is being modulated at m_R bits/hertz 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 (5) to be:

$$S = Ge^{-2G}$$

A plot of this equation is shown as Figure 1. In particular, as will be seen in this figure, maximum packet throughput is achieved at $G = 1/2$ and is equal to $1/2e \simeq 0.184$ 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.

$$D = \left[\frac{G}{S} - 1\right] R + 1 + \beta$$

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:

$$R = 1 + 2\beta + \delta + \alpha$$

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. To determine a typical value of β, consider a channel rate of 10 kbits/sec and a packet length of 1000 bits the value of β for a transmission path distance of say 5 km is

$$\beta = \left(\frac{5}{300000}\right) \Big/ \left(\frac{1000}{10000}\right) \simeq 16\times10^{-6}$$

and can therefore in practice be neglected.

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 (15) 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.

Concluding 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 (16) 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 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 (17), (18), (19). 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 (20), (21) 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. Similar results were later obtained by Yu (22).

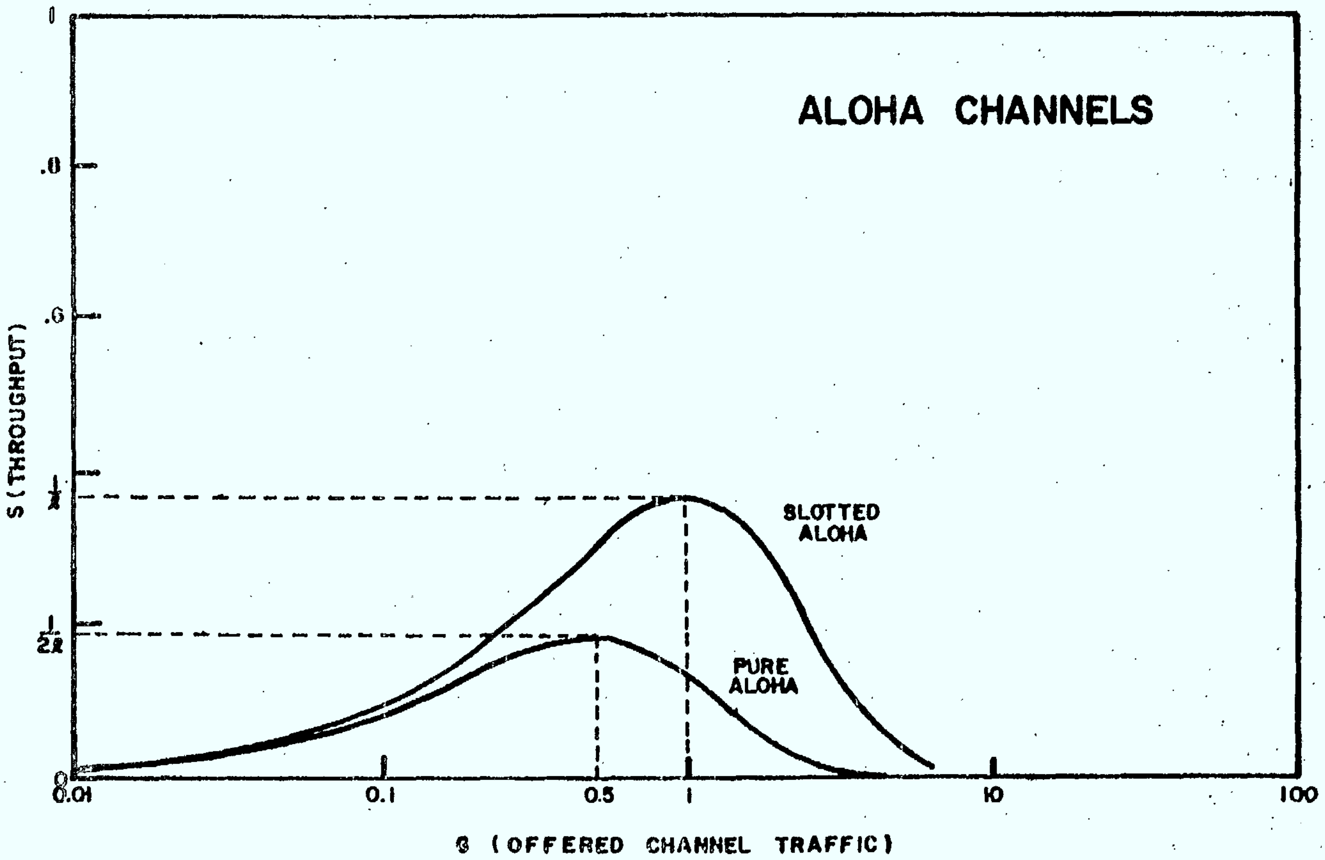

ALOHA CHANNELS
S (THROUGHPUT)
1
.8
.6
$\frac{1}{e}$
$\frac{1}{2e}$
0
0.01
0.1
0.5
1
10
100
G (OFFERED CHANNEL TRAFFIC)
SLOTTED ALOHA
PURE ALOHA

FIGURE 1

The conclusion is unmistakable, namely that messages in any given radio access scheme should be broken down into fixed length packets.

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 (23) 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 (23) the throughput equation for the channel becomes:

$$S = Ge^{-G}$$

Again, as shown in Figure 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 that:

$$D = \left(\frac{G}{S} - 1\right) R + 1 + \beta$$

or *

$$D = \left(\frac{G}{S} - 1\right)\left(\beta + \frac{k+1}{2}\right) + 1 + \beta$$

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

Lam (15) proved that for each value of S, an optimum value of k could be selected which would minimize the packet dealy. Again, neglecting the propagation delay β, the equation can be written as:

$$D = \left(\frac{G}{S} - 1\right)\left(\frac{k+1}{2}\right) + 1$$

Concluding 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.

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 (24), (25), (26) 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 terminal "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 described in this presentation is given in (25). 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 (24) showed that the throughput S was:

$$S = \frac{Ge^{-\beta G}}{G(1+2\beta) + e^{-\beta G}}$$

The graph of this equation is shown in Figure 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. If we neglect β, the propagation delay, then the equation can be written as:

$$S = \frac{G}{1 + G}$$

In this case, throughput achieved by maximizing S with respect to G as shown in Figure 2 is practically equal to one (100%) and would represent perfect channel utilization.

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 retransmission 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 (24) is:

$$D = \left(\frac{G}{S} - 1\right)\left(1 + \alpha + \delta\right) + 1$$

Concluding 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 $G \to 1$). 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.

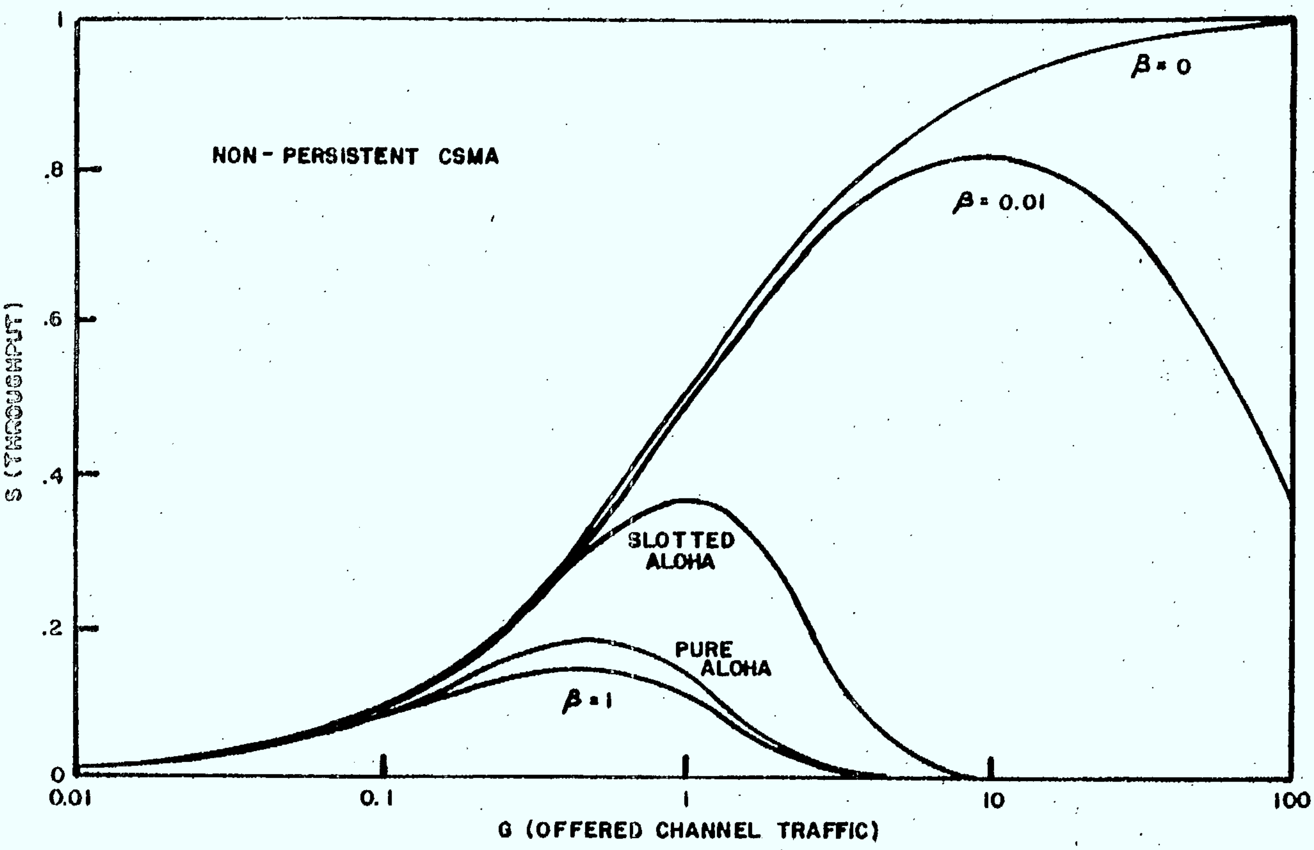

NON-PERSISTENT CSMA
S (THROUGHPUT)
G (OFFERED CHANNEL TRAFFIC)
β = 0
β = 0.01
SLOTTED ALOHA
PURE ALOHA
β = 1
1
.8
.6
.4
.2
0
0.01
0.1
1
10
100

FIGURE 2

"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. The six distinct types of reservation schemes that are known to the author are as follows:

Reservation I

A description of one of the earliest reservation random access schemes is due to Roberts (27). 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.

Reservation II

In 1976 Tobagi (28) 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 (29) suggested a reservation technique 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.

Reservation IV

A variation of the above random access scheme, suggested by Binder (30), 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.

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 (31) 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 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 (32). 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.

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 (33) 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.

CSMA Variation

In this scheme proposed by Hansen (34) and later refined (35) by the same author, each terminal is assigned a specific sensing slot on a frame of L slots that is periodically repeated. If any terminal with a packet to transmit finds the channel idle, it transmits its packet; otherwise transmission is delayed according to some retransmission delay probability distribution function. In the case where there is only one user assigned to each slot the system becomes conflict free.

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 (36) 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 (37) 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 (38) 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.

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 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 (39). 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. However, this scheme

PACKET RADIO NETWORK

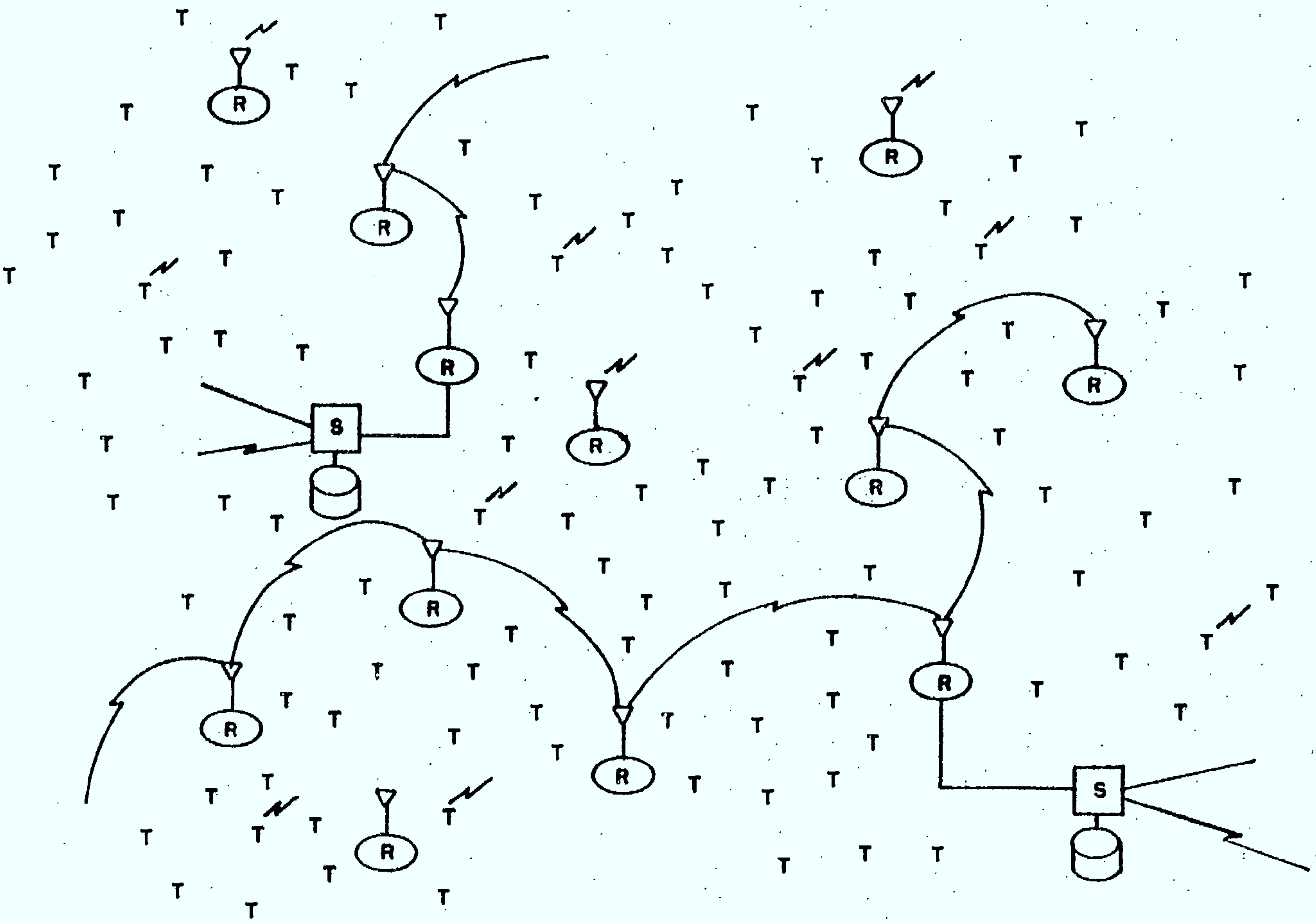


FIGURE 3

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 (40) 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 (41). 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 (42) 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 (43), 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 (44). 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 (45) 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.

Experimental Packet Radio Networks

The ALOHA system (1), at the University of Hawaii, as was mentioned previously, 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 (3) 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 (2) 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 (4).

A number of packet radio experimental networks exist in Canada on a largely localized basis, in universities and research laboratories. Recently the Canadian Amateur Radio Federation in collaboration with Montreal Amateur Radio Club announced the formation of AMPAC (Amateur Packet Network).

3. ANALOG LAND-MOBILE CELLULAR SYSTEMS

Current Status

Mobile telephone service was introduced into Canada and the U.S.A. in 1947. Existing systems operate in the VHF and UHF bands on the single base station with many mobiles concept. Because of the close relationship that has been maintained between both countries, similar frequency allocations and common system standards prevail that allow mobile telephone users to operate in all North-American cities and on many of the principal routes.

In Canada, the Department of Communications continuously reviews the adequacy of existing spectrum allocations. In the late 1960's, several studies, indicated that in the early 1980's that there would be severe congestion of the VHF land mobile bands in some of Canada's largest cities.

Partly as a result of this forecasted congestion and also because it wished to expand opportunities in mobile communication, Canada announced new spectrum allocations for mobile communications in the 406-960 MHz band (46). In particular, the 806-890 MHz band was reallocated from UHF television to mobile communications without initially specifying the sub-allocations. Forecasts are that the first Canadian cellular systems for analog communications will undergo field tests commencing in 1983.

In the U.S.A. the Federal Communications Commission (FCC) had earlier (in 1974) reallocated on a natiowide, 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.

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 (47).

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 countrywide coverage is achieved.

The author has also heard of plans for cellular systems in Austria, Denmark, Finland, Norway, Sweden, etc. Let us now look at some of the underlying concepts of these systems.

THE CELLULAR SYSTEM CONCEPT

Conventional land mobile systems use fixed transmitter stations (called base stations) usually located well above ground level. The 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 is easy to show 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 improvement comes from this multiple reuse of the same channel. This leads to a measure of spectrum efficiency in terms of traffic handled per unit of bandwidth per unit of area.

Another technique that improves the traffic handling capability of cellular systems is called trunking. Trunking involves the grouping of channels and the connection of terminals requiring service to unused or idle channels. Trunking is therefore a resource sharing procedure that reduces the probability of a call being blocked and thereby increases the system's traffic carrying capacity.

Cellular Structure

The boundaries of each cell are dependent upon many factors such as terrain, antenna heights and the distance to adjacent cell sites. However, for analysis and modelling purposes, cells are assumed to be hexagonal in shape, since the hexagon along with the triangle and the square are the only regular polygon that completely covers a plane area. The hexagon however is the best approximation to the circular coverage shape of omni-directional antennae over a given area.

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

(1) the co-channel reuse ratio D/R which is the ratio of the distance between co-channel base stations transmitters to the cell radius, and

(2) C the number of distinctly unique sets of frequencies.

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

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

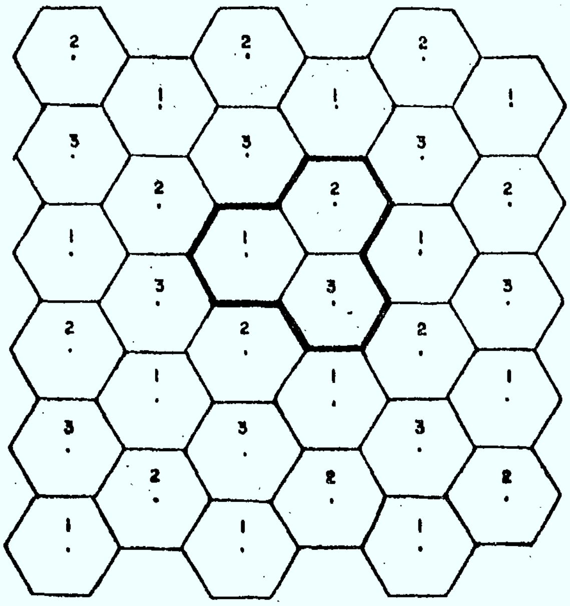
where C can take only the selected values:

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

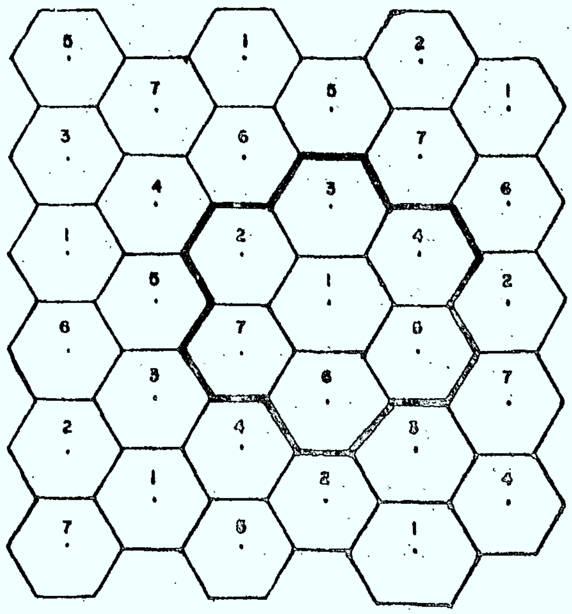
Figure 4 shows the four cellular structures that are obtained by letting C = 3, 7, 9 and 12. Each cell has a set of distinct frequencies that will not interfere with frequencies assigned to adjacent cells. The frequencies (channels) within each cell are used for communications between the base station (which is connected to the telephone network) and the mobile terminals. Also shown in dark lines in Figure 4 are arrangements of the different sets of frequencies among the cells. By repeating this pattern, the same set C of frequencies can be reused to cover a given area.

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.

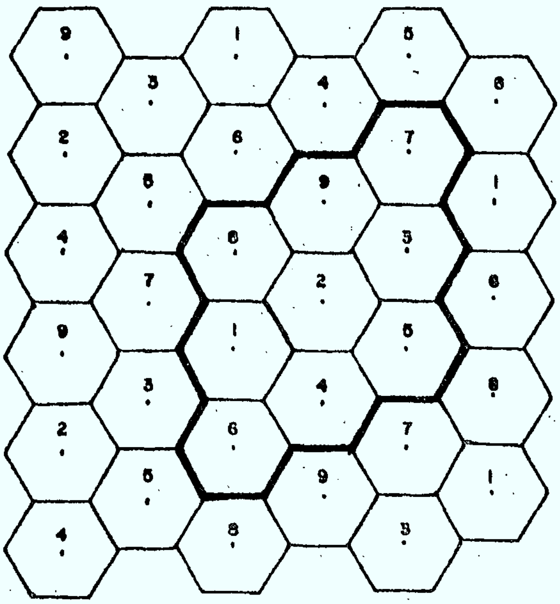
The traffic handling capability of a given cellular network can be greatly increased by a technique called splitting. This involves the addition of new cells midway between existing cells reducing the transmitter power and,



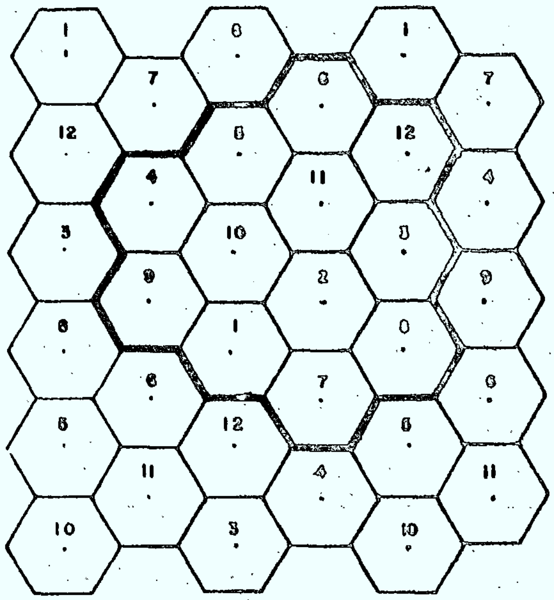
• CELL-SITE LOCATION
C = 3 , D/R = 3



• CELL-SITE LOCATION
C = 7 , D/R = 4.6



• CELL-SITE LOCATION
C = 9 D/R = 5.2



• CELL-SITE LOCATION
C = 12 , D/R = 6

FIGURE 4

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).

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 distinct address and automatically "tune in" (listen) on the paging channel when not in use. Thus, the mobile terminal being sought is readily

* 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.

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 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.

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 (48, 49, 50) and fall into three classes:

- static assignment schemes
- dynamic assignment schemes
- hybrid assignment 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 neighbouring 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 re-allocate 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 (51), (52), (53), (54), (55) 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 studied so far.

Computational Considerations

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 have presented computational difficulties, Graph Theory can be applied. In particular, Sengoku (56) 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:

$$P(i) = \frac{(\rho/L)^i \nu_i}{\sum_{j=0}^{k} (\rho/L)^j \nu_j}$$

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:

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

The paper (56) 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.

Concluding Remarks

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 to every cell on a static basis while reserving a pool of 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 (55) for example, studied the case of 10 channels available for assignment and found the optimum division to be 8 static channels and 2 dynamic channels.

In a recent paper Kahwa et al (57) simulated several hybrid assignment schemes 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.

COMPARISON OF CELLULAR SYSTEMS

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 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 II) 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, of course, practical physical and economical limits to the cell size.

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* 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

SOME CHARACTERISTICS OF CELLULAR SYSTEMS

(TABLE I)

CHARACTERISTICS	7 CELLS	19 CELLS
SERVICE AREA (km^2)	450	450
CELL RADIUS (km)	≈ 5	≈ 3
CHANNEL WIDTH (kHz)	2 x 30	2 x 30
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

APPROXIMATE SPECTRUM EFFICIENCIES OF CELLULAR SYSTEMS

(HYPOTHETICAL EXAMPLE)

TABLE II

4. PACKET RADIO IN CELLULAR ENVIRONMENTS

In the previous sections we discussed both cellular architectures and packet radio schemes as distinct concepts. Let us know turn to the potential of these two concepts in possible joint schemes resulting from the merger of "the channel reuse" potential of cellular systems and the traffic handling capacities of packet radio schemes.

The first study was carried out by Schiff (58) who analyzed the throughput of a packet radio cellular system under three different channel assignment schemes. The analysis was based on the pure ALOHA random access techniques for those cases in which each cell base station was within range of its own mobile terminals, as well as the mobile terminals of neighbouring cells only.

We summarize below some of the principal results obtained by Schiff.

CHANNEL ASSIGNMENT ARRANGEMENTS

1st Arrangement

Let us assume that to operate anywhere in the cellular network each mobile terminal must operate (transmit/receive) on all available channels.

Then further if we assume that all of these mobile terminals are uniformly distributed throughout the network, then every cell base station will handle an equal amount of traffic.

Arranging the network so that the traffic in one cell is not interferred with by the traffic in any adjacent cells we can immediately apply the pure ALOHA result and write the carried traffic S_i in terms of the offered traffic G_i in cell i as:

$$S_i = G_i \, e^{-2G_i}$$

The total system throughput per channel will then be given by:

$$S = \frac{L}{M} S_i$$

where M is the number of channels and L the total number of cells in the system. The relationship between the total system throughput per channel, S, and the total system offered traffic per channel, G, is then:

$$S = Ge^{-\frac{2MG}{L}}$$

Measures of system performance is "the probability of successful packet transmission" and "the average number of packet transmissions required to successfully transmit a packet."

The probability of successful packet transmission will be given by:

$$P_s = e^{-\frac{2MG}{L}}$$

The average number of attempts required to successfully transmit a packet on this environment is $\frac{1}{P_s}$ or

$$\bar{n} = e^{\frac{2MG}{L}}$$

If instead of a pure ALOHA technique we use the Non Persistent Carrier Sense Multiple Access scheme, the relationship between S_i and G_i will be given by:

$$S_i = \frac{G_i\, e^{-\beta G_i}}{G_i(1+2\beta) + e^{-\beta G_i}}$$

where β is the ratio of propagation delay to packet transmission time. If, in this equation, we take the limit of β when it goes to zero, we find:

$$S_i = \frac{G_i}{1+ G_i}$$

by direct substitution the expressions for P_s and $\overline{n}$ are

$$P_s = (1+ \frac{MG}{L})^{-1}$$

$$\overline{n} = (1 + \frac{MG}{L})$$

2nd Arrangement

As an alternative to the fixed frequency plan described above one can envisage a cellular system arrangement in which the same channel is assigned to all cell base stations. This means that a given cell base station can receive packets transmitted from the mobile units located in its own cell as well as packets transmitted by the mobiles that are in adjacent cells. The stipulation being that a base station in any given cell only acknowledges packets transmitted by mobile units within its own cell, which can be accomplished by using an appropriate packet header.

For the pure ALOHA scheme Schiff obtained the following relationship between S_i and G_i:

$$S_i = G_i\, e^{-2IG_i}$$

where I is the number of cells in the interferring cell group formed by a given cell and its neighbours within range.

For this arrangement P_s and $\overline{n}$ are:

$$P_s = e^{\frac{-2IG}{L}}$$

and

$$\bar{n} = e^{\frac{2IG}{L}}$$

The corresponding equations for the CSMA are:

$$S_i = \frac{G_i}{1 + IG_i}$$

$$P_s = \left(1 + \frac{IG}{L}\right)^{-1}$$

$$\bar{n} = \left(1 + \frac{IG}{L}\right)$$

Concluding Remarks

The standard assumptions that terminals are always within range of each other is not always realistic and should be relaxed.

No published investigation as far as the author is aware has been made of the throughput of cellular packet radio network operating under the variety of random access schemes such as CSMA and employing hybrid or dynamic channel assignment methods. It would probably be easier to simulate the performance of such systems than to obtain "closed form" analytical results.

THE TRANSMISSION OF VOICE BY PACKET RADIO TECHNIQUES IN CELLULAR NETWORKS

As we mentioned much of the current cellular technology emphasizes analog FM. This emphasis, we feel, is overdone and has many shortcomings that have not been fully weighed. In this regard there are many other modulation techniques, such as spread spectrum and narrowband digitized voice that can increase the traffic carrying capacity of mobile voice communications systems.

Several companies (59) currently build hand-held portables, utilizing digital voice techniques and offering high reliability and security, for law enforcement and military applications. To date, one of the major problems in digitizing voice in mobile applications involves the selection of an encoding and modulation technique that will result in an intelligible transmission over a conventional 25 KHz or 30 KHz radio channel.

As far as encoding is concerned there are two main types of speech encoding methods. These are:

(1) waveform reconstruction methods which result in waveforms that look like the original ones (PCM, Delta Modulation, Adaptive DM, Continuous variable slope delta modulation ...), and

(2) analysis-synthesis methods which result in waveforms that sound like the original ones (Linear Predictive Coding).

Using either of these, in the case of digital voice, the resulting bit stream is then transmitted after modulation using either direct modulation of the carrier or a sub-carrier modulation technique. Recently a number of modulation techniques have been proposed, that appear to result in narrow emission spectra compatible with the allowable channel spacing. One such technique named "Tamed Frequency Modulation" (60) is claimed to achieve a very narrow radiated spectrum, in that for a modulation rate of 30 kbps the width of the corresponding radiated spectrum main lobe is of the order of 30 KHz (1 bit/hz).

As has been known for a long time, to simply digitize voice will not lead to any spectrum savings unless the voice digitization rate is low. Furthermore, we have known for some time how to take advantage of statistical properties of voice (see for example (61)) to avoid dedicating all of the transmission facilities to users during their "pauses". One such technique based on these statistical properties is known as TASI (Time Assignment Speech Interpolation) (62) which has been implemented on a submarine cable over the Atlantic ocean. Digital variations of the original TASI concept, such as Digital Speech Interpolation (DSI) (63) and Speech Predictive Encoding (SPEC) (64) have also been implemented. All of these are called "interpolation or compression techniques", and provide a 2 to 1 bandwidth compression or doubling of the effective number of analog channels on the facility.

Another technique that can take advantage of the bursty behaviour of speech is known as packet speech. This technique operates by using a speech detector that detects the bursts of speech and an encoder that digitizes these and fomats them in packets (see Figure 5) and a buffer to store the packets that have been generated. This approach eliminates all of the silent intervals (see Figure 5) within and between calls and, therefore, gives better sharing of the communications channel. However, some reduction in efficiency always occurs since added overhead per packet is always required. Depending on the type of network and its traffic control procedures, this overhead can consist of an abbreviated header (65) of about 16 bits or a more elaborate header of say 128 bits containing details of the coding algorithm, a time reference, the address of both the source and destination speaker, error control bits, etc.

The overall packet length should, however, be optimized to minimize the effect of transmission errors, delay, etc. Other design considerations that are important include the routing schemes flow control procedures, and the error detection and control.

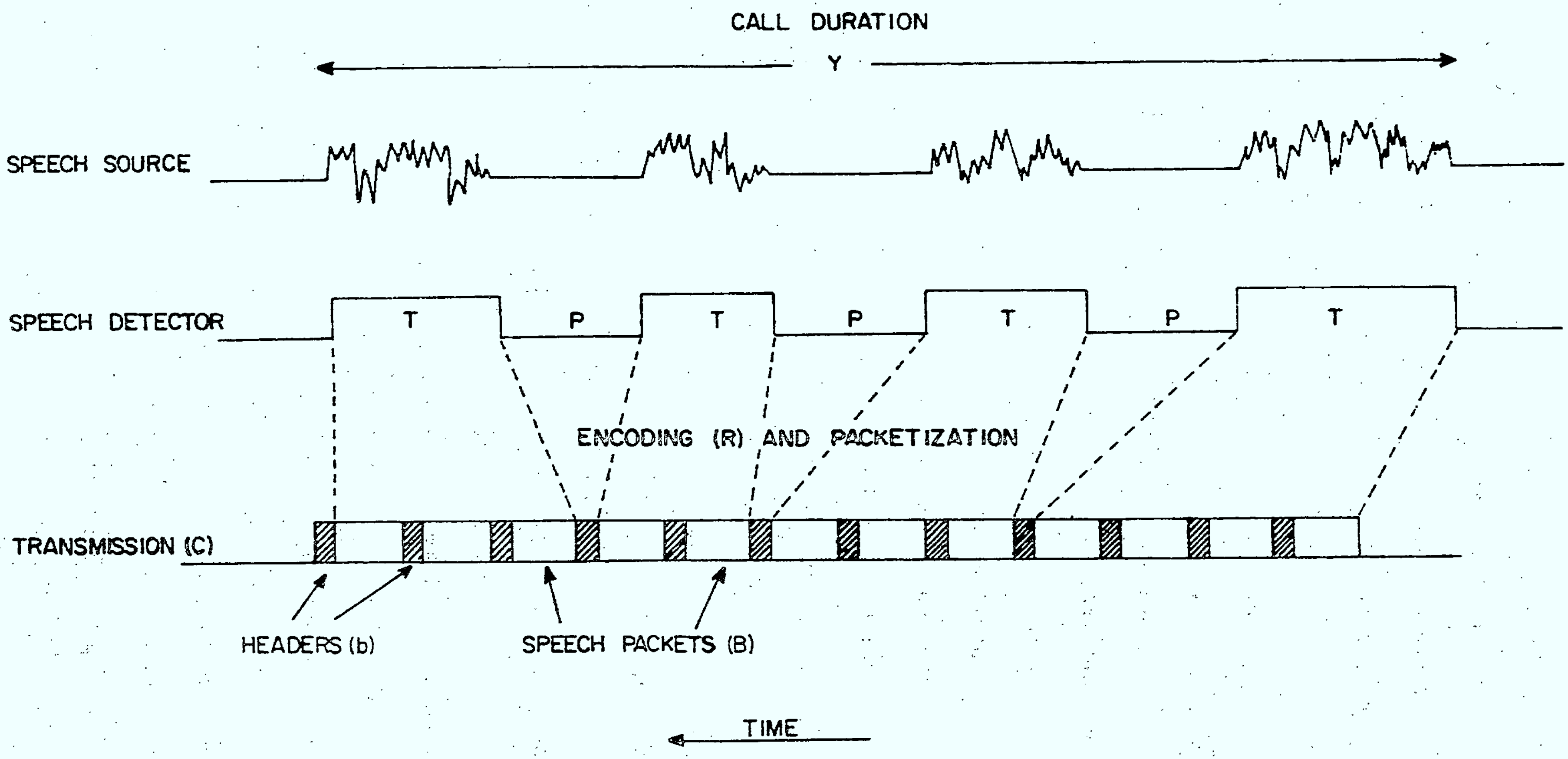
CALL DURATION
Y
SPEECH SOURCE
SPEECH DETECTOR
T
P
T
P
T
P
T
ENCODING (R) AND PACKETIZATION
TRANSMISSION (C)
HEADERS (b)
SPEECH PACKETS (B)
TIME

FIGURE 5

The results that have been obtained to date on the transmission of speech by packet come from experiments on the ARPANET (66) and, in summary, are that in order to maintain 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 250 msec.

It turns out that in practice quite surprisingly that:

- packets of lengths varying from 10 to 50 msec of speech intelligibility can be lost without seriously affecting the voice quality, and that
- serious degradation begins to be observed only after the amount of packets being lost exceeds 5% of the total number of packets being generated.

A Model for Packet Speech

Measurements have been carried out on the characteristics of speech during telephone conversations that have shown that human speech is bursty in nature. Brady (67) has confirmed that the actual channel utilization during a telephone conversation is only about 40%, and that a typical call consists of several talkspurts between pauses (see Figure 5).

Examination of a typical voice record have confirmed that the duration of talkspurts and pauses are exponentially distributed. Wang (68) and Kekre et al (69), (70) have used the exponential distribution to investigate the possibility of integrating voice and data on the same channel. Recently, Weinstein (71) analyzed the performance of a packet speech link and showed that the fraction of lost packets due to the unavailability of the communications channel is independent of the probability density function of the talkspurt duration.

On the basis of these and our own studies, we suggest that a model of packet speech for land mobile communications can be constructed as follows*.

The probability density function of the duration of pauses is:

$$P(t) = ae^{-at} \qquad \text{with} \qquad \overline{P} = \frac{1}{a}$$

The probability density function for the duration of the talkspurts is:

$$T(t) = be^{-bt} \qquad \text{with} \qquad \overline{T} = \frac{1}{b}$$

and the probability density function for the distribution of the silent periods between calls on the channels is:

$$S(t) = ce^{-ct} \qquad \text{with} \qquad \overline{S} = \frac{1}{c}$$

Assuming that the number of pauses per call is geometrically distributed then, the probability $F(k)$ that there are exactly k pauses in an average call, is

$$F(k) = \theta(1-\theta)^{k-1} \qquad k = 1,2,\ldots.$$

where $\overline{k} = \frac{1}{\theta}$ is the average number of pauses per call.

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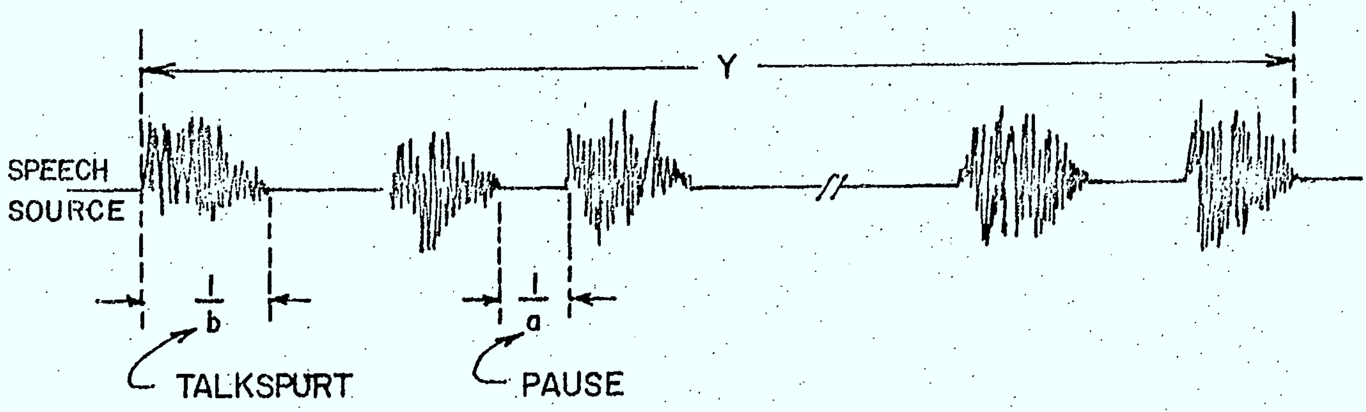
* See also the graphical representation (Figure 6).

MEASUREMENT DATA
(TELEPHONE CHANNELS)



AVERAGE PAUSE DURATION $= \frac{1}{a} = 1.802$ sec

AVERAGE TALKSPURT DURATION $= \frac{1}{b} = 1.366$ sec

MEAN NUMBER OF PAUSES PER CALL $= \bar{k} = 56.5$

MEAN CALL DURATION $= \bar{y} = 180.36$ sec

Figure 6

The probability density function Y(t) of the random variable Y (duration of a call) can therefore be obtained as the convolution of k negative exponentials with parameter a, and k-1 negative exponentials with parameter b.

This is a straightforward exercise in the Lapalce transform domain, and denoting

$$\mathcal{L}\{Y(t)\} = Y(s)$$

$$Y(s) = \sum_{k=1}^{\infty} \theta(1-\theta)^{k-1} \frac{a^{k+1}\, b^{k}}{(s+a)^{k+1}(s+b)^{k}}$$

which can be rewritten after a few manipulations as:

$$Y(s) = \frac{a^2 b\theta}{(s+a)\ (s-\alpha_1)\ (s-\alpha_2)}$$

where

$$\alpha_{1,2} = -\frac{a+b}{2} \pm \sqrt{\left(\frac{a+b}{2}\right)^2 - ab\theta}$$

The probability density function Y(t) is given by the inverse transform of Y(s):

$$Y(t) = a^2 b\theta \left\{ \frac{e^{-at}}{ab(\theta-1)} + A_1\, e^{+\alpha_1 t} + A_2\, e^{+\alpha_2 t} \right\}$$

where A_1 and A_2 are given by:

$$A_1 = \frac{1}{(\alpha_1+a)\ (\alpha_1 - \alpha_2)}$$

and

$$A_2 = \frac{1}{(\alpha_2+a)\ (\alpha_2-\alpha_1)}$$

The average call $\overline{Y}$ duration is

$$\overline{Y} = \int_0^\infty t\ Y(t)dt$$

and substituting we find

$$\overline{Y} = \frac{1}{a\theta} + \frac{1+\theta}{b\theta}$$

Using the values obtained by Brady (67) we obtain the results as shown in Figure 6.

Assuming that the voice waveform is being digitized at a rate R (bits/sec), then the average number of packets generated during a talkspurt, where the average talkspurt duration $\overline{T} = \frac{1}{b}$, is

$$\overline{N}_t = \frac{R\overline{T}}{B}$$

where B is the length of the data block or packet that is being generated at the source encoding point*.

---

* NOTE: b is the overhead that must be added to each source data packet of length B before it is transmitted. So what is transmitted over the channel is a packet of length B+b.

The average number of packets generated per call will be:

$$\overline{N}_c = \overline{N}_t \; (\overline{k}+1)$$

The factor (k+1) arises because $\overline{k}$ is the mean number of pauses per call and there will always be one more talkspurt than there are pauses. Substituting for $\overline{N}_t$ gives

$$\overline{N}_c = \frac{R\overline{T}(\overline{k}+1)}{B}$$

The average number of packets generated per mobile station during the busy hour is simply $\overline{N}_c\lambda$ where λ is the average number of calls being generated by each mobile voice terminal during its busiest period.

To avoid overflow, the rate at which packets are being transmitted must be greater than or equal to the rate at which they are being generated (otherwise extensive buffering would be required). This bound can be written as follows, assuming that C is the transmission rate of the channel, then

$$C \geq \frac{R(B+b)}{B}$$

Delay Considerations

The total end to end (that is human to human) delay using packet speech transmission should not exceed 250 msec, or the speech is found to be uncomfortable to humans. In a typical radio system using packet speech, the various sources of delay are:

- packetization delay
- transmission delay
- propagation delay

- processing delays
- queuing delays

The packetization delay, i.e. the time it takes to form a packet of size B, when encoding speech at a rate R is given by:

$$D_p = \frac{B}{R}$$

The transmission delay D_t or the time required to transmit a packet of size B + b on a channel operating at a rate C is:

$$D_t = \frac{B + b}{C}$$

and the total end-to-end delay D^*(neglecting propagation delay) is

$$D = 2D_p + D_t$$

The propagation delay is a function of the distance travelled by a packet and becomes significant on transcontinental paths (≈50msec) or satellite paths (≈250msec). However, for the distances involved in land mobile communications, the propagation delay is only 54 μsec, for a 10 mi path, and can be neglected.

The processing delays account for a negligible amount of delay since they include the speech detection delays, header insertion and removal and logic operations.

The queuing delays which can constitute a significant element of the total end-to-end delay, are a function of the channel utilization, the network topology, the routing procedures, etc. For land mobile networks with

* The factor 2 D_p arises because of the packet requiring a time D_p to be assembled and a time D_p to be disassembled at the receiver.

at most two hops (mobile to base and base to mobile) such queuing delays should not give a significant contribution to overall delay.

Example:

Assuming a digitization rate R of 2.4 kbps (LPC) or 16 kbps (CVSD), a transmission rate C of 30 kbps and 50 msec of speech intelligibility per packet (also called packet duration), we obtain the following data:

$$B = \begin{cases} 120 \text{ bits (LPC)} \\ 800 \text{ bits (CVSD)} \end{cases}$$

$$b = 16 \text{ bits (abbreviated header)}$$

$$B + b = \left.\begin{cases} 136 \text{ bits (LPC)} \\ 816 \text{ bits (CVSD)} \end{cases}\right\} = \text{transmitted packet length}$$

$$D_p = 50 \text{ msec} \quad \} = \text{time taken to assemble data and source}$$

$$D_t = \left.\begin{cases} 4.5 \text{ msec (LPC)} \\ 27.2 \text{ msec (CVSD)} \end{cases}\right\} = \text{time taken to transmit packet length } B + b$$

$$D = 2D_p + D_t \left.\begin{cases} 104.5 \text{ msec (LPC)} \\ 127.2 \text{ msec (CVSD)} \end{cases}\right\} = \text{average total end-to-end time neglecting propagation delay, processing and queueing delays}$$

LPC ≡ Linear predictive coding

CVSD ≡ continuous variable slope delta modulation

Note: LPC is more costly than CVSD but its bits/λz capabilities are substantially better.

Concluding Remarks

Speech waveforms can be economically digitized using CVSD codecs, etc., and translated into packets (called voice packets). Even though there has been indication of the potential for integrating, both packet speech and packet data on the same network, no studies have so far been carried out in the published literature on integrated voice/data packet radio networks. We hope that the results reported herein will stimulate further studies in the area.

In this regard we are convinced that further study of packet radio mobile system will confirm that smaller number of radio channels than are currently required for trunked cellular systems would be practical and easier to implement. As has been mentioned previously the performance parameter of conventional land mobile systems is the blocking probability however in a packet speech radio system we suggest that a more appropriate performance measure would be "the fraction of lost speech, due to channel unavailability."

THE OPTIMAL PACKET SIZE FOR PACKET RADIO CHANNELS

In packet radio systems transmission errors result from ignition noise, fading and packet overlaps. The contribution of fading which causes burst errors is far more significant in packet radio channels as a source of error than any other type of error. Yet there are no published investigation of fading errors in packet radio networks.

Various error control procedures designed to operate over fading channels have been proposed. It has been suggested by Cavers (72) that the receiver could monitor the channel and request the transmitter to stop sending when the signal drops below a certain threshold level. Several other studies attempt to describe and quantify errors on mobile radio channels (73), (74), (75).

However the available results assume that the sending station transmits a continuous bit stream to the mobile terminal. No results have so far been published on the error characteristics of packet speech on mobile radio channels. We will now discuss some of our recently obtained results on this problem.

<u>Channel Model</u>

Two types of errors namely random errors and burst errors can be considered when dealing with mobile radio channels. The probability P_B that one or more errors occur in a given packet is*

$$P_B = 1 - (1-P_\ell)^B$$

where P_ℓ is the average bit error rate and B the packet size. This equation is valid for a random error channel in which bit errors occur independently of each other. In the case of burst errors arising as a result of fading, this equation is not valid. Eventhough fading contributes more to error rate on a packet radio channel than thermal noise, etc., its effects have been untreated in the literature and we now give some of our recently obtained results on this problem (80).

In order to derive an equivalent expression to P_B for the fading case, consider Figure 7 which shows that whenever the received signal level falls below a certain threshold a burst of errors of variable duration occurs.

With reference to Figure 7, consider the transmission of a packet of data of duration T_B sec over the fading channel, and it is immediately obvious that we can write the following expression for the fraction of time η that a packet or part of a packet will overlap a fade period, as

---

* $(1 - P_\ell)^B$ is the probability that a packet of size B contains no errors.

$$\eta = \frac{1}{T_o} \left\{ \sum_{i=1}^{M} \tau_i + T_B \right\}$$

In the equation for N, M is the number of fade periods within a time interval of length T_o and τ_i is the duration of the i^{th} fade period. Assuming that any degree of overlapping between a packet and a fade period will result in at least one detected error, then every such affected packet will have to be retransmitted.

If we take the limit when T_o goes to infinity, the probability P_B that a given packet will have at least one detected error will be proportional to η. Hence,

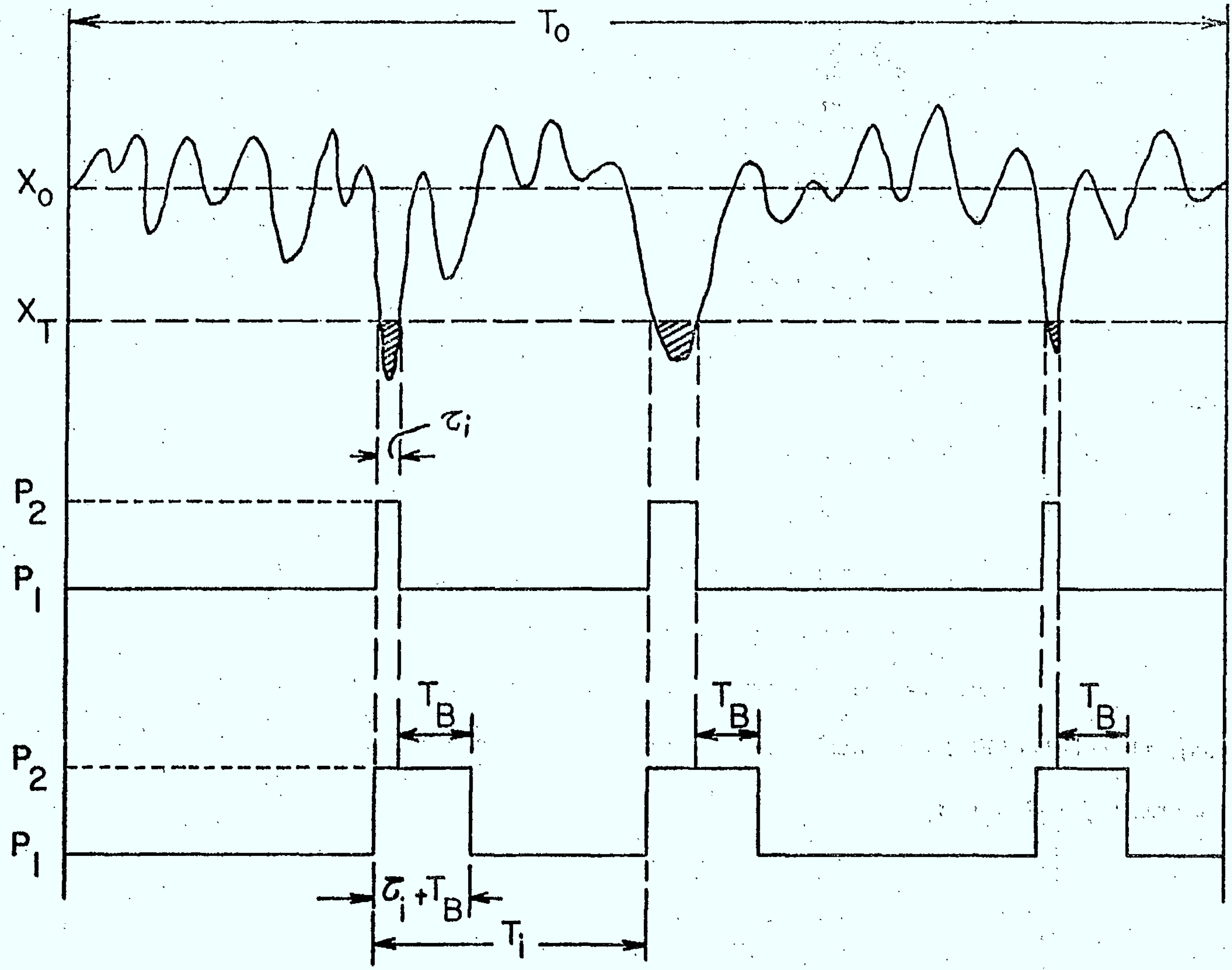
$$P_B = \gamma \left\{ T_B \lim_{T_o \to \infty} \frac{M}{T_o} + \lim_{T_o \to \infty} \frac{1}{T_o} \sum_{i=1}^{M} \tau_i \right\}$$

where γ is a constant of proportionality.

The first limit expression in the above equation for P_B represents the threshold level crossing rate, N_t, while the second limit expression represents the probability that the received signal level is less than or equal to an arbitrarily specified threshold level x_t. The equation can therefore be rewritten as

$$P_B = \gamma \left\{ T_B N_t + P(x \leq x_t) \right\}$$

where following Jakes (76), N_t is defined as the expected rate at which the envelope crosses the threshold level x_t in the positive direction. In general N_t and $P(x \leq x_t)$ are given by



X_o	=	Mean power of the received signal
X_T	=	Threshold level of the receiver
P_1	=	Error probability when the signal is above $X_T \approx 0$
P_2	=	Error probability when the signal is below $X_T = 0$
T_B	=	Packet duration
τ_i	=	Fade duration
T_i	=	Interfade intervals
M	=	Number of fade intervals in an interval of length T

FIGURE 7

$$N_t = \sqrt{\frac{2\pi x_t}{x_o}} \; \frac{vf_o}{C} \; \exp\left[-\frac{x_t}{x_o}\right]$$

and

$$P(x \leq x_t) = 1 - \exp\left[-\frac{x_t}{x_o}\right]$$

where:

f_o - carrier frequency

v - vehicle speed

c - speed of light

x_o - received mean power

If we consider medium term mean value variations, the local mean signal level will be lognormally distributed (77) and for $x_o >> x_t$ we find for $P(x \leq x_t)$

$$P(x \leq x_t) = \frac{x_t}{\overline{x}_o} \; \exp\left[\frac{\sigma}{4.3}\right]^2$$

where

$\overline{x}_o$ = mean value of the mean of the received power

σ = standard deviation of $\overline{x}_o$ (in dB)

The corresponding expression for N_t, denoted by $\overline{N}_t$, is given by:

$$\overline{N}_t = \sqrt{\frac{2\pi x_t}{\overline{x}_o}} \cdot \frac{vf_o}{C} \cdot \exp\left\{\frac{3}{8} \cdot \left[\frac{\sigma}{4.3}\right]^2\right\}$$

Substituting equations we find P_B for the fading case is

$$P_B = \gamma\left\{T_B \cdot \frac{vf_o}{C} \cdot \sqrt{\frac{2\pi x_t}{\overline{x}_o}} \exp\left\{\frac{3}{8} \cdot \left[\frac{\sigma}{4.3}\right]^2\right\} + \frac{x_t}{\overline{x}_o} \cdot \exp\left[\frac{\sigma}{4.3}\right]^2\right\}$$

In order to express P_B in terms of the packet length it is only necessary to replace T_B the packet duration, by

$$T_B = \frac{B+b}{R}$$

where B, b and R are as defined previously.

Substituting this result gives

$$P_B = \gamma \left\{ \frac{(B+b)}{R} \frac{vf_o}{C} \sqrt{\frac{2\pi x_t}{\overline{x}_o}} \exp\left[\frac{3}{8} \cdot \left[\frac{\sigma}{4.3}\right]^2\right] + \frac{x_t}{\overline{x}_o} \exp \left[\frac{\sigma}{4.3}\right]^2 \right\}$$

The constant of proportionality, γ, can be obtained by noting that the packet error rate should be equal to the bit error rate P_ℓ, when in the limit, T_B goes to zero.

An approximate expression for P_ℓ, obtained when $\overline{x}_o >> x_t$, is given by

$$P_\ell = P_{\ell max} \frac{x_t}{\overline{x}_o} \cdot \exp \left[\frac{\sigma}{4.3}\right]^2$$

where $P_{\ell max}$, the error rate below the threshold level x_t is usually taken to be equal to ½.

From the above expression for P_B, we find

$$P_\ell = 2\gamma P_\ell$$

or $$\gamma = \tfrac{1}{2}$$

The final expression for the packet error rate will then be:

$$P_B = P_\ell + \frac{(B+b)}{2 \cdot R} \cdot \frac{vf_o}{C} \cdot \sqrt{\frac{2\pi x_t}{\overline{x}_o}} \cdot \exp\left[\frac{3}{8} \cdot \left[\frac{\sigma}{4.3}\right]^2\right]$$

Using the expression for the packet error rate P_B in a fading environment it is possible to derive an expression for optimal packet size as a function of a bit rate, the vehicle speed, the relative signal level, the frequency, etc.

One other important result is now within our reach in addition to being able to determine the optimal packet size and this is an expression for the throughput degradation of packet radio channels that takes into account packet collisions AND fading (80). In the case of a pure ALOHA channel the throughput - traffic equation previously given would become in the presence of fading

$$S \doteq Ge^{-2G}\ (1-P_B)$$

where P_B is as given above. For other random access techniques similar expressions can be obtained.

4.3.2 Concluding Remarks

It is a relatively straightforward exercise, following for example Chu (78) to determine the optimal packet size that minimizes the expected packet throughput time in a fading environment.

In such a model as previously mentioned on the base to mobile channels we can assume that transmissions errors are only caused by fading, i.e., errors due to ignition noise need not be considered. Errors due to packet collisions can also be neglected on the base to mobile packet radio channel. This would however not be the case on the mobile to base channel. We can further assume that the acknowledgement traffic carried by a separate channel arrives at the fixed station reliably, and we can also ignore propagation delays.

Since transmission errors due to channel fading will increase the probability of unsuccessful transmission in a packet radio channel, a study of the throughput degradation for a representative set of random access schemes has been undertaken. It turns out that models for the throughput degradation a pure ALOHA, slotted ALOHA and CSMA packet radio channels due to fading and collisions are fairly easy to derive.

INTERFERENCE TO UHF TELEVISION RECEIVERS FROM MOBILE TERMINALS

The allocation of the top 14 UHF (namely channels 69 - 83 UHF-TV) channels to mobile communications has led to a potential image interference problem to UHF-TV channels 58 to 61 (that operate between 734 MHz and 758 MHz - See Figure 8).

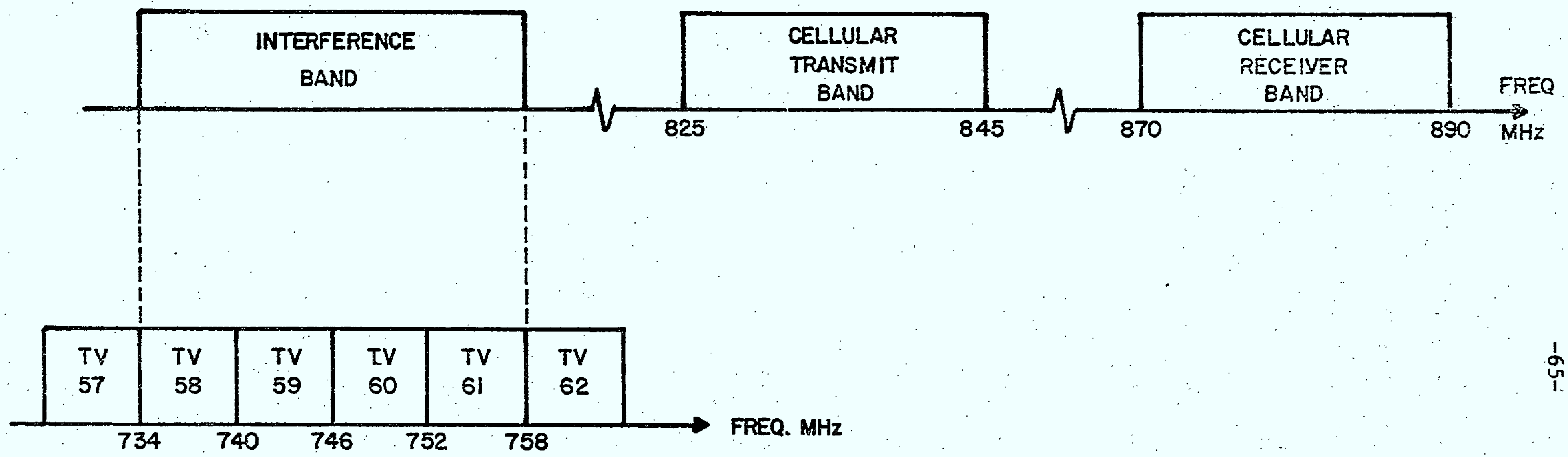
This problem arises because the band allocated to cellular transmit frequencies (see also figure 8) 825 MHz - 845 MHz will "beat" with the local oscillator of the television set to produce interference signal products in the band occupied by channels 58 - 61 UHF-TV. Similar statements can be made for the other portions of the mobile UHF band between 806MHz - 810 MHz.

Fisher (79) suggested that it would not be prudent to assign UHF-TV channels 58 through 61 for use in areas served by cellular systems. His results are that there is interference to TV receivers operating in channels 58 - 61 from such cellular system mobile transmitters, especially if these TV receivers are at the grade B contour of their reception area. This analysis, however, did not assign any probabilities to the likelihood that such interference would occur. We investigated this problem and have found that the probability of no interference even under very dense concentration of mobiles (>10 per square mile) is 99% or better. Assignments of UHF television in this band should therefore proceed on schedule.

Let us now discuss our analysis methodology underlying this result.

The Probability of Interference Model

Let us now derive an expression for the probability that a randomly selected TV receiver will suffer from interference caused by mobile transceivers in an area served by a high capacity cellular system. This model comes from the recent work of da Silva (81).



CELLULAR FREQUENCY PLAN

FIGURE 8

The model is based on the assumption that several mobile transceivers are geographically scattered over a given interference area. The radius of this circular area, denoted by R_i, corresponds to the area in which a mobile transmitter can cause interference to a UHF TV receiver located in the center of the area. Denoting the geographical density of mobile stations by D, then the average number $\bar{n}$ of mobiles in a circular area, A, of radius R_i will be given by:

$$\bar{n} = \Pi R_i^2 D = AD$$

A good assumption is that the number of mobile stations within this area is Poisson distributed with mean $\bar{n}$. That is the probability of finding n mobiles in this area will be given by:

$$P(n) = \frac{\bar{n}^n e^{-\bar{n}}}{n!}$$

Denoting by ρ the average traffic load per mobile station, the probability P_{NI} of no interference when there are n stations in the interference area is given by:

$$P_{NI}(n) = P(n)\ (1-\rho)^n$$

Summing over all possible values of n we obtain:

$$P_{NI} = \sum_{n=0}^{\infty} P(n)\ (1-\rho)^n$$

$$P_{NI} = e^{-\bar{n}\rho}$$

Several authors have indicated that a typical interference area is about 0.1 mi^2 and that an average mobile density of 10 mobiles per mi^2 would be characteristic of a mature cellular system. Using these figures in the above equations we can find that $\bar{n}$, the average number of mobiles within an interference area, is of the order of 1.

It is interesting to note that for typical values of ρ between 0.01 and 0.02 erlangs/mobile the probability of no interference varies between 99% and 98%.

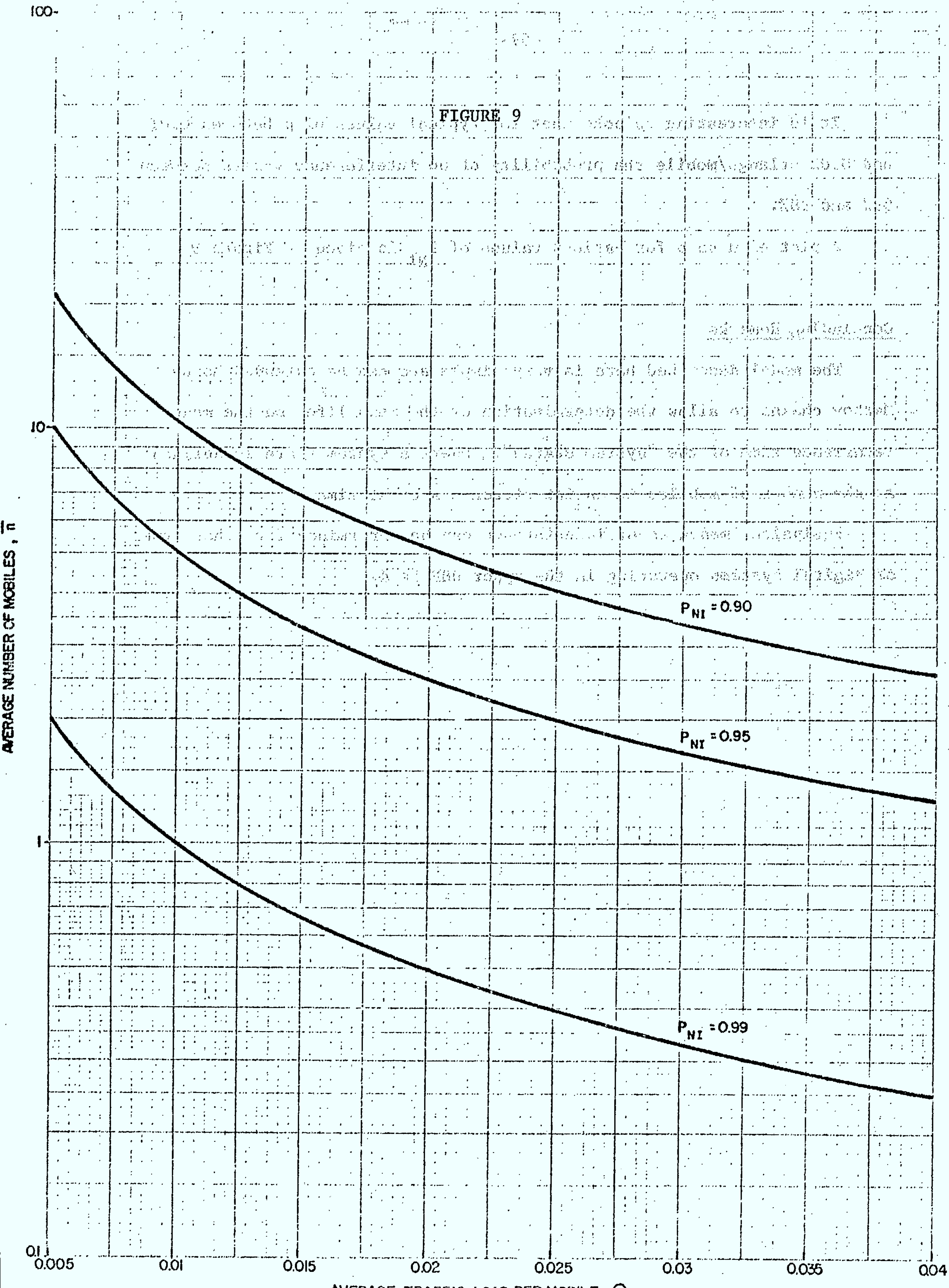
A plot of $\overline{n}$ vs ρ for various values of P_{NI} is given in Figure 9.

Concluding Remarks

The model described here is very simple and can be extended using Markov chains to allow the determination of the mean life and the mean recurrence time of the "system states". Where a system state is defined as the number of mobiles in an interference area at time t.

Equivalent measures of interference can be determined for other types of digital systems operating in the upper UHF band.

FIGURE 9


100
10
1
0.1
0.005
0.01
0.015
0.02
0.025
0.03
0.035
0.04
AVERAGE NUMBER OF MOBILES , n̄
P_NI = 0.90
P_NI = 0.95
P_NI = 0.99

5. AMATEUR PACKET RADIO IN CANADA

In September of 1978, Canada amended its radio regulations to introduce a new type of amateur radio operator certificate (82) (83). This certificate is called the Amateur Digital Radio Operator's Certificate and holders of this certificate (there are currently more than 50 digital operators and the number is growing), can conduct packet and digital radio experiments on man/man, man/machine and machine/machine communications in the amateur VHF bands and higher. The allocations and types of emissions open to this new class of amateur experiments are summarized in Table III.

It is expected that these new allocations will promote closer collaboration among radio and computer hobbyists and foster a better understanding of resource sharing concepts among Canada's experimental hobbyists.

TABLE III

FREQUENCY BAND (MHz)	EMISSION TYPES	RESTRICTIONS
145.5 - 145.8	A0, A1, A2, A3, A4 F1, F2, F3, F4 P0, P1	PULSE EMISSIONS 15 kHz PEAK POWER 100 WATTS AVERAGE POWER 10 WATTS
220.1 - 220.5 220.5 - 221.0 221.0 - 223.0 223.0 - 223.5	OPEN OPEN OPEN OPEN	PACKET EMISSIONS 10 kHz, SHARED PACKET EMISSIONS 100kHz, SHARED PACKET EMISSIONS 25 kHz, EXCLUSIVE PACKET EMISSIONS 100kHz, SHARED
433.0 - 434.0 434.0 - 434.5	OPEN A0, A1, A2, A3, A4, A4 F1, F2, F3, F4, P0, P1, P2, P3	PACKET EMISSIONS 100kHz, EXCLUSIVE PULSE EMISSIONS 30 kHz PEAK POWER 100 WATTS AVERAGE POWER 10 WATTS
1215.0 - 1300.0 2300.0 - 2450.0 3300.0 - 3500.0 5650.0 - 5925.0 10000.0 - 10500.0	A0, A1, A2, A3, A4, A5 F1, F2, F3, F4, F5 P0, P1, P2, P3, P4, P5, P9	PEAK POWER 2.5 k WATTS AVERAGE POWER 25 WATTS
24000.0 - 24010.0 24010.0 - 24250.0	OPEN A0, A1, A2, A3, A4, A5 F1, F2, F3, F4, F5 P0, P1, P2, P3, P4, P5, P9	PACKET SHARED PEAK POWER 2.5 k WATTS AVERAGE POWER 25 WATTS

AMATEUR PACKET RADIO ALLOCATIONS

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