Government of Canada
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PACKET SPEECH FOR LAND MOBILE CHANNELS

by

J. Schwarz daSilva

June 1979

Telecommunication
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Service de la Réglementation
des Télécommunications

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ABSTRACT

This report describes the results of a study to evaluate the feasibility of applying packet speech concepts to land mobile systems.

One particular random access technique (NPCSMA) is investigated and it is shown that for typical values of the system parameters packet speech compares favourable to conventional trunked land mobile system.

An expression is derived for the maximum number of active users that can be supported by a single 30 KHz channel, as a function of the voice digitization rate and the lost packet level.

It is further shown that the number of base-to-mobile channels does not need to be equal to the number of mobile-to-base channels.

TABLE OF CONTENTS

INTRODUCTION

In the past few years considerable interest has been demonstrated in digital voice techniques [1] [2], packet radio random access schemes [3] [4] and cellular structures for land mobile communications [5] [6].

A number of experiments have been carried out over the ARPANET [7], to demonstrate the feasibility of transmitting packetized voice. Also the integration of packetized data and packetized voice has been advocated by some researchers [8] who have attempted to quantify the performance measures of such integrated networks and provide some design guidelines.

Closely related to the idea of packetized voice is the concept of speech interpolation which permits a number of voice sources to share a number of channels through voice-activated switching by taking advantage of the gaps or pauses that naturally occur during a conversation. One of the earliest [9] speech interpolation systems known as TASI (Time Assignment Speech Interpolation) was a pure analog system whose performance parameter was the fraction of speech lost due to freeze-outs. A freeze-out occurs when a given talkspurt finds no channel idle. Recently a digital version of speech interpolation known as DSI has been advocated in particular for satellite circuits [10].

On the other hand a considerable body of knowledge [11] [12] ... exists for the so called packet radio systems whose potential have been exploited uniquely for data transmission purposes. Packet radio concepts have also been applied to the control channels of the Chicago [13] and Tokyo [14] cellular systems. It has also been suggested by Norway [15] that packet radio concepts could be efficiently applied to the signalling channels of the maritime mobile service.

However, to the author's knowledge no attempt has been made so far to apply the concepts of digital speech interpolation and packetized voice to increase the traffic carrying capacity of land mobile systems.

In this study we show that for typical values of the system parameters, a land mobile system that employs packetized voice concepts, compares favourably to the conventional trunked land mobile system. We derive an expression for the maximum number of active users that can be supported by a single 30 KHz uplink channel (mobiles to base) and show that the number of downlink (base to mobiles) channels does not need to be equal to the number of uplink channels.

i.e. radio frequency

Operational Concept

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:

$$P_B = \frac{\rho^M/M!}{\sum_{n=0}^{M} \rho^n/n!} \qquad \ldots\ldots (1)$$

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:

$$\rho_c = \rho\left[1-P_B\right] \qquad \ldots\ldots (2)$$

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.

The same channels could be shared, perhaps 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 the model of Figure (1) 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).

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 nobody else is in the process of transmitting a voice packet.

Fortunately enough such access protocols have already been extensively studied [16], in the context of packetized data transmission. 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 previously collided packets per packet transmission time it can be shown [16] that "S" and "G" are related by:

$$S = \frac{G\, e^{-\delta G}}{G(1+2\delta) + e^{-\delta G}} \qquad \ldots\ldots (3)$$

where "δ" the normalized propagation delay is given by:

$$\delta = \frac{\tau}{T_p} \qquad \ldots\ldots (4)$$

where "τ" is the one way propagation delay between any two terminals and "T_p" is the packet transmission time. It can also be shown [16] that when a packet is ready for transmission, the probability "θ" that the channel is busy is given by:

$$\theta = \frac{G(1+\delta) - 1 + e^{-\delta G}}{G(1+2\delta) + e^{-\delta G}} \qquad \ldots\ldots (5)$$

From equation (3) we can establish that a packet will be successfully transmitted with probability "ξ" given by:

$$\xi = \frac{S}{G} = \frac{e^{-\delta G}}{G(1+2\delta) + e^{-\delta G}} \qquad \ldots\ldots (6)$$

For the case of voice packets we suggest a slight modification of the NPCSMA protocol, namely, we do no attempt to retransmit a packet that has collided. This implies that in the above equations "G" will denote the offered channel traffic and "S" the actual successfully carried traffic.

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 "G" the normalized offered channel traffic. We must now relate "G" 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 [17] with means "$\bar{T}$" and "$\bar{P}$". Hence the probability density functions for the random variables "T" and "P" are given by:

$$f_T(t) = \frac{1}{\bar{T}} \exp \left(- \frac{t}{\bar{T}}\right) \qquad \ldots\ldots (7)$$

and

$$f_P(t) = \frac{1}{\bar{P}} \exp \left(- \frac{t}{\bar{P}}\right) \qquad \ldots\ldots (8)$$

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:

$$P \left[RT \leq nB \right] = 1 - \exp\left(- \frac{nB}{R\bar{T}}\right) \qquad \ldots\ldots (9)$$

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

$$P \left[(n-1)B < RT \leq nB \right] = \exp \left(- \frac{(n-1)B}{R\bar{T}}\right) - \exp \left(- \frac{nB}{R\bar{T}}\right) \qquad \ldots\ldots (10)$$

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

$$\bar{n} = \left\{ 1 - \exp \left(- \frac{B}{R\bar{T}}\right) \right\}^{-1} \approx \frac{R\bar{T}}{B} \qquad \ldots\ldots (11)$$

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:

$$a = \frac{\bar{T}}{\bar{T} + \bar{P}} \qquad \ldots\ldots (12)$$

The source activity ratio can also be interpreted as the probability that an active speech source is issuing a talkspurt at some random time. Studies [17] have shown that "a" is typically of the order of 0.4.

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:

$$T_p = \frac{B+b}{C} \qquad \ldots\ldots (13)$$

where "C" denotes the channel transmission rate.

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

$$g = \eta\, a\, \frac{R}{C} \cdot \frac{B+b}{B} \qquad \ldots\ldots (14)$$

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

$$G = \eta\, a\, \frac{R}{C} \cdot \frac{B+b}{B} \cdot \frac{N}{M} \qquad \ldots\ldots (15)$$

Traffic Model

Consider Figure (2) 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 of 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 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:

$$\alpha = \theta^k \qquad k=1,2,.... \qquad ...... (16)$$

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

$$(1-\alpha) = \sum_{i=1}^{i=k} \theta^{i-1}(1-\theta) = 1-\theta^{k} \qquad \ldots\ldots (17)$$

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 talkspurt 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:

$$\emptyset_u = \alpha + (1-\alpha)(1-\xi) \qquad \ldots\ldots (18)$$

To determine the fraction of lost packets on the downlink channels we assume that packets arrive at the base station via "M" channels. Since "θ" is the probability that any one of these channels is busy, if there are "L" downlink channels (L<M) the fraction of lost packets, "$\emptyset_d$", is obtained from:

$$\emptyset_d = \frac{\sum_{i=L+1}^{M} (i-L)\binom{M}{i}\theta^{i}(1-\theta)^{M-i}}{M\theta} \qquad \ldots\ldots (19)$$

To obtain the above expression 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. 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.

End-to-End Delay

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:

$$D = \frac{3B}{R} \qquad \ldots\ldots (20)$$

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.

Discussion and Results

Before applying the previous equations to a specific set of system parameters it is worthwhile to mention 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 [17] 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 [2] [18] 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 250 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. Hence denoting by "$\emptyset$" the total fraction of lost packets we have from equations (19) and (18):

$$\emptyset = \emptyset_u + \emptyset_d \leq \begin{cases} 0.005 \text{ for } R \approx 2.4 \text{ kbps} \\ 0.2 \text{ for } R \approx 16 \text{ kbps} \\ 0.5 \text{ for } R \approx 32 \text{ kbps} \end{cases} \qquad \ldots\ldots (21)$$

From equation (15) it can be clearly seen that "G" 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 (3) and (4), are highly dependent on the values of "G". 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 [19], 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 [20] is all that is required to properly address the packets.

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:

$$K = \frac{B/R - T_p}{t_s} \qquad \text{...... (22)}$$

which, for all practical purposes is a very large number. Indeed from Table 1, if we assume a channel speed of 40 kbps and a sensing time of 0.03 msec, we see that in the worst case "K" 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 1 which were obtained for two temporal packet lengths of 10 msec and 50 msec (with $\eta=0.02$ and $a=0.4$) indicate, as expected, that for a given channel speed (40 kbps), as 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. We will show that rather on the contrary, the voice digitization rate should be kept as high as possible. Consider a single radio channel supporting an amount of traffic "G" given by equation (15). If we 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:

$$\emptyset = \emptyset_u = \alpha + (1-\alpha)\ (1-\xi) \approx 1-\xi \qquad \text{...... (23)}$$

Now from Figure (4) we see that, for $\delta = 0.0001$, "G" is related to "ξ" by:

$$G \leq \begin{cases} 1 & \text{for} \quad \xi \geq 0.5 \\ 0.25 & \text{for} \quad \xi \geq 0.8 \\ 0.005 & \text{for} \quad \xi \geq 0.995 \end{cases} \qquad \ldots\ldots (24)$$

which implies from (21) and the data of Table 1, that the maximum number of voice sources that can be supported is given by:

$$N \leq \begin{cases} \frac{1}{g} = 148 & \text{for} \quad R = 32 \text{ kbps} \\ \frac{0.25}{g} = 71 & \text{for} \quad R = 16 \text{ kbps} \\ \frac{0.005}{g} = 6 & \text{for} \quad R = 2.4 \text{ kbps} \end{cases} \qquad \ldots\ldots (25)$$

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 equation (20) to keep the packets as short as possible.

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 equation (25) 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 "θ" we select the appropriate number "L" of downlink channels. As an example assume that the digitization rate R is equal to 32 kbps and that we can tolerate a total fraction of lost packets of the order of 50%. From Figure (4) we find that G should be less than 1 and from Figure (3) we see that the value of "θ" corresponding to G=1 is of the order of 0.5. Hence using equation (19) with M=100, we see from Figure (5) that in order to keep "ϕ_d" below 0.001, the number "L" of required downlink channels is of the order of 60. We can then achieve a spectrum saving of the order of 40% which for 30 KHz channels represents about 1.2 MHz. Additional savings in spectrum can be obtained by allowing "ϕ_d" to increase while keeping "ϕ" below 50%.

Conclusion

Based on the results discussed above it appears that the concept of packet speech can be advantageously applied to land mobile channels. We have shown that for a given number of uplink channels, the maximum number of voice sources that can be supported will, under some assumptions attain a value of 148 M which is a three fold increase over what can be achieved with conventional analog land mobile channels. We have further shown that, as opposed to a conventional land mobile system where the same amount of spectrum is allocated in both directions, in a packet speech system the amount of spectrum required for the downlink channels represents about 60% of what is required in a comparable analog land mobile system.

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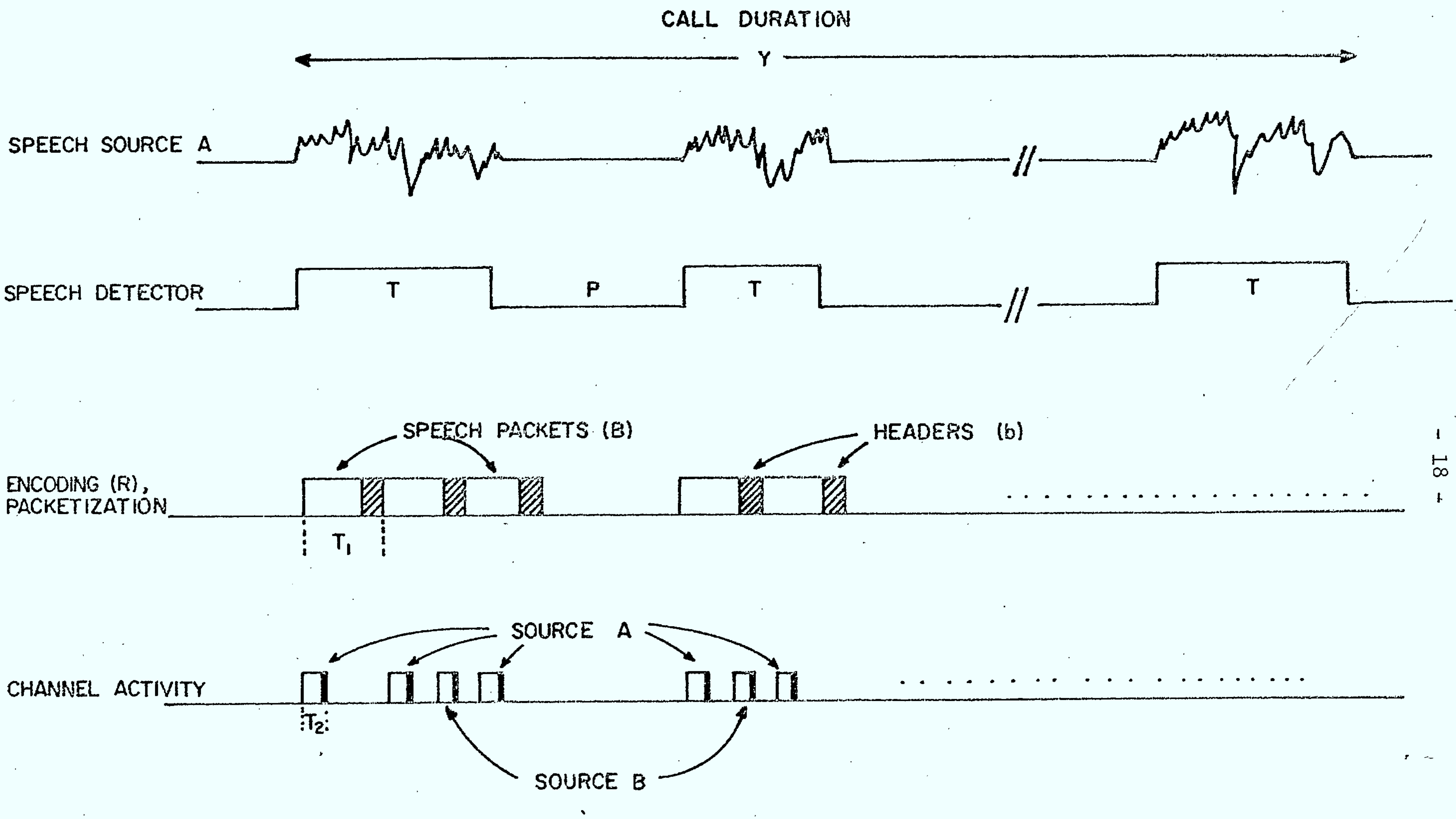
	B/R = 10 msec				B/R = 50 msec			
R(kbps)	B(bits)	B+b(bits)	T_p(msec)	g($x10^{-3}$)	B(bits)	B+b(bits)	T_p(msec)	g($x10^{-3}$)
2.4	24	40	1	0.8	120	136	3.4	0.54
4.8	48	54	1.35	1.08	240	256	6.4	1.02
16	160	176	4.4	3.52	800	816	20.4	3.26
32	320	336	8.4	6.72	1600	1616	40.4	6.46

TABLE 1

PACKET SPEECH MODEL

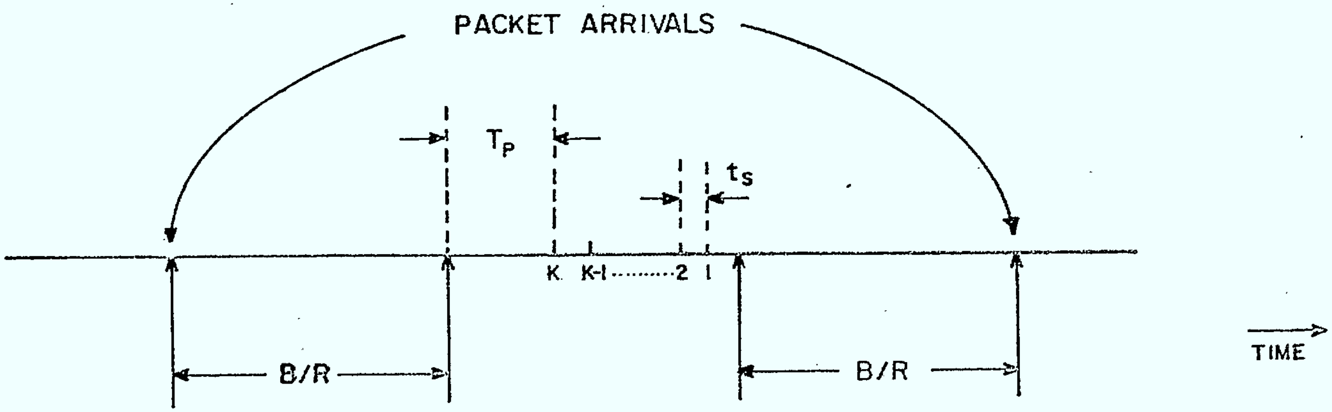


$T_2 = \frac{C}{R} T_1$

TIME

FIGURE 1

TRAFFIC MODEL

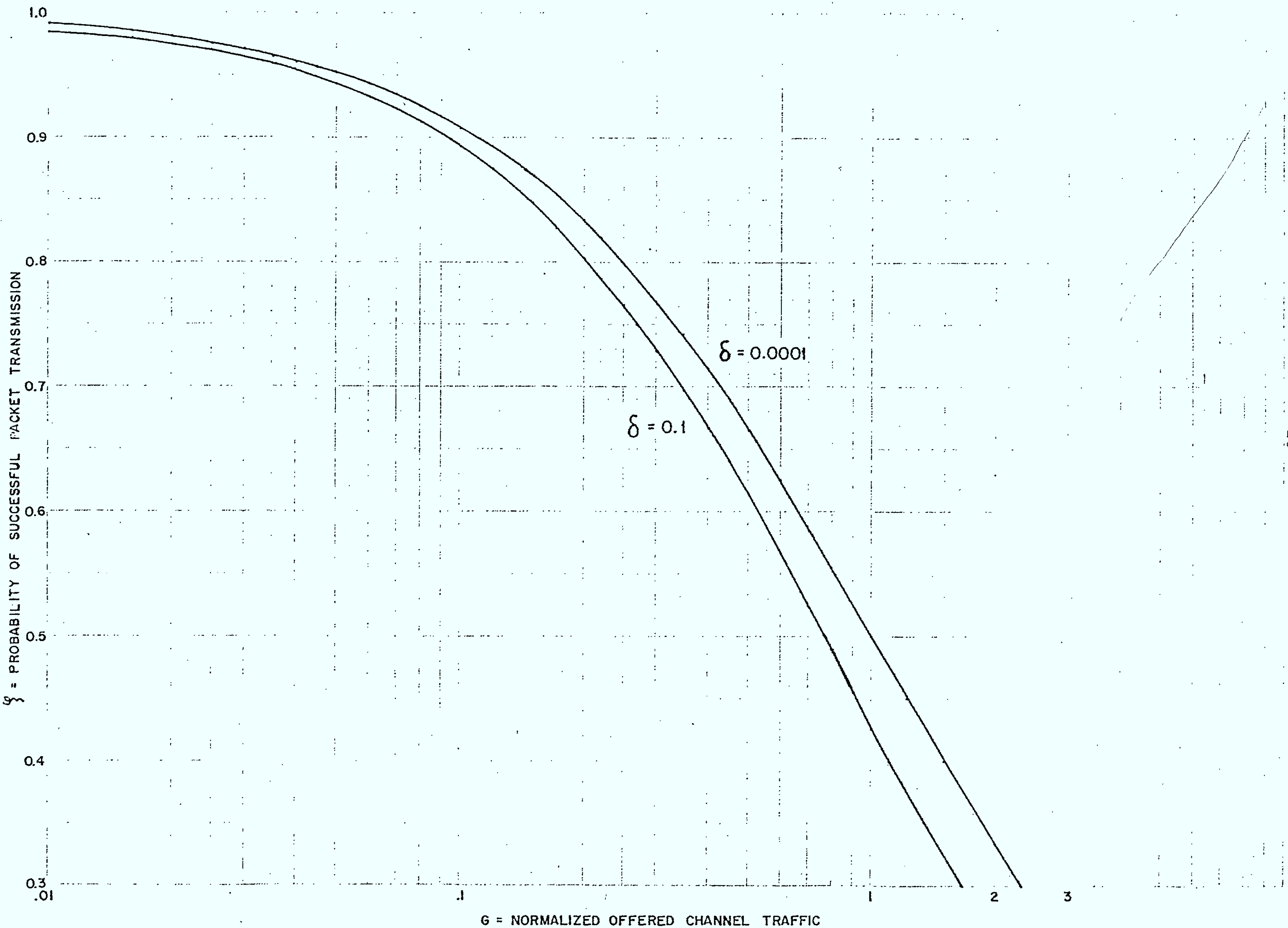


t_S = SENSING PERIOD

T_P = PACKET TRANSMISSION TIME

B/R = TEMPORAL PACKET LENGTH

FIGURE 2


1.0
0.9
0.8
0.7
0.6
0.5
0.4
0.3
ξ = PROBABILITY OF SUCCESSFUL PACKET TRANSMISSION
δ = 0.0001
δ = 0.1
.01
.1
1
2
3
G = NORMALIZED OFFERED CHANNEL TRAFFIC
FIGURE 3

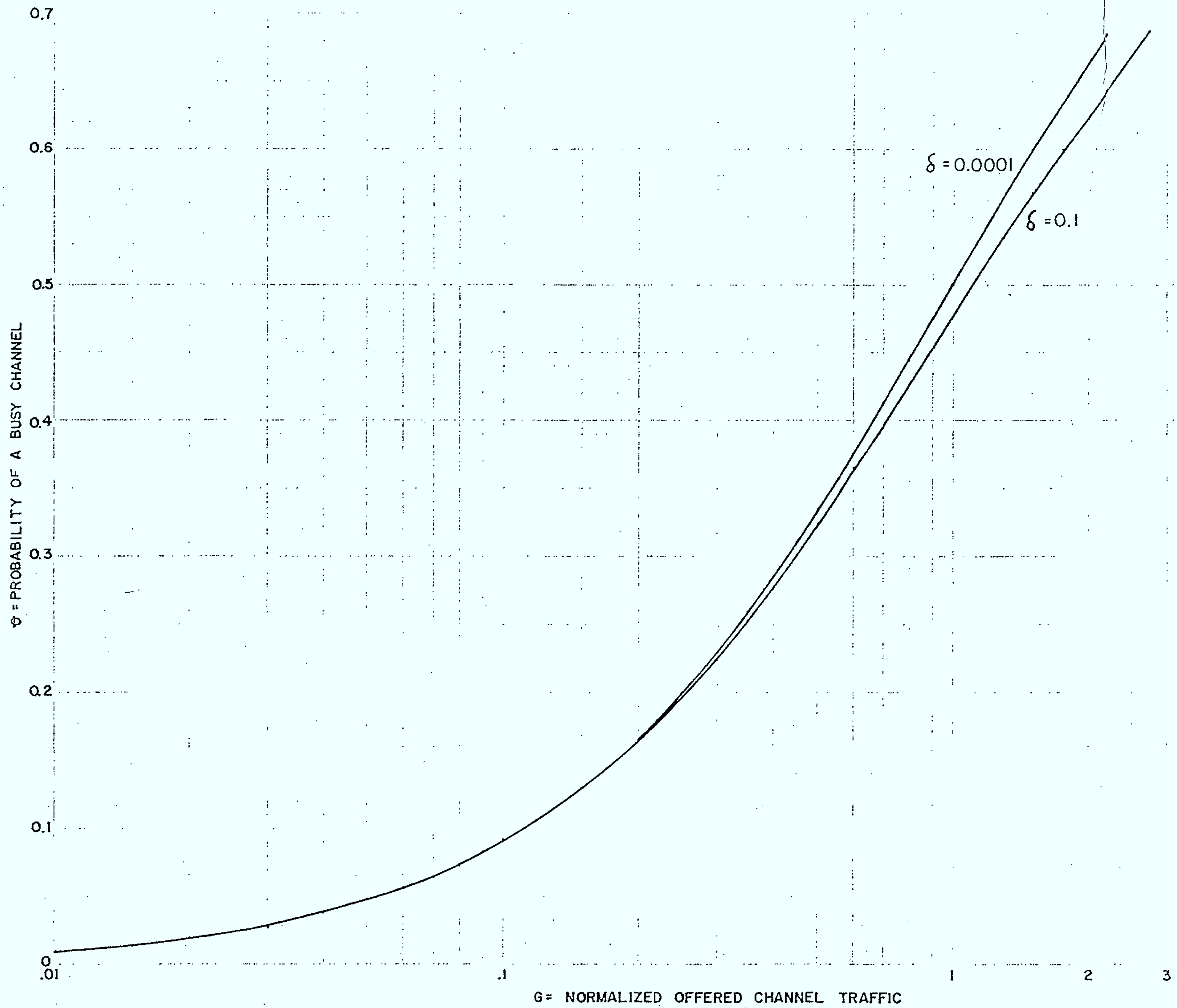

0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
Φ = PROBABILITY OF A BUSY CHANNEL
δ = 0.0001
δ = 0.1
.01
.1
1
2
3
G = NORMALIZED OFFERED CHANNEL TRAFFIC

FIGURE 4

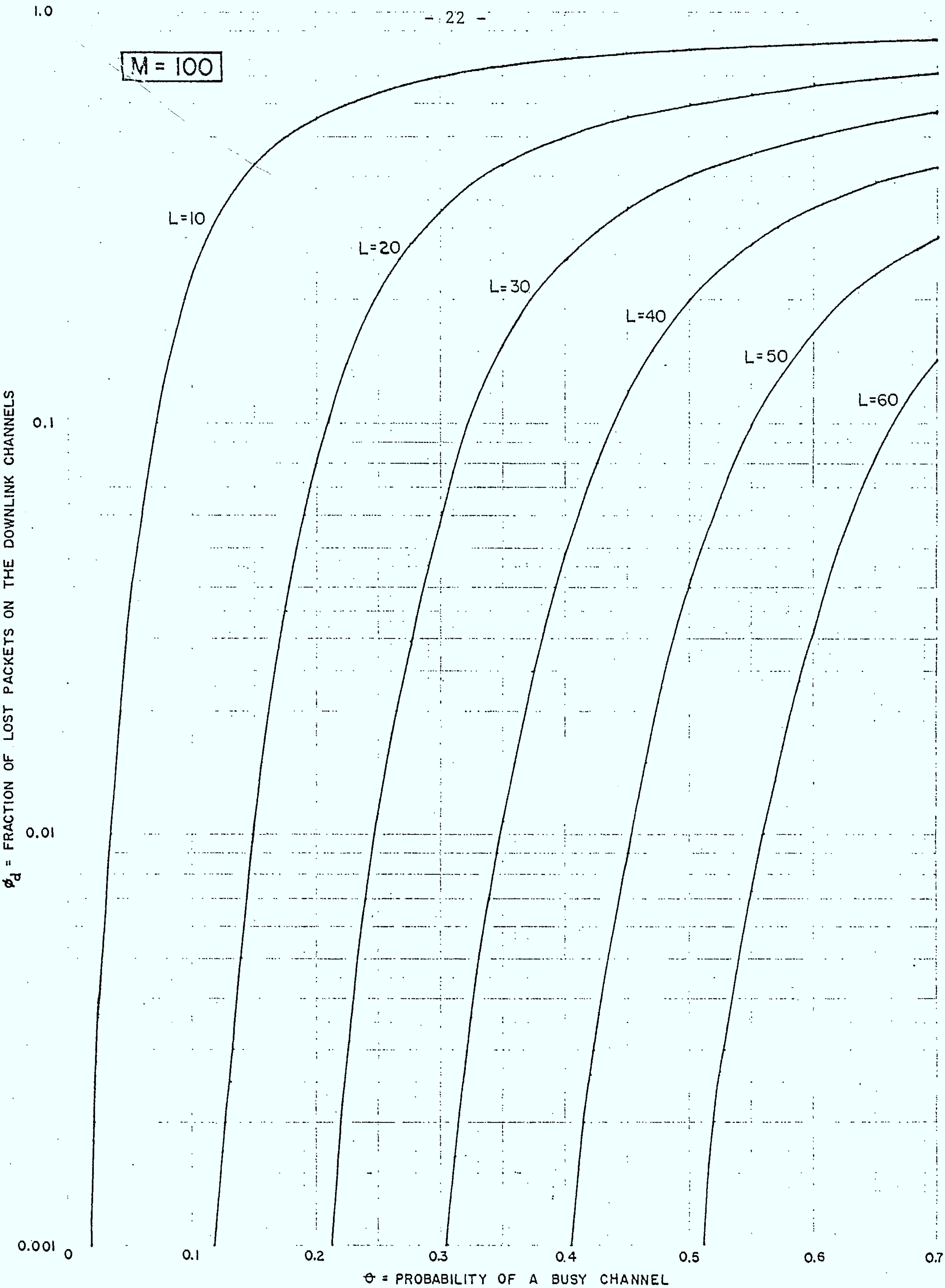

M = 100
L=10
L=20
L=30
L=40
L=50
L=60
1.0
0.1
0.01
0.001
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
ϕ_d = FRACTION OF LOST PACKETS ON THE DOWNLINK CHANNELS
θ = PROBABILITY OF A BUSY CHANNEL

FIGURE 5

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