Integration of Voice and Data in Spread Spectrum Mobile Network

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A Thesis

in

The Department

of

Electrical Engineering

Presented in Partial Fulfillment of the Requirements
for the Degree of Master of Engineering at
Concordia University
Montréal, Québec, Canada

December 1986

ISBN 0-315-37092-0

ABSTRACT

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The delay, throughput and blocking characteristics of a population of Md data users and Mv voice users operating in a mobile network envoirnment are evaluated. Frequency hopping and forward error correction are used ~~in~~ like user simultaneous transmission. QPR and MFSK signals are investigated as candidates for data modulation.

The interaction between the queueing aspects and the modulation aspects of the network are considered. The potential error correction techniques and adaption of backoff strategies in the frequency hopped network is investigated.

ACKNOWLEDGEMENTS

I wish to acknowledge my heartfelt gratitude to Professor A.K. Elhakeem for suggesting the problem, for his guidance and assistance during the entire preparation of this thesis.

I gratefully acknowledge the encouragement and full support provided by my parents, sisters and brother in law.

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LIST OF SYMBOLS

Rd	data rate.
tp	service time.
Np	bits per packet.
PG	processing gain.
G	total input traffic to the network.
A	amplitude of the transmitted signal.
a_m, b_m	transmitted symbols in QPR system.
$\hat{a}_m$,$\hat{b}_m$	estimate of the transmitted symbols.
c_n d_n	precoded symbols in the QPR system.
D	delay.
D/A	digital to analog converter.
E_b	energy per bit.
E_b/N_J	energy per bit to jammer spectral density.
E_b/N_o	energy per bit to additive noise spectral density.
FEC	forward error correction.
FH/QPR	frequency hopping quadrature partial response.
FH/MFSK	frequency hopping M ary frequency shift keying.
$H_T(\omega)$	QPR pulse shaping transmitter filter.
$H_R(\omega)$	receiver filter in frequency domain.
$h_T(t)$	transmitter filter in time domain.
$h_R(t)$	receiver filter in time domain.

I_{s_1}	intersymbol interference of the signal.
I_{J_1}	intersymbol interference of the jammer.
J	jammer power per slot.
J_T	total jammer power.
M	number of jammed channels.
Md	number of data users.
Mv	number of voice users.
N	number of frequencies available.
N_p	service time.
n(t)	additive white Gaussian noise of two sided spectral density $\eta_o/2$.
$N_c(t)$, $N_s(t)$	statistically independent low-pass white Gaussian noise processes with single-sided noise spectral density N_o W/Hz.
P(t)	rectangular pulse.
Pbe	probability of bit error.
Pdp	probability of data user in pause state.
Pds	probability of data user in successful transmitting state.
Pvb	probability of voice packet loss.
Pvs	probability of voice user generating a successful packet.
P_f	probability that a request fails to secure a message channel.
Ps	probability of successful transmission on SS channel.
P_I, P_Q	probability of error in the in-phase and quadrature channels.
Q(x)	Gaussian probability integral.

$r(t)$	is the received signal.
S	average transmitted power.
SS	spread spectrum.
$s(t)$	is the transmitted signal.
$s_F(t)$	is the faded component of the transmitted signal.
$Si(\ )$	sine integral function defined as $Si(x) = \int_0^x \frac{\sin y}{y}\, dy$.
T	bit duration.
ω_{o}	oscillator frequency.
ω_k	is the kth hopping frequency.
W_{ss}	spreading bandwidth.
$Z_Q(t),\ Z_I(t)$	quadrature and in-phase demodulated signals.
$Z_{Qm},\ Z_{Im}$	sampled values of $Z_Q(t)$, $Z_I(t)$.
γ_s	transmission coefficient of the signal channel.
γ_J	transmission coefficient of the jammer channel.
$\beta_s(\tau),\ \beta_J(\tau)$	independent zero mean complex Gaussian random processes representing the low-pass equivalent impulse response of the channel.
θ_k	is the random phase generated by the frequency synthesizer in the kth interval.
ϕ_k	is the random phase of the jammer.
$\delta(\cdot)$	Dirac delta function.
$\rho(\tau)$	multipath delay spread of the channel.
$\eta_o/2$	two-sided spectral density of additive white Gaussian noise.

σ_N^2	variance of the noise.
σ_Q^2	variance of the signal in quadrature channel.
ρ	fraction of the total channels jammed.
c	number of frequency channels.

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CHAPTER ONE

INTRODUCTION

The potential for communicating with non fixed points over the horizon without the use of wires was first recognised following the invention of the radio in the late 1800 and its development in the early 1900s. The first use of this potential was to vessels at sea as an aid to navgation and safety. Since those early days, the use of mobile communication has spread dramatically. Today it is used not only for ships at sea, but with land vehicles, aircraft and even people using portable communication equipment(mobile telephones).

The telecommunication radio service in general is being developed at a rapid rate. In the case of land mobile service, there is an increase in the number of users and also in the type of services being provided.

Over the years, the system designers have set various objectives for large scale mobile service based on the interests of the public, mobile customers, and mobile telephone operating companies. The basic objectives to be met include large subscriber capacity, efficient use of the spectrum, nationwide compatability, widespread availability, adaptability to traffic density and service to vehicles and portables.

The integration of services such as mobile telephones, dispatch, paging, information distribution, road and traffic data, monitoring and control are being developed.

The system must be capable of growing to serve many customers in a local service area, yet the provision must not be contingent on the continual enlargement of the allocated spectrum. The need to operate and grow indefinitely within an allocation of hundreds of channels has led to the cellular concept.

Instead of covering an entire local area from one land transmitter with high power at a high elevation, several transmitters of moderate power are distributed throughout the coverage area. Each site then primarily covers some nearby sub area or zone called cell. The cells cover hexagonal area.

Each cell is served by a distinct set of channel frequencies to avoid interference problem. Cells sufficiently apart may use the same channel set. Through frequency reuse, a cellular system in one coverage area can handle a number of simultaneous calls greatly exceeding the total number of allocated channel frequencies.

This thesis describes a cellular mobile network employing frequency hopped spread spectrum for integrating voice and data.

Spread spectrum communication systems have been widely studied and have been mainly used for military applications. Although they were not considered to be band width efficient for civilian applications, they provide a certain degree of immunity to intentional or nonintentional jamming. These systems have the capability of combating multipath fadding, as well as allowing simultaneous multiple user operation.

Chapter two describes the proposed frequency hopped spread spectrum mobile network. It is assumed that there are Md data users and Mv voice users in one cell of the mobile network. The voice and data packets are transmitted in packets. The data packet is backed off probabilistically if the load on the network is high. Coherent QPR signals are used for the modulation and Spread Spectrum with frequency hopping is used for the transmission of the signals. The delay, throughput and the probability of voice packet loss for Md data users and Mv voice users is evaluated taking into consideration the interaction between the queueing aspects and the modulation aspects.

In chapter three, the performance of multifrequency central transmitter, fixed channel assignment scheme and the dynamic channel assignment scheme are compared to the scheme described in chapter two.

Chapter four describes a FH spread spectrum technique using noncoherent MFSK modulation scheme. The analysis to determine the probability of bit error for the modulation scheme with multiple access is also presented. The delays, throughputs and the probability of voice packet loss is determined as described in chapter two are determined for different traffic loads.

The described network and the analysis differs from the other proposals by the inclusion of the following aspects:

Considering the interaction between the queueing aspects and the modulation aspects of the network.

Studying the exact phenomena of bit errors, traffic variations, resulting packet errors, nature of voice and data traffic, effect of bulk and interactive data considering the silence and talk periods of voice.

The potential error correction techniques and the adaption of the backoff strategies in FH networks is also investigated.

CHAPTER TWO

COMBINED ERROR/QUEUEING ANALYSIS OF A MOBILE NETWORK EMPLOYING FREQUENCY HOPPING/QPR, SIGNALS

2.1 Introduction

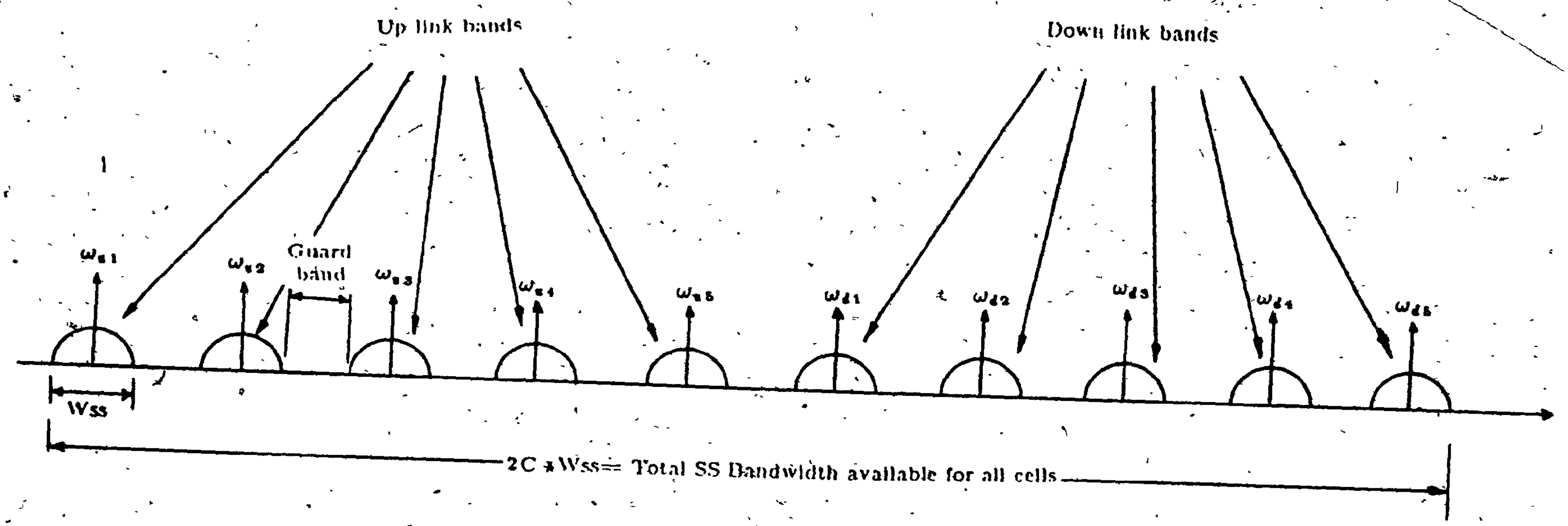
In this chapter, a cellular mobile network employing frequency hopping (FH) as a spread spectrum technique for jamming rejection and Quadrature partial response signalling for baseband modulation is proposed and described.

The transmitter buffer is modelled by a Markovian type state diagram.

The blocking, delay and the throughput for the network is analysed for combined voice and data transmission.

2.2 The SS Cellular Mobile Network Description

The network is configured as a cellular FH/QPR spread spectrum mobile network. This does not preclude however the potential use of other DS systems and other modulation techniques (MFSK, OQPSK or DPSK). The total available spread spectrum band width ($2C*W_{ss}$) is divided equally between the nonoverlapping up link (from mobile units to the base stations) and the down link (from the base station to the mobile units). The up and the down links are each divided into nonoverlapping bands each centered at the carrier frequency of the applicable cell (Figure 2.1 & 2.2). Each cell base station is assigned two center frequencies for the up and down links (ω_{ui}, ω_{di}) and these two frequencies (and the associated SS band) are reused far enough from the



Nonoverlapping SS bands of various neighbouring cells

$\omega_{u1}, \omega_{u2}, \ldots, \omega_{um}$ are uplink carrier frequencies used by mobile in cells 1......C

$\omega_{d1}, \omega_{d2}, \ldots\ldots, \omega_{dc}$ are down link carrier frequencies used by base stations in cells 1......C

Figure 2.1

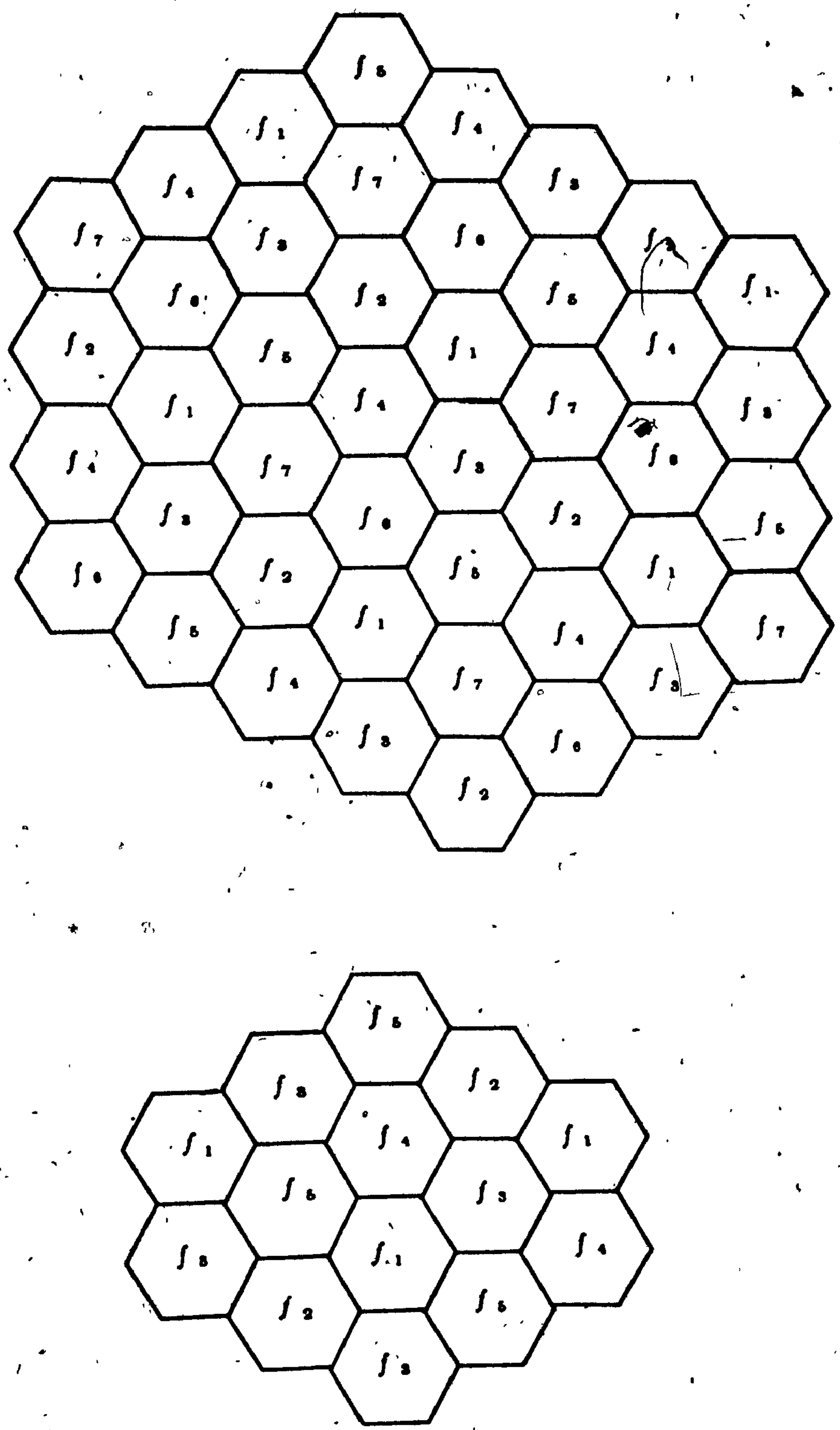


Figure 2.2

Frequency allocation to the cells.

applicable cell (Figure 2.2) depending on the reuse distance and the number of centre station frequencies considered. The number of basic SS bands selected in this network is 5, however it may be increased to 7 or 11 on the expense of reducing the SS band per cell per link (W_{ss}). On one hand it reduces the processing gain PG which is a measure of the system capability to reject simultaneous like user interference (thus degrading the achieved data rates and increasing the delay and voice blocking etc), on the other hand, the center frequency reuse distance is increased, thus limiting the existing of interference of up link or down link signals.

Each voice or data user is identified by a unique signature code (a string of pseudorandom bit sequence)

This code may be of the linear maximal type or Gold codes[1]. The later is preferred because it provides a large number of user codes with reasonable corelation properties[1].

The user code controls the frequency synthesiser of both the transmitter and the receiver of the mobile units(Figure 2.6 & 2.5) thus generating one frequency hop ω_{hi} per a certain interval T_h . If T_h is large compared to the information bit duration T_b a slow FH results. If T_h is less than T_b we obtain a fast FH system. All mobile units within a certain cell use the up link band (Figure 2.1) of this cell thus sharing the frequency hops of this band. However the probability of two or more transmitters of mobile units having the same ω_{hi} at the same time is very small by the semirandom nature of the codes controlling the FH of each transmitter.

At the mobile transmitter (Figure 2.6) the FH signal is mixed with the appropriate base station crystal output (as will follow later) and the information carrying base band modulated signal (FSK or QPR etc) after which it is

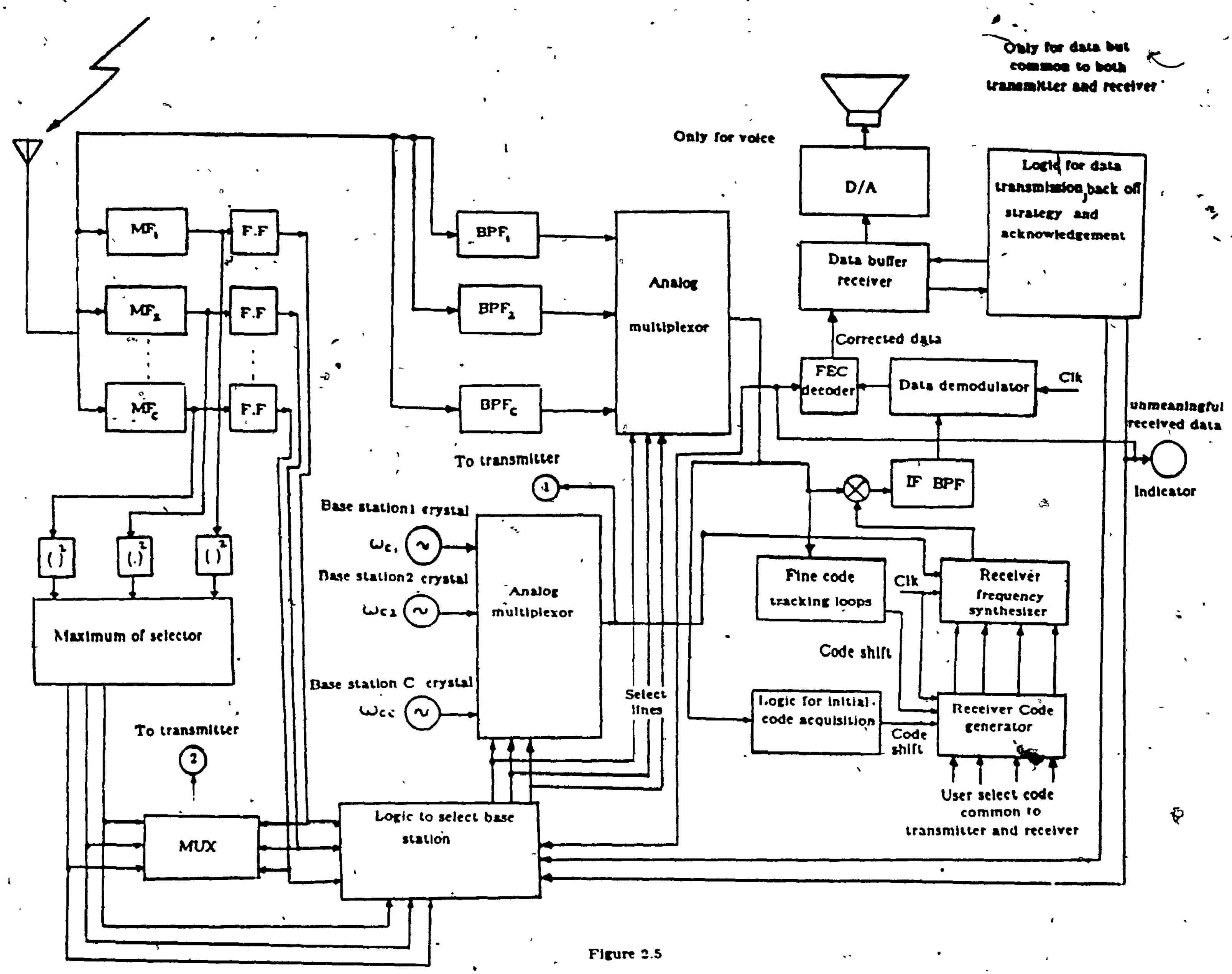


Figure 2.5

Block diagram of mobile receiver.

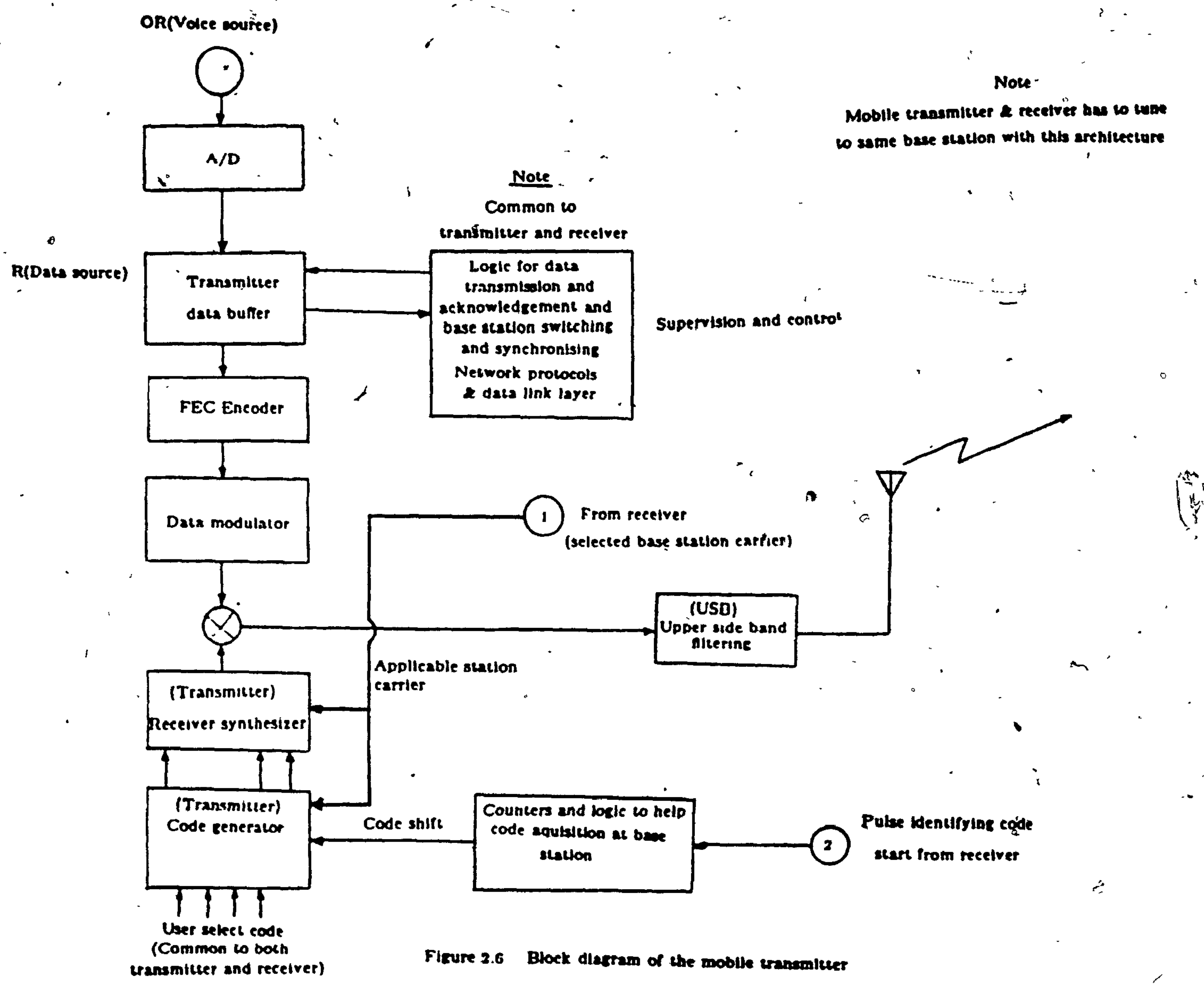


Figure 2.6 Block diagram of the mobile transmitter

upconverted and transmitted.

2.2.1 Interference Control Mechanism

Data mobile units are free to transmit their FH packets to the base station they select and they continue to do so as long as there is no 'NO STATION AVAILABLE' message received from that base station. The base station generates this signal whenever the number of on going voice calls and average level of data users traffic exceeds a certain limit. However voice calls being typically composed of thousands of packets will need a permission to start.

The voice mobile user will act first as a data user and so we call it (data like voice user). He will ask the base station he selected using FH packets for permission to transmit and repeat the request until acknowledgement reception from the base station. The base station looks at its buffers and accumulators and if the number of on going voice calls and data traffic permits, the voice request will be granted (of course given that the destination voice user and the destination base station are both free as will follow, otherwise a 'TRUNK BUSY',signal will be sent in the down link by the source base station to the source mobile). If the buffer status of the base station does not permit new voice calls, a request denial (blocking) will be sent by the base station to the mobile user (i.e 'STATION OVER CROWDED'). The voice user will then try other base stations (done automatically during dial up period). Also to further limit SS interference, both data and data like users will back off probabilistically whenever they encounter a collision (as indicated by absence of acknowledgement from base station after a time out period). This means that each data or data like voice user will transmit with a probability 1/KD in each one of the KD time slots (packet times) following a collision (packet failure).

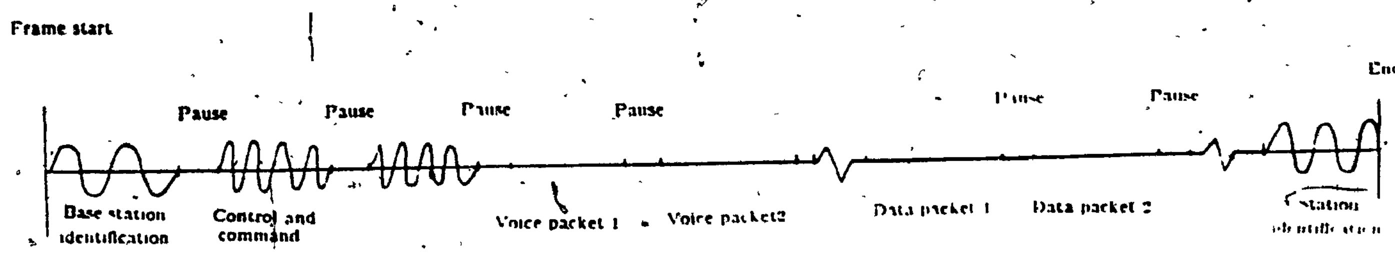


Figure 2.3

Sequential transmission of spread spectrum packets in a typical frame in the down link (from base station to all mobiles) All data and voice packets are of equal length The length of base station identification burst or control and command are fixed a priori and are multiples or equal to information carrying packets

This limits the traffic but does not improve the voice quality once the voice call is permitted since voice transmission is based on loss. (Backoff in an ongoing voice call may result in prohibitive delays).

The above strategy controls only the uplink traffic volume received from the mobiles of a certain cell. It is possible for the base station to also handle traffic coming from other base stations (long distance traffic). To allow for this traffic to be accomodated a percentage of the down link capacity (time slots) of any base station is usually reserved thus affecting the number of local uplink voice calls and local data traffic that can be permitted from mobile users in the vicinity of the cell.

2.2.2 Royality And Location Problems

Rather than asking the mobile unit to transmit powerful tones to the neighbouring base stations for the purpose of location and identification and power control, the more powerful base station transmits tones identifying themselves in the begining of each down link transmission frame (Figure 2.3). The mobile users will monitor the identification bursts continously (through matched filters or phase locked loops MF1,MF2,....MFC. of Figure 2.5) and select the station giving the maximum identifying tone at the mobile receiver. This means that the presence of a mobile in a certain cell does not mean that it will forward its transmission to this base station. Rather good reception in channel fading will be the factor.

This means that the mobile based on the terrian and fading conditions etc, selects the appropriate base station to tune to. As the mobile voice user moves towards a better cell, its loyality block (Figure 2.6) will dial the better station to get new reservation, inform the on going base station, then switch very smoothly to the new base station thus reducing the familiar zone crossing problems in a classic cellular system.

The above scheme reduces the multiplicative effect of fading on power differential problem (near far problem)[2] since each mobile unit selects the best station apart from its present location. However finding a destination mobile will be harder. Also extra power control can be accomodated by designing the mobile transmitters to adjust their power depending on the power of the best received base station tone (as above) such that all mobile signals have the same received powers at a certain base station. Communications between base stations have to exist and statistics of mobile movements have to be reviewed especially if a mobile is hidden intentionally or unintentionally. Needless to say that all these problems are common to all cellular mobile systems.

2.2.3 The Back Bone Support Network

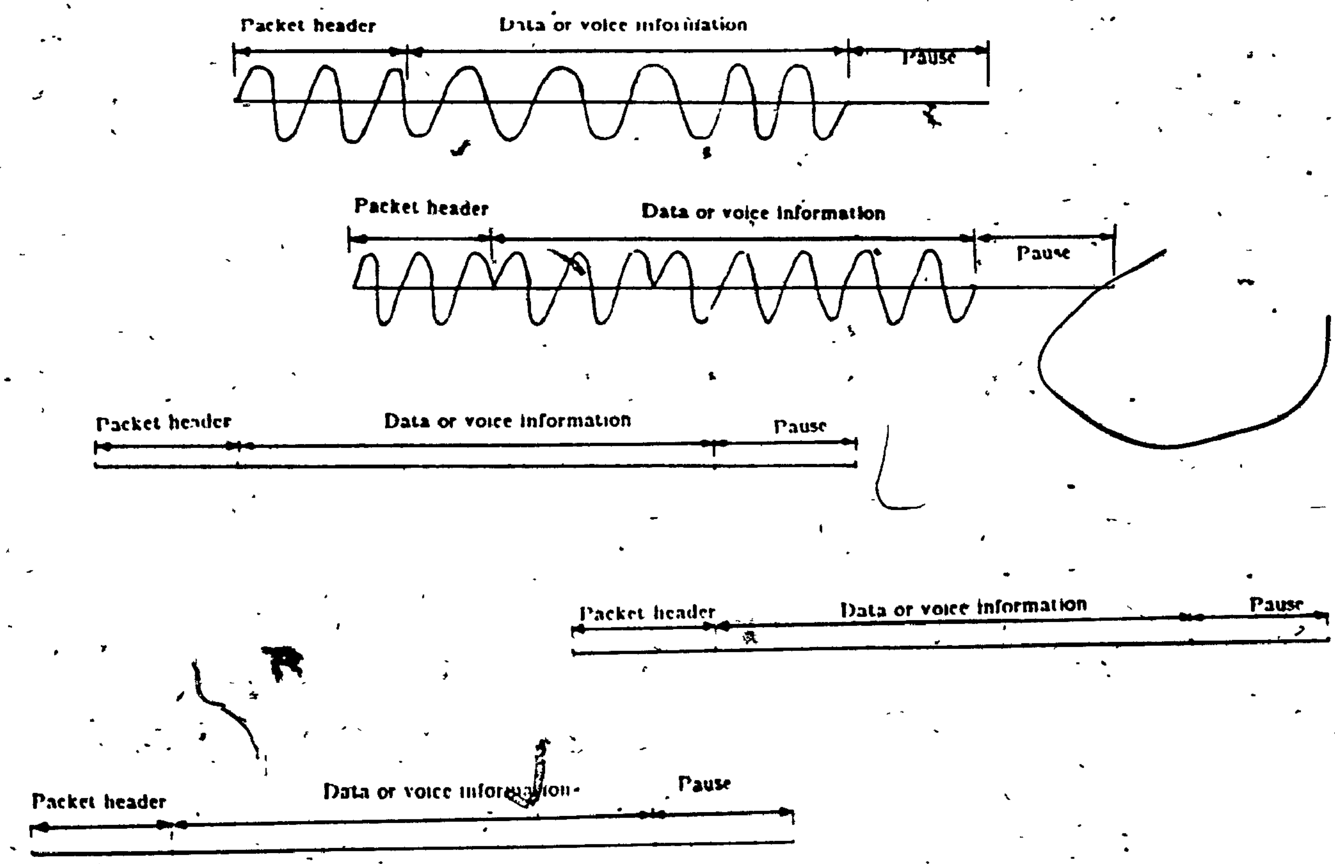
A very efficient backbone network handling the interbase stations communications has to exist. This is common to any cellular system handling wide areas and consisting of many cells. Traditionally this has been the public network or a satellite based multiple access. It is also possible to use high bandwidth backbone optical networks [3] for such purposes. This inter station communication should alleviate the contention for the right to handle the mobile units specially around cell borders. The search for a destination mobile is handled by the base station transmitting 'YOU ARE CALLED' messages in its cell several times. If there is no reply, this base station interrogates other base stations who issue a search for the destination mobile in their cells. While the mobile unit selects a certain base station to transmit to, the down link receiver signal to this unit may come from other base station. This possible independence of the transmitter and the receiver of the unit is justified for bursty traffic. In this case interstation communications will be really needed whenever he mobile crosses the cell boundary while receiving from a different source mobile.

The whole link from the source station to the destination station to the destination mobile has to change and the fact that the mobile has the same control over the selection of the station adds to the problem. The mobile receiver detects the best new station, informs the transmitter who issues a 'CONNECT ME' request to the new station then tries the next best etc. A smooth transition will then take place. Another need for inter station communications arises in the case where SS or more mobile units in the neighbouring cells choose accidentally the same SS code. In this case base stations will help the mobile users by issuing the appropriate 'CHANGE CODE' message to the appropriate user.

2.3 The Mobile Unit Transmitter And Receiver Structure And Relations To The Network

The mobile transmitter emits a FH/QPR (or FH/MFSK) packets, but before this is done, information bits have to be collected, grouped with the packet overhead bits and then encoded for the possible error correction at destination (FEC encoder of Figure 2.6). The control overhead bits are provided by the box 'Supervision Protocols'. This box is common to both the transmitter and the receiver and provides overhead containing messages such as: 'VOICE CALL REQUEST', BASE #J', 'ORDINARY DATA PACKET', 'ACK SWITCH', (i.e. station switching confirmed) etc.

Also as the block extracts the overheads of the arriving packets (from



Note
Pause intervals facilitate processing at the receiver.

Figure 2.4

Typical voice and data packets arriving from mobiles at the base station. All packets carry the same modulation selected (FH/FSK) or (FH/QPR)....etc.

Note that some of the arriving SS packets are synchronised with base station frame start base station) and issues the appropriate commands to both the transmitter and the receiver to adjust to their carriers, codes, transmission times,....etc. Such arriving overheads contain messages such as: 'STATION OVER CROWDED', 'VOICE CALL GO AHEAD TO TRANSMITTER #J', 'DATA CALLS WAIT', etc.

Both the physical layer protocols (like the data back off and retransmission strategy) and the data link protocols (for handling the acknowledgements and hand shaking between users) are handled at the mobile user level and not at the base stations. Also both are incorporated in the mobile transmitter packet overheads and implemented by the supervisor in the control box (Figure 2.6).

The 'Pause' intervals in the transmitted packets (Figure 2.4) facilitates the processing at the mobile units and base stations. The output of the FSK or QPR or DPSK modulator is mixed with the FH waveform coming of the synthesiser (Figure 2.6) and another mixer handles the selected carrier coming from the receiver. Finally the (upper side band) USB is filtered for transmission.

As mentioned previously, the user may (on less frequent basis) change his code to improve his performance through the manual code select. However, extra user identification bits for the multiuser per code should be incorporated within the packet overhead. At a certain base station, all SS packets arriving from all mobile (preferred) be synchronized i.e. their code starts should be the same. This eases the initial code acquisition problem at the base station receivers and make the packet detectable at first arrivals, so increasing the capacity of the network. The mechanism for achieving this is to make the mobile receiver pick the periodically transmitted carrier of their selected base station (Figure 2.3) and tune their transmitter code to start immediately after the 'Pause' of the down link station signal. The counters and the logic box of figure 6 is intended for such purposes. However, it is possible of course due to the differing locations of the mobiles that the mobile users codes starts at the base station are not the same. This implies the necessity of the base station having an ordinary search code scheme [4] for all users operating in its cell.

The receiver of Figure 2.5, consists of a few matched filter banks MF1 through MFC (or alternately phase locked loops PLLs). Each one of this recognizes the tone and pause of one of the base stations in the network (Figure 2.3). Recall the mobile receiver picks the base station giving the maximum tone power at the mobile receiver and adjusts his transmitter power accordingly. The squaring devices $(.)^2$, and the base station logic achieve such a purpose. While the latches and the multiplexer (MUX) sample only that (MF) pulse of the best station and feed it to the transmitter (Point(2)) for synchronization purposes. Actually two analog multiplexors are used at the receiver, the first selects best of C down links SS bands of the C potential base stations while the second multiplexer selects the corresponding station carrier crystal.

The selected received band will be mixed with the receiver dehopping signal to yield the base band information carrying signal which is then demodulated and FEC decoded and D/A converted (for voice or analog data) (figure 2.5).

The logic and fine tracking boxes and also the down link pulse coming from the station (Point(2)) should help to bring the local code in synchronism with the received signal code. Some FEC decoders come with failure to correct indicator and this signal may be fed to the base station selector (Figure 2.5) to switch to a new station. Also this selection may be affected by unmeaningful data as detected by the supervision and control box.

This box can be implemented via software. However for high speed applications or large SS bandwidths hardware implementation may be necessary.

A final note regarding the down link signal (Figure 2.3) is in order, it is seen there the sequential transmission of SS voice and data packets from the base station to all destinations plus the pause periods (aiding SS codes acquisition at mobile receivers) and the base station identifying tone at the begining of each frame.

Any network control commands issued from the base stations to the various users (such as the ones already mentioned in this section) are included in the voice and data SS packets sequentially transmitted.

However, other network supervisory signals and frame housekeeping such as number of voice and data packets in the current frame (Figure 2.3), station status,parity check bits,....etc. are all included in each frame control overhead (Figure 2.3).

2.4 Blocking - Delay - Throughput Analysis of the Proposed Protocol

Because all up link and down link SS bands are non overlapping (figure 2.1), the up link signals from a set of mobile users in one cell will not interfere with the uplink SS signals of the mobile users in the neighbouring cell. Similarly, the down link SS signals transmitted from one base station will not interfere with the that of the neighbouring cell. Also from figure 2.3 it is evident that the SS packets are sent sequentially from each base station to its mobile users (no contention or mutual interference).

The above implies that the up link from mobile users to base station is the only place where we should expect SS interferences (overlap of mobile users signals both in time and possibly frequency).

However, different from ALOHA or carrier sense techniques[5], [7] where a partial overlap of two signals means complete destruction the very nature of SS signals allow more than one signal to coexist at the same time. The interference rejection capability of SS signals [6] amounts finally to having a few random bit errors per packet which can be corrected employing forward error correction techniques. However SS means sacrificing a bandwidth much larger than that of the basic information signal and also FEC sacrifices a little bit of the capacity to correct errors. Apart from some recent proposals ignoring some of these effects, all those considerations should and will be considered in the process of calculating the useful throughput and other performance measures of the proposed network.

It is assumed that a total SS bandwidth per cell up link is equal to Wss, a number of data and voice users equal to Md, Mv per cell respectively. All results are normalized with the length between two embedded points in the Markov chains involved which is the same (in our case) as the service time of one data packet at the basic information rate. Assuming a data rate Rd and number of bits per packet equal to Np the service time equals,

$$t_p = \frac{1}{R_d} . N_p \tag{2.1}$$

If the mobile transmits all the time at this full rate Rd, the probability of generating a packet per packet time is 1. (He cannot generate two packets or more per packet time). We divide the packets into small divisions and assume Poisson traffic in each of the small divisions for the data users reflecting itself finally in a probability of generating a packet per packet time equal to λ_d.

Assuming in general QPR data modulation and if NF is the number of frequencies, then

$$NF = W_{ss} . 2T_b \tag{2.2}$$

The SS interference rejection capability is usually expressed as the processing gain (PG) which equals in this case

$$PG = \frac{W_{ss}}{W_d} \tag{2.3}$$

We assume (for analysis convenience) that voice users have the same Np, td, Rd and finally the same PG as that of data users.

For data users, we also adopt the following random back off strategy at the transmitter buffer

1. The (head of the line) packet in the buffer is transmitted.

2. If an acknowledgement is received for this packet, it is dropped out of the buffer, if after a time out period the acknowledgement is not received (means unsuccessful transmission) then retransmission is tried in only one slot out of the Kd packet times (slots).

The above is equivalent (on the average) to saying that retransmission is tried in each of the succeeding slots, with the probability equal to,

$$q = \frac{1}{Kd} \tag{2.4}$$

and remain delayed (refrain from transmission) with the probability (1-q).

The behavior of the transmitter buffer can be analyzed through studying the Markovian state diagram of figure 2.7.

Excluding the idle state, the states of each transmitter buffer are described by the integers i, j (at the time frame (slot) number 1).

$$S_l = S_l(i, j) \tag{2.5}$$

where i=1,2,3,....L represents the buffer contents in packets

j=0,1,2 indicating a transmitting or collided or delayed transmitter respectively.

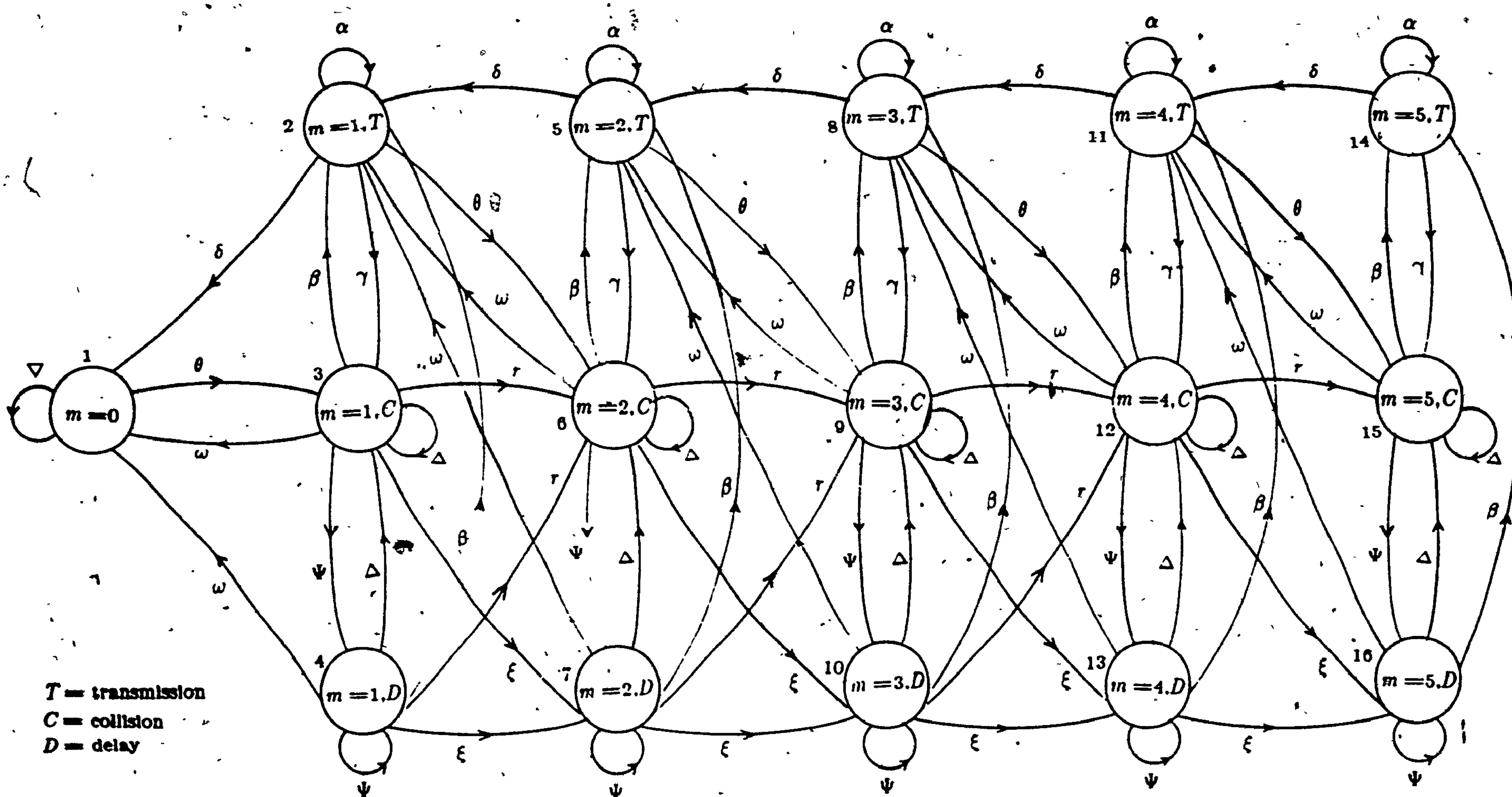


Figure 2.7 State diagram of one of the data users in a FH mobile network with backoff

In figure 2.7, the assumption is made that sensing collisions and acknowledgement is instantaneous thus permitting the existing of links such as the one from state (1,C) to (2,C) and from (2,D) to (2,C)....etc. This assumption is almost typical in the recent literature.

Based on this we have the following definitions:

$$\Psi = (1 - \lambda_d)(1 - \frac{1}{K_d}) \tag{2.6}$$

Which is the probability that the user will not generate a new packet and will back off from transmitting old packets

$$\omega = \frac{1}{K_d}(1 - \lambda_d)P_s \tag{2.7}$$

Which is the probability that the user will try transmitting in this period, will not generate new packets and will succeed in transmitting on the SS channel with probability (Ps) the head of line packet (HOL)

$$\theta = \lambda_d(1 - P_s) \tag{2.8}$$

Which is the probability that the user will generate a new packet and will not succeed in transmitting the (HOL) packet.

$$\delta = P_s(1 - \lambda_d) \tag{2.9}$$

Which is the probability of success on the channel and generating no new packets

$$\alpha = \lambda_d P_s \tag{2.10}$$

Which is the probability of success and generating a new packet in the mean time.

Similarly the remaining transition probabilities, i.e.

$$\beta = \lambda_d \frac{P_s}{K_d} \tag{2.11}$$

$$\gamma = (1 - P_s)(1 - \lambda_d) \tag{2.12}$$

$$\Delta = (1 - P_s)(1 - \lambda_d)/K_d \tag{2.13}$$

$$r = \lambda_d (1 - P_e)/K_d \tag{2.14}$$

$$\xi = \lambda_d (1 - \frac{1}{K_d}) \tag{2.15}$$

$$\nabla = \lambda_d + P_e (1 - \lambda_d) \tag{2.16}$$

The two dimensional state S (.,.) is now mapped to the single dimension state Y (.) using

$$Y_l\,(3i + j - 3) = S_l(i, j) \tag{2.17}$$

Where l=0,1,2,...,L.

j=0,1,2

L is the number of the applicable time slot and the state Y(1) is reserved for the idle state.

The mapping of (2.17) is well described in figure 2.7 where we show both the two dimensional state S (.,.) and its corresponding Y(.).

For illustration purposes we write the equilibrium equation for the idle state Y(1).

$$Y_{l+1}(1) = \nabla Y_l(1) + \delta Y_l(2) + \omega Y_l(3) + \omega Y_l(4) \tag{2.18}$$

Similarly for the other states

$$Y_{(l+1)}(2) = \alpha Y_l(2) + \beta Y_l(3) + \beta Y_l(4) + \delta Y_l(5) + \omega Y_l(6) + \omega Y_l(7) \tag{2.19}$$

$$Y_{l+1}(3) = \theta Y_l(1) + \gamma Y_l(2) + \Delta Y_l(3) + \Delta Y_l(4) \tag{2.20}$$

$$Y_{l+1}(4) = \Psi Y_l(3) + \Psi Y_l(4) \tag{2.21}$$

$$Y_{l+1}(5) = \alpha Y_l(5) + \beta Y_l(6) + \beta Y_l(7) + \delta Y_l(8) + \omega Y_l(9) + \omega Y_l(10) \tag{2.22}$$

$$Y_{l+1}(6) = \theta Y_l(2) + r Y_l(3) + r Y_l(4) + \gamma Y_l(5) + \Delta Y_l(6) + \Delta Y_l(6) \tag{2.23}$$

$$Y_{l+1}(7) = \xi Y_l(3) + \xi Y_l(4) + \Psi Y_l(6) + \Psi Y_l(7) \tag{2.24}$$

$$Y_{l+1}(14) = \alpha Y_l(14) + \beta Y_l(15) + \beta Y_l(16) \tag{2.25}$$

$$Y_{l+1}(15) = \theta Y_l(11) + r Y_l(12) + \Psi Y_l(14) + \Delta Y_l(15) + \Delta Y_l(16) \tag{2.26}$$

This set of equations will be complimented by the condition.

$$Y_l(1) + Y_l(2) + Y_l(3) + \_\_\_\_\_\_ = 1 \tag{2.27}$$

At equilibrium we can write

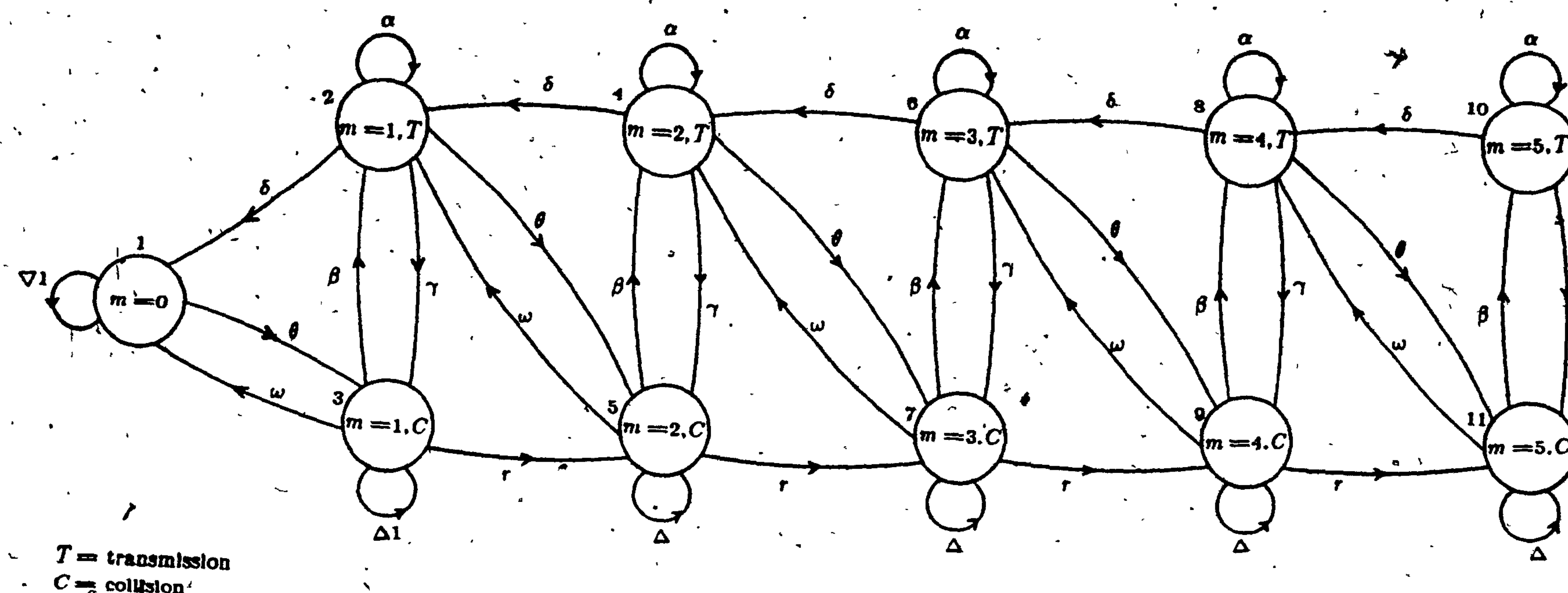


Figure 2.8

State diagram of one of the data users in a FH mobile network with backoff with Kd=1.

$$Y_{l+1}(j) = Y_l(j) \tag{2.28}$$

for all j=1,2,....

Once the transition probabilities ω, θ,.... etc are known the system equations can be solved to yeild the equilibirium probabilities Y(j).

Various performance measures can then be evaluated, such as the delay D, so

$$D = \sum_{K=1}^{\infty} K(Y(3K-1) + Y(3K) + Y(3K+1)) \tag{2.29}$$

The probability of data user transmitting a packet or colliding on the SS channel at equilibirium (i.e. not idle or delayed) is given by (at equilibirium).

$$\overline{Pdp} = 1 - Y(1) - \sum_{K=1}^{\infty} Y(3K+1) \tag{2.30}$$

and the probability of a data user be successfully transmitting state is

$$Pds = \sum_{K=1}^{\infty} Y(3K-1) \tag{2.31}$$

Similarly for KD=1, the transmitter buffer can be represented as the Markovian state diagram as shown in Figure 2.8.

ω, θ, δ, α, γ, β are the same as defined before.

$$\nabla 1 = (1 - \lambda_d) + \lambda_d Ps \tag{2.32}$$

$$\Delta 1 = (1 - \lambda_d)(1 - Ps)/Kd + (1 - \frac{1}{Kd}) \tag{2.33}$$

The equilibrium equation for the idle state can be written as

$$Y_{(l+1)}(1) = \nabla 1 Y_l(1) + \delta Y_l(2) + \omega Y_l(3) \tag{2.34}$$

Similarly for other states:

$$Y_{(l+1)}(2) = \alpha Y_l(2) + \beta Y_l(3) + \delta Y_l(4) + \omega Y_l(5) \tag{2.35}$$

$$Y_{(l+1)}(3) = \Delta 1 Y_l(3) + \theta Y_l(1) + \gamma Y_l(2) \tag{2.36}$$

$$Y_{(l+1)}(4) = \alpha Y_l(4) + \beta Y_l(5) + \delta Y_l(6) + \omega Y_l(7) \tag{2.37}$$

$$Y_{(l+1)}(5) = \Delta 1 Y_l(5) + \theta Y_l(2) + r Y_l(3) + \gamma Y_l(4) \tag{2.38}$$

$$Y_{(l+1)}(10) = \alpha Y_l(10) + \beta Y_{(l)}(11) \tag{2.39}$$

The set of equations are complimented by the condition:

$$Y_l(1) + Y_l(2) + Y_l(3) + \ldots = 1 \tag{2.40}$$

and at equilibrium

$$Y_{(l+1)}(j) = Y_l(j) \tag{2.41}$$

for all j=1,2

The delay D, Pdp and Pdf can be obtained in similar way as in (2.29), (2.30) and (2.31).

The value of Ps is not a given constant. This probability of a packet success on the SS channel is a complicated function of the current traffic on the network, the modulation technique (and hence the probability of error), etc. and it will be computed shortly.

The voice traffic of each user is assumed to consist of a talk burst followed by a silence burst.

The performance will be evaluated for a current number of voice users Mv each with a probability of generating a new voice burst (talk plus silence) during the basic packet time tp equal to λ_v .

If this is added to the data traffic, the total input traffic to the network will be given by

$$G = M_d \lambda_d + M_v \lambda_v . \tag{2.42}$$

Moreover it is assumed that voice users are operating on loss basis, i.e. an unsuccessful voice packet (blocked) are lost and will not be buffered.

During any packet time the talking voice user can be in three states, having a talk burst and successfully transmitting the corresponding packet (Pvs), or having a talk period and unsuccessful transmission (Pvf), or having a silence period(Pvd) (A completely idle voice user is not considered here). An analyst wishing to include the effect of completely idle state will only have to change Mv and Md, this is effectively done by assigning different values for Md, Mv. Each talk period is followed by a silence period. However for the combined effect on the channel of a number of such users, their reflections can be modeled as having random voice and talk bursts (not necessarily related).

The probability density function of the silent period is [8]

$$P_s(t) = ae^{-at} \tag{2.43}$$

Where the mean silent period is 1/a .

The probability density function (Pdf) of the talk burst is given by

$$P_t(t) = be^{-bt} \tag{2.44}$$

Where b is the mean talk burst length

The Pdf of the voice burst (talk + silence) is given by

$$P_t = P(t)*Q(t) = ab/(b-a)(e^{-at} - e^{-bt}) = ce^{-ct} \tag{2.45}$$

or approximately (implying a worst case Poisson distribution [8])

$$1/c = (1/a + 1/b) \tag{2.46}$$

will be the length of the burst.

Normalizing with respect to the total packet time and noting that the case of continuous presence on the channel, voice user is taken (implying λ_v = 1 in (2.42)) we obtain

$$1/c = 1/a + 1/b = 1 \tag{2.47}$$

Since the total time is normalized to 1 and the probability of finding a talking packet is approximately equal to the duration of the talk period relative to the whole period (now 1) we obtain:

$$\mu = (1/b)/1 \tag{2.48}$$

μ is the probability of having a talking packet ($\lambda_v = \mu$)

$$z = (1/a)/1 \tag{2.49}$$

z is the probability of having a silent period

Given the average talk and silence periods (1/b),(1/a) we first normalize and then compute μ, z from (2.48), (2.49).

At a certain level of traffic conditions (data and voice) Ps will be evaluated. It follows that the probability of the voice user generating a successful packet will be given by

$$Pvs = \mu P_s \tag{2.50}$$

and the probability of packet blocking is

$$Pvb = \mu(1 - P_s) \tag{2.51}$$

and the probability of the user generating a packet at all on the SS channel was given in (2.48).

Because of the complete independence of the transmission trials of the Md and Mv users to have a SS packet on the channel the probability of having n packets from Mv, Md users will be given by

$$P(n) = \sum_{i=1}^{Md} \sum_{j=1}^{Mv} \binom{Md}{i} (P_{dp})^i . (1 - P_{dp})^{Md-i} \binom{Mv}{j} (\lambda_v)^j . (1 - \lambda_v)^{Mv-j} \tag{2.52}$$

n = (i+j), n= 1,2,.....,(Md+Mv)

Where Pdp is given by equation (2.30). However having a number of simultaneous " n " packets on the channel does not mean having "n" interferences occupying the same frequency hop of the applicable base station receiver. Assuming random control of the FH synthesizers, the probability of any hop overlapping with another interfering user hop is (1/PG). Given "n" packet users, the conditional probability of having Ks interference tones out of all n users hitting in the base band modulation (MFSK or QPR or DPSK,........,etc.) of the intended receiver is given by,

$$P(Ks/n) = \binom{n}{Ks}(1/PG)^{Ks} \quad (1 - 1/PG)^{n-Ks} \qquad (2.53)$$

Where Ks = 0,1,2,........,n.

As a worst case it assumed that there is complete overlap of the interfering users packets in the time domain thus implying interference in each bit of the packet considered.

The conditional probability of bit demodulation error at the intended receiver is defined as P(e/Ks,n) and is

$$P(e/Ks,n) = \sum_{Ks=1}^{n} Pbe.P(Ks/n) \qquad (2.54)$$

Pbe is the bit error probability and is explained later.

At the intended receiver, Forward error correction (FEC) will correct few random errors of each packet. If (12, 23) code[10] is selected, the minimum distance will be d=4 and the probability of making a decoding error per packet of 23 bits is given by

$$P(Block/Ks,n) = \sum_{j=4}^{23}\binom{23}{j}(P(e/Ks,n))^{j}(1 - P(e/Ks,n))^{23-j} \qquad (2.55)$$

where error randomness is assumed.

This implies that the FEC will divide the packet into nb blocks and try to correct the random errors in each block. Now nb equals

$$nb = (Np/23) \qquad (2.56)$$

Where (.) stands for the integer part. Np is 512.

A packet consisting of nb blocks and assuming randomness of occurance of blocks constituting the packet, the following packet detection (given n,Ks) is obtained

$$P(Correct\ packet/Ks,n) = P(Block/n,Ks)^{nb} \qquad (2.57)$$

Averaging over (n,Ks) from (2.53), (2.52) into (2.57), the final average probability of packet success (taken every thing into consideration) is given as

$$P_s = \sum_{n=0}^{Md+Mv} \sum_{Ks=0}^{n} P(correctPacket/Ks,n)P(Ks/n).P(n) \qquad (2.58)$$

It is easily seen that this should be nothing but the same Ps that was assumed. Equations (2.7) - (2.16) and equations (2.50)- (2.51) were used in the computations leading to the right hand side of (2.58).

This means that there is an iterative procedure. Starting with an initial Ps and going through all the equations and finally ending with a better estimate from (2.58).

In spread spectrum, the bandwidth is much larger than that of the information signal. This implies that any throughput obtained using spread spectrum has be refered to the bandwidth used (i.e. Wss). During all calculations Wd has been used as a reference and the resultant is that the throughput obtained should be divided by (Wss/Wd) i.e. multiplying throughput by the coding rate ((12/23) in this case). Finally the SS network uplink effective useful throughput becomes

$$SSS = (12/23).(1/PG).(Ps.G) \qquad (2.59)$$

where PG and Ps and G are given by equations (2.3), (2.58), (2.42) respectively.

A voice packet blocking stems from an error in any of the np blocks constituting this packet.

If the probability of voice blocking (per block) is Pvb, then the relation to the voice blocking per packet is

$$P'vb = \sum_{K=0}^{nb} \binom{nb}{K} (Pvb)^K \quad (1 - Pvb)^{nb-K} \qquad (2.60)$$

In the course of computation, Pvb will result first (equation (2.51)) in which case (2.60) will become a transcidental equation in Pvb and undoubtly (Pvb<P'vb).

2.5 Analysis to calculate the bit error probability

The bit error probability of a communication system employing frequency hopping (FH) as a spread spectrum technique, for jamming rejection and Quadrature partial response signalling for base band modulation are evaluated as in [12].

The modulator and demodulator shown in Figure 2.9 are the same as those presented in [13]. However, with a frequency selective fading and the narrowband tone jamming channel far from being an ideal channel we choose for analysis's convenience to concentrate the QPR pulse shaping filter at the transmitter. The transmitter and receiver filters are defined in the frequency domain as .

$$H_T(\omega) = \begin{cases} 4T\cos\omega T & |\omega| < \pi/2T \\ 0 & \text{otherwise} \end{cases} \tag{2.61}$$

$$H_R(\omega) = \begin{cases} 1 & |\omega| < \pi/2T \\ 0 & \text{otherwise} \end{cases} \tag{2.62}$$

and in time domain as (T is the bit duration)

$$h_T(t) = \frac{4}{\pi}\left[\frac{\cos(\pi t/2T)}{1 - t^2/T^2}\right] \tag{2.63}$$

$$h_R(t) = \frac{\sin(\pi t/2T)}{\pi t} \tag{2.64}$$

Here the receiver filter $H_R(\omega)$ merely bandlimits the noise.

The transmitted signal can be expressed as

$$s(t) = \sqrt{2}\,A\sum_{k=-\infty}^{\infty}\Big\{\sum_{n=-\infty}^{\infty} c_n h_T[t-2nT]\cos[(\omega_k+\omega_o)t+\theta_k] + \sum_{n=-\infty}^{\infty} d_n h_T[t-2nT]\sin[(\omega_k+\omega_o)t+\theta_k]\Big\}P(t-2kT) \tag{2.65}$$

where A is the signal amplitude, c_n, d_n are the precoded bits in quadrature and in phase channels taking on values ± 1, these are obtained by splitting the

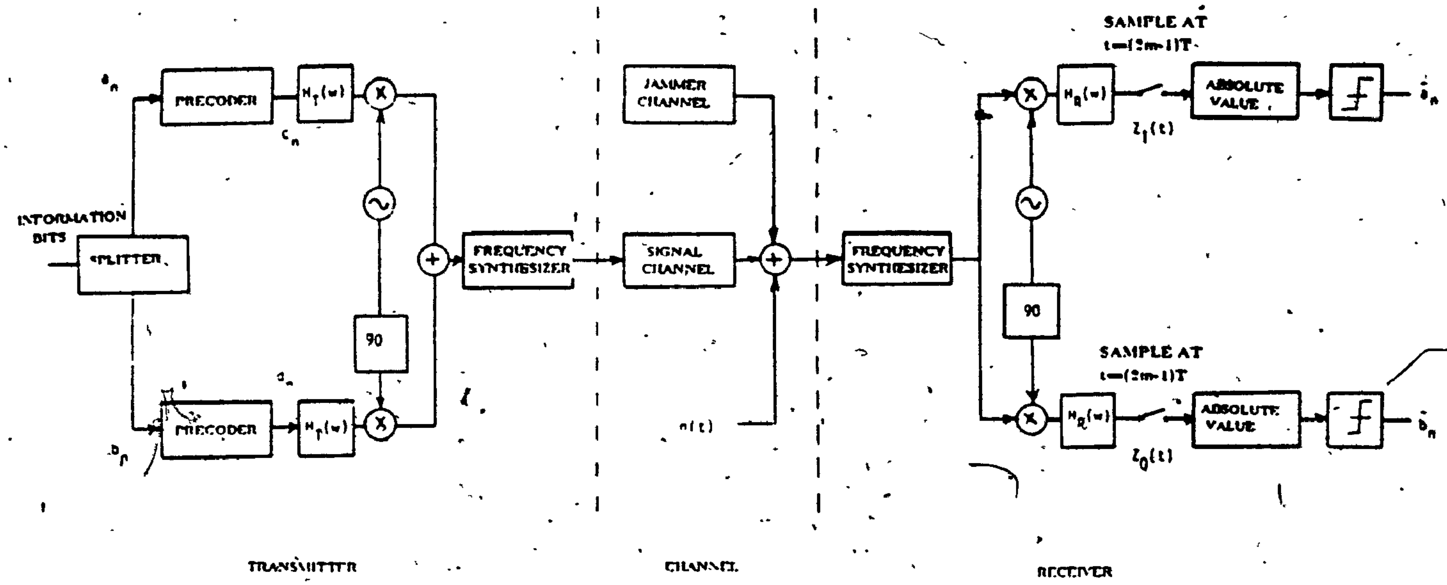


Figure 2.9 Block diagram of a FH/QPR system. *(Source: Ref.[12])*

information bit array into odd and even arrays in a well known manner, $h_T(t)$ is the transmitter filter defined above, ω_o is the oscillator frequency, ω_k is the kth hopping frequency, and θ_k is the random phase generated by the frequency synthesizer in the kth interval (bit interval).

$$P(t) = \begin{cases} 1 & 0 \leq t \leq 2T \\ 0 & \text{elsewhere} \end{cases} \tag{2.66}$$

It is to be noted that the first summation with index k is one way of representing a FH signal, while the infinite summation represent the generation process of a QPR signal as is well known.

The channel model which assumes wide-sense-stationary-uncorrelated-scattering (WSSUS) is used in the analysis. This is a representative model for practical radio links and troposcatter channels [15],[19]. If $x(t) \cos(\omega_o t + \theta)$ is the input to this channel, then the corresponding output is given by

$$y(t) = \mathrm{Re}\left\{[x(t) + \gamma\int_{-\infty}^{\infty} \beta(\tau)\, x(t-\tau)\, d\tau] \exp j(\omega_o t+\theta)\right\} \tag{2.67}$$

where γ is the transmission coefficient for the fading channel. $\beta(\tau)$ is defined as a independent, zero mean, complex Gaussian random process representing the low-pass equivalent impulse response of the frequency selective channel. The correlation function of $\beta(\tau)$ is denoted from [14],[15] as

$$\frac{1}{2}E\{\beta(\tau_1)\beta^*(\tau_2)\} = \rho(\tau_1)\, \delta(\tau_1-\tau_2) \tag{2.68}$$

and assuming [15],[17],[18]

$$E\{\beta(\tau_1)\beta(\tau_2)\} = 0 \tag{2.69}$$

where $\rho(\tau)$ for practical WSSUS channels is a real function of τ and $\delta(\tau)$ is the dirac delta function. As in [14],[15]-[16] $\rho(\tau)$ is the multipath-delay spread of the channel chosen as

$$\rho(\tau) = \begin{cases} \frac{1}{2T}\left(1 - \frac{|\tau|}{2T}\right) & |\tau| \leq 2T \\ 0 & \text{otherwise} \end{cases} \tag{2.70}$$

and this limits the intersymbol interference to just adjacent symbols. The received signal is given by

$$r(t) = s(t) + s_F(t) + j(t) + j_F(t) + n(t) \tag{2.71}$$

where s(t) is the specular component of the intended signal, given by

$$s(t) = \mathrm{Re}\left[\sqrt{2}A \sum_k P(t-2kT) \left\{ \sum_{n=-\infty}^{\infty} c_n h_T(t-2nT) - j \sum_{n=-\infty}^{\infty} d_n h_T(t-2nT) \right\} \cdot \exp j\,[(\omega_k+\omega_o)t+\theta_k] \right] \tag{2.72}$$

$s_F(t)$ is the faded part of the intended signal, given by

$$s_F(t) = \gamma_s \,\mathrm{Re}\left[\sqrt{2}\,A \sum_k \int_{-\infty}^{\infty} \beta_s(\tau) \left[P(t-\tau-2kT) \cdot\left\{ \sum_{n=-\infty}^{\infty} c_n h_T(t-\tau-2nT) - j \sum_{n=-\infty}^{\infty} d_n h_T[t-\tau-2nT] \right\} \right] d\tau \cdot \exp j\,[(\omega_k+\omega_o)t+\theta_k] \right] \tag{2.73}$$

similarly, the specular received components of the jamming signal is given by

$$j(t) = \lambda_J \,\mathrm{Re}\left\{ \sqrt{2J} \exp j\,[(\omega_k + \omega_o)t + \phi_k] \right\} \tag{2.74}$$

and the faded jamming component is given by

$$j_F(t) = \lambda_J \,\mathrm{Re}\left\{ \gamma_J \sqrt{2J} \int_{-\infty}^{\infty} \beta_J(\tau)\, d\tau \cdot \exp j\,[(\omega_k+\omega_o)t + \phi_k] \right\} \tag{2.75}$$

$\eta(t)$ is the additive White Gaussian noise of two-sided spectral density $\eta_o/2$. For the partial-band multi-tone jamming policy assumed, we let the total power of the jammer be denoted as J_T and the number of jamming tones by M. With the total jamming power of J_T equally divided among the M jamming tones then each tone has power of $J=J_T/M$. The variable λ_J takes a a value of 1 or 0, depending on whether the jammer is present or absent respectively.

Assuming that a coherent frequency synthesizer is available at the receiver. That is it is capable of estimating and correcting for the phase errors caused by the transmitter synthesizer, the channel and the receiver synthesizer. The dehopped signal then can be expressed as

$$y(t) = \mathrm{Re}\Big\{ \sqrt{2}A \Big[\sum_{n=-\infty}^{\infty} c_n\, h_T(t-2nT) - j \sum_{n=-\infty}^{\infty} d_n\, h_T[t-2nT]$$

$$+ \gamma_s \sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} c_n \int_{-\infty}^{\infty} \beta_s(\tau)\, P(t-\tau-2kT)\, h_T(t-\tau-2nT)\, d\tau$$

$$- j\gamma_s \sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} d_n \int_{-\infty}^{\infty} \beta_s(\tau)\, P(t-\tau-2kT)\, h_T[t-\tau-2nT]\, d\tau \Big]$$

$$\cdot \exp j\omega_o t \Big\}$$

$$+ \lambda_J \mathrm{Re}\Big\{ [\, \sqrt{2J} + \gamma_J\sqrt{2J} \int_{-\infty}^{\infty} \beta_J(\tau)\, d\tau\,] \sum_k P(t-2kT)\, \delta\omega_o\omega_k$$

$$\cdot \exp j(\omega_o t+\phi_k) \Big\} \tag{2.76}$$

then filtering by $H_R(\omega)$ to give the following quadrature and inphase signals,

$$Z_Q(t) = y(t)(\sqrt{2}\cos\omega_o t) * h_R(t)$$

$$= A \sum_{n=-\infty}^{\infty} c_n\, h_T(t-2nT)$$

$$+\gamma_s A \sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} c_n \int_{-\infty}^{\infty} \beta_s(\tau) P(t-\tau-2kT) h_T[t-\tau-2nT] d\tau * h_R(t)$$

$$+ \lambda_J\sqrt{J}\cos\phi$$

$$+ \lambda_J\gamma_J\sqrt{J}\cos\phi \sum_k P(t-2kT)\, \delta\omega_o\omega_k \int_{-\infty}^{\infty} \beta_J(\tau)\, d\tau * h_R(t)$$

$$+ N_c(t) \tag{2.77}$$

$$Z_I(t) = y(t)(\sqrt{2}\sin\omega_o t) * h_R(t)$$
$$= A\sum_{n=-\infty}^{\infty} d_n h_T[t-2nT]$$
$$+\gamma_s A\sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} c_n \int_{-\infty}^{\infty} \beta_s(\tau)P(t-\tau-2kT)h_T[t-\tau-2nT]d\tau * h_R(t)$$
$$+ \lambda_J\sqrt{J}\sin\phi$$
$$+ \lambda_J\gamma_J\sqrt{J}\sin\phi\sum_k P(t-2kT)\ \delta\omega_o\omega_k\int_{-\infty}^{\infty}\beta_J(\tau)d\tau * h_R(t)$$
$$+ N_s(t) \qquad (2.78)$$

The baseband signals $Z_Q(t)$ and $Z_I(t)$ are then sampled to give

$$Z_{Qm} = Z_Q[(2m-1)T]$$
$$= A(c_m + c_{m-1}) + \lambda_J\sqrt{J}\cos\phi$$
$$+ \gamma_s A\sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} c_n\int_{-\infty}^{\infty}\beta_s(\tau)P[(2m-1)T-\tau-2kT]$$
$$\cdot h_T[(2m-1)T-\tau-2kT]d\tau * h_R[(2m-1)T]$$
$$+ \lambda_J\gamma_J\sqrt{J}\cos\phi\sum_k P[(2m-1)T-2kT]\delta\omega_o\omega_k$$
$$\cdot\int_{-\infty}^{\infty}\beta_J(\tau)d\tau * h_R[(2m-1)T]$$
$$+ N_c[(2m-1)T] \qquad (2.79)$$

$$Z_{Im} = Z_I[(2m-1)T]$$
$$= A(d_m + d_{m-1}) + \lambda_J\sqrt{J}\sin\phi$$
$$+ \gamma_s A\sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} d_n\int_{-\infty}^{\infty}\beta_s(\tau)\ P[(2m-1)T-\tau-2kT]$$

$$\cdot h_T[(2m-1)T-\tau-2kT]d\tau * h_R[(2m-1)T]$$

$$+ \lambda_J \gamma_J \sqrt{J} \sin\phi \sum_k P[(2m-1)T-2kT]\ \delta\omega_o\omega_k$$

$$\cdot \int_{-\infty}^{\infty} \beta_J(\tau)d\tau * h_R[(2m-1)T]$$

$$+ N_s[(2m-1)T] \tag{2.80}$$

To evaluate the probability of bit errors, one needs to find the mean and variances of various terms of Z_{Qm} and Z_{Im}. However because of the many parameters involved we choose to fix them first then remove the conditioning later. Before performing this step though we will assume that inter-symbol interference exists only between neighboring adjacent bits and so consider only the quantities c_{-2}, c_{-1}, c_0, c_1, c_2 in equations (2.78), (2.79). With this assumption and assuming given values c_{-2}, c_{-1}, c_0, c_1, c_2, d_{-2}, d_{-1}, d_0, d_1, d_2 and inserting the known $h_R(t)$, $P(t)$ in (2.78), (2.79) we see that the 3rd term becomes a deterministic term. Also the first term of (2.78), (2.79) is deterministic once the data bits c_m, c_{m-1}, d_m, d_{m-1} are assumed. For the computation of probability of bit errors we compute the probability of bit errors in the I and the Q banks of the receiver. To this objective a normal density is assigned to the decision variables Z_{Qm}, Z_{Im} and the needed variances of faded signal, faded jammer and the noise are presented as shown later. The transmitted data sequences a_l and b_l are then recovered by the well known detection criteria for QPR signals (Table 2.1).

$$\hat{a}_l = \begin{cases} +1 & \text{if } |Z_{sl}| > A \\ -1 & \text{if } |Z_{sl}| < A \end{cases} \quad \hat{b}_l = \begin{cases} +1 & \text{if } |Z_{cl}| > A \\ -1 & \text{if } |Z_{cl}| < A \end{cases} \tag{2.81}$$

Table 2.1.

Transmitted Symbol b_l	Received Symbols c_l	c_{l-1}	Duobinary Value $c_l + c_{l-1}$
-1	+1	-1	0
-1	-1	+1	0
+1	+1	+1	+2
+1	-1	-1	-2

To evaluate the probability of error, we need the following results. The conditional variances of the faded signal, faded jammer and of noise. Considering first the variance of the faded signal

$$\sigma_{Q_{s_1}}^2 = E\Big\{\mathrm{Re}\Big[\gamma_s A \sum_k \delta\omega_o \omega_k \sum_{n=-\infty}^{\infty} c_n \int_{-\infty}^{\infty} \beta_s(\tau_1) P(t-2kT-\tau_1)$$

$$\cdot h_T(t-\tau_1-2nT)d\tau_1 * h_R(t)\Big]$$

$$\cdot \mathrm{Re}\Big[\gamma_s A \sum_{l=-\infty}^{\infty} \delta\omega_o\omega_l \sum_{m=-\infty}^{\infty} c_m \int_{-\infty}^{\infty} \beta_s(\tau_2)\, P(t-2lT-\tau_2)$$

$$\cdot h_T(t-\tau_2-2mT)d\tau_2 * h_R(t)\Big]\Big\} \tag{2.82}$$

using the relation $\mathrm{Re}(a)\mathrm{Re}(b) = \frac{1}{2}\mathrm{Re}(ab^*) + \frac{1}{2}\mathrm{Re}(ab)$ and equations (2.67),(2.68) the above becomes

$$\sigma_Q^2 = \frac{\gamma_s^2 A^2}{2} \int_{-\infty}^{\infty}\int_{-\infty}^{\infty}\int_{-\infty}^{\infty}\int_{-\infty}^{\infty} E[\beta_s(\tau_1)\beta_s^*(\tau_2)]$$

$$\cdot \sum_k \delta\omega_o\omega_k \sum_{n=-\infty}^{\infty} c_n P(l_1-2kT-\tau_1)\; h_T(l_1-\tau_1-2nT)\; h_R(t-l_1)$$

$$\cdot \sum_{l=-\infty}^{\infty} \delta\omega_o\omega_l \sum_{m=-\infty}^{\infty} c_m P(l_2-2lT-\tau)\; h_T(l_2-\tau-2mT)\; h_R(t-l_2)$$

$$\cdot\, d\tau_1 d\tau_2 dl_1 dl_2 \tag{2.83}$$

Substituting eq. (2.68) in the above results in

$$\sigma_Q^2 = \frac{\gamma_s^2 A^2}{2} \int_{-\infty}^{\infty}\int_{-\infty}^{\infty}\int_{-\infty}^{\infty} \rho(\tau) \sum_k \delta\omega_o \omega_k$$

$$\cdot \sum_{n=-\infty}^{\infty} c_n P(l_1-2kT-\tau)\, h_T(l_1-\tau-2nT)\, h_R(t-l_1)$$

$$\cdot \sum_{l=-\infty}^{\infty} \delta\omega_o \omega_l \sum_{m=-\infty}^{\infty} c_m P(l_2-2lT-\tau)\, h_T(l_2-\tau-2mT)\, h_R(t-l_2)$$

$$\cdot d\tau\, dl_1\, dl_2 \tag{2.84}$$

$$\sigma_Q^2 = \gamma_s^2 A^2 \int_{-2T}^{2T} \rho(\tau) \Big[\sum_k \delta\omega_o \omega_k \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} c_n P(\lambda-2kT-\tau)$$

$$\cdot\, h_T(\lambda-\tau-2nT) h_R(t-\lambda) d\lambda \Big]^2 d\tau \tag{2.85}$$

Substituting eq. (2.69) the multipath-delay spread of the channel which limits the intersymbol interference to just adjacent symbols. Then we get the following four cases

$$\sigma_Q^2 = \frac{\gamma_s^2 A^2 16 T^3}{2\pi^4} I_{s_i}; \qquad i=1,2,3,4. \tag{2.86}$$

case (i) $\omega_{-1} = \omega_o$, $\omega_1 = \omega_o$

$$I_{s_1} = \int_{-2T}^{0} \left(1+\frac{\tau}{2T}\right) \left[\sum_{n=-2}^{2} c_n \int_{-2T+\tau}^{4T+\tau} h_T(\lambda-\tau-2nT) h_R(t-\lambda) d\lambda \right]^2 d\tau$$

$$+ \int_{0}^{2T} \left(1-\frac{\tau}{2T}\right) \left[\sum_{n=-2}^{2} c_n \int_{-2T+\tau}^{4T+\tau} h_T(\lambda-\tau-2nT) h_R(t-\lambda) d\lambda \right]^2 d\tau \tag{2.87}$$

case (ii) $\omega_{-1} \neq \omega_o$, $\omega_1 \neq \omega_o$

$$I_{s_2} = \int_{-2T}^{0} \left(1+\frac{\tau}{2T}\right) \left[\sum_{n=-2}^{2} c_n \int_{\tau}^{2T+\tau} h_T(\lambda-\tau-2nT) h_R(t-\lambda) d\lambda \right]^2 d\tau$$

$$+\int_{0}^{2T}\left(1-\frac{\tau}{2T}\right)\left[\sum_{n=-2}^{2}c_n\int_{\tau}^{2T+\tau}h_T(\lambda-\tau-2nT)h_R(t-\lambda)d\lambda\right]^2 d\tau \tag{2.88}$$

case (iii) $\omega_{-1}\neq\omega_o,\ \omega_1=\omega_o$

$$I_{s_3}=\int_{-2T}^{0}\left(1+\frac{\tau}{2T}\right)\left[\sum_{n=-2}^{2}c_n\int_{\tau}^{4T+\tau}h_T(\lambda-\tau-2nT)h_R(t-\lambda)d\lambda\right]^2 d\tau$$
$$+\int_{0}^{2T}\left(1-\frac{\tau}{2T}\right)\left[\sum_{n=-2}^{2}c_n\int_{\tau}^{4T+\tau}h_T(\lambda-\tau-2nT)h_R(t-\lambda)d\lambda\right]^2 d\tau \tag{2.89}$$

case (iv) $\omega_{-1}=\omega_o,\ \omega_1\neq\omega_o$

$$I_{s_4}=\int_{-2T}^{0}\left(1+\frac{\tau}{2T}\right)\left[\sum_{n=-2}^{2}c_n\int_{-2T+\tau}^{2T+\tau}h_T(\lambda-\tau-2nT)h_R(t-\lambda)d\lambda\right]^2 d\tau$$
$$+\int_{0}^{2T}\left(1-\frac{\tau}{2T}\right)\left[\sum_{n=-2}^{2}c_n\int_{-2T+\tau}^{2T+\tau}h_T(\lambda-\tau-2nT)h_R(t-\lambda)d\lambda\right]^2 d\tau \tag{2.90}$$

where

$$h_T(\lambda-\tau-2nT)\,h_R(t-\lambda)=\frac{\cos\dfrac{\pi(\lambda-\tau-2nT)}{2T}}{\left[1-\dfrac{(\lambda-\tau-2nT)^2}{T^2}\right]}\,\frac{\sin\dfrac{\pi(t-\lambda)}{2T}}{(t-\lambda)} \tag{2.91}$$

sampled at $t=(2m-1)T$. Using trignometric identities and knowing m takes on integer values (2.91) may be written as

$$h_T(\lambda-\tau-2nT)\,h_R(t-\lambda)=\frac{\text{Num}}{\text{Denom}} \tag{2.92}$$

where

$$\text{Num}=\left\{\left(.5+.5\cos\frac{\lambda\pi}{T}\right)\cos\frac{\pi\tau}{2T}+2\sin\frac{\pi\lambda}{T}\sin\frac{\pi\tau}{2T}\right\}$$
$$\cdot\cos n\pi\cos k\pi \tag{2.93}$$

$$\begin{aligned}\text{Denom}=&-\lambda^3+\lambda^2[(2m-1)T+2(\tau+2nT)]\\&+\lambda[T^2-(\tau+2nT)-2(\tau+2nT)(2m-1)T]\\&+(2m-1)T[(\tau+2nT)^2-T^2]\end{aligned} \tag{2.94}$$

Similarly for the jammer case it can be shown that

$$\sigma_J^2 = \frac{\gamma_J^2 J}{\pi^2} I_{J_i}; \quad i=1,2,3,4. \tag{2.95}$$

case (i) $\omega_{-1} = \omega_o, \omega_1 = \omega_o$

$$I_{J_1} = Si^2(6\omega T) \tag{2.96}$$

case (ii) $\omega_{-1} \neq \omega_o, \omega_1 \neq \omega_o$

$$I_{J_2} = [Si(4\omega T) - Si(2\omega T)]^2 \tag{2.97}$$

case (iii) $\omega_{-1} \neq \omega_o, \omega_1 = \omega_o$

$$I_{J_3} = [Si(6\omega T) - Si(2\omega T)]^2 \tag{2.98}$$

case (iv) $\omega_{-1} = \omega_o, \omega_1 \neq \omega_o$

$$I_{J_4} = Si^2(4\omega T) \tag{2.99}$$

where

$$Si(x) = \int_0^x \frac{\sin z}{z} dz \tag{2.100}$$

Next the variance of the noise is

$$\sigma_N^2 = \frac{N_o}{2}\left[\frac{1}{2\pi}\int_{-\pi/2T}^{\pi/2T} |H_R(\omega)|^2 d\omega\right] = \frac{N_o}{4T} \tag{2.101}$$

In addition to the above results, to characterize the performance of the system by the probability of bit error as a function of bit energy to noise ratio, bit energy to jamming noise ratio we need the following, a relation between signal amplitude A to the average transmitted power S.

$$S = (\sqrt{2}A)^2 \left[\frac{1}{2T} \int_0^{2T} \sum_{n=-\infty}^{\infty} h_T^2(t-2nT)dt \right]$$

$$= \frac{2A^2}{2T} \left[\frac{1}{2\pi} \int_{-\pi/2T}^{\pi/2T} | H_T(\omega) |^2 d\omega \right] = 4A^2 \qquad (2.102)$$

Based on all the above, the conditional probability of bit errors in the Q and I channels is given as

$$P_Q(e \mid \phi, \lambda_J, c_{-2}, c_{-1}, c_0, c_1, c_2, \omega_{-1}, \omega_1) =$$
$$\frac{3}{4}[Q(arg1)+Q(arg2)] - \frac{1}{4}[Q(arg3)+Q(arg4)] \qquad (2.103)$$

where

$$Q(x) = \int_x^{\infty} \frac{1}{\sqrt{2\pi}} e^{\frac{-z^2}{2}} dz \qquad (2.104)$$

$$arg1 = \frac{A + \lambda_J \sqrt{J} \cos \phi}{\sigma_T} \qquad (2.105)$$

$$arg2 = \frac{A - \lambda_J \sqrt{J} \cos \phi}{\sigma_T} \qquad (2.106)$$

$$arg3 = \frac{3A + \lambda_J \sqrt{J} \cos \phi}{\sigma_T} \qquad (2.107)$$

$$arg4 = \frac{3A - \lambda_J \sqrt{J} \cos \phi}{\sigma_T} \qquad (2.108)$$

$$\sigma_T = \left[\frac{N_o}{4T} + \frac{\gamma_s^2 A^2 16 T^3}{\pi^4} I_{S_1} + \frac{\lambda_J \gamma_J J}{\pi^2} I_{J_1} \right]^{1/2} \qquad (2.109)$$

Following the same procedure,

$$P_I(e \mid \phi, \lambda_J, c_{-2}, c_{-1}, c_0, c_1, c_2, \omega_{-1}, \omega_1) =$$
$$\frac{3}{4}[Q(arg1)+Q(arg2)] - \frac{1}{4}[Q(arg3)+Q(arg4)] \qquad (2.110)$$

the arguments in this case are the same as defined above but with $\cos \phi$ replaced by $\sin \phi$. Defining $\rho = M/N$ as a fraction of the total slots jammed, $N_J = J_T/W$ as the effective jammer power spectral density in the total frequency hopping band W. Using the following relation $S=4A^2$ (average transmitted power) and $J=J_T/M$ as defined above then defining $E_b=ST_b$ as bit

energy. Based on these definitions, we may rewrite the arguments in terms of bit energy yielding

$$\mathrm{arg1} = \frac{1 + \dfrac{\sqrt{2}\lambda_J}{\sqrt{\rho E_b/N_J}}\cos\phi}{\left[\dfrac{N_o}{E_b} + \dfrac{\gamma_s^2 16T^3}{2\pi^4} I_{s_1} + \dfrac{2\lambda_J\gamma_J^2}{\pi^2\rho E_b/N_J} I_{J_1}\right]^{1/2}} \tag{2.111}$$

$$\mathrm{arg2} = \frac{1 - \dfrac{\sqrt{2}\lambda_J}{\sqrt{\rho E_b/N_J}}\cos\phi}{\left[\dfrac{N_o}{E_b} + \dfrac{\gamma_s^2 16T^3}{2\pi^4} I_{s_1} + \dfrac{2\lambda_J\gamma_J^2}{\pi^2\rho E_b/N_J} I_{J_1}\right]^{1/2}} \tag{2.112}$$

$$\mathrm{arg3} = \frac{3 + \dfrac{\sqrt{2}\lambda_J}{\sqrt{\rho E_b/N_J}}\cos\phi}{\left[\dfrac{N_o}{E_b} + \dfrac{\gamma_s^2 16T^3}{2\pi^4} I_{s_1} + \dfrac{2\lambda_J\gamma_J^2}{\pi^2\rho E_b/N_J} I_{J_1}\right]^{1/2}} \tag{2.113}$$

$$\mathrm{arg4} = \frac{3 - \dfrac{\sqrt{2}\lambda_J}{\sqrt{\rho E_b/N_J}}\cos\phi}{\left[\dfrac{N_o}{E_b} + \dfrac{\gamma_s^2 16T^3}{2\pi^4 I_{s_1}} + \dfrac{2\lambda_J\gamma_J^2}{\pi^2\rho E_b/N_J} I_{J_1}\right]^{1/2}} \tag{2.114}$$

Averaging over all possible combinations of data bits

$$P_b(e \mid \phi,\lambda_J,\omega_{-1},\omega_1) = \frac{1}{32}\sum_{i=1}^{32} P_b(e \mid \phi,\lambda_J,c_{-2},c_{-1},c_0,c_1,c_2,\omega_{-1},\omega_1) \tag{2.115}$$

where the values of c_{-2}, c_{-1}, c_o, c_1, c_2 take all possible 32 combinations i.e. { (1,1,1,1,1), (1,1,1,1,-1), (1,1,1,-1,1) ... etc }. Averaging over λ_J, ϕ yields

$$P_b(e \mid \omega_{-1},\omega_1) = \frac{M}{N}\frac{1}{2\pi}\int_0^{2\pi} P(e \mid \lambda_J{=}1,\phi,\omega_{-1},\omega_1)\, d\phi$$

$$+ \left(1-\frac{M}{N}\right) P(e \mid \lambda_J=0, \omega_{-1}, \omega_1) \tag{2.116}$$

Finally, for the four possible cases of intersymbol interference $\delta\omega_o\omega_k$ assumed (i.e. conditions of ω_{-1}, ω_1 compared to ω_o).

$$P_b = \frac{1}{N^2} P(e \mid \omega_{-1}=\omega_o, \omega_1=\omega_o) + \frac{N-1}{N^2} P(e \mid \omega_{-1}=\omega_o, \omega_1 \neq \omega_o)$$

$$+ \frac{N-1}{N^2} P(e \mid \omega_{-1} \neq \omega_o, \omega_1=\omega_o) + \left(\frac{N-1}{N}\right)^2 P(e \mid \omega_{-1} \neq \omega_o, \omega_1 \neq \omega_o) \tag{2.117}$$

Figure 2.10 shows the probability of bit error (Pbe) as a function of M. It has been plotted for γ_s and γ_J at 0.2 and E_b / N_0 at 20 dB.

$$E_b/N_J = \frac{S/2J_0}{N/M} \tag{2.118}$$

Let $S/2J_0$ be 1.0 and M=2000, N is given as

$$N = Ks/32 * 2000$$

Where Ks is the number of interferers actually hitting the signal.

From the Figure 2.10 the Pbe is found corresponding to the value of N and E_b / N_J.

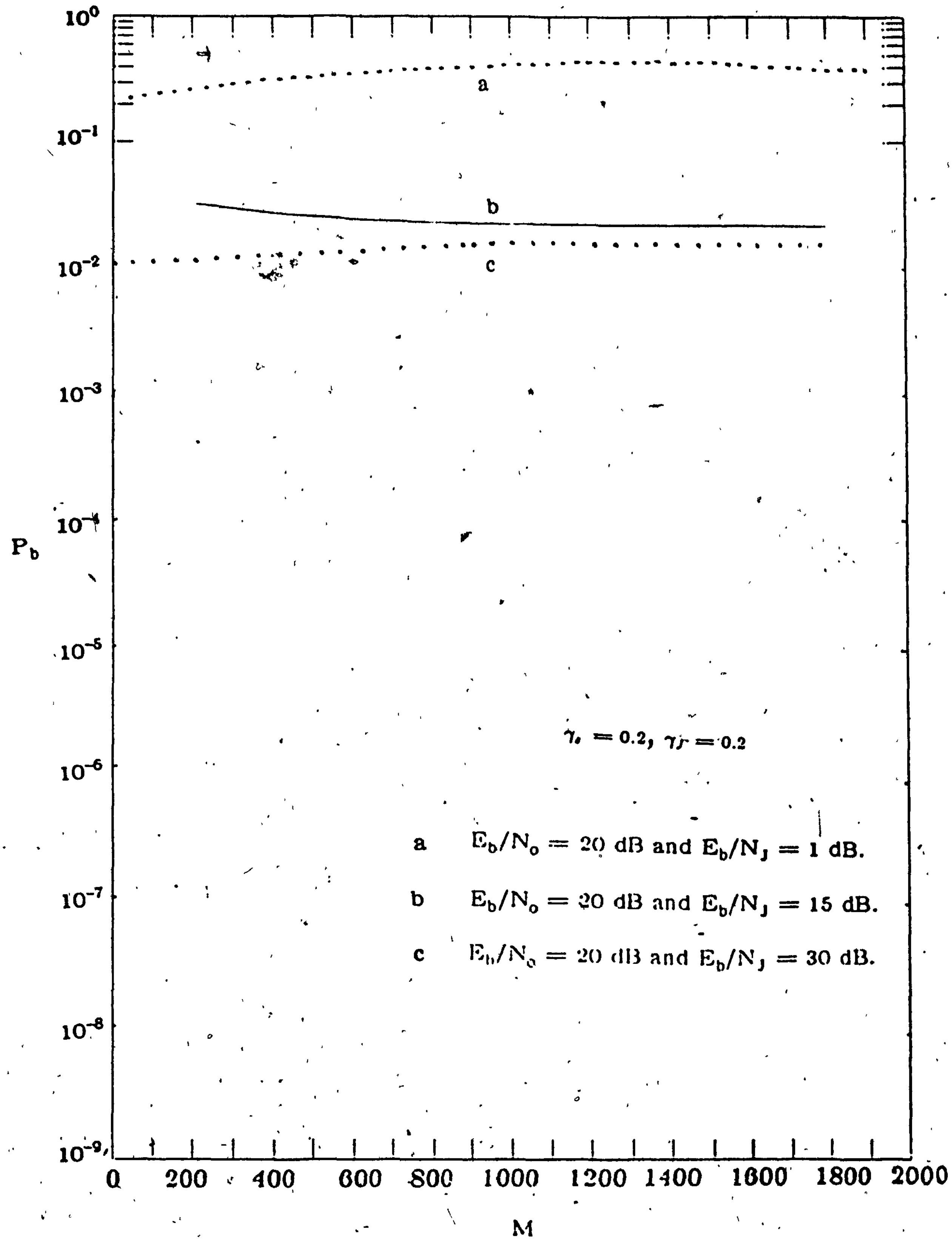


Figure 2.10 P_b versus M *(Source: Ref.[12])*

G	0.156	0.281	0.469	0.594	0.719	0.906	1.03	1.28
Ps	0.272	0.247	0.213	0.192	0.172	0.146	0.129	0.101

Table 2.2

Ps for a given total traffic (λ total). The number of voice users is varied and the number of data users is fixed at five. KD=1, λ data =0.3, and μ =0.1, PG=16.

G	0.188	0.344	0.469	0.594	0.719	0.902	1.03	1.28
Ps	0.294	0.264	0.242	0.220	0.199	0.171	0.154	0.123

Table 2.3

Ps for a given total traffic (λ total). The number of voice users is varied and the number of data users fixed at five. Kd=2, λ data =0.3, and μ =0.1, PG=16.

G	0.188	0.344	0.469	0.594	0.719	0.906	1.03	1.28
Ps	0.312	0.278	0.255	0.233	0.211	0.181	0.163	0.130

Table 2.4

Ps for a given total traffic (λ total). The number of voice users is varied and the number of data users is fixed at five. Kd=4, λ data =0.3, and μ =0.1, PG=16.

G	0.25	0.406	0.531	0.656	0.781
Ps	0.249	0.110	0.061	0.031	0.014

Table 2.5

Ps for a given total traffic (λ total). The number of data users is varied and the number of voice users is fixed at five. KD=2, λ data=0.1, and μ =0.5, PG=16.

G	0.25	0.406	0.531	0.656	0.781	0.989
Ps	0.259	0.186	0.143	0.109	0.083	0.052

Table 2.6

Ps for a given total traffic (λ total). The number of data users is varied and the number of voice users is fixed at five. KD=4, λ data=0.1, and μ =0.5, PG=16.

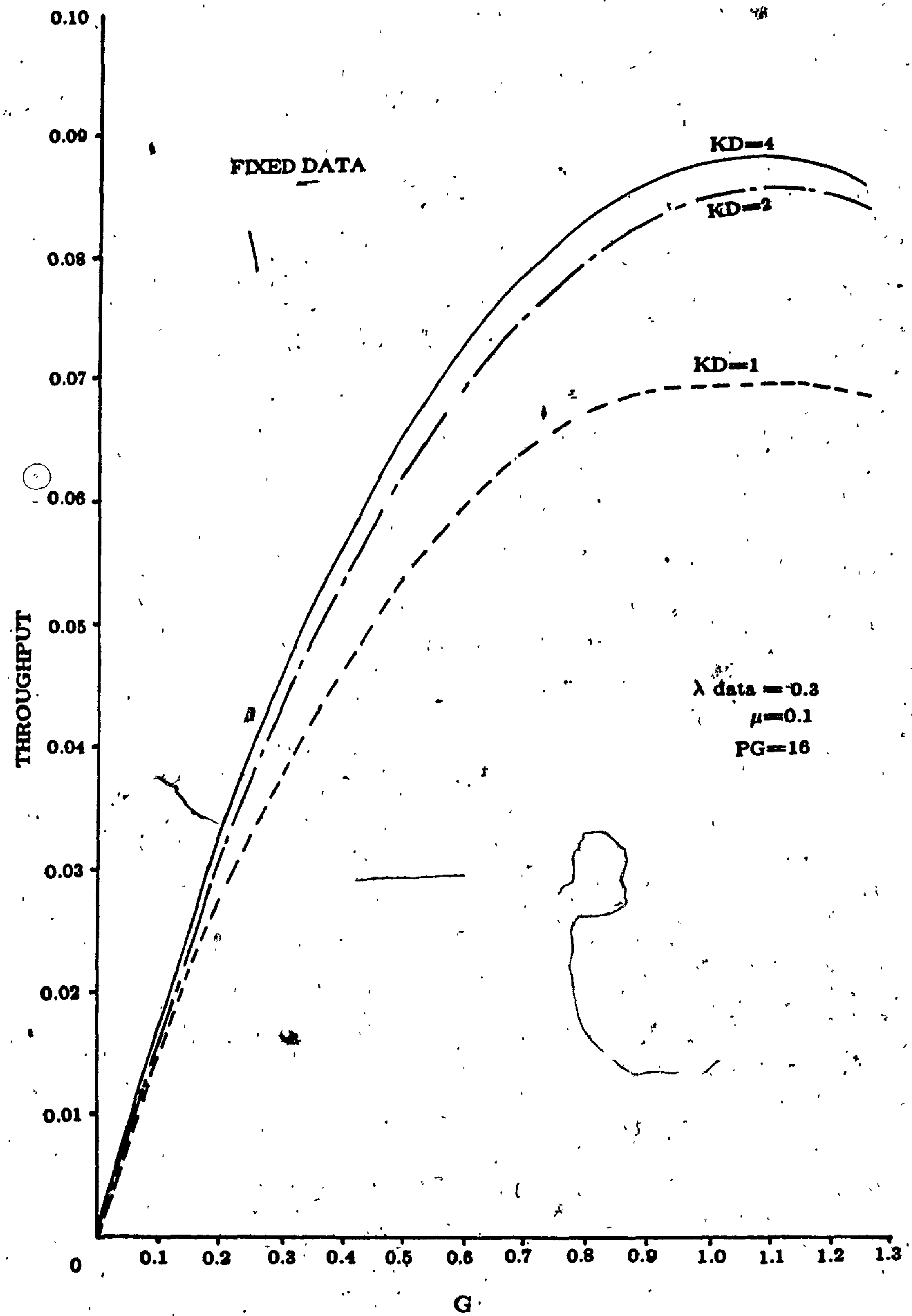


Figure 2.11 Throughput for the given traffic load. The number of voice users is varied and the data users is fixed at five.

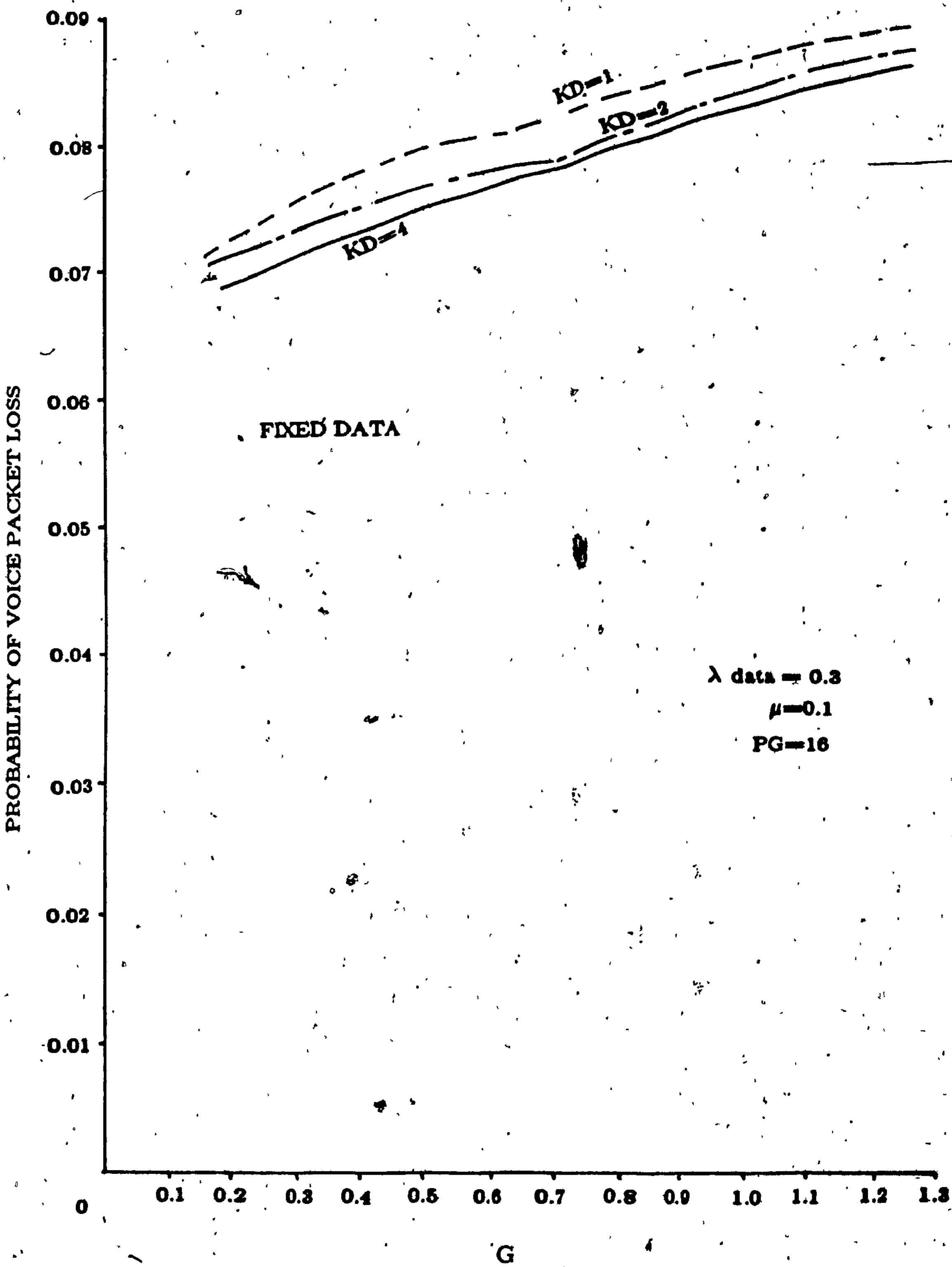


Figure 2.12 Blocking for the given traffic load. The number of voice users is varied and the data users is fixed at five.

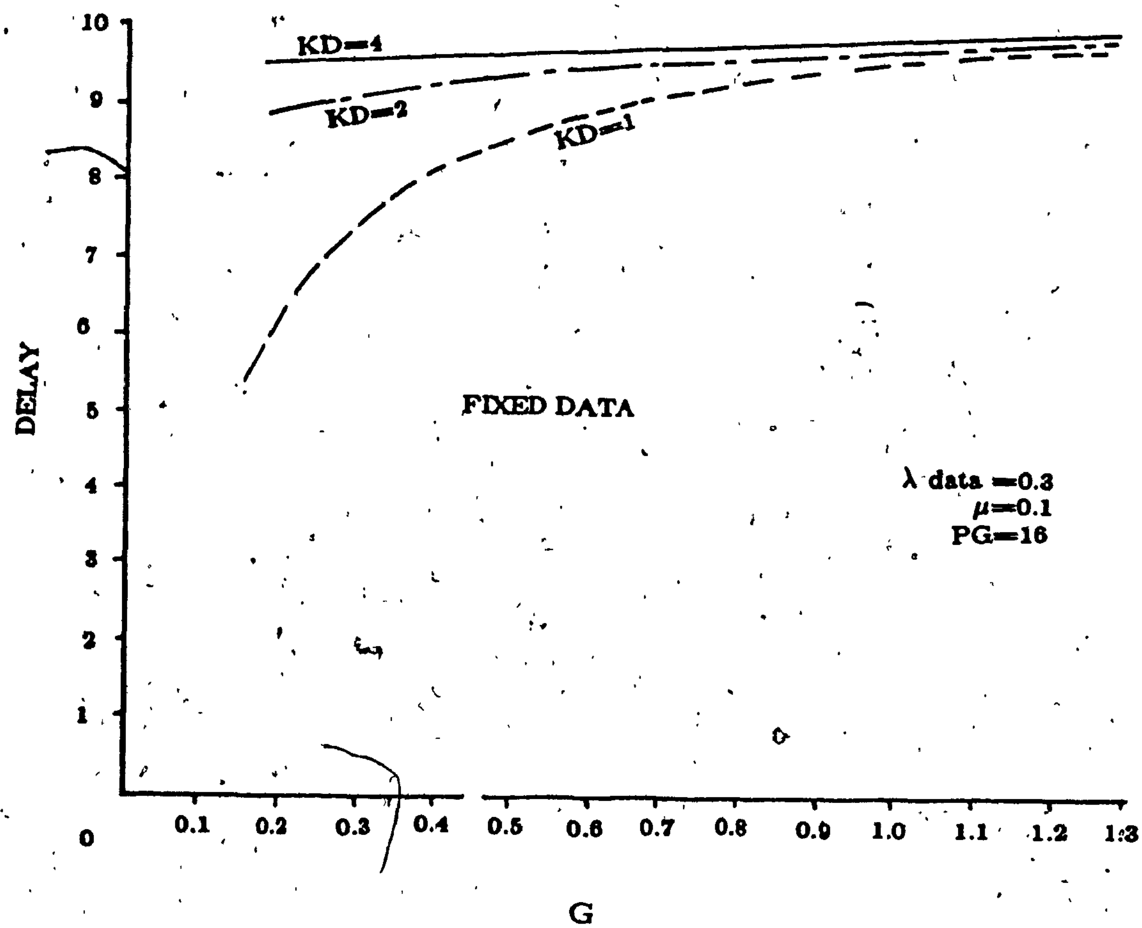


Figure 2.13 Delay for the given traffic load. The number of Voice users is varied and the data users is fixed at five.

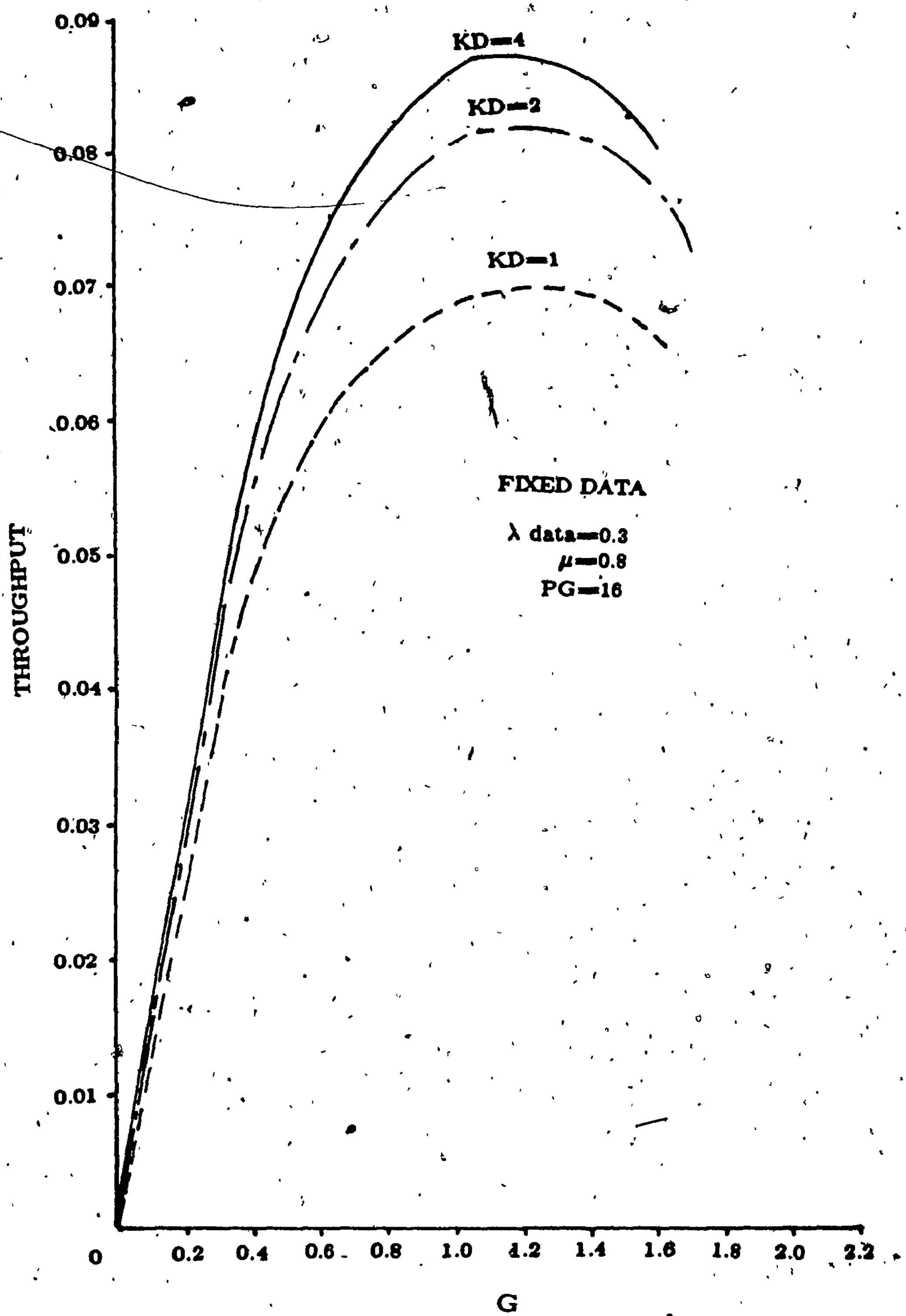


Figure 2.14 Throughput for the given traffic load. The number of voice users is varied and the data users is fixed at five.

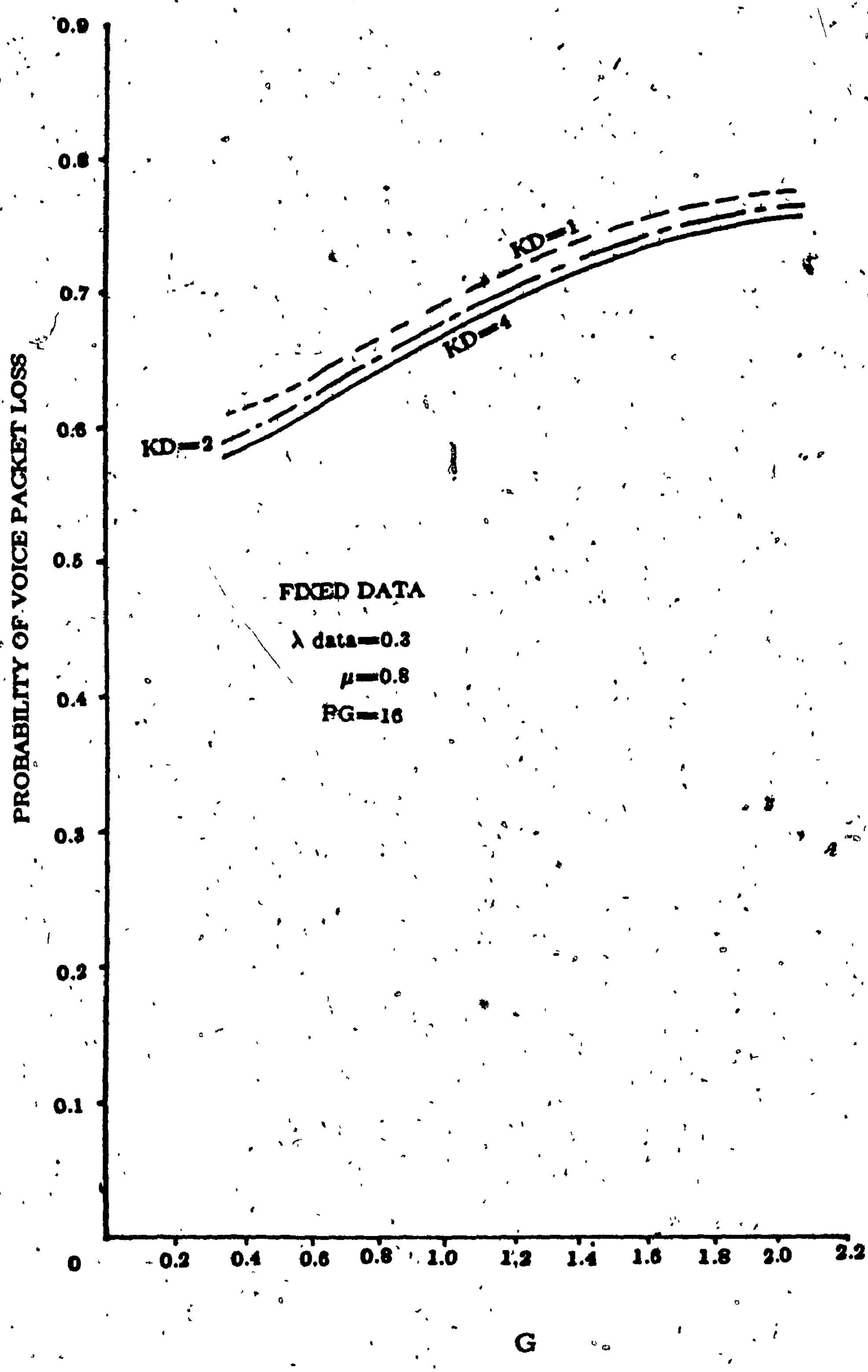


Figure 2.15 Blocking for the given traffic load: The number of voice users is varied and the data users is fixed at five.

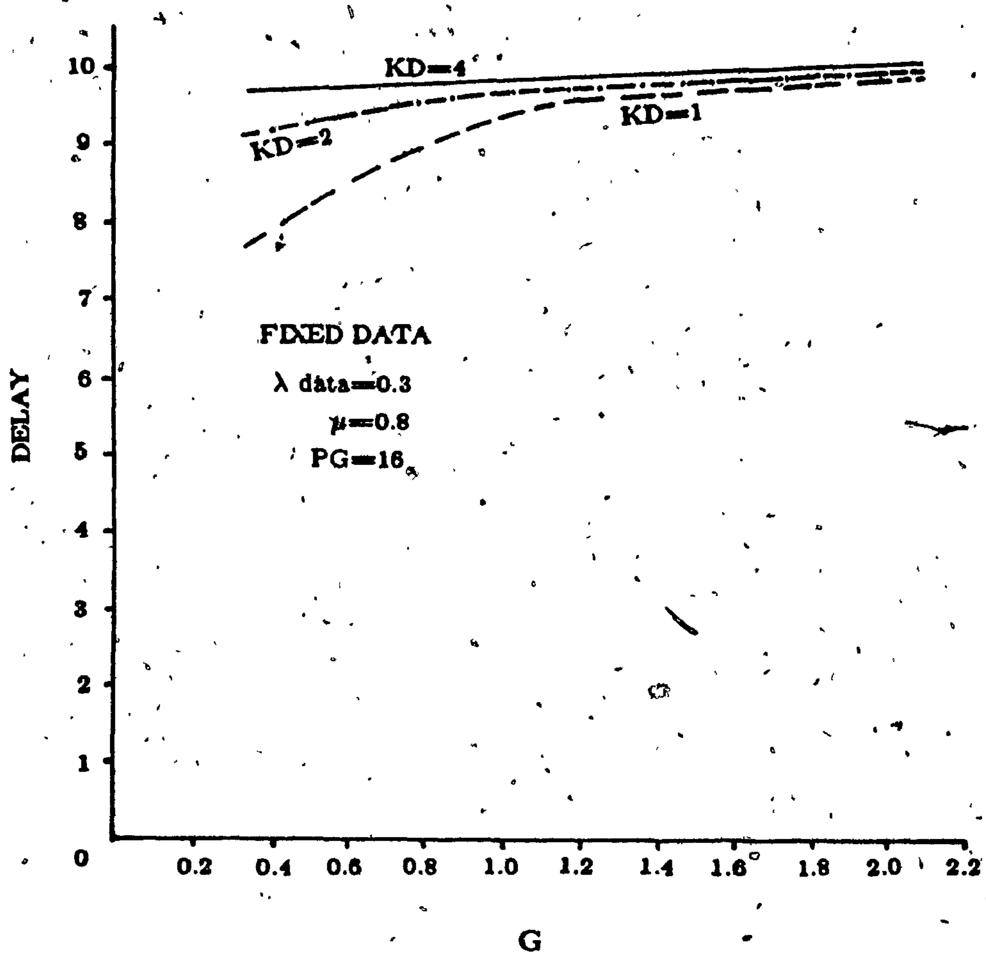


Figure 2.16 Delay for the given traffic load. The number of Voice users is varied and the data users is fixed at five.

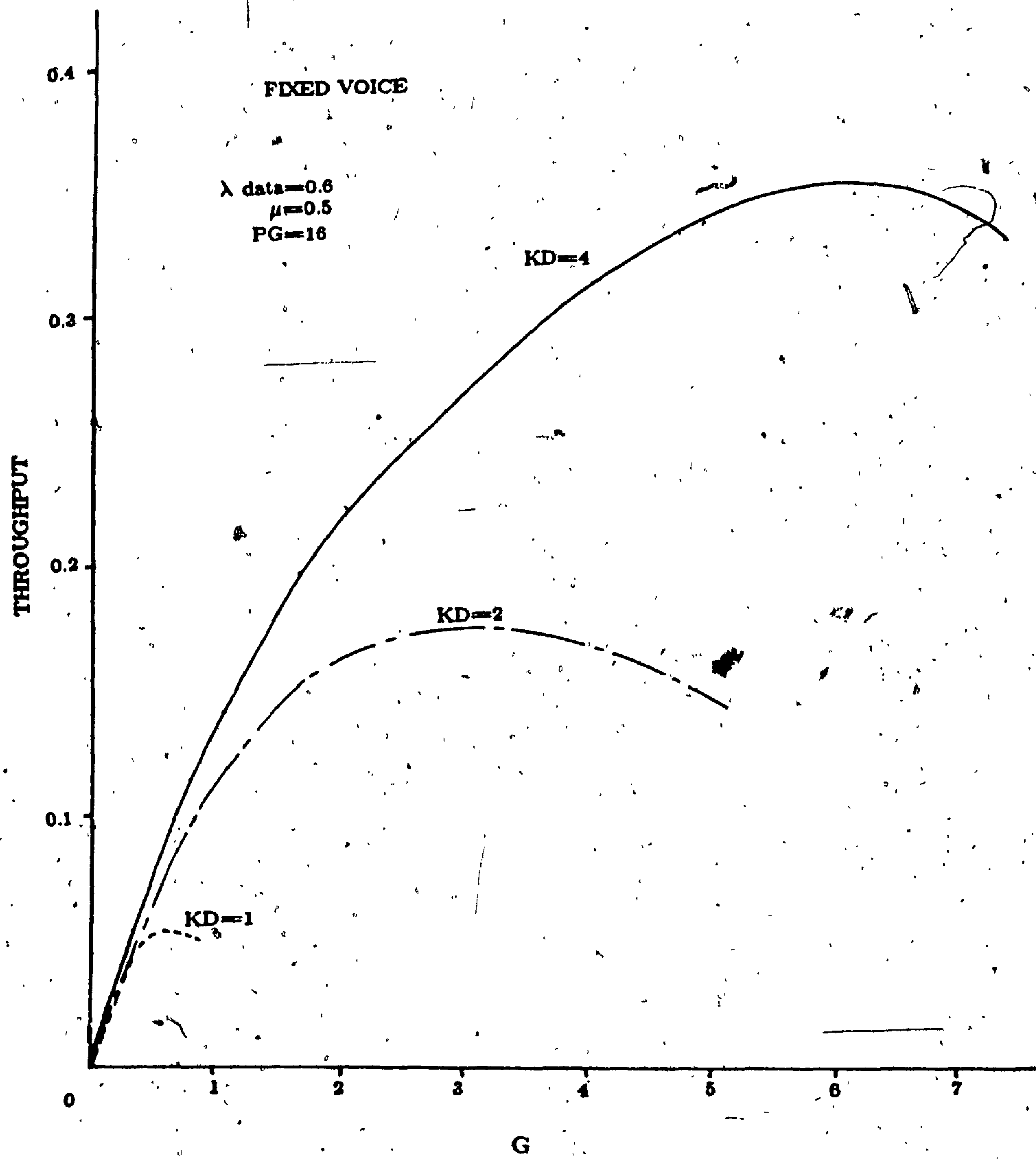


Figure 2.17 Throughput for the given traffic load. The number of data users is varied and the voice users is fixed at five.

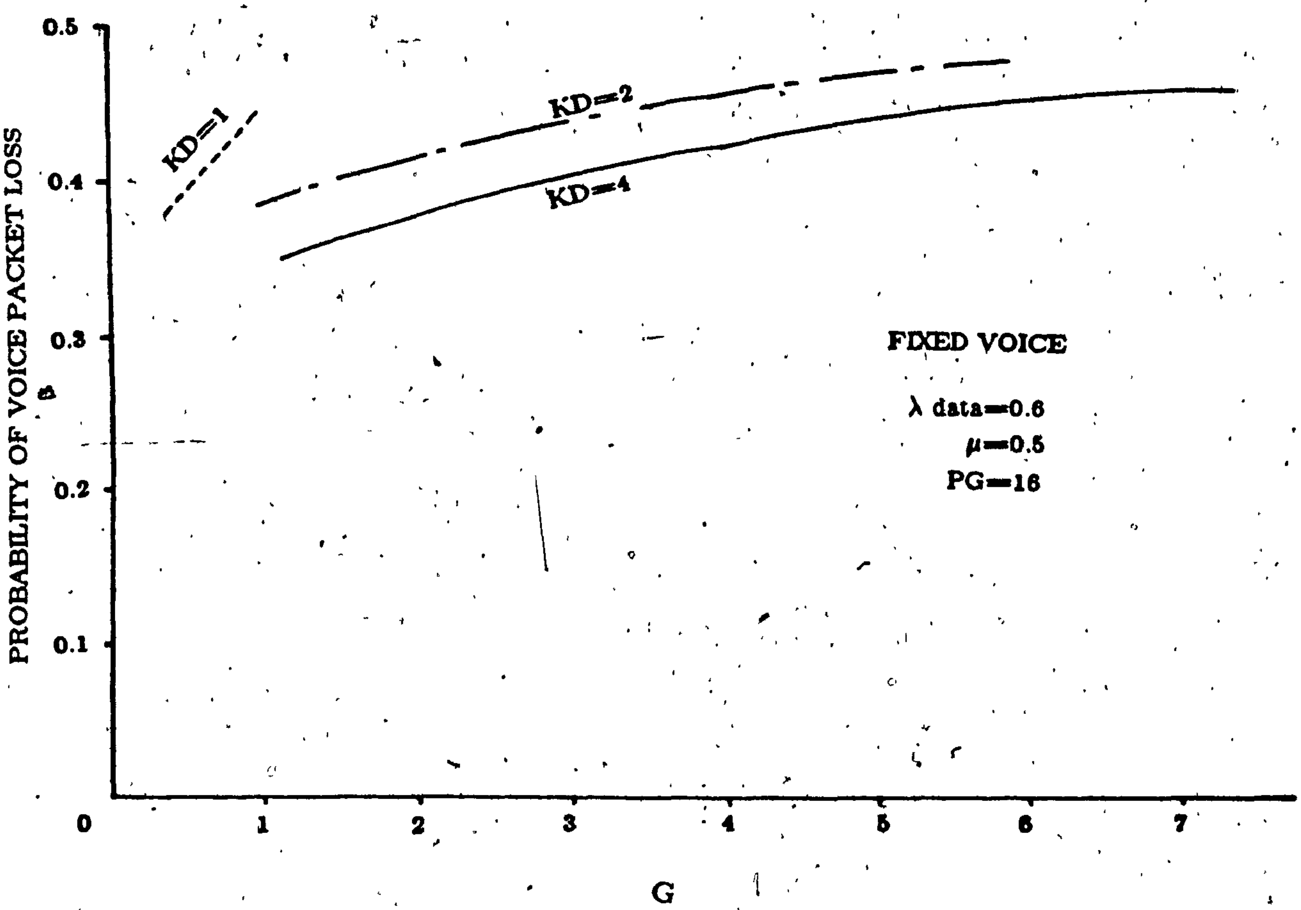


Figure 2.18 Blocking for the given traffic load. The number of data users is varied and the voice users is fixed at five.

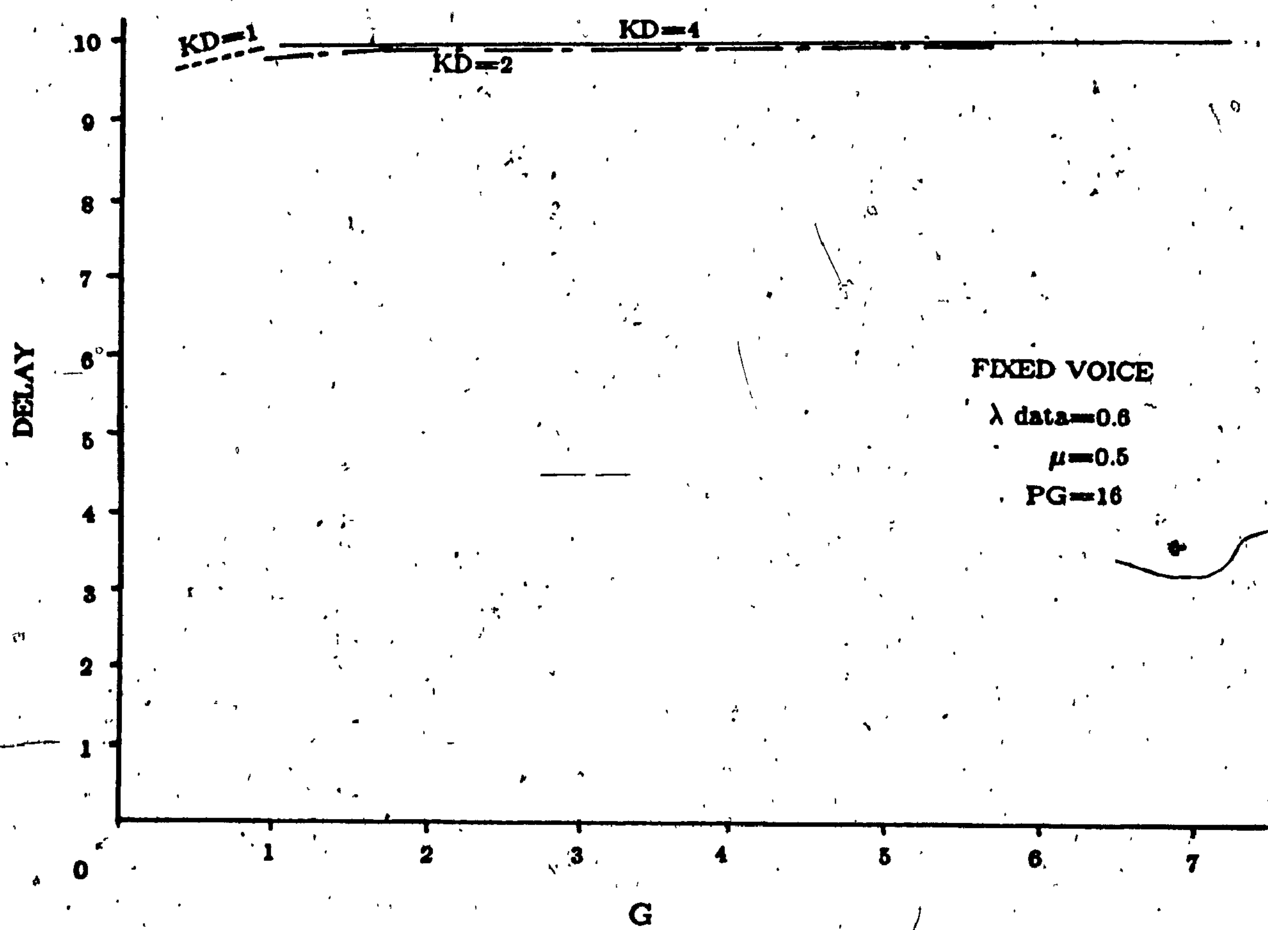


Figure 2.10 Delay for the given traffic load. The number of data users is varied and the voice users is fixed at five.

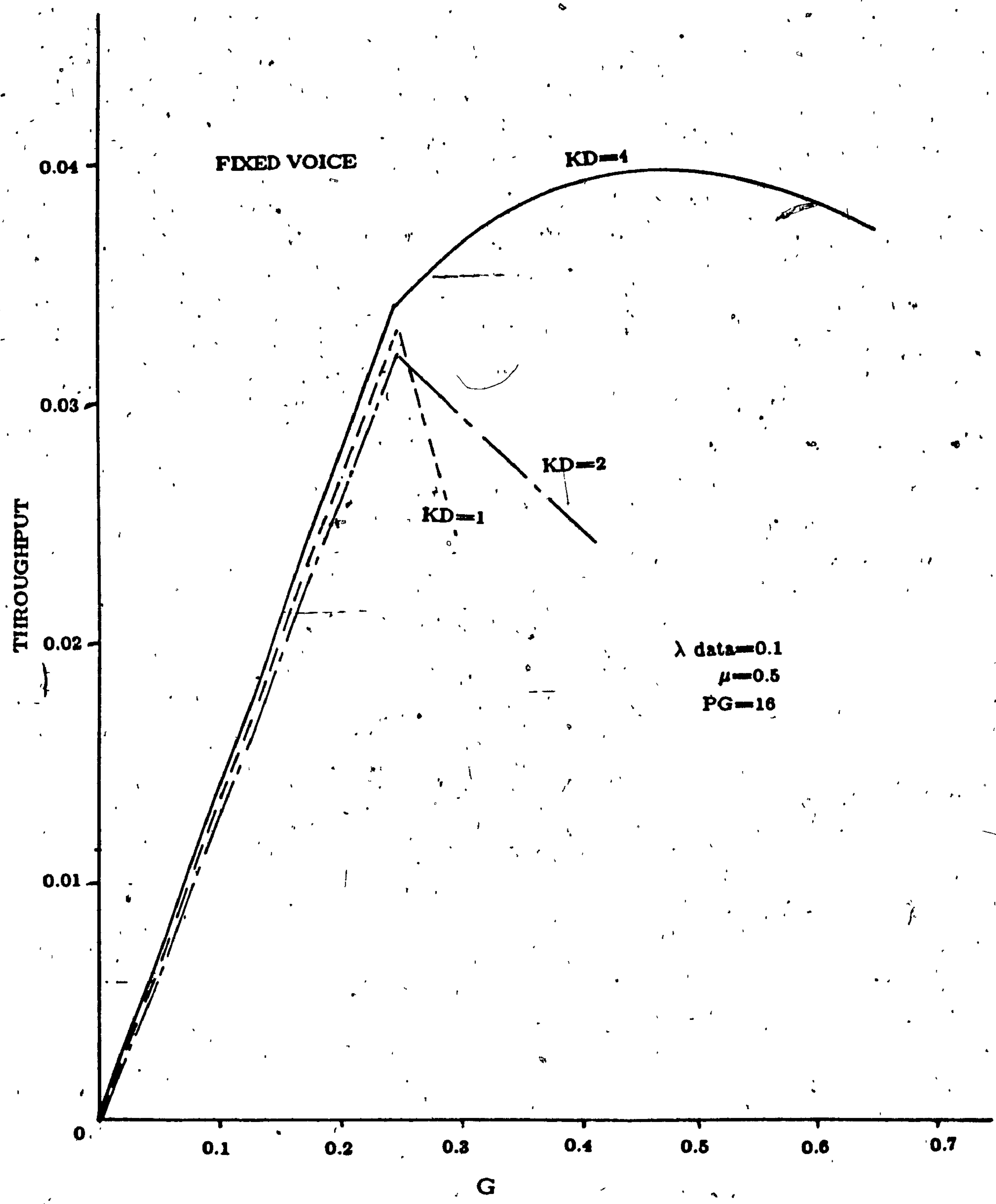


Figure 2.20 Throughput for the given traffic load. The number of data users is varied and the voice users is fixed at five.

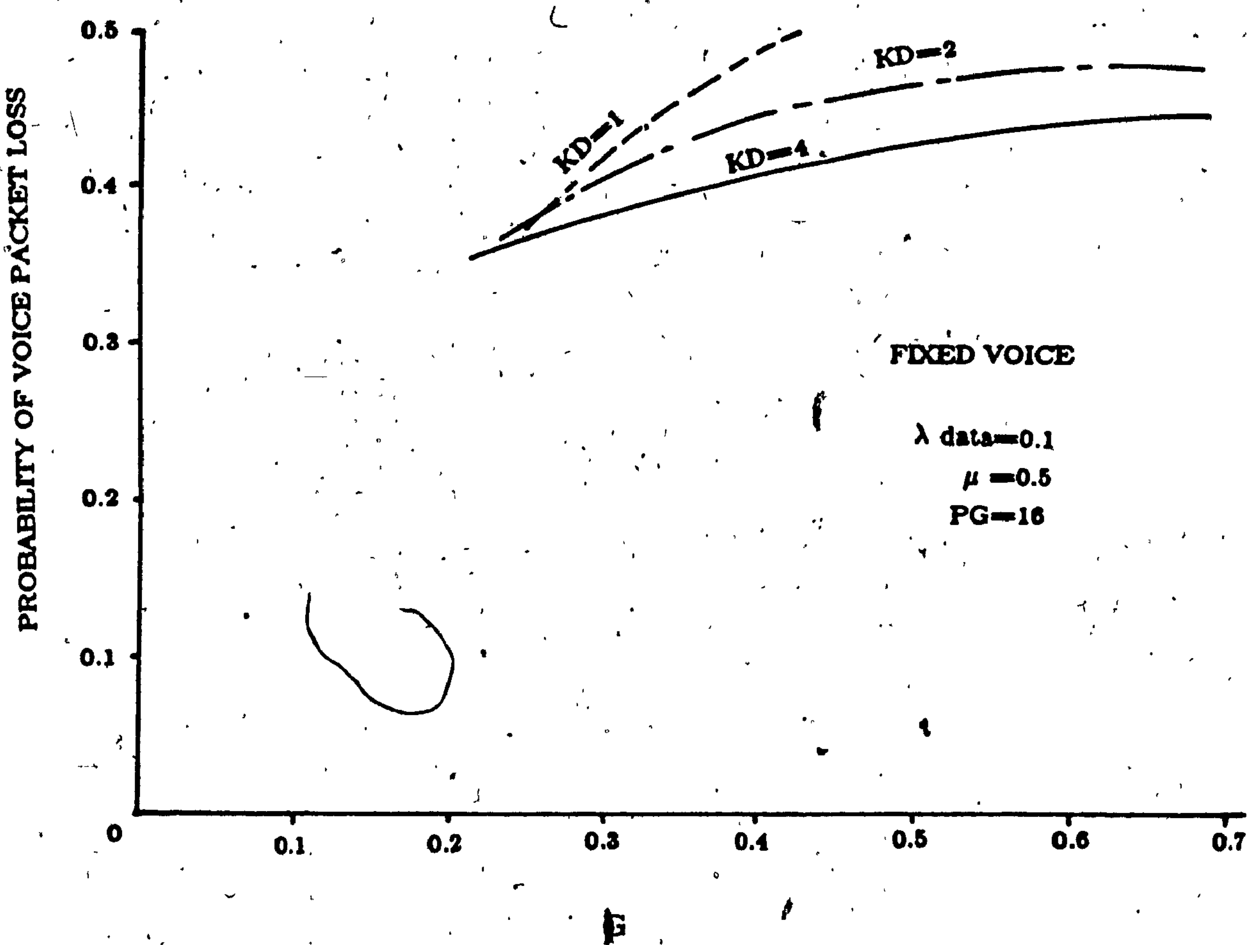


Figure 2.21 Blocking for the given traffic load. The number of data users is varied and the voice users is fixed at five.

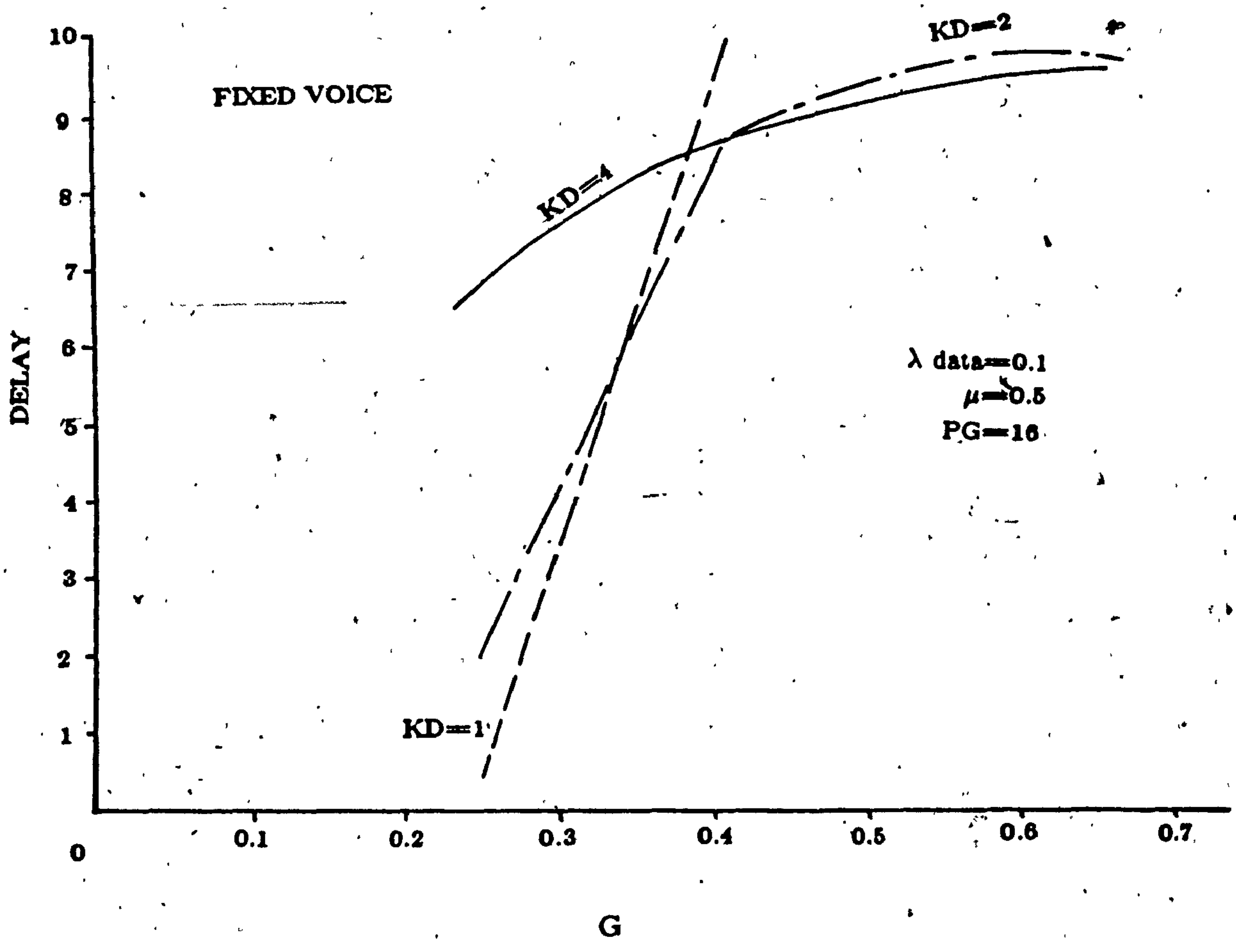


Figure 2.22 Delay for the given traffic load. The number of data users is varied and the voice users is fixed at five.

2.6 Numerical Results

Figures 2.11, 2.12 and 2.13 depict the throughput, delay and the blocking for the given total input traffic. The number of data users Md is fixed and the number of voice users Mv is varied, λ_d is 0.3 and the number of data users is fixed at five.

Figures 2.14, 2.15 and 2.16 depict the throughput, delay and the blocking for the given total input traffic. The number of data users Md is fixed and the number of voice users Mv is varied, λ_d is 0.3 and the number of data users is fixed at five.

Figures 2.17, 2.18 and 2.19 depict the throughput, delay and the blocking for the given total input traffic, given that the number of voice users Mv is constant and the number of data users Md is varying. In this case μ is 0.5 and the number of voice users is fixed at five.

Figures 2.20, 2.21 and 2.22 depict the throughput, delay and the blocking for the given total input traffic, the number of voice users is kept constant at five and μ is 0.1 and the number of data users is varied.

In all figures values of Kd in the range (1,2 & 4) were selected.

Figure 2.11 when compared to Figure 2.14 reveals that the throughput decreases when μ is increased from 0.1 to 0.8.

Figure 2.12 when compared to Figure 2.15 reveals that the blocking increases when μ is increased from 0.1 to 0.8.

Figure 2.13 when compared to Figure 2.16 reveals that the delay increases when μ is incresed from 0.1 to 0.8.

Figure 2.17 when compared to Figure 2.20 reveals that the throughput decreases when λ data is decreased from 0.6 to 0.1.

Figure 2.18 when compared to Figure 2.21 reveals that the blocking decreases when λ data is decreased from 0.6 to 0.1.

Figure 2.19 when compared to Figure 2.22 reveals that the delay decreases when λ data is decreased from 0.6 to 0.1.

Figure 2.17 when compared to Figure 2.19 reveals the familiar throughput - delay tradeoffs. The throughput is maximum for KD=4 and the delay is the maximum for the case of KD=1.

Figure 2.12 shows that for mostly voice load, KD=4 is preferred followed by KD=2 and KD=1.

Comparing Figures 2.11 and 2.20, Figure 2.20 suggests that backoff is necessary (KD=4) for a better throughput.

Comparing Figures 2.12 and 2.21 it is found that for a better probability of packet loss, the backoff should be employed (KD=4) in Figure 2.21 where the voice load is more pronounced.

Comparing the curves of Figure 2.22 it is seen that at low traffic loads, there is no need of backoff. However as the load increases it is necessary to reduce the delay. Most of the traffic originates from the voice users. Comparing to Figure 2.19 it is seen that the voice to data ratio than in Figure 2.22. The difference in performance becomes more pronounced for the case of KD=1,2,4 in Figure 2.22.

CHAPTER THREE

COMPARISION OF MULTIFREQUENCY CENTRAL TRANSMITTER, FIXED CHANNEL ASSIGNMENT, DEMAND ACCESS FIXED CHANNEL ASSIGNMENT AND DYNAMIC CHANNEL ASSIGNMENT SCHEMES

3.1 Introduction

In this chapter the results of chapter two are compared to other schemes such as the Multi frequency central transmitter, Fixed channel assignment, Demand assigned multiple access fixed channel assignment, and Dynamic channel assignment.

3.2 Multi frequency central transmitter

In this type of configuration [20] each mobile has c different frequency channels over which to communicate and is blocked only if all the c channels are in use. A central base station is connected to each of the non mobile users.

For analysis the assumptions made are:

The arrivals of initiations or requests for service form a Poisson process of Λ arrivals per second.

For a dispersed array the traffic is considered uniform over the whole urban area, and hence the arrival rate for each cell is $\lambda = \Lambda / N$, where N is the total number of zones.

The duration of the message is an exponential random variable with the mean length T.

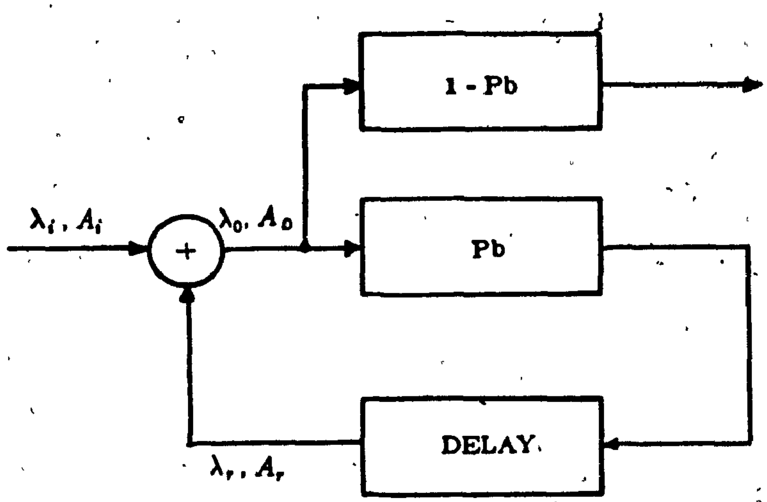


Figure 3.1

The location of the mobile unit is known to the central processing unit.

The cell size is very large and the average message length is short so that the possibility of a mobile unit crossing a cell boundry during communication does not occur.

The blocked calls are abondoned.

Let λ_i be the packet arrival per unit time. Let T be the packet transmission time. Hence the traffic input to the system can be said to be A_i erlangs where

$$A_i = \lambda_i T \tag{3.1}$$

Let the packet retransmission per unit time be λ_r . The number of packets in the channel will be

$$\lambda_0 = \lambda_i + \lambda_r \tag{3.2}$$

The actual traffic in the channel can be said to be

$$A_0 = A_i + A_r \tag{3.3}$$

Where

$$A_r = \lambda_r T$$

The above can be modelled as in Figure (3.1)

$$A_0 = A_i + A_r = A_i + A_0 Pb$$

Where Pb is the probability of blocking.

$$A_i = A_0(1 - Pb) \tag{3.4}$$

If c frequency channels are available per mobile then Pb is given as [27] (it is assumed that the blocked calls are lost)

$$Pb = B(c, A_0) = \frac{A_0^c}{c!} / (\sum_{k=0}^{c} A^k / K!) \tag{3.5}$$

The traffic carried per frequency channel is

$$L = A_i / c \tag{3.6}$$

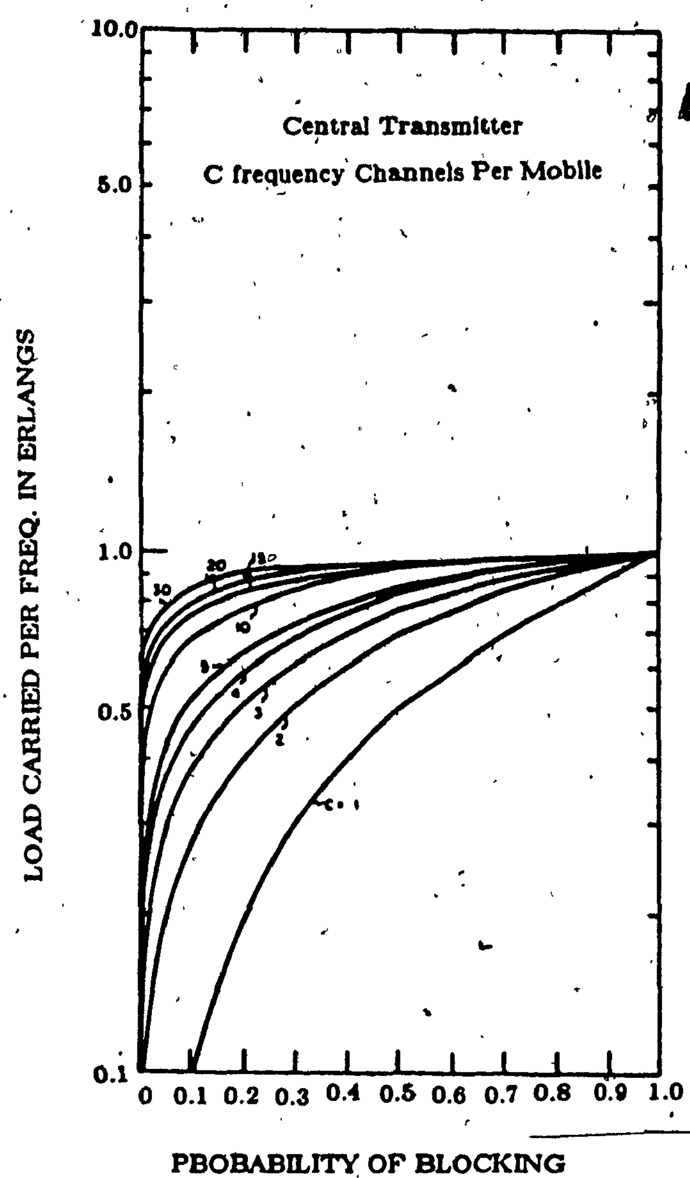


Figure 3.2 *(Source: Ref.[20])*

Load carried per frequency in erlangs for given probability of blocking.

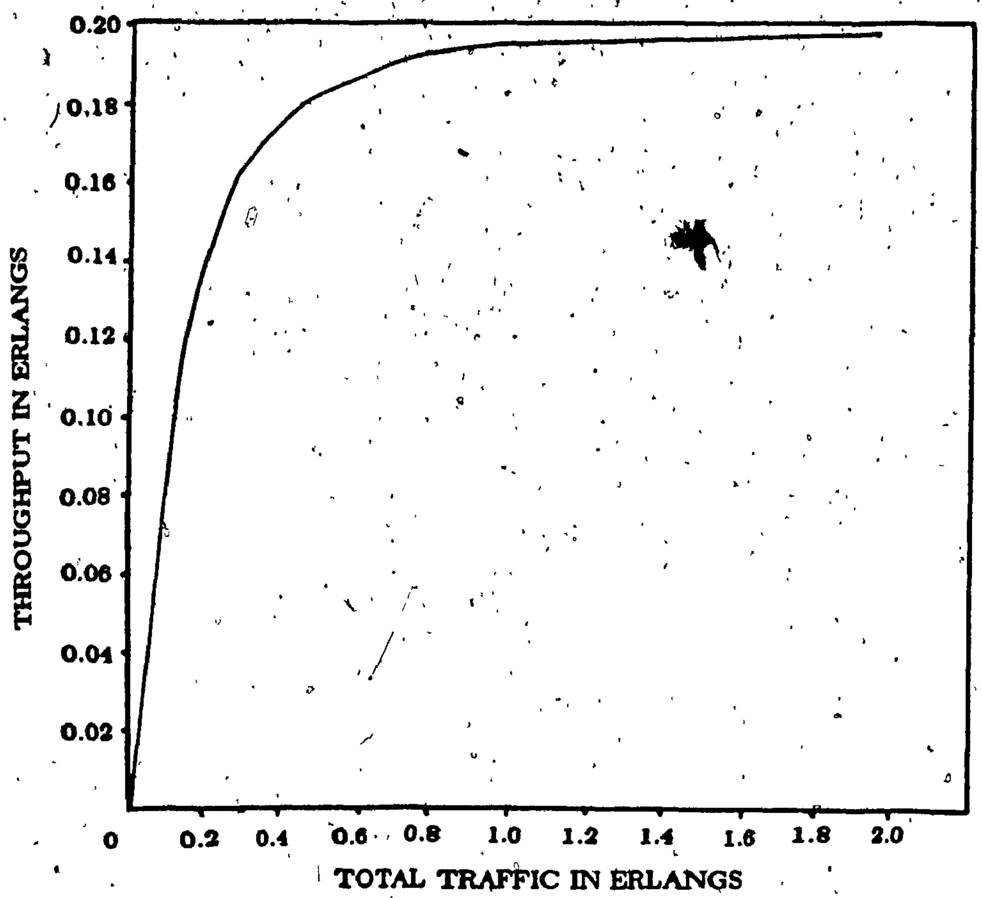


Figure 3.3

Throughput in erlangs for given total traffic in erlangs.

From Figure 3.2 the offered traffic A_0 and the throughput A_t are calculated and drawn as shown in figure 3.3.

3.3 Fixed channel assignment

In the fixed channel assignment[20], the base station of each cell is alloted a group of channels. The neighbouring cells are assigned different groups of channels. No channel belongs to more than one group. A same group of channels may be alloted to cells which are sufficiently seperated geographically so that transmissions from these cells do not interfere with one another. Each user transmitts only in some channel appropriate to his own cell.

In this particular scheme the total frequency available is divided into m seperate frequency groups each containing j frequency channels. A cell is assigned to one of the m frequency groups. Hence each base station can communicate with as many as j mobile units in its cell simultaneously using j different channels in its group. The number m is choosen as the smallest number that satisfies the interference buffering requirements.

The oblivious advantage of this scheme is that each base station is equipped to communicate on j of the mj frequencies rather than all mj. The disadvantage is that the mobile unit must be equipped with channels with groups of m to try any where in the city.

Based on the same assumptions as in the multifrequency central transmitter the analysis is given below:

If the number of frequency groups is m, c is the number of frequency channels over which to communicate then

$$c = im \tag{3.7}$$

where i is an integer.

Then the total traffic offered per zone is given as

$$a = A / N \quad erlangs \tag{3.8}$$

Where A is the total traffic intensity and is given as λr erlangs. N is the total number of zones and λ is the arrival of the initiations which form a poisson process of λ arrivals per second. r is the duration of the message with mean length r.

The probability of blocking as per Erlang B formula [27] is defined as

$$Pb = B(i, a) = B(c/m, A/N) \tag{3.9}$$

The traffic carried is defined as

$$A' = A(1 - Pb) = A(1 - B(c/m, A/N)) \tag{3.10}$$

The traffic carried per channel is

$$L = A' / c = A(1 - B(c/m, A/N))/c \tag{3.11}$$

From the Figure 3.4, the traffic offered and the throughput is calculated and Figure 3.5 shows the same. The Figure 3.6 shows the throughput as a function of the offered traffic for FH/QPR network as in chapter two.

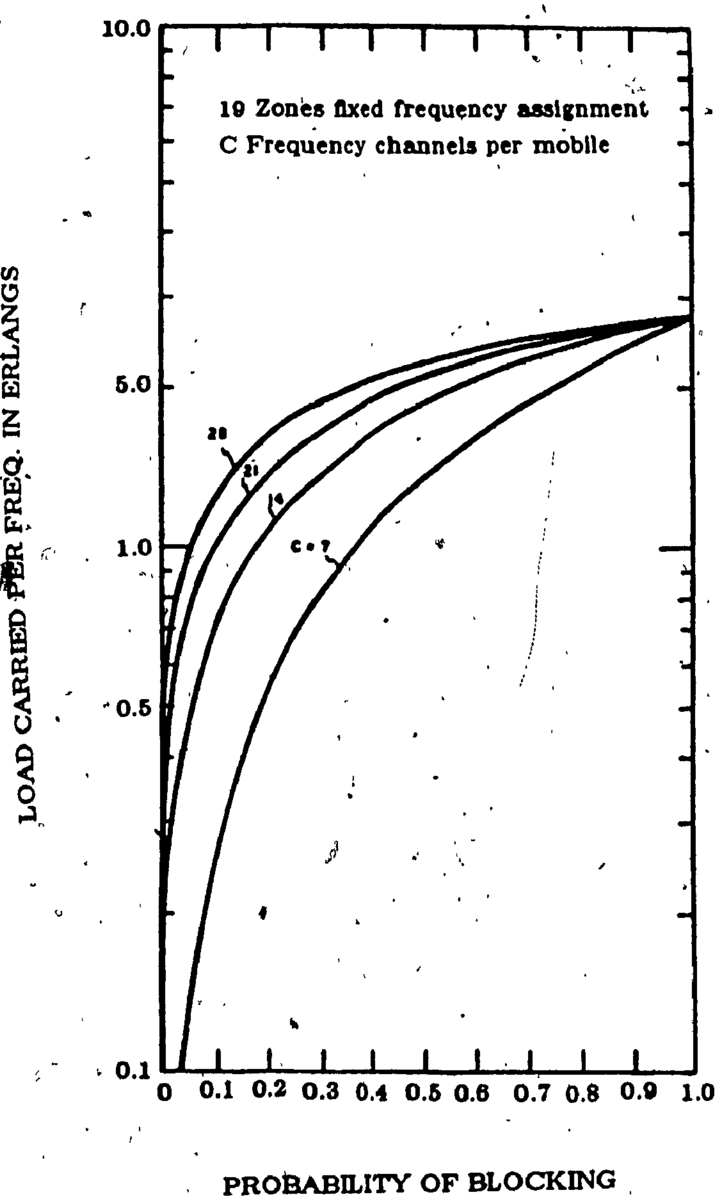


Figure 3.4 *(Source: Ref. [20])*

Traffic carried for fixed frequency plan

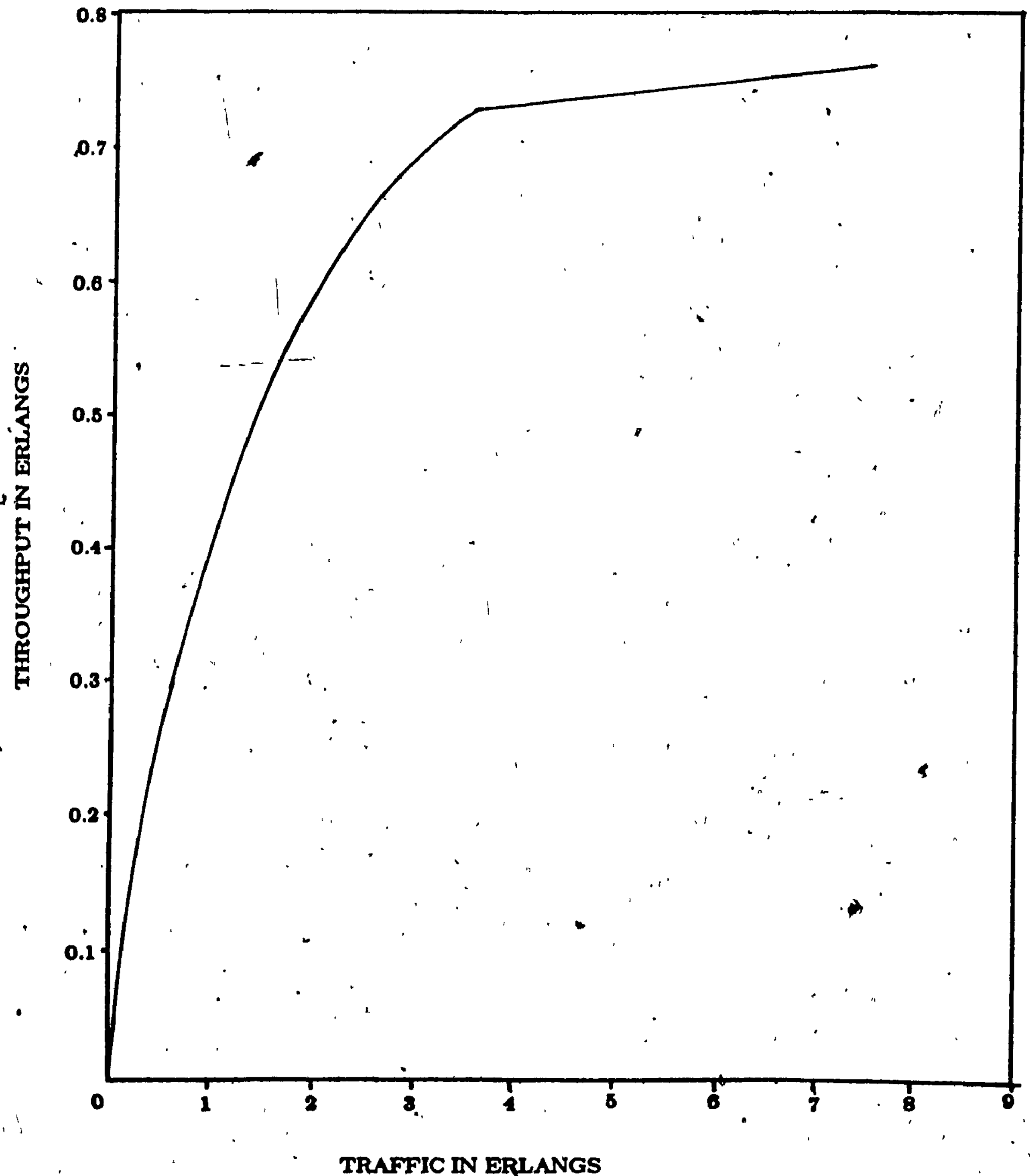


Figure 3.5

Throughput in erlangs for the given traffic in erlangs.

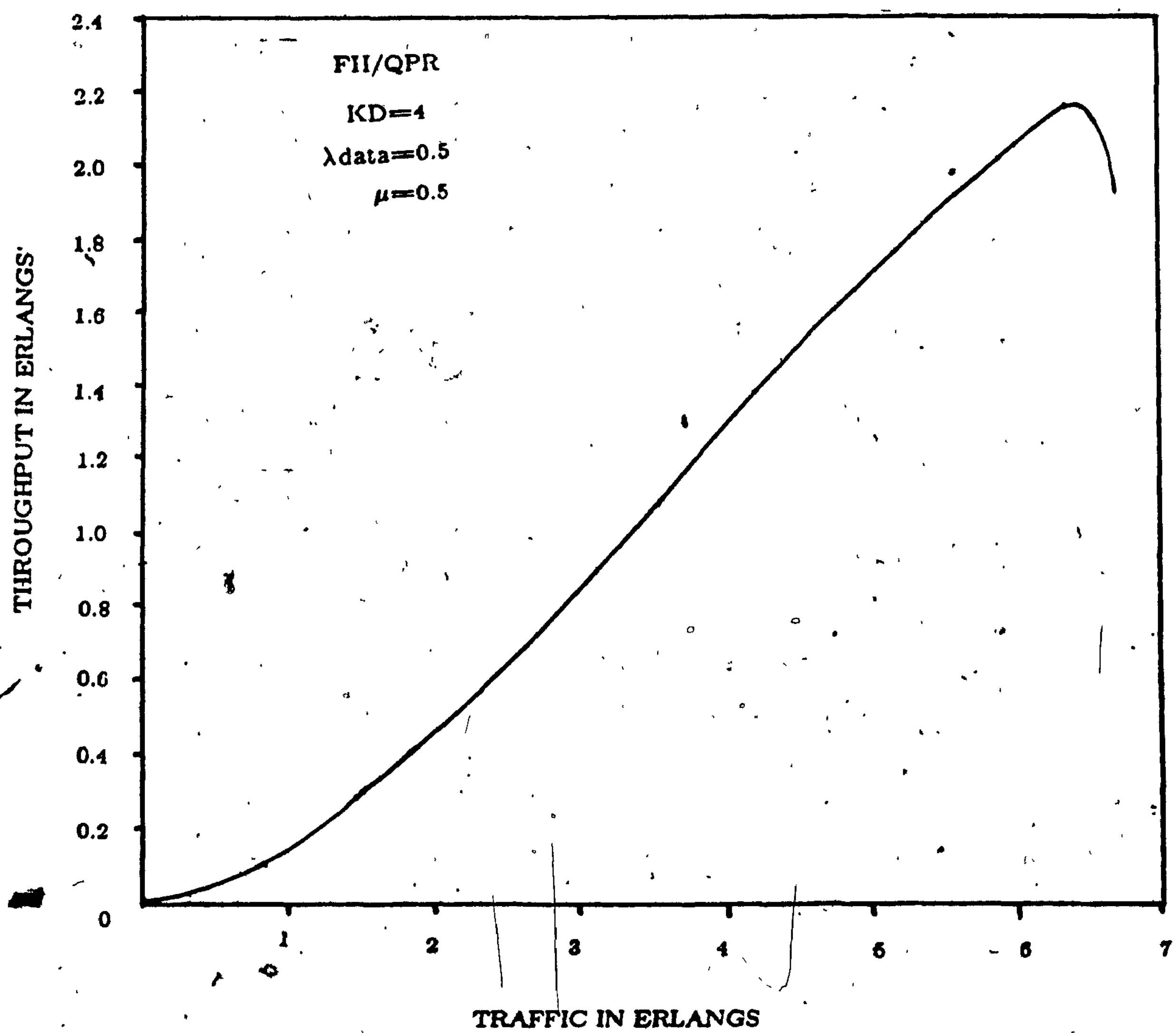


Figure 3.6

Throughput in erlangs for the given traffic in erlangs. FH/QPR, KD=4, λ data=0.5, μ =0.5.

3.4 Demand assigned multiple access fixed channel assignment scheme

In this type of assignment scheme [21], the users transmit messages in the form of packets in the random access ALOHA mode. Due to occasional destructive collision among packets and due to idle times on channels, the efficiency of the scheme is not very high.

One of the methods of accessing a message channel is to send a request for the message channel on a seperate request channel. If one request channel is dedicated to a base station pair then accessing a base station will not be a problem. The utilisation of the band width resources will be low as there will be large number of low duty factor users.

Another method is to use a demand access multiple access scheme with collision type request channels which are shared in random access in ALOHA type mode. A ready user randomly selects one of the request channels on which it transmits to a central controller a request for assignment of a message channel. The request contains the identification of the user and that of the base station by which the user may be served. In most of the standard Fixed channel assignment schemes this is the nearest base station. If the request is received correctly and at least one message channel is available then a message channel is assigned by the base station by sending a positive acknowledgement and the message channel number. Otherwise a request is again retransmitted after some random rescheduling delay.

Let P_f be the probability that a request fails to secure a message channel. The new requests in each cell are generated as per Poisson point process with rate Λ_{new}. If successive retransmission of a request are independent then the total request generation can be assumed as Poisson. Let Λ be the total request rate. Hence

$$\Lambda_{new} = \Lambda(1 - P_f) \tag{3.12}$$

With ALOHA type request channels, the probability that a request does not collide with requests from a particular cell is given by exp ($-2\,\Lambda\,\tau_s\;/\;\eta$ Ca). τ_s is defined as the request duration and η is 1 or 2 depending on weather it is unslotted or slotted type request channels being used. Ca is the number of request channels.

Considering the base station of a particular cell or in m cells around it may cause interference at the base station, but requests in any cell outside of the m belts cannot cause any interference. If T' is the number of cells that can cause interference at a base station, by hexagonal geometry it is clear that

$$T' = 1 + \sum_{i=1}^{m} 6i \quad or \tag{3.13}$$

$$T' = 1 + 3m(m+1) \tag{3.14}$$

The probability that a particular request does not collide is given by

$$Pnc = \exp(-2\lambda\tau_s T'/\eta Ca) \tag{3.15}$$

The rate at which the new requests are generated must be the same as the rate at which the circuits are assigned in steady state. If T_m is the average message holding time, the traffic carried by each base station is $\Lambda_{new}\,T_m$. If Cm is the number of message channels per base station, then the traffic carried per channel per base station is

$$\rho_m = \Lambda_{new}\,T_m / Cm \tag{3.16}$$

The system bandwidth utilisation can be defined as the average fraction of the system bandwidth that is busy for message transmission and is

$$S = \rho_m (1 - \upsilon_s) \tag{3.17}$$

Where

$$\upsilon_s = Ca.Ba/(Ca.Ba + N.Cm.Bm) \tag{3.18}$$

Ca is the number of request channels each having a bandwidth of Ba and Cm is the number of message channels each having a bandwidth of Bm. N is the number of message channels required in the system.

From (3.12),(3.16),(3.17) and using algebraic manupulation (3.15) can be written as

$$Pnc = \exp(-2.S.\,\phi_T\,.T'/(N\,\upsilon_s\,\eta(1 - P_f\,))) \tag{3.19}$$

Where ϕ_T is defined as $\tau\,/T_m$ and τ is the request channel duration if it were transmitted in a message channel.

The traffic carried by each base station is defined as

$$a_c = \rho_m\,.Cm = S.Cm\,/(1 - \upsilon_s\,) \tag{3.20}$$

If a_0 is the corresponding traffic offered, the probability of blocking is given as

$$Pb = B\,(Cm\,,a_0) \tag{3.21}$$

Where B(-,-) is given by the Erlang loss probability [27] as

$$B\,(s\,,a\,) = (a^s\,/S\,!)/\sum_{k=0}^{s}(a^k\,/K\,!) \tag{3.22}$$

The relation between the carried and the offered traffic is

$$a_c = a_0(1 - B\,(Cm\,,a_0)) \tag{3.23}$$

The probability of failure to get a message channel in a particular try is given by

$$P_f = 1 - (1 - Pb\,).Pnc \tag{3.24}$$

For a given S, P_f will be a function of υ_s. The value of υ_s is choosen such that P_f is minimum.

Figure 3.7 shows the minimised probability of failure as a function of band width utilization S. The Figure 3.8 shows the fractional allocation υ_s as a function of band width utilization S.

From Figure 3.8, the fractional allocation υ_s for the band width utilization S for five message channels is determined. From Figure 3.7, the minimised probability of failure corresponding to the band width utilization S is determined.

Substituting the values of S, υ_s in (3.17) ρ_m is determined. Normalising T_m to 1, Λ_{new} is determined from (3.16). Λ is obtained from (3.12). The throughput per channel as a function of the total input are plotted as shown in Figure (3.9).

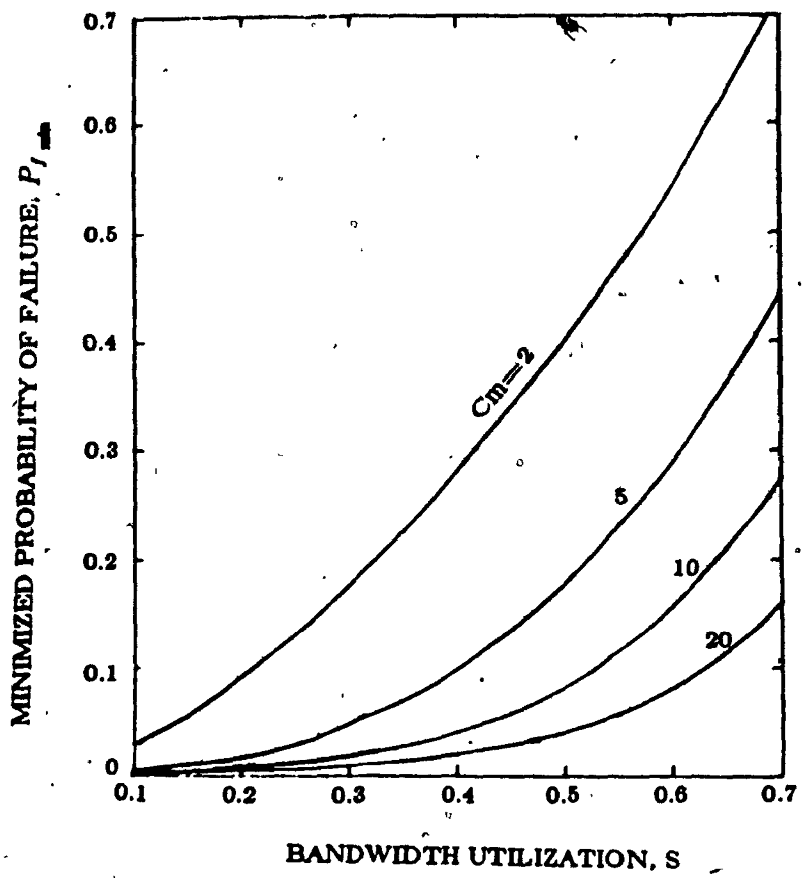


Figure 3.7 *(Source: Ref.[21])*

Minimised probability of failure $P_{f_{min}}$ as a function of bandwidth utilisation S. FCA scheme, ϕ_T =0.01, η =2, n=2, m=1, various Cm.

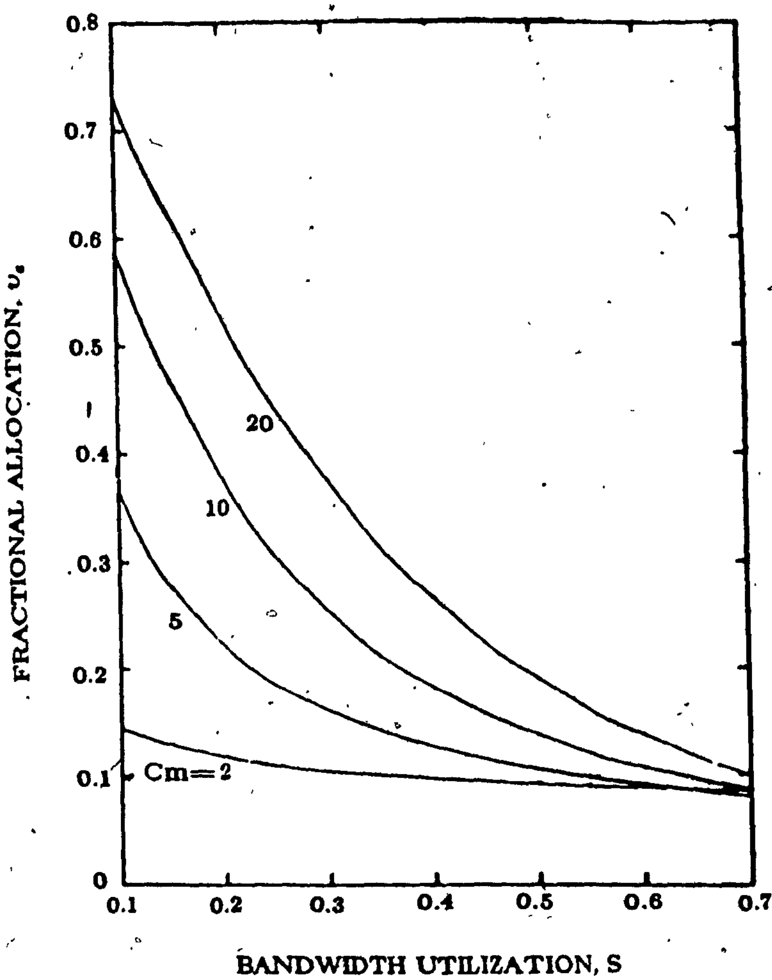


Figure 3.8 *(Source: Ref.[21])*

Fractional allocation υ_s that minimises the probability of failure P_f as a function of bandwidth utilization S. FCA scheme, $\phi_T = 0.01$, $\eta = 2$, n=2, m=1, various Cm.

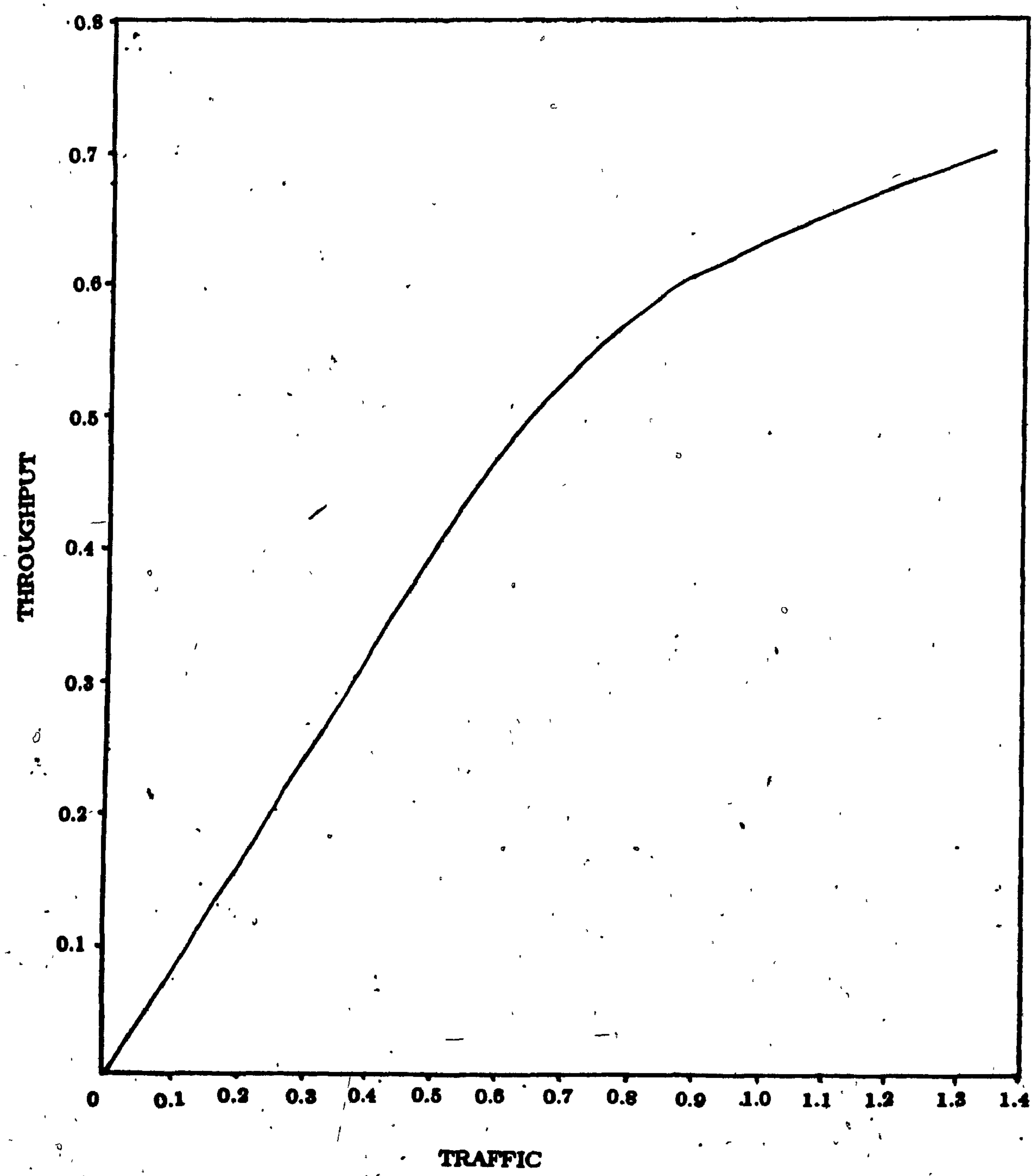


Figure 3.9

Throughput for the given traffic.

3.4.1 Voice and data type channels

Consider there are two types of messages: voice and data in the channel. For voice a moderate interference rejection is acceptable, where as for data a high interference rejection is needed. Let type A message represent data and type B message represent digitally encoded voice messages.

If during the system design, the interference rejection is taken into consideration capable of catering for type A messages, it will lead to low bandwidth utilisation and or high probability of failure. On the other hand if the system is designed to meet only the interference rejection criterion of type B messages then it will cause unacceptable interference to type A messages.

It has been suggested [21], in such a situation it would be better if type B messages be served by any of the nearest three base stations and type A messages by the nearest base station.

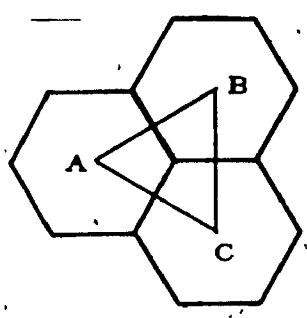


Figure 3.10 *(Source: Ref. [21])*
Contiguous cells configuration.

The Figure (3.10) shows the typical three contiguous cells with centre A,B,C. For any user within the triangle ABC, the base stations A,B,C will be near. The probability of getting a message channel in A or at B or at C is

$$1 - P_f = P(A \bigcup B \bigcup C)$$

$$1 - P_f = P(A) + P(B) + P(C) - P(AB) - P(BC) - P(CA) + P(ABC) \quad (3.25)$$

P(A) is the probability that a request succeeds in getting a message channel at A, P(AB) represents the probability that a request succeeds in getting message channels at A,B; P(A,B,C) represents the request channel succeeding in getting message channels at A,B and C. Similarly for P(B), P(C), P(BC) and P(CA). If the request rate is the same in all the cells, and by the symetry of the cellular structure

$$P(A) = P(B) = P(C) \quad (3.26)$$

$$P(AB) = P(BC) = P(CA) \quad (3.27)$$

Can be written as

$$1 - P_f = 3(P(A) - P(AB)) + P(ABC) \quad (3.28)$$

Proceeding in the same manner as before

$$P(A) = (1 - B(Cm, a_0)).exp(-2SPhi_T T_1'/(N v_s \eta(1 - P_f))) \quad (3.29)$$

$$P(AB) = (1 - B(Cm, a_0))^2.exp(-2S\phi_T T_2'/(N v_s \eta(1 - P_f))) \quad (3.30)$$

$$P(ABC) = (1 - B(Cm, a_0))^3.exp(-2S\phi_T T_3'/(N v_s \eta(1 - P_f))) \quad (3.31)$$

Where a_0 is the offered traffic to each base station. It can be obtained from (3.23).

T_l' represents the total number of cells containing l contiguous cells and m belts of cells around them for l=1,2 and 3. T_1' is the same as in (3.14).

$$T_2' = 2 + \sum_{j=1}^{m} (2 + 6j)$$

$$or \quad T_2' = (m + 1)(3m + 2) \quad (3.32)$$

$$T_3' = 3 + \sum_{j=1}^{m} (3 + 6j)$$

$$or \quad T_3' = 3(m + 1)^2 \quad (3.33)$$

Let α_1 be the fraction of new requests belonging to type A messages and α_2 be the rest belonging to type B messages. Hence

$$\alpha_1 + \alpha_2 = 1 \quad (3.34)$$

If $P_{f\,1}$ is the probability of failure for type A messages and $P_{f\,2}$ is the probability of failure for type B messages. If Λ_{new} and Λ are the new and total request rates per cell, then

$$\alpha_1.\Lambda_{new}/(1 - P_{f\,1}) + \alpha_2.\Lambda_{new}/(1 - P_{f\,2}) = \Lambda \tag{3.35}$$

Using this equation and proceeding as before, the probability of success of A and B type messages is

$$1 - P_{f\,1} = (1 - B(Cm\,,a_0)).exp\,(-2S\,\phi_T\,T'(\alpha_1/(1 - P_{f\,1}) + \alpha_{f\,2}/(1 - P_{f\,2}))/(N\,\upsilon_s\,\eta)) \tag{3.36}$$

$$1 - P_{f\,2} = 3(P(A) - P(AB)) + P(ABC) \tag{3.37}$$

$$P(A) = (1 - B(Cm\,,a_0).exp\,(-2S\,\phi_T\,T_1'(\alpha_1/(1 - P_{f\,1}) + \alpha_2/(1 - P_{f\,2}))/(N\,\upsilon_s\,\eta)) \tag{3.38}$$

$$P(AB) = (1 - B(Cm\,,a_0)^2\exp(-2S\,\phi_T\,T_2'(\alpha_1/(1 - P_{f\,1}) + (\alpha_2/(1 - P_{f\,2}))/(N\,\upsilon_s\,\eta)) \tag{3.39}$$

$$P(ABC) = (1 - B(Cm\,,a_0)^3.exp\,(-2S\,\phi_T\,T_3'(\alpha_1/(1 - P_{f\,1}) + (\alpha_2/(1 - P_{f\,2}))/(N\,\upsilon_s\,\eta)) \tag{3.40}$$

Solving (3.36), (3,37) $P_{f\,1}$ and $P_{f\,2}$ are determined which are functions of υ_s. υ_s is choosen such that the linear combination of $P_{f\,1}$ and $P_{f\,2}$ are minimised. The overall probability of failure C_f is

$$C_f = \alpha_1.\beta_1.P_{f\,1} + \alpha_2.\beta_2.P_{f\,2} \tag{3.41}$$

Where β_1 and β_2 are weighting factors. C_f is sealed $0 < C_f < 1$. Let β_1 and β_2 be such that

$$\alpha_1.\beta_1 + \alpha_2.\beta_2 = 1 \tag{3.42}$$

The figure (3.11) shows the probability of failures of type A messages and type B messages and also the overall probability of failure with respect to the fractional allocation υ_s.

The figure (3.12) shows the bandwidth utilisation and the fractional allocation υ_s. The throughput is calculated as described before and Figure (3.13) is drawn.

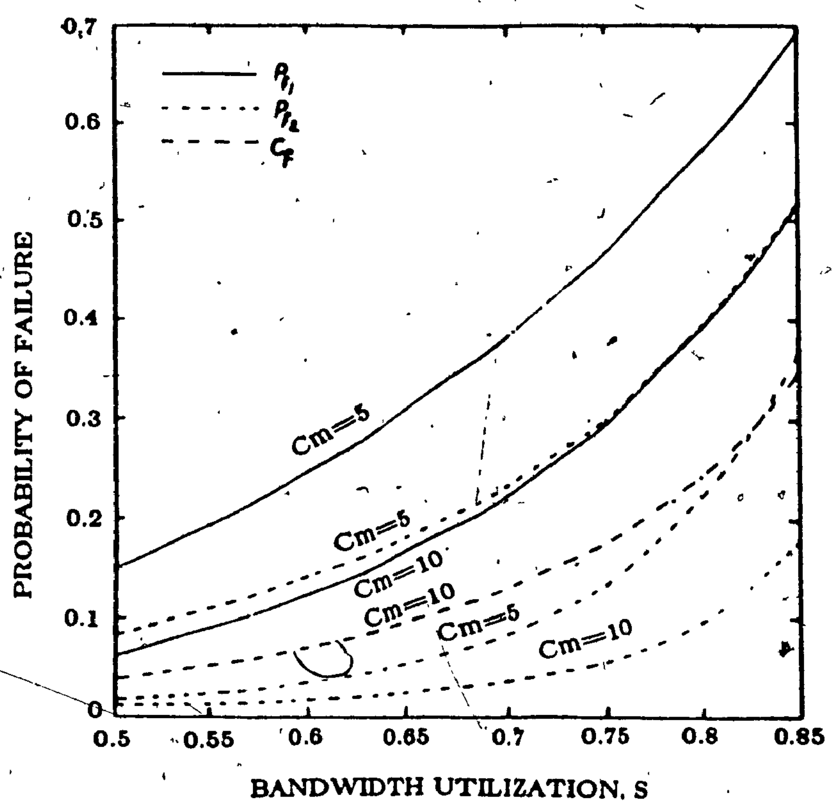


Figure 3.11 *(Source: Ref.[21])*

Probability of failures P_{f1} and P_{f2} for the two types of messages and the overall probability of failure C_f when the latter is minimized with respect to the fractional allocation υ_s. Mixed scheme, $\phi_T = 0.01$, $\eta = 2$, n=3, m=1, $\alpha_1 = \alpha_2 = 0.5$, $\beta_1 = \beta_2 = 1$, various Cm.

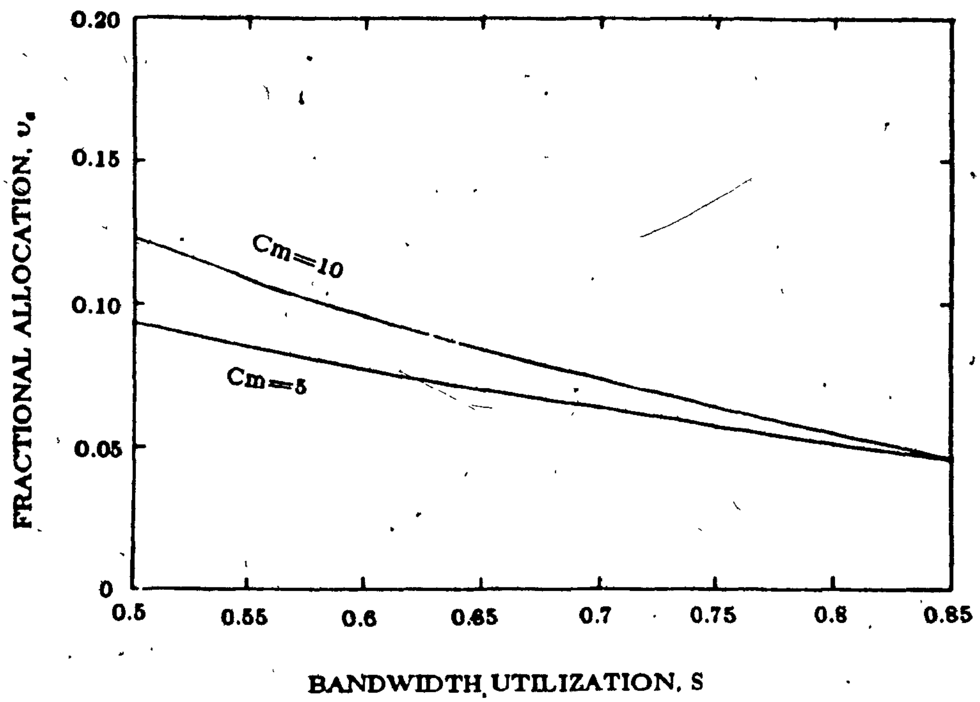


Figure 3.12 *(Source: Ref.[21])*

Fractional allocation υ_s that minimizes overall probability of failure C_f . Mixed scheme, $\phi_T = 0.01$, $\eta = 2$, n=3, m=1, $\alpha_1 = \alpha_2 = 0.5$, $\beta_1 = \beta_2 = 1$, various Cm.

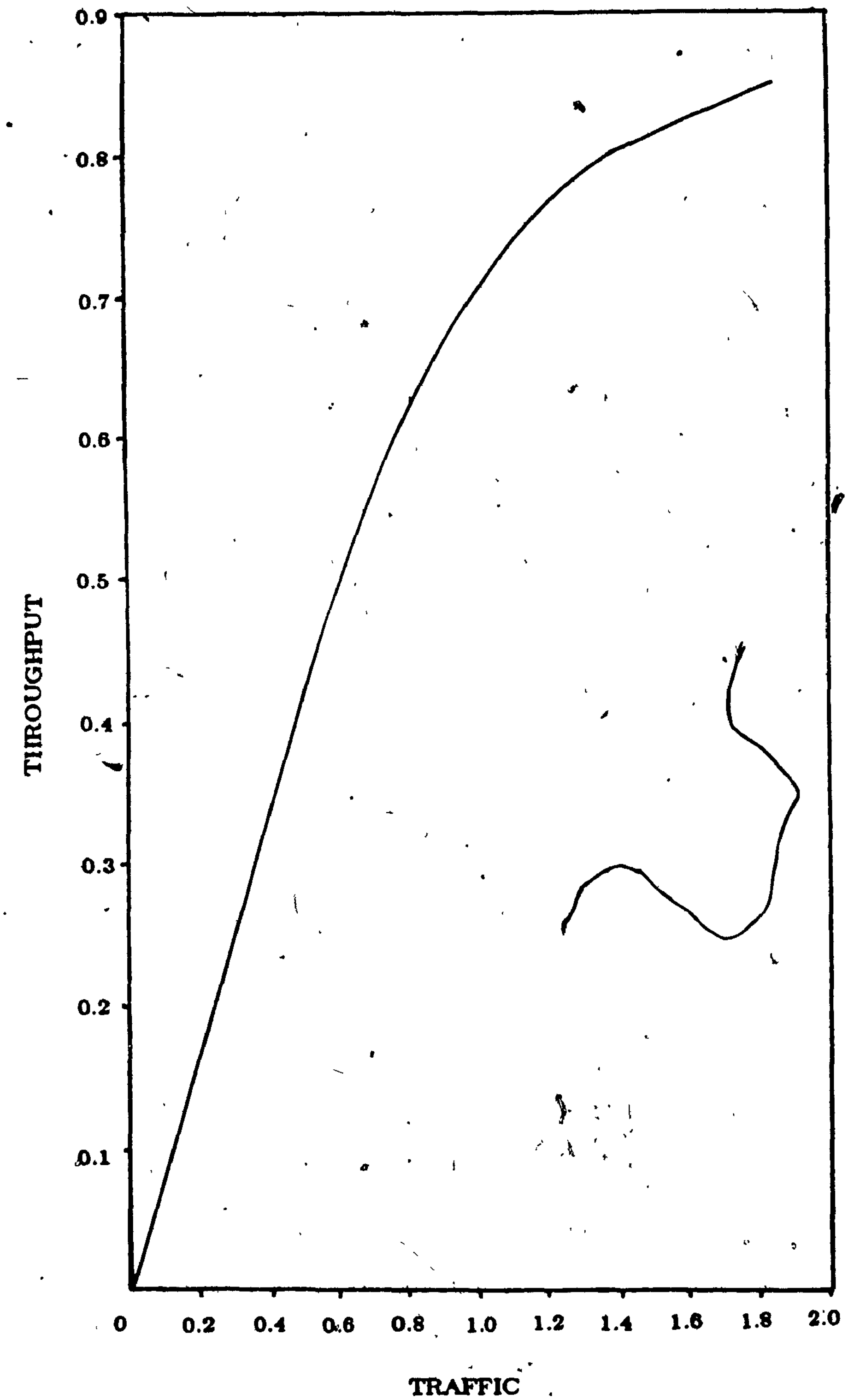


Figure 3.13
Throughput for the given traffic.

3.5 Dynamic Channel Assignment

In the dynamic channel assignment[22], all channels are kept in a central pool and any channel can be used by any cell. This system takes advantage of the fact mobile unit and the base station communication and control is effected through the central processing unit on which all dispatchers and base stations terminate. The central processor keeps track of frequencies in use in the base stations and to what mobile units and it assigns channels on dynamic basis. The processor allows a channel in use in one cell to be used simultaneously in another cell in the system if the seperation distance between the two cells is greater than a minimum distance to avoid co channel interference, Figure (3.14).

For each system cycle, each subsriber who is either 'on' the system or is attempting a call is checked for his activity status.

New call attempts are processed along the centre branch of the flow chart as shown in Figure (3.15). The first step in processing an attempt is to determine which base station should be used to serve the call. The base station nearest to the subscriber has the strongest received signal from the subscriber and is the only one with the adequate signal strength to serve.

A radio channel which satisfies some channel assignment criteria is now processed. The channel search is done channel by channel starting from channel 1.

If the particular channel is being used at the base station to be assigned, then the channel is rejected and the next channel is checked. If the channel is not in use at that base station, a check is made if the channel in use at a base station closer than the reuse interval away from the base station to be

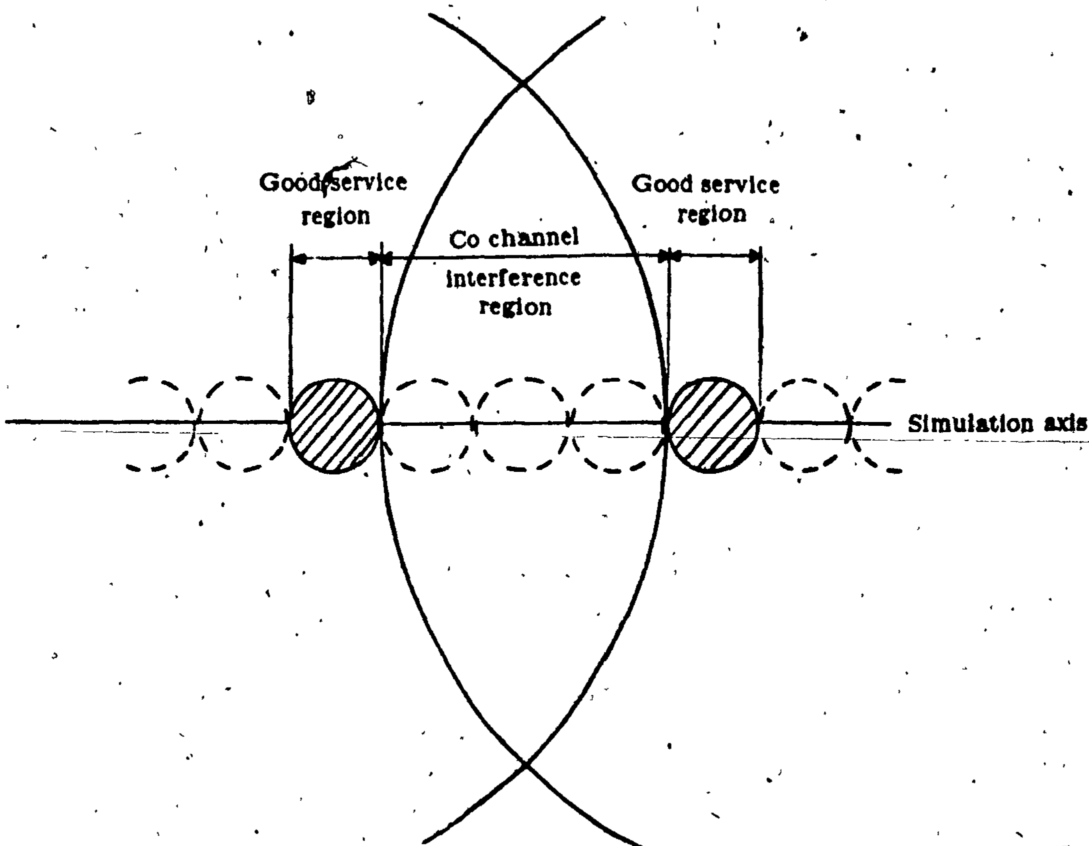


Figure 3.14 *(Source: Ref.[22])*

Illustration of channel reuse.

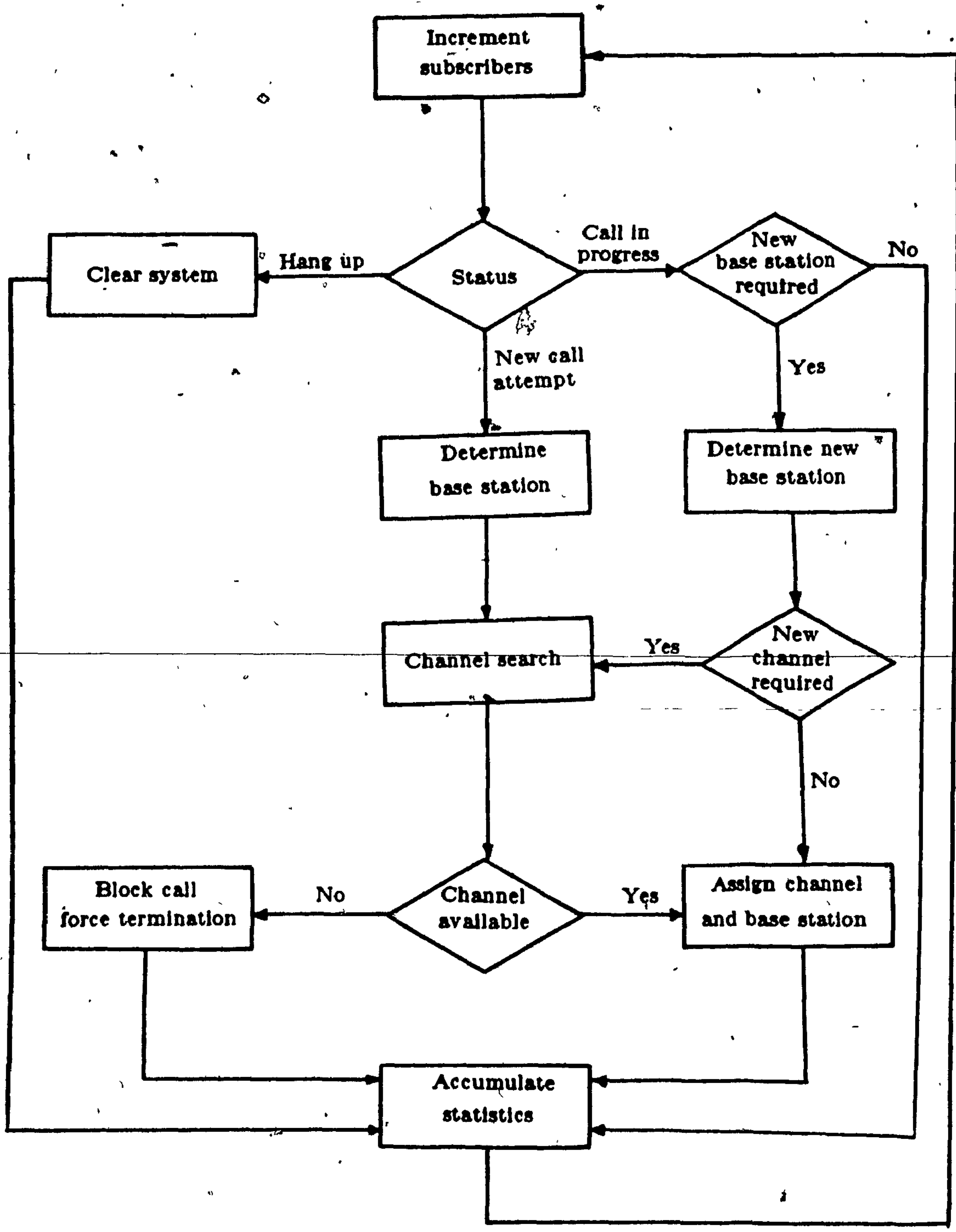


Figure 3.15 *(Source: Ref.[22])*

Flow chart of channel allotment.

assigned. If the channel is in use, the channel is rejected and the next channel is checked starting again at the begining of the channel check procedure. If the channel is not in use at any base station less than the reuse interval from the base station to be assigned, then further checks on that channel are made and it is compared with other channels, if any, which have met this reuse criterion. If during the channel search, a channel is found which is not in use at any base station in the system, the channel is alloted and the channel search is terminated. Otherwise, the channels which meet the reuse criteria are compared to find the most desirable one.

If after checking all radio channels in the system, no available channel is found, the call attempt is refused service and the call is blocked. The blocked calls are removed from the system.

For calls in progress, the suitability of the current base station assignment is checked. This is done to see if the vehicle has moved to some other base station than the one serving it and has a stronger signal at the other base station. If it has not moved into such a situation, then its current base station and channel assignment are acceptable and no further action is taken. If however, the vehicle has moved closer to another base station, that closer station must be used to serve the call. As shown in the flow chart Figure 3.15, the assigned channel is checked at the new base station to be used and at all the base stations within a reuse interval of the new base station. If the channel satisfies the conditions, the new base station and the old channel are assigned to serve the call and the old base station is cleared off the call. If the old channel is in use within a reuse interval of the new base station, then a new channel search is initiated which is same as new call attempts. If a subsititute channel is not found, the call is forced to terminate.

Channel number \ Base station	1	2	---	k-7	k-6	k-5	k-4=k-D_R	k-3	k-2	k-1	k	k+1	k+2	k+3	k+4=k+D_R	k+5	k+6	k+7
1							X				X					X		
2									X						X			
⋮																		
i				X													X	
i+1				X											X			
i+2						X												X
i+3						X										X		

Figure 3.16 *(Source: Ref.[22])*

The channel base station matrix. X indicates channel in use at base station. k is the assigned base station.

For call terminations, the channel is cleared at the assigned base station.

In the computer simulation [22], the base station activity is stored in a two dimensional matrix M(i,k), where i denotes the channel number, and k denotes the base station number. The matrix is shown in Figure (3.16).

For example if the base station k is assigned, then neither channel 1 or 2 may be assigned as neither meet the criteria discussed before. Some of the assignment strategies are discussed below and the results as simulated [32] are presented.

3.5.1 First Available

In this assignment strategy, the first available channel is assigned during the channel search as described before. This is the simplest and the cheapest strategy to implement but may not be the optimum assignment strategy.

3.5.2 Mean Square (MSQ)

In this assignment scheme, the channel activity is checked at base stations in between one and two reuse intervals, i.e. if D_R is the allowable reuse interval then the activity between D_R and $2D_R$ away from the assigned base station.

This is done as the channel in use a distance greater than $2D_R$ away would allow the use of that channel at another base station within that distance. The mean square assignment strategy minimises the quantity

$$1/n \sum_{j=1}^{n} D_i^2 \qquad D_R < D_j < 2D_R$$

D_j is the distance between the assigned base station and the base station using the channel within the specified interval and n is the number of base stations using the channel within this interval. If n=0 for some channels, it means that the channel is not in use between D_R and $2D_R$ on either side of the assigned base station, then the first such channel encountered is assigned to serve the call.

3.5.3 Nearest Neighbour (NN)

In this type of scheme, the channel that is in use at a base station nearest to the assigned base station, but still at least a reuse interval D_R away is assigned. It minimises the distance D over the available channels where D is the distance to the first base station. If more than one channel has thd same minimum D, then the first channel encountered is assigned withoud regard to the distance to the first base station using that channel in the opposite direction.

3.5.4 Nearest Neighbour + 1 (NN + 1)

This assignment strategy is similar to the nearest neighbour strategy except that it finds the minimum D for the available channels with

$$D > D_R + 1$$

IF such a channel does not exist, then a channel with D=D_R is assigned. This allows more callers to keep to their assigned channel when they cross a

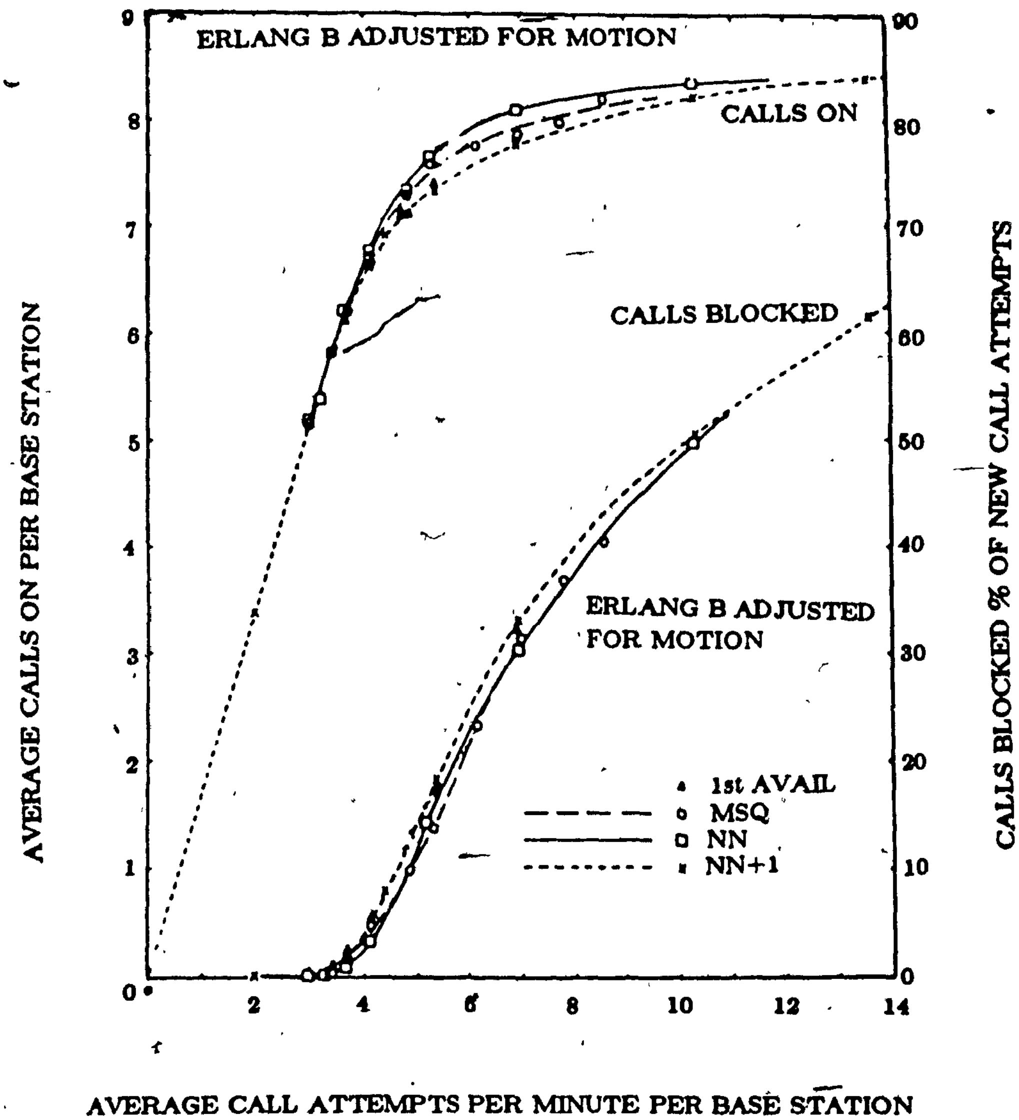


Figure 3.17 *(Source: Ref.[32])*

Performance of 40 channel mobile radio system.

radio coverage cell boundry.

The Figure (3.17) shows the average call attempts and the average calls on a base station for the various methods described.

3.6 Conclusions

Comparing the throughputs for the Multifrequency central transmitter and Fixed channel assignment schemes (Figures 3.3 and 3.5) to the throughput of the proposed scheme shown in Figure 3.6, it is seen that the throughput is higher in Figure 3.6.

In the case of Demand assigned multiple access fixed channel assignment scheme the throughput as a function of the offered load is shown in Figure 3.9. For voice and data type channels, the throughput as a function of the offered traffic is shown in Figure 3.13. Comparing these to the Figure 3.6 it is seen that the proposed scheme is comparable to the other schemes.

The Dynamic channel assignment scheme has been described. The proposed scheme is better as it facilitates the transmission of voice and data.

CHAPTER FOUR

QUEUEING/ERROR ANALYSIS OF A MOBILE NETWORK EMPLOYING FREQUENCY HOPPING MFSK

4.1 Introduction

In this chapter, an noncoherent detected FH/MFSK signal is considered for baseband modulation for voice or data in the mobile network as described in chapter two.

The analysis to evaluate the bit error probability for the noncoherent, FH/MFSK signal is given.

The packet length is assumed to be 255 bits.

4.2 Analysis

The analysis of the mobile network is done in the same manner as described in chapter two.

Assuming in general MFSK data modulation of size M, the bandwidth of this data set is given as

$$W_d = M.Rd/(Log_2.M) \tag{4.1}$$

The SS interference rejection capability expressed as the processing gain is

$$PG = \frac{W_{ss}}{W_d} = W_{ss}.Log_2.M/(M.Rd) \tag{4.2}$$

Rd is the data rate.

At the intended receiver, Forward error correction (FEC) will correct a few random errors of each packet (255 bits). The code selected is (255, 128) as given in [10] and the minimum distance will be 19.

The conditional bit error probability P(e/Ks,n) is given later. Ks is total number of actually interfering tones lying within the data bandwidth of the applicable receiver and n is the number of packets.

The probability of making a decoding error per packet of 255 bits is given as

$$P(Correctpacket/Ks;n) = \sum_{j=0}^{9} \binom{255}{j} (P(e/Ks,n))^j (1-P(e/Ks,n))^{255-j} \qquad (4.3)$$

Where error randomness is assumed.

Averaging over (n,Ks) in the similar manner described in chapter two, the final average of packet success is given as

$$Ps = \sum_{n=0}^{Md+Mo} \sum_{Ks=0}^{n} P(Correctpacket/Ks.n).P(Ks/n).P(n) \qquad (4.4)$$

P(Ks/n) and P(n) are the same as given in chapter two.

The coding rate is 1/2 and hence the effective useful throughput becomes

$$SSS = (1/2)(1/PG)(Ps.G) \qquad (4.5)$$

Where the processing gain PG, and the packet success Ps is given in (4.2) and (4.4) respectively.

4.3 Analysis for bit error of noncoherent detected MFSK signals.

The conditional bit error probability P(e/Ks,n) for a noncoherent detected FH/MFSK signal will be computed at one of the many receivers of the applicable base station. As a first order analysis, it is possible to model the multi access interference as a narrow band multitone jamming[4],[6] thus providing a simplified yet worst case probability of bit errors (upper bound). It has been observed that applying worst case jamming results to multiaccess envoirnment have yeilded unreasonably poor results. Moreover, these loose upper bounds leads to unstable networks, not consistent with the graceful degradation the

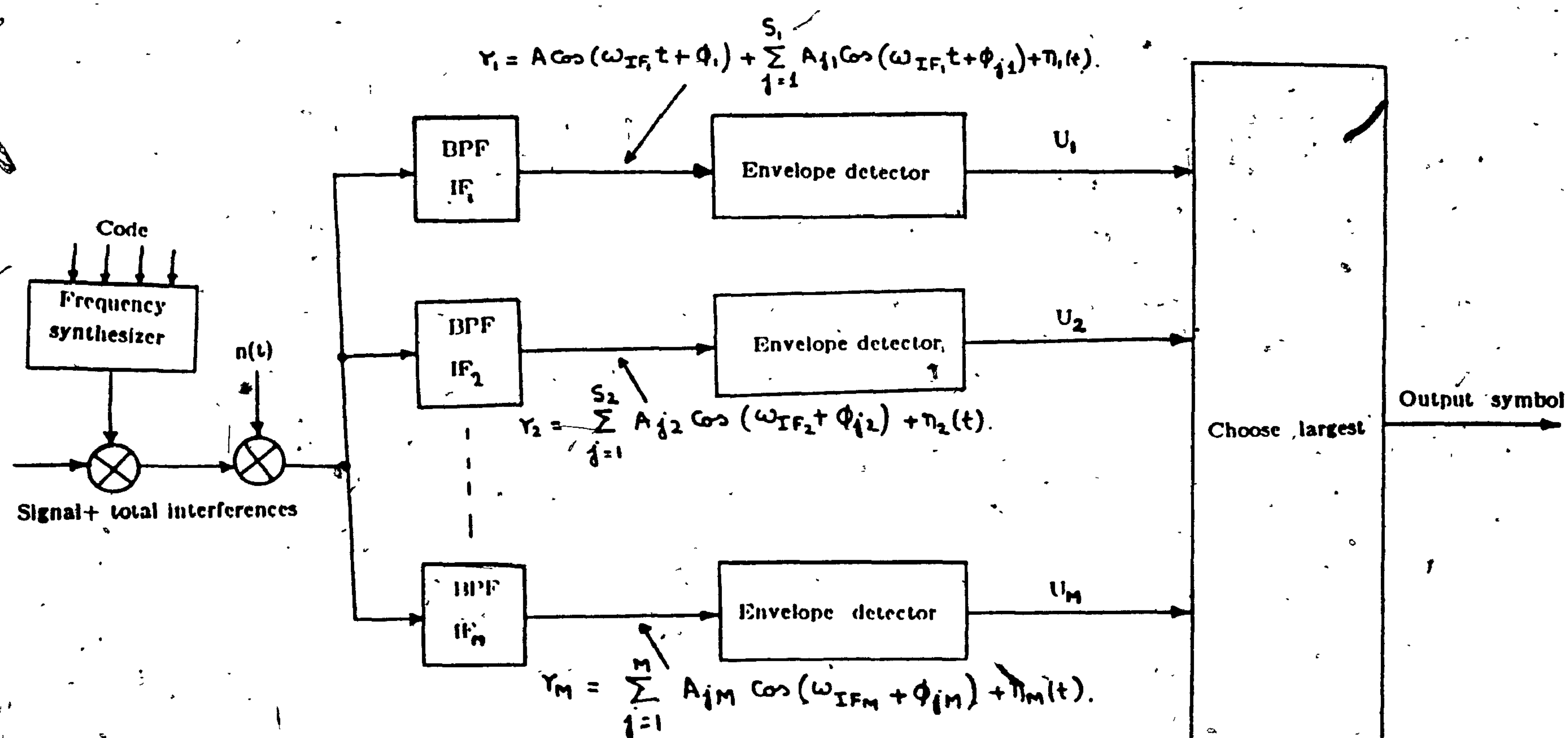


Figure 4.1

Envelope detection of MFSK signals following dehopping at one of the M base station receivers

Current symbol transmitted is assumed in BPF(IF_1)

$S_1+S_2+\ldots+S_M=Ks=$ total number of interference tones in the M-ary receiver leads to unstable networks, not consistent with the graceful degradation the spread spectrum networks should achieve as we increase the load. The following analysis assumes that in a certain packet time, there are n total number of packets, Ks is the total number of actually interfering tones lying within the data bandwidth of the applicable receiver.

Figure 4.1 shows the sequence of events taking place at one of the base station receivers following dehopping. Each of the (S_1+S_2+ ......$+S_M$ = Ks) like user sinusoidal interferences has a random phase ϕ uniformly distributed in the range $(0,2\pi)$. The receiver AWGN will reflect itself finally in noises$n_1,n_2,....$, at the output of each IF matched filter (BPF) with variance σ_n^2 and the thermal signal to noise ratio is defined as

$$SNR_{th} = A_i^2/(2\sigma_n^2) \tag{4.6}$$

Where A is the intended symbol amplitude

More~~over most analy~~sis in the literature have neglected these thermal like effects which is a flaw since SNR may drop below the believed 10 or 20 db in the SS case (at the M ary demoulator). The total number of interference tones Ks is automatically randomly divided among the M band pass filters and as a start, a slow frequency hopping equal to the symbol rate is assumed. It is assumed that all noises n_1 (t),n_2 (t), ..., and random phases ϕ_1, ϕ_{j1}, ϕ_{j2} are mutually uncorrelated and all noises have zero mean and variance σ_n^2.

The following analysis starts by finding the conditional symbol correct detection probability given S_1,S_2,S_3, (i.e. Ps(C|S_1,S_2,....)), then the conditioning is removed by summing over all possible values of S_1, S_2, ..., such that (S_1 + S_2 + ...= Ks).

This Ps (C| S_1, S_2,....) is well known to be

$$Ps(C \mid S_1,S_2,....)=\int_0^{\infty} Pr(\overline{U_2<U_1,U_3<U_1,....,U_m<U_1} \mid U_1=u_1)P(u_1)du_1 \tag{4.7}$$

Where U_1, U_2,.... are the decision variables following envelope detection and sampling and $P(u_1)$ is the probability density function of the decision variable. These are functions of the number of interferences and are mutually independent, so,

$$Ps(C \mid S_1, S_2, \ldots, S_m) = \int_0^{\infty} Pr((U_2 \mid S_2) < (U_1 \mid S_1)) \ldots Pr((U_m \mid S_m) < (U_1 \mid S_1))$$
$$*P(U_1 \mid S_1 = u_1) P(u_1) du_1 \qquad (4.8)$$

In the following, to ease the evaluation of (4.8) by taking a receiver which compares the first bank to the remaining (M-1) signals not containing the signal. This leads to sub optimum receiver and symbol error probability obtained will serve as an upper bound. To evaluate the integral part of (4.8), the condition of u_1 is removed and multiply the probability with the integrand thus obtain Ps(C|....). To evaluate the probabilities within the integrand of (4.8), the distribution of U_1 is found, U_2, U_3,...... U_m.

The signal containing the transmitted symbol (following BPF_1) is given by,

$$r_1 = A_1 cos\,(\omega_{IF_1} + \phi_1) + \sum_{j=1}^{S_1} A_{j1} \cos(\omega_{IF_1} + \phi_{j1}) + n_1(t)$$

The first term represents the desired signal of amplitude A_1 and random phase of angle ϕ_1 distributed between 0 and 2π. The second term represents the sum of S_1 statistically independent pulsed sinusolds, j^{th} member having an amplitude of A_{j1} and random phase of ϕ_{j1}. The last term is additive gaussian noise.

$$r_1 = E_1 cos\,(\omega_{IF} + \theta_1) + n_1(t)$$

The first term represents the sum of all statistically independent interfering pulsed sinusolds each having a random phase angle distributed between 0 and 2π.

Since the signals are matched filter, envelope detected, it is conveinient to write the above as,

$$r_1 = U_1 \cos(\omega_{IF} t + \psi_1) \tag{4.9}$$

Where U_1 represents the envelope of a signal consisting of a fixed amplitude A_1 plus gaussian noise. The conditional $P(U_1(S_1))$ can be computed from,

$$P(U_1/S_1) = \int_0^{A_1 + \sum_{j=1}^{S_1} A_{j1}} P(U_1/E_1,S_1) P(E_1/S_1) dE_1 \tag{4.10}$$

The density $P(U_1/E_1,S_1)$ is well known to be Ricelan distributed as

$$P(U_1/E_1,S_1) = \frac{U_1}{\sigma_n^2} e^{-(U_1^2 + E_1^2)/2\sigma_n^2} I_o\left(U_1 \frac{E_1}{\sigma_n^2}\right) \tag{4.11}$$

$0 < U_1 < \infty$

the second pdf of (4.10) is given in [11] and is

$$P(E_1/S_1) = \int_0^\infty X E_1 J_0(XE_1) \prod_{j=1}^{S_1} J_0(A_{j1}X) J_0(A_1 X) dX \tag{4.12}$$

Substituting (4.12), (4.11) into (4.10) and exchanging the order of integration, we get,

$$P(U_1/S_1) = (U_1/\sigma_n^2) e^{-U_1^2/2\sigma_n^2} \int_0^\infty X \prod_{j=1}^{S_1} J_0(A_{j1}X) J_0(A_1 X)$$

$$*\left(\int_0^{A_1 + \sum_{j=1}^{S_1} A_{j1}} E_1 e^{-E_1^2/2\sigma_n^2} J_0(E_1 X) I_0(U_1 E_1/\sigma_n^2) dE_1 \right) dX \tag{4.13}$$

The integrand of the inner integration becomes so small for large values of

$$E_1 > (A_1 + \sum_{j=1}^{S_1} A_{j1})$$

so the limit can approximately set at ∞ in which case the integration yeilds,

$$\sigma_n^2 e^{-\sigma_n^2/2(X^2 - U_1^2/\sigma_n^4)} J_0(XU_1) \tag{4.14}$$

Substituting (4.14) in (4.13), we get,

$$P(U_1/S_1) = U_1 \int_0^\infty X e^{-x^2\sigma_n^2/2} \prod_{j=1}^{S_1} J_0(A_{j1}X) J_0(A_1 X/\sigma_n) J_0(U_1 X) dX \tag{4.15}$$

$$P(U_1'/S_1) = U_1' \int_0^\infty y e^{-y^2/2} J_0(U_1' y) \prod_{j=1}^{S_1} J_0(A_{j1} y/\sigma_n) J_0(A_1 y/\sigma_n) dy \tag{4.16}$$

Similarly for the i^{th} receiver bank not containing the signal it is easy to see that,

$$P(U_i'/S_i) = U_i' \int_0^\infty z e^{-z^2} \prod_{j=1}^{S_i} J_0(A_{ki} Z/\sigma_n) J_0(U_i' Z) dZ \tag{4.17}$$

The conditional probability of making a subdecision error $(U_i' > U_1')$ in (4.8) is given by,

$$P(U_i' > U_1'/S_1, S_i) = \int_0^\infty \int_{U'}^\infty P(U_1'/S_1) P(U_i'/S_i) dU_i' dU_1'$$

$$P(U_i' > U_1'/S_1, S_i) = \int_0^\infty \int_0^{U_i'} P(U_1'/S_1) dU_1' P(U_i'/S_i) dU_i' \tag{4.18}$$

Substituting (4.17), (4.16) into (4.18) exchanging the order of integration and reducing, we ~~get,~~

$$P(U_i' > U_1'/S_1, S_i) = \int_0^\infty dU_i' \int_0^\infty U_i' e^{-y^2/2} J_1(U_i' y) \prod_{j=1}^{S_1} J_0(A_{j1} y/\sigma_n)$$

$$J_0(A_1 y/\sigma_n) dy . \int_0^\infty z U_i' e^{-z^2/2} \prod_{K=1}^{S_i} J_0(A_{ki} z/\sigma_n) J_0(U_i' z) dz \tag{4.19}$$

Integrating and collecting all terms in U_i we get,

$$-\int_0^\infty J_1(yU_i') z U_1^{-2} J_0(zU_i') dU_i' = \delta_z'(z-y) \tag{4.20}$$

$$P(U_i' > U_1'/S_1, S_i) = -\int_0^\infty e^{-y^2/2} \prod_{j=1}^{S_1} J_0(A_{j1} y/\sigma_n) J_0(A_1 y/\sigma_n)$$

$$d(e^{-y^2/2} \prod_{K=1}^{S_i} J_0(A_{Ki} y/\sigma_n)/dy) dy \tag{4.21}$$

Assuming all interferences have the same power as the signal, i.e,

$$A_{ki} = A_{ji} = A_1 \tag{4.22}$$

and performing integration by parts, we get,

$$P(U_i' < U_i'/S_1,S_i) = 1/2 + (A_1/\sigma_n)(S_1-S_{i+1})/2\int_0^{\infty} e^{-z^2} J_0^{S_1+S_i}(A_1 z/\sigma_n) J_1(A_1 z/\sigma_n)dz \quad (4.23)$$

The correct symbol detection probability is now upper bounded by

$$\overline{Ps}(C/S_1,S_2,....) = \prod_{i=2}^{M} P(U_i' < U_1'/S_1,S_i) \quad (4.24)$$

On the other hand if one wishes to compute the exact symbol probability, using soft decisions from (4.18), (4.19)

$$P(U_i' < U_1'/U_1'=\beta S_1,S_i) = \int_0^{\beta} U_i' \int_0^{\infty} z e^{-z^2/2} \prod_{K=1}^{S_i} J_0(A_{ki} z/\sigma_n) J_0(U_i' z) dz dU_i'$$

$$P(U_i' < U_1'/U_1'=\beta,S_1,S_i) = \beta\int_0^{\infty} e^{-z^2/2} J_1(\beta z) \prod_{K=1}^{S_i} J_0(A_{ki} z/\sigma_n) dz \quad (4.25)$$

Substituting (4.25), (4.16) into (4.8),

$$Ps(C/S_1,S_i,....) = \int_0^{\infty} \prod_{i=2}^{M-1} (\beta\int_0^{\infty} e^{-z^2/2} J_1(\beta z_i) \prod_{K=1}^{S_i} J_0(A_{ki} z/\sigma_n) dz_i).$$

$$\beta\int_0^{\infty} y e^{-y^2/2} J_0(\beta y) \prod_{j=1}^{S_1} J_0(A_{j1} y/\sigma_n) J_0(A_{iy} y/\sigma_n) dy d\beta \quad (4.26)$$

Having evaluated the conditional symbol detection of (4.14), it is an easy exercise now to average (enumerate) over all possible values of the number of interfering tones in each bank, i.e. S_1, S_2, ...S_m, i.e.

$$Ps(e) = 1 - \sum_{S_1=0}^{Ks} \sum_{S_2=0}^{Ks} ...... \sum_{s_m=0}^{Ks} 1/Q.\prod_{i=2}^{M} P(U_i' < U_1'/S_1,S_i) \quad (4.27)$$

$(S_1 + S_2 + ...S_m = \text{Ks})$

Equation (4.27) assumes that all situations for the distribution of Ks tones into S_1, S_2,...... are equally likely with probability 1/Q where Q is the number of all situations.

The probability of bit errors is related to the symbol error probability of (4.27) by (if orthogonal MFSK is assumed)

$$P(e/Ks,n) = Ps(e).M/(2(M-1)) \quad (4.28)$$

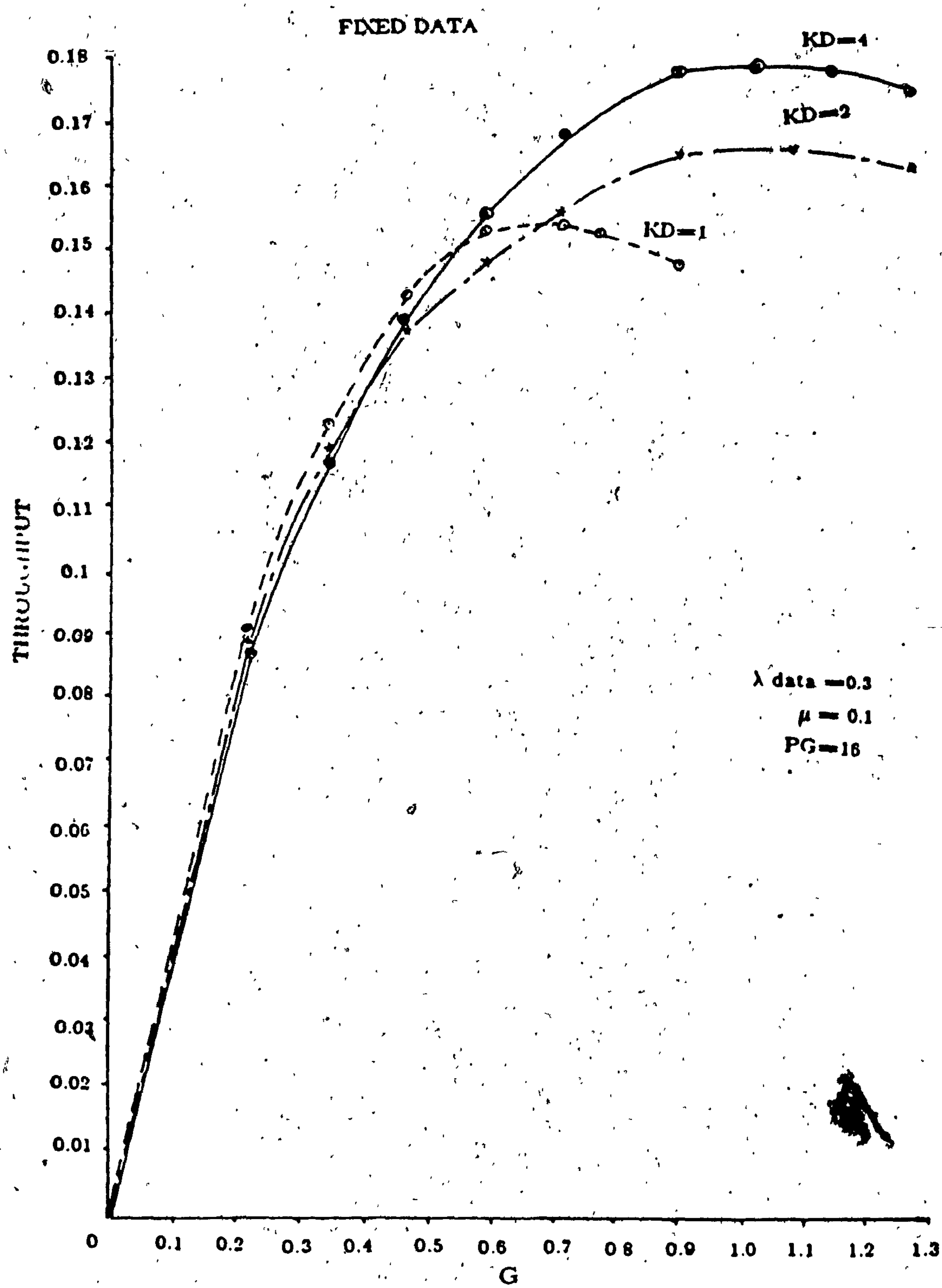


Figure 4.2 Throughput for the given traffic load. The number of voice users is varied and the data users is fixed at five.

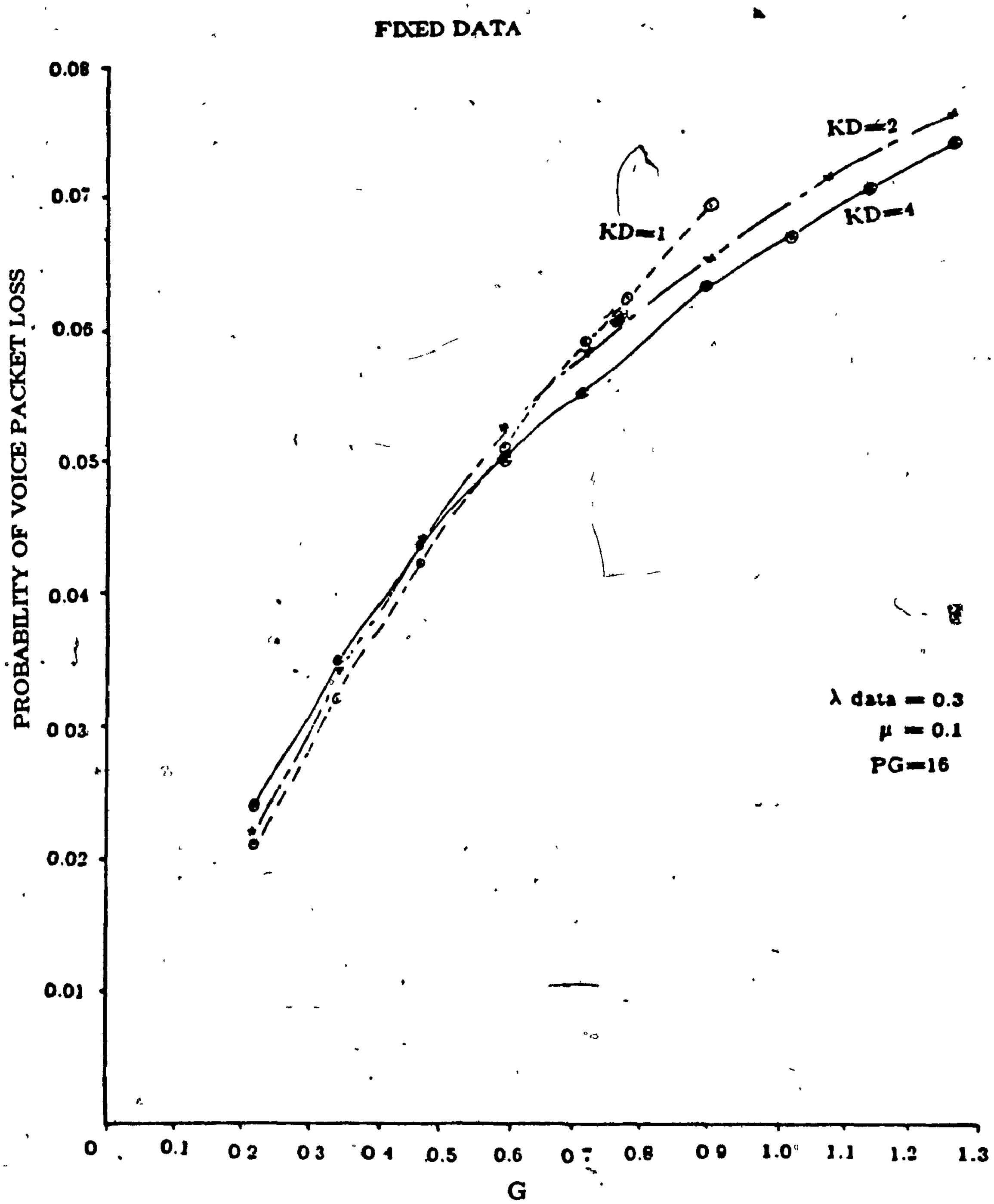


Figure 4.3 Blocking for the given traffic load. The number of voice users is varied and the data users is fixed at five.

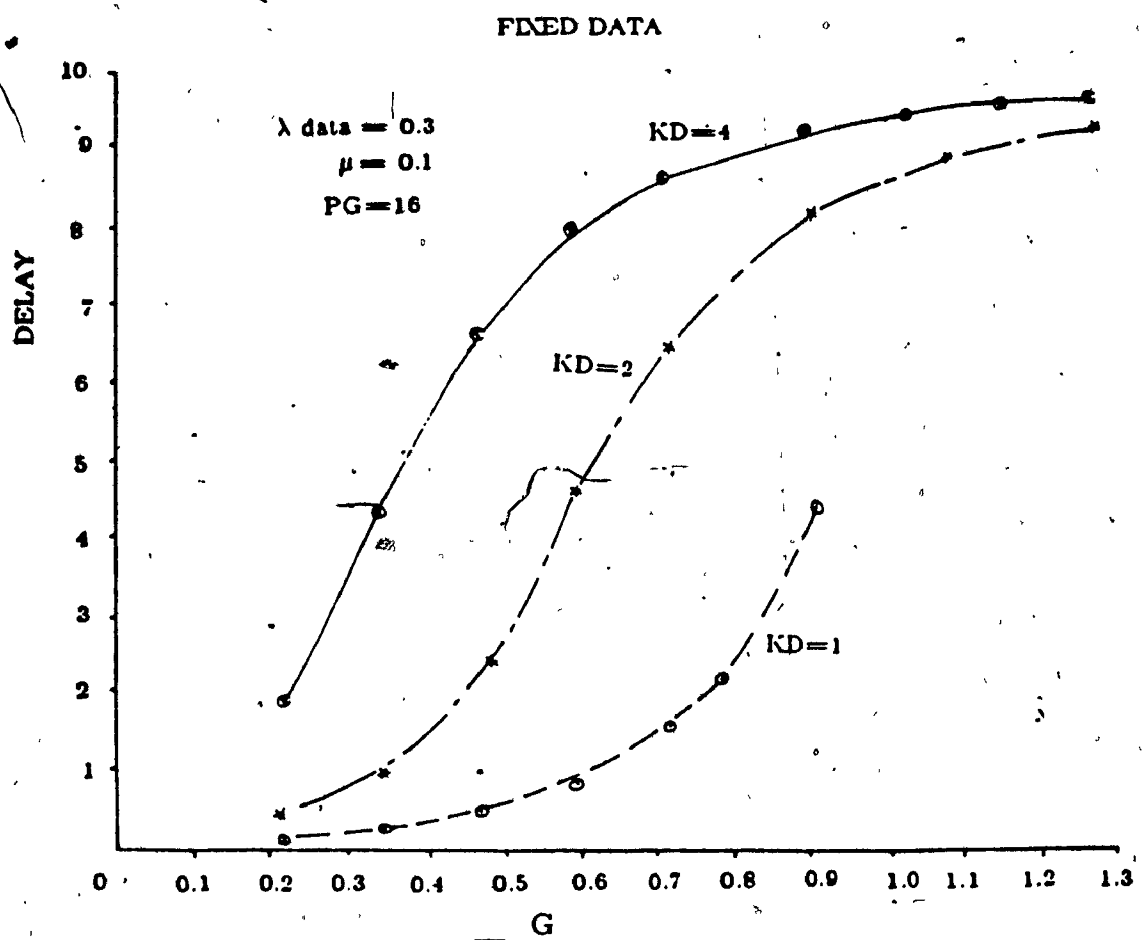


Figure 4.4 Delay for the given traffic load. The number of Voice users is varied and the data users is fixed at five.

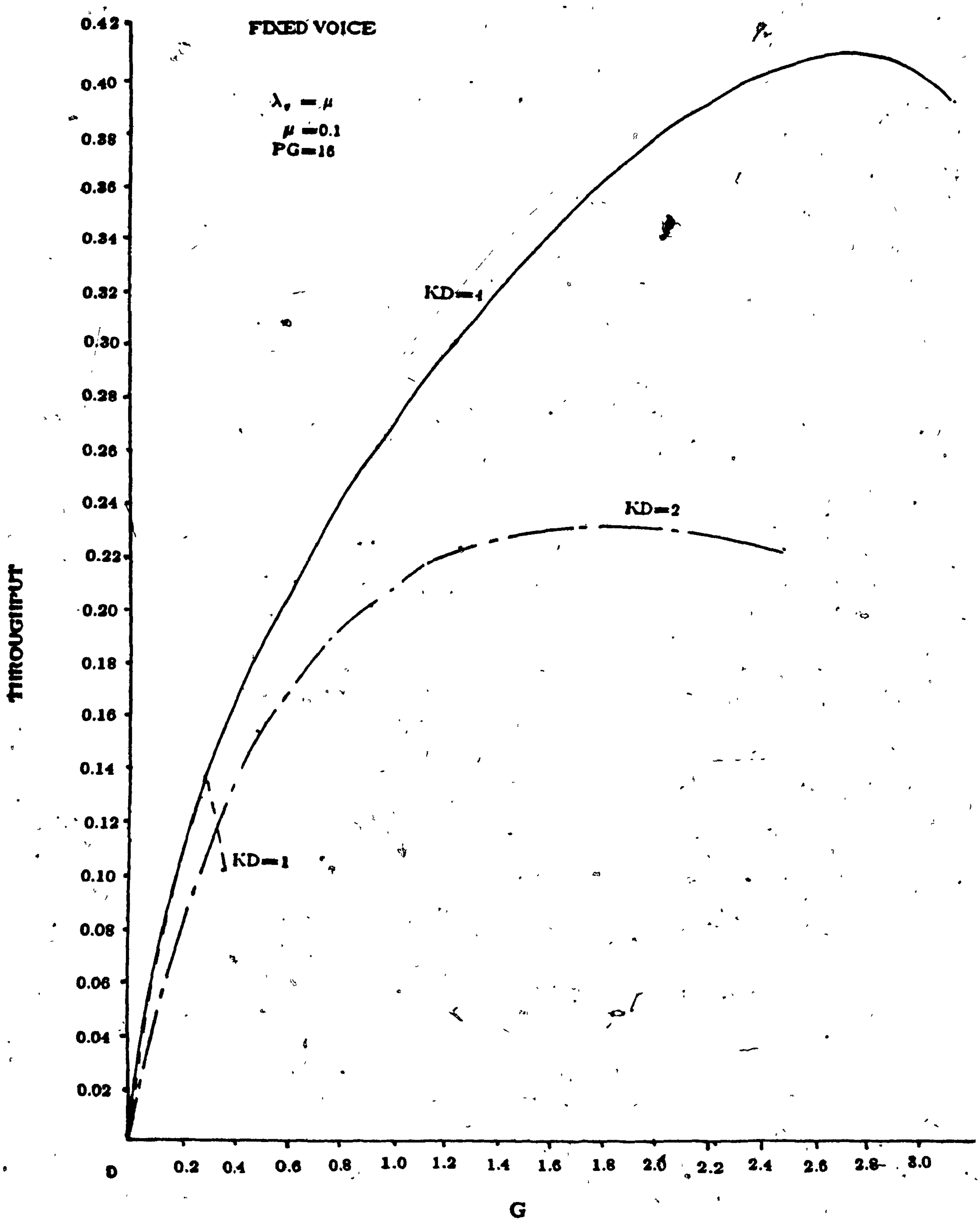


Figure 4.5 Throughput for the given traffic load. The number of data users is varied and the voice users is fixed at five.

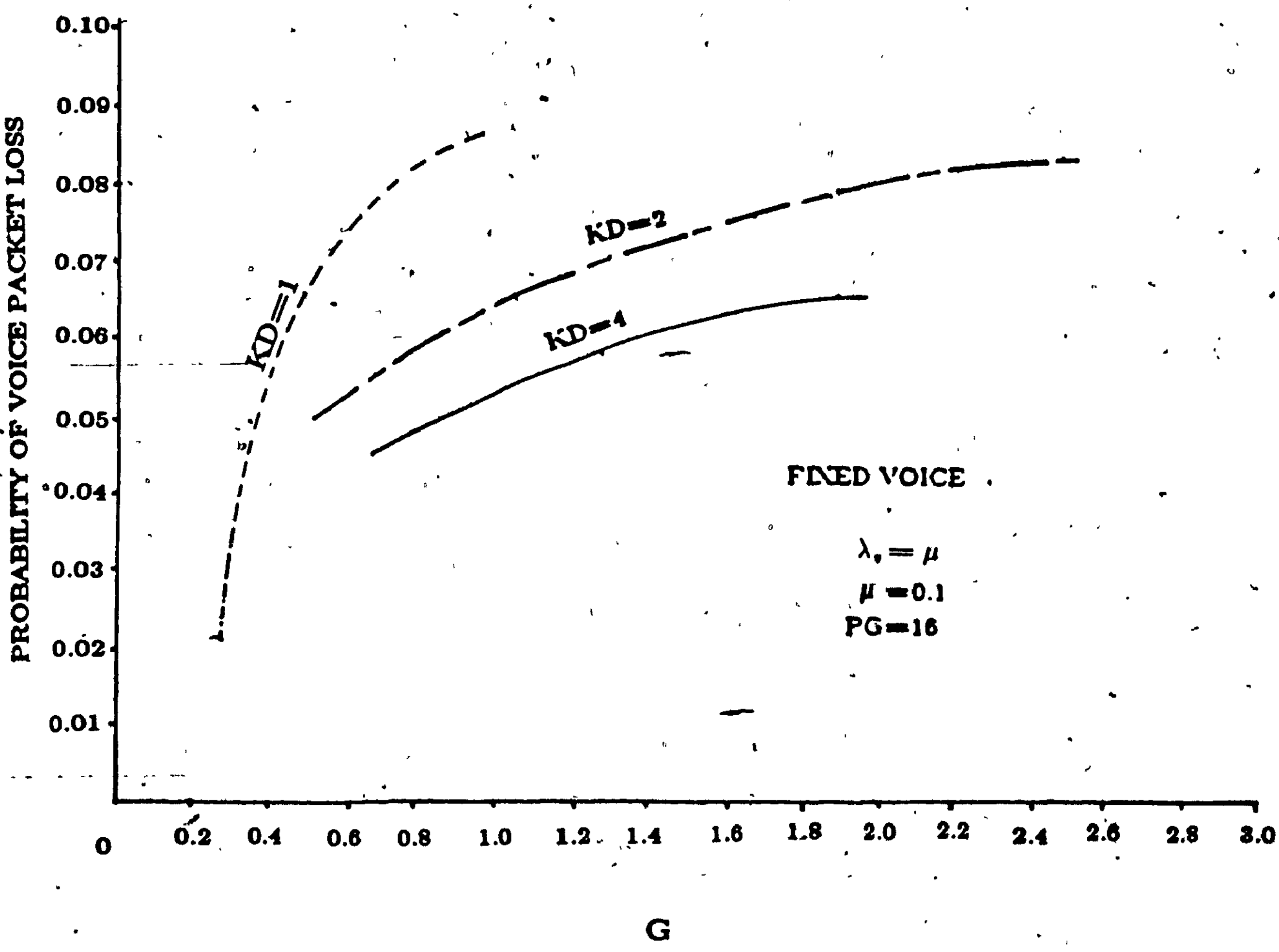


Figure 4.6 Blocking for the given traffic load. The number of data users is varied and the voice users is fixed at five.

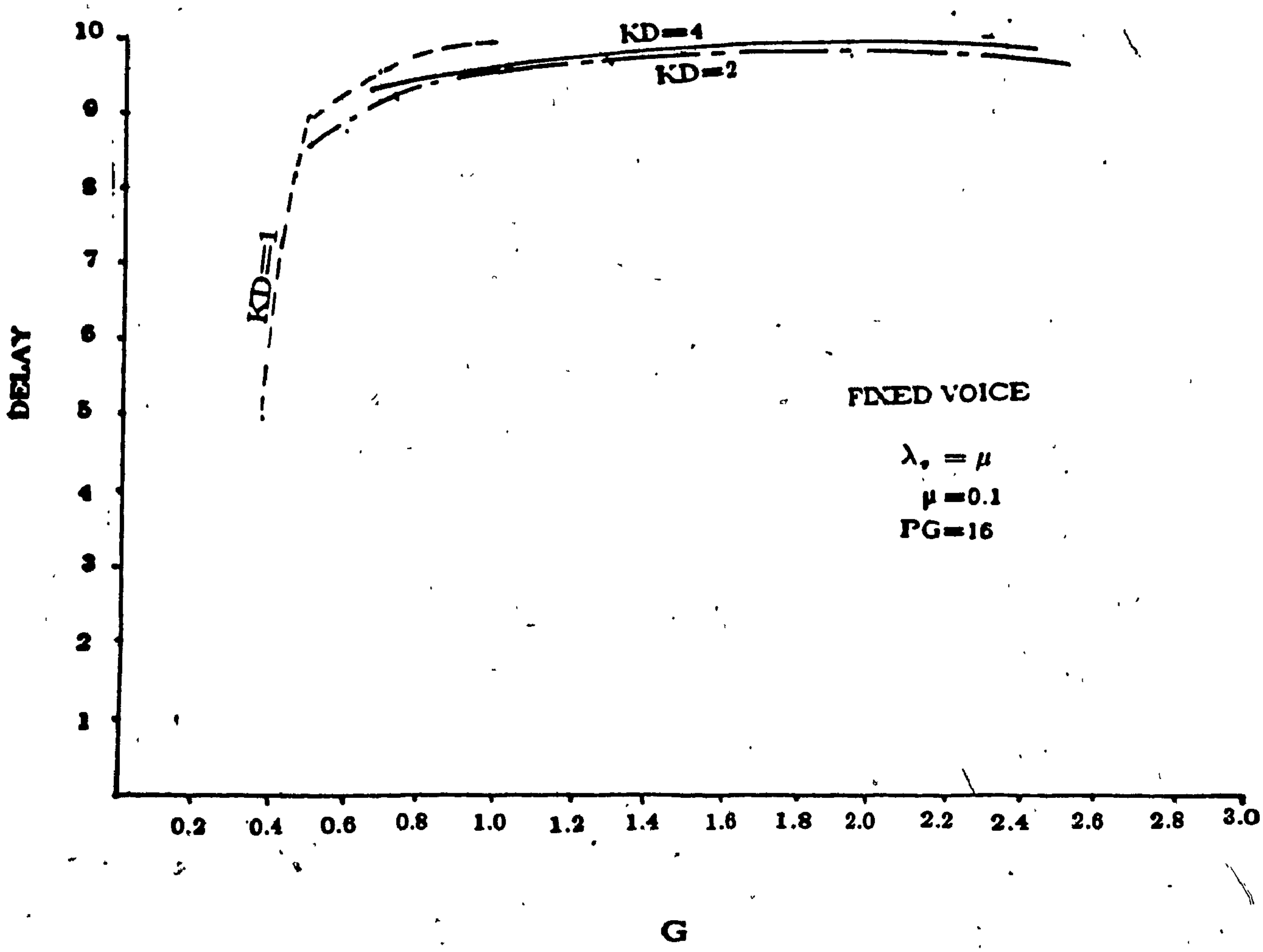


Figure 4.7 Delay for the given traffic load. The number of data users is varied and the voice users is fixed at five.

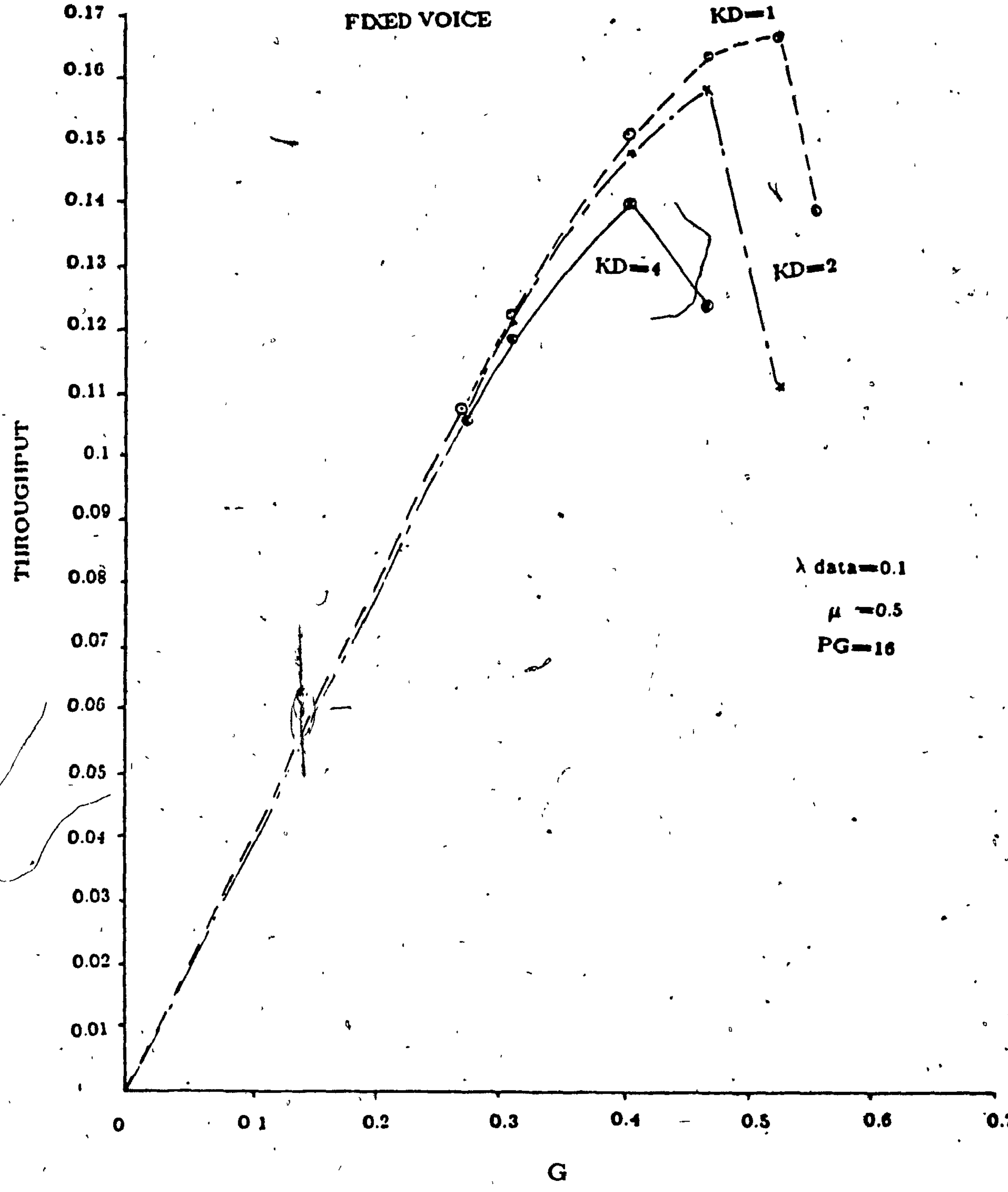


Figure 4.8 Throughput for the given traffic load. The number of data users is varied and the voice users is fixed at five.

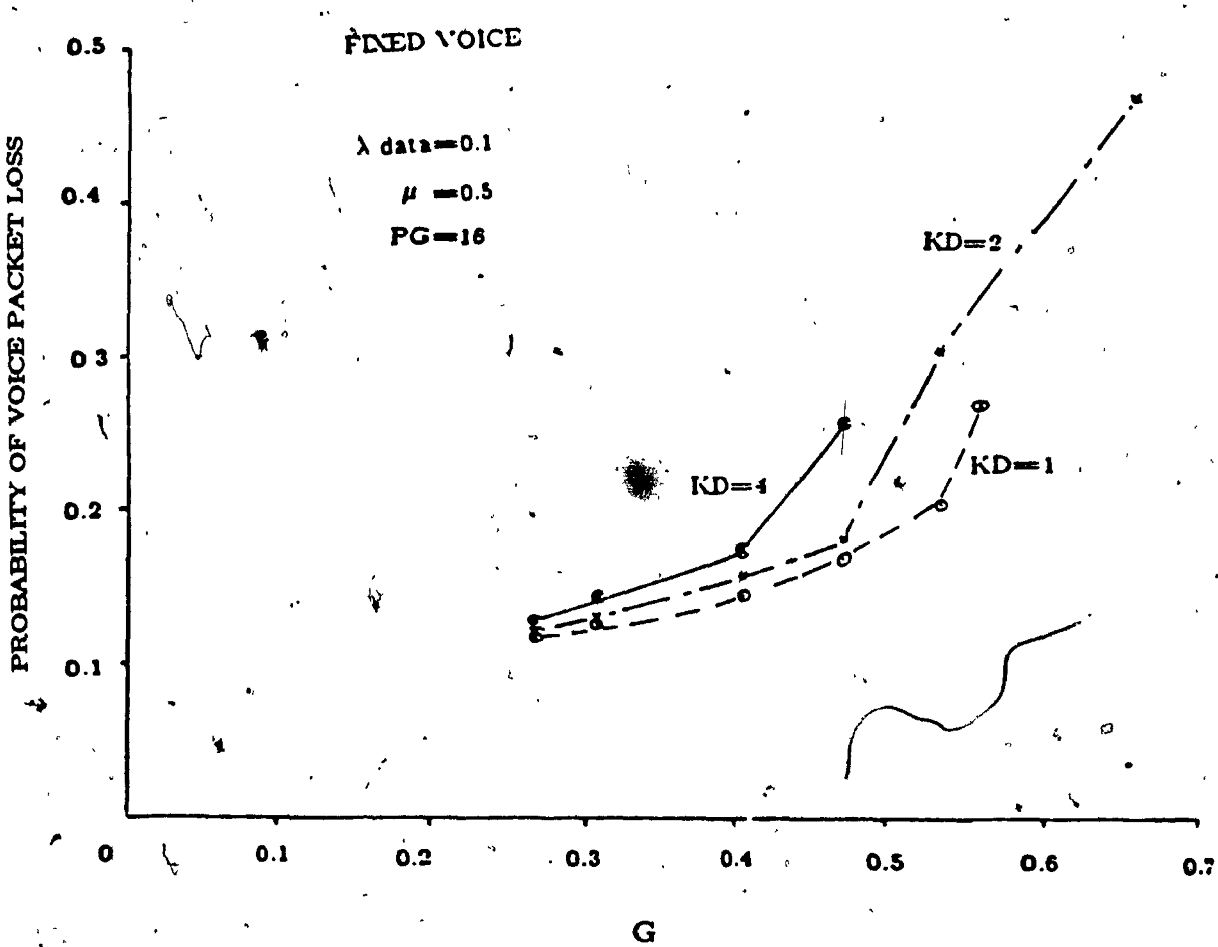


Figure 4.9 Blocking for the given traffic load. The number of data users is varied and the voice users is fixed at five.

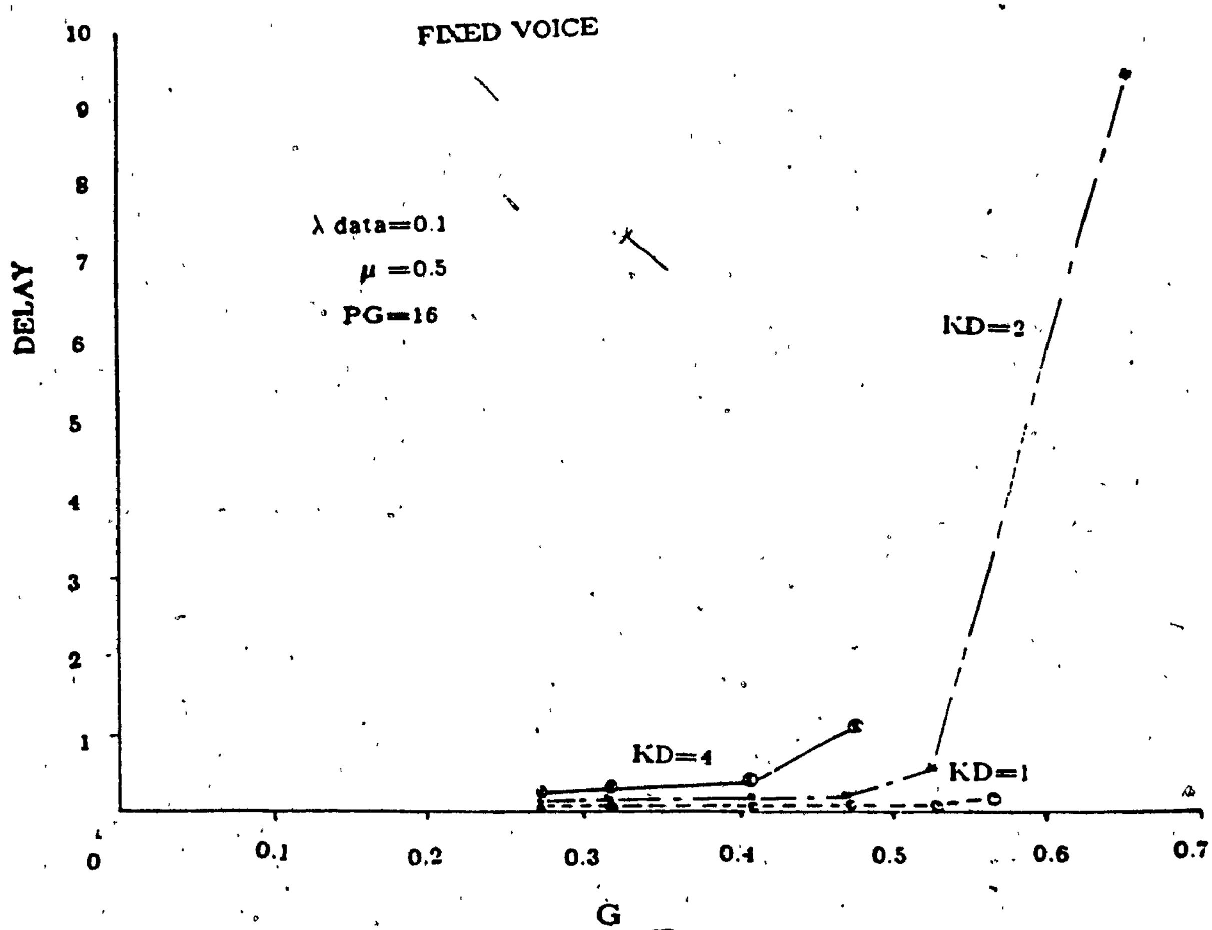


Figure 4.10 Delay for the given traffic load. The number of data users is varied and the voice users is fixed at five.

4.4 Numerical results

Figures 4.2, 4.3 and 4.4 depict the throughput, delay and the blocking for the given total input traffic. The number of data users Md is fixed and the number of voice users Mv is varied, λ_d is 0.3 and the number of data users is fixed at five.

Figures 4.5, 4.6 and 4.7 depict the throughput, delay and the blocking for the given total input traffic, given that the number of voice users Mv is constant and the number of data users Md is varying. In this case μ is 0.1 and the number of voice users is fixed at five.

Figures 4.8, 4.9 and 4.10 depict the throughput, delay and the blocking for the given total input traffic, the number of voice users is kept constant at five and μ is 0.5 and the number of data users is varied.

In all figures values of KD in the range (1,2 & 4) were selected.

Comparing the two curves of Figure 4.7 we can clearly see that at low traffic loads, there is no need for back off (KD=2, KD=4). However as the load increases, back off will be necessary to reduce the delay. The curves of Figure 4.10 depict the same effect except that the difference in performance between the cases of KD=2, KD=4 becomes more pronounced because of the fact that due to the increase of the voice load (μ=0.5). This means that most of the traffic originates from voice users where no back off exists. (The effect of voice traffic is more pronounced).

Figure 4.5 reveals that the throughput increases when the backoff is introduced. In Figure 4.5 the case of KD=4 is better than KD=2 and KD=1.

Also comparing Figures 4.2 and 4.8, reveals the dependancy of performance on the type of load (data or voice). While Figure 4.2 suggests (to improve the throughput) employing backoff only at high loads. Figure 4.8 suggests for the same objectives having no backoff strategy at all.

Comparing Figures 4.3 and 4.9 we arrive at the same conclusion with respect to the blocking probability.

In Figure 4.4, KD=1 is unstable for higher traffic, thus leading to non existance of equilibrium solution.

Comparing Figures 4.4 to 4.7 it is seen that for mostly voice load, (Figure 4.4) the delay for KD=4 is higher than that of KD=2 and KD=2 is higher than KD=1. This is expected as the throughput is higher for KD=4 followed by KD=2 and KD=1. However this tradeoff is not very pronounced in Figure 4.7 where the load is mostly data. Upto a certain traffic KD=1 gives better delay while beyond this traffic value KD=2 & 4 gives better delay results.

Figure 4.3 shows that for mostly voice traffic, KD=4 is preferred followed by KD=2.

In Figures 4.8, 4.9 and 4.10 the load is mostly data but the fixed voice proportion is higher than the previous figures and hence the backoff is not very effective here. The performance is lowest for KD=4 followed by KD=2 and followed by KD=1.

CHAPTER FIVE

CONCLUSION

A cellular mobile network employing frequency hopped Spread Spectrum has been studied for integrating voice and data.

In chapter two the mobile network is described. The analysis was done for a cell in the network in which it is assumed that there are Md data users and Mv voice users and that they have the same modulation scheme.

The voice and data were encoded into packets and transmitted. The data packet was backed off from transmission and the various cases such as KD=1,2 and 4 were considered in the analysis. The voice packet was also modelled in a Markovian state diagram.

The modulation scheme for both the data and voice packets was coherent QPR with channel fading. The throughput, delay, probability of voice packet loss and the probability of successful transmission of a packet (Ps) was evaluated considering the queueing aspects and the modulation aspects of the network.

The probability of successful transmission of a packet (Ps) decreases with the increase of the load. In the case where there were fixed data users and the voice users was varied, for the corresponding value of traffic, Ps was higher when the backoff strategy was applied and was the best for KD=4. When the voice users was fixed and the data users varied, Ps improved when backoff strategy was employed.

In the case of throughput for the given traffic (for fixed data and variable voice traffic), it is found that the backoff increases the throughput. It is the best for KD=4 followed by KD=2.

The probability of voice packet loss is the minimum for the backoff strategy KD=4 followed by KD=2 and KD=1. The delay is the maximum in the case of KD=4 followed by KD=2 and KD=1.

When the voice traffic was fixed and the data traffic was varied, the probability of voice packet loss can be reduced by using backoff strategies for data beyond a certain traffic in the network. The delay in the network can also be reduced if backoff strategies are applied to the data packeys beyond a certain traffic in the network.

In chapter three, the proposed network is compared to other schemes such as the multifrequency central transmitter, fixed channel assignment scheme, demand access fixed channel assignment and dynamic channel assignment schemes.

The proposed SS network has advantages such as selective addressing capability, code division multiplexing for multiple access, low density power spectra for signal hiding, message screening for eavesdroppers, high resolution ranging and interference rejection.

In chapter four a similar scheme as described in chapter two is analysed. In this case non coherent detected MFSK signals were used for modulation of the voice and data packets. The analysis to calculate the probability of bit error for the MFSK signals in multiple access is given.

The value of Ps reduces as the total traffic in the network is increased. Comparing the results of chapter two for the same given conditions, it is found that the Ps is better for the scheme proposed in chapter four.

For fixed data and variable voice traffic, the throughput was higher for the case where no backoff was used upto a certain traffic and beyond this traffic the throughput was better for the backoff strategy KD=4 followed by KD=2.

The probability of voice packet loss is better for the case where there is no backoff (KD=1) at low traffic loads and for higher traffic conditions, the backoff reduces the voice packet loss. The delay is the minimum for no backoff and is the most for KD=4 followed by KD=2.

For fixed voice and variable data traffic, the backoff strategy increases the throughput considerably. When the fixed voice traffic is more dominant than the data traffic, the backoff strategy reduces the throughput.

The probability of voice packet loss is the minimum for no backoff at low traffic loads. However as the traffic is increased, the backoff strategy reduces the loss. When the fixed voice traffic is more prominent than the data traffic, the backoff strategy increases the probability of voice packet loss.

The delay is the minimum for no backoff strategy at low traffic loads. As the traffic increases, the delay can be reduced using backoff strategies. When the fixed voice traffic is more prominent than the data traffic, the backoff strategy increases the delay.

It can be concluded that the throughput, delay, probability of voice packet loss and the probability of successful transmission of a packet on a channel depend on the type of traffic, amount of traffic, the backoff strategy for data and the type of modulation scheme used.

Further extensions of the work can be done considering the channel fading effects while calculating the bit error rate for multiple access for MFSK signals. The Doppler effect due to the movement of the vehicle can be considered while evaluating the various performance measures.

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