Link Layer Analysis of Integrated Hybrid CDMA Networks for Mobile and Satellite Communication Channels

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A Thesis

in

The Department

of

Electrical and Computer Engineering

Presented in Partial Fulfillment of the Requirements

for the Degree of Doctor of Philosophy at

Concordia University

Montréal, Québec, Canada

July 1995

ISBN 0-612-05071-8

ABSTRACT

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Code Division Multi-Access (CDMA) is being promoted as the method of choice in next generation of Mobile, Personal, and Cellular Communications due to its resistance to fading and its feasibility as an asynchronous method for channel sharing in wireless environments. Most of the current literature compares CDMA to other access schemes using performance in the physical layer without addressing the effect of ARQ for data on the physical layer. This thesis addresses mainly this point in a comparison of CDMA to TDMA and in introduces hybrid adaptive schemes with superior performance in the link layer.

A study of a variable-frame Direct Sequence CDMA and an adaptive Time Hopping with Direct Sequence has shown these systems to outperform regular DS-CDMA and TDMA systems for voice traffic in a fading-free environment and for either voice and data transmission in nonshadowed Rician fading.

The delay and throughput performance of satellite-switched Slow Frequency Hopping CDMA network for simultaneous voice and data transmission was analyzed and compared to that of a DS-CDMA system. Two ARQ schemes were suggested for data while Forward Error Correction using the same encoder is used for voice packets. The queueing analysis assumes priority for voice and two models for voice traffic are used (Markovian and IPP). The probability of successful packet transmission was derived for all systems as a function of traffic load allowing one to evaluate the systems using delay, throughput, and voice packet loss as figures of merit. Numerical results show that while voice delay is minimal in all cases, DS CDMA is much more effective then SFH CDMA in all cases. One interesting result is that SFH systems perform better with Stop and Wait (S/W) schemes and achieve a higher maximum throughput.

DS-CDMA systems were also compared to TDMA systems in Rician and log-Normal shadowed environments using the same model above, and CDMA was shown to achieve lower delay for data, and a much lower voice packet loss when data traffic is moderate to low and when shadowing is negligible.

It is shown through various assumptions and comparisons that DS-CDMA is the best approach for integrated wireless multiaccess networks that suffer from severe multipath fading and small levels of shadowing, and that hybrid CDMA/TDMA could improve the performance further. For typical narrowband satellite channels where shadowing is more of a problem than multipath, CDMA may not be the best choice.

ACKNOWLEDGEMENTS

I would like first tc thank Professor Ahmed K. Elhakeem for his help and guidance as well as support and criticism. I would like to thank all the professors in the examining committe for their constructive criticism that contributed to the final layout of the thesis. I am also indebted to Prof. Mehmet-Ali for providing many suggestions and insight.

Most of all, I would like to thank my wife Sondes Zouid who has been the main reason why I eventually finished my Ph. D. Her sacrifices and moral support are qualities that certainly cannot be described by words or equations.

DEDICATION

This thesis is dedicated to my mother Zohra Ismail and to my late father Ali who did not live to see me finish, but whose soul is certainly with us.

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Chapter 1

Introduction

Despite the tremendous evolution in optical fiber communications and all the excitement about the "information superhighway," wireless personal communications are also gaining a lot of momentum recently [1, 2, 3]. Whether the transmission medium is mobile Satellite, indoor, or cellular outdoor channels, personal communications has become an area of extensive research and development. The distinct features of such channels - namely the mobility of its users and the effects of high noise and fading levels, degrades the performance of modems. CDMA systems on the other hand use spread spectrum to combat fading which allows them to be feasible for commercial use.

CDMA, or Spread Spectrum Multiple Access (SSMA), has already gained momentum even in the industry and is being presented as an alternative for the next generation cellular and micro-cellular personal communications. Its use has been promoted even in optical communication (optical CDMA) and personal Ka-band satellite communications. Many personal and satellite commercial systems being proposed for the late nineties use CDMA, either alone or in addition to some form of time-division, frequency-division, or random access [4, 5, 6, 7]. There is now a standard for CDMA: EIA/TIA IS-95 based on the system described in [8, 9].

One problem that has been addressed only recently in wireless communications is the need to transmit multi-media information and the effects that it may have on the performance of various kinds of traffic on the channel. While there were actually studies of wireless Local Area Networks (LAN's) and standards developed for them [4], none -to our knowledge prior to the start of this work- addressed integrated-services wireless CDMA networks, either for narrowband or broadband applications.

Multiple access methods that are being considered for wireless networks include Time Division Multiple Access (TDMA), Code Division Multiple Access (CDMA), Frequency Division Multiple Access (FDMA) and varieties or hybrids of one or more of the above methods. Objective comparisons of the spectrum efficiency or capacity of these systems are difficult to make since it is difficult to have similar assumptions for different systems. Comparisons between optimized systems that disregard implementation feasibility might lead to similar capacities of CDMA and TDMA for example. On the other hand, final cost and feasibility are bound to change with time as technology evolves. Even a claim that a system is more feasible than another in a certain environment might be invalid if an evolution in the hardware renders some parameters (such as system complexity) irrelevant. It is for this reason that one is faced with a necessity of assuming various specific environments and constraints in any study, and that the conclusions of the study should be bounded by and subject to those assumptions. As an example, when we conclude that CDMA is more successful than TDMA in some fading environments, we recognize that TDMA's performance can be improved using more complex equalization systems but we assume that any equalization to make TDMA's performance equal to that of CDMA is more costly than the additional cost (if any) of CDMA over TDMA, at least at the present time.

1.1 Objectives and Contribution of The Thesis

There has been a wealth of contributions in the literature on the capacity and performance of CDMA systems in terms of probability of error and their merits over Frequency Division Multiple Access (FDMA) in mobile communications (especially fading channels) have already been shown. A more detailed look into past contributions in this field (see Chapter 2), will reveal that not a lot of the existing literature compares CDMA to TDMA in terms of data throughput and delay, which are the most important figures of merit for data transmission, or investigates these properties as a function of a varying traffic load, an entity that affects the probability of error in CDMA channels.

Given the discussion at the beginning of this chapter, it is clear however that there needs to be an assessment, both analytical and empirical, of (1) how much and (2) under what conditions CDMA is more a feasible wireless multiple access scheme for integrated data/voice transmission for commercial applications. A study of just the physical layer is not adequate in answering these two questions. In this work, we address analytically parts of both questions by studying the performance of various CDMA schemes and hybrids (some of them we introduce for the first time) and comparing them to TDMA using delay, voice loss and data throughput as figures of merit.

In this thesis, the relationship between link-layer parameters and physical layer parameters for CDMA systems was formalized, which allowed us to assess ARQ systems and their effect on Link-layer performance for both voice and data traffic. The goal is to study the performance of the systems in an integrated environment that gives priority to packetized voice and allows data to be retransmitted if transmission fails using standard ARQ schemes. New hybrid systems that adjust to the traffic are introduced to achieve a better maximum throughput in higher traffic loads and lower delay under light traffic. The study also includes a comparison between Slow Frequency Hopping (SFH) CDMA and Direct Sequence (DS) CDMA.

1.2 Methodology

The main aspects of the approach of the analysis are summarized below:

1. Analytical derivations are attempted in this thesis as experimentation was not feasible. Simulations to corroborate the findings are important and are left for future research.

2. A unified approach will assume constant limiting factors that are common to all systems being compared: the total system bandwidth, the maximum number of connected users, and the available power at the receiver which is assumed to have been adjusted optimally (admittedly not an easy feat for CDMA systems)..

3. Using queueing theory, current literature on the probability of error of direct sequence and Frequency Hopping multi-access systems under fading (which assumes a given number of simultaneous users), and equations derived to link the two mostly in the CDMA radiolink layer, we will express all queueing parameters in terms of total traffic load on the CDMA channel (including retransmitted traffic) for all systems. Voice traffic is analyzed using Markovian and Interrupted Poisson Process (IPP) statistics.

4. Hybrid CDMA/TDMA systems are introduced to take advantage of the more favorable performance of TDMA under higher traffic. They are designed to converge to TDMA (or FDMA) as traffic rises and to use spread spectrum most of the time under light traffic conditions.

5. Various system architectures apply to the analysis. However, the wireless network is mostly assumed to be comprised of a hub and a number of users that transmit to the hub (and receive from it) signals that are approximately orthogonal and have equal power levels, the analysis could apply to one cell in a multibeam satellite system where the interference from other beams is

neglected. The non-orthogonality of the codes results in multiaccess interference approximated as Gaussian noise for direct sequence CDMA and as random frequency hits for Frequency Hopping. The probability of error given this multiaccess interference, channel Gaussian noise, and fading effects (Rician or Rayleigh fading, or Satellite Log-Normal Shadowed Rician fading)) is computed and used in the equations of throughput and delay.

6. In our comparison of the systems involved, the same system model and assumptions are applied except the multiple access method and its effects on the protocols (such as the need of source addresses in TDMA and lack thereof in some CDMA systems).

7. The analysis will concentrate on different issues in various stages and evolve accordingly: first a simplified M/G/1 analysis of adaptive data transmission systems compares the performance of TDMA, CDMA and adaptive hybrids of the two, then a more exact analysis of integrated CDMA systems in negligible fading, then more relevant fading environments will be assumed.

1.3 Contents of Thesis

Due to the complexity of the systems involved and the impossibility of a comprehensive study involving exact analysis of all parameters involved, some of which do not have a realistic analytical model yet, the thesis is comprised of the following partial and complementary steps: Chapters 2 and 3 present the necessary background that is used in the analysis: Chapter 2 summarizes recent attempts at designing higher-layer protocols for wireless systems in general, stressing on aspects that affect the analysis at the link-layer level, while Chapter 3 discusses physical layer issues concerning CDMA systems which were dominant in the literature until recently and which are needed for our analysis.

In Chapter 4, five Systems are compared using delay, throughput and voice loss as criteria: TDMA, a regular direct sequence CDMA using Gold codes, and three hybrid systems: CDMA/TDMA combining CDMA with TDMA in a fixed manner, CDMA with Time Hopping where every user hops in a number of assigned slots that is inversely proportional to the traffic load, and a variable-frame CDMA that uses more and more slots to separate the users as the average traffic load rises. This study assumes users can send either voice or data but not both, and ARQ issues are neglected by assuming negligible time-outs between retransmissions. The systems introduced (the last two) have proven to be effective and to outperform TDMA and regular DS-CDMA systems in Rician fading channels. Parts of this chapter were published in [10].

Chapter 5 extends the study to integrated prioritized traffic and compares the performance of Satellite-Switched DS and SFH CDMA systems using FEC for voice packets and one of two ARQ schemes for data: Type I Hybrid Stop-and Wait (S/W) and Go-Back-N (GBN). In this chapter, fading effects are neglected and a more realistic model for voice traffic (IPP) is used assuming one of three different levels of priority for voice packets in the transmitter. Voice packets are multiplexed asynchronously (ATM-style) in the mobile transmitter which is shared by more than one user (a typical application is a mobile public-transportation vehicle). It is shown that DS systems outperform SFH systems by a large margin in the environment considered. Parts of this chapter is to be published soon [11].

In Chapter 6, A comparison is made between DS-CDMA systems and TDMA systems assuming the same assumptions used in Chapter 5 except that satellite fading channels are assumed. It is shown that CDMA systems outperform TDMA systems especially in fading channels and in low and medium traffic. As the multi-path varies with the bandwidth and the delay spread assumed, the degree at which

CDMA systems outperform TDMA systems varies accordingly. Although voice traffic is assumed to be close to capacity to favor TDMA and compensate for the disadvantage that a fixed assignment TDMA would suffer from due to the assumptions adopted.

The thesis then ends with conclusions and suggestions for future research. The appendices in the end contain derivations of some of the equations used in Chap 5. The next section contains a glossary of the terms and abbreviations used throughout the thesis.

1.4 List of Terms and Abbreviations:

ACK: Acknowledgment, a signal meaning that packet is successfully received.

ARQ: Automatic Repeat Request, data retransmission protocols.

ATM: Asynchronous Transfer mode.

AWGN: Additive White Gaussian Noise.

BCH code: A channel coding algorithm named after the initials of its three inventors.

BFSK: Binary Frequency Shift Keying.

BPSK: Binary Phase Shift Keying.

CDMA: Code Division Multiple Access.

DS: Direct Sequence.

FDMA: Frequency Division multiple Access.

FEC: Forward Error Correction.

FH; Frequency Hopping.

GBN: Go Back N, a data retransmission protocol.

GEO: Geostationary satellites.

GSM: Global System for mobile Communications, 2'nd generation mobile communication system.

IPP Interrupted Poisson Process.

LAN: Local Area Network.

LEO: Low Earth Orbit Satellite.

LSRM: Lognormal Shadowed Rician Model.

MFSK: Multiple Frequency Shift Keying.

MMPP: Markov Modulated Poisson Process]

M/G/1: Markovian arrivals, Generally-distributed service times, and one server.

NACK: No Acknowledgment: a signal saying that the packet was not successfully received.

PG: Processing Gain.

PDF: Probability Distribution Function.

PSK: Phase Shift Keying.

QoS: Quality of Service.

SFH: Slow Frequency Hopping

SNR: Signal to Noise Ratio.

SR: Selective Repeat, a data retransmission protocol.

S/W: Stop and Wait, a data retransmission protocol.

TDMA: Time Division Multiple Access

TH: Time Hopping.

VSAT: Very Small Aperture terminals.

UMTS: Universal Mobile Telecommunication Service.

WAN: Wide Area Network.

Chapter 2

CDMA Networks, A Probability of Error Analysis

2.1 Introduction

This chapter will discuss briefly, but as clearly as possible, the most recent developments in the area of Code Division Multiple Access with an emphasis on performance under fading. For the sake of completeness and for the benefit of the readers that are less acquainted with the details of spread spectrum, most topics will be introduced virtually from scratch to explain the terminology and the assumptions. The first section will deal with Direct Sequence (DS) multiple access, the second one with Frequency Hopping (FH) issues, and the last will discuss channel models for terrestrial and satellite mobile channels. This chapter is an overview of the physical layer models that will be used in Chapters 4-6 and will, as such, provide all the necessary background and motivation behind the approach taken in this thesis. A significant effort was made to follow a consistent terminology, but due to the wealth of parameters to be discussed, we were forced to use the same symbols for some different parameters . For this reason, a list of symbols is provided at the beginning of this and every chapter.

2.2 List of Symbols and Parameters Used in This Chapter

A_j: Amplitude of the received signal in Receiver j.

E_b/N_0: signal to channel noise ratio.

M: Number of interfering users.

SNR: Signal to total noise/interference ratio.

U: Number of connected users sharing the channel.

ω_c: The carrier frequency.

λ_{ij}: the crosscorrelation between any DS code i and DS code j.

$d_i(t)$: Binary data waveform used in Transmitter i.

$m_i(t)$: DS encoded message waveform for User i.

k: Number of code chips per bit.

$q_i(t)$: DS coding sequence for user i.

$x(t)$: Received signal.

$y(t)$: Decorrelated signal in the receiver.

2.3 Direct Sequence CDMA for Mobile Communications

2.3.1 Fundamentals of DS modulation and demodulation

In Direct Sequence CDMA (DS- CDMA), a pseudorandom sequence of bits is multiplied or XORed with the sequence of data bits. In other words, for User#i, let

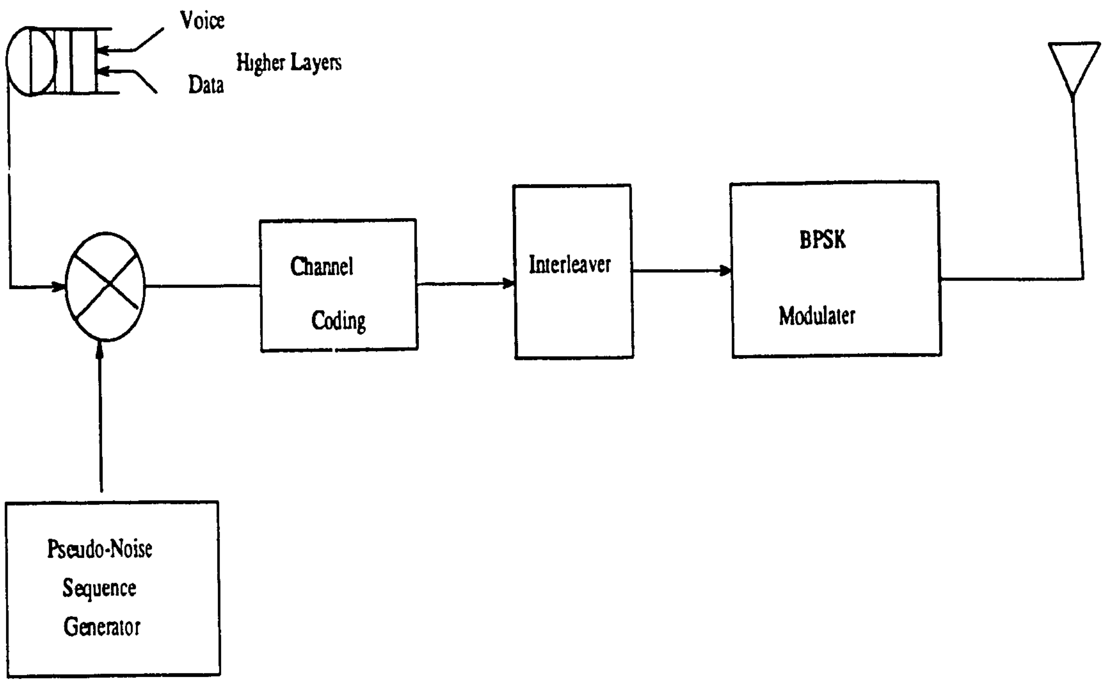

Figure 2.1: An example of a DS-CDMA transmitter

The hub here is in the satellite which decodes all signals and recodes them with the destination's code or routes the packet to other networks.

$d_i(t)$ be the binary data waveform (a sequence of 1 or -1 with a period of T_b), then the transmitted encoded message is given by

$$m_i(t) = d_i(t)q_i(t) \tag{2.1}$$

where $q_i(t)$ is the coding sequence corresponding to the i'th User. The bit rate of the resulting coded waveform is the same as the chip rate of the code sequence whose period is $T_c \approx 1/W$. The modulation assumed for DS-CDMA is Binary Phase Shift Keying (BPSK), which a method of choice due to the possibility of coherent detection.. In this type of CDMA, there is no need to synchronize the bit packets (as is the case with TDMA, for example) and the various uplink signals are sent independently with no coordination as far as timing goes. Figure 2.1 shows a block diagram of a simplified DS-CDMA transmitter. In the down-link each station receiver must obtain phase, bit and code coherence with the desired transmitted

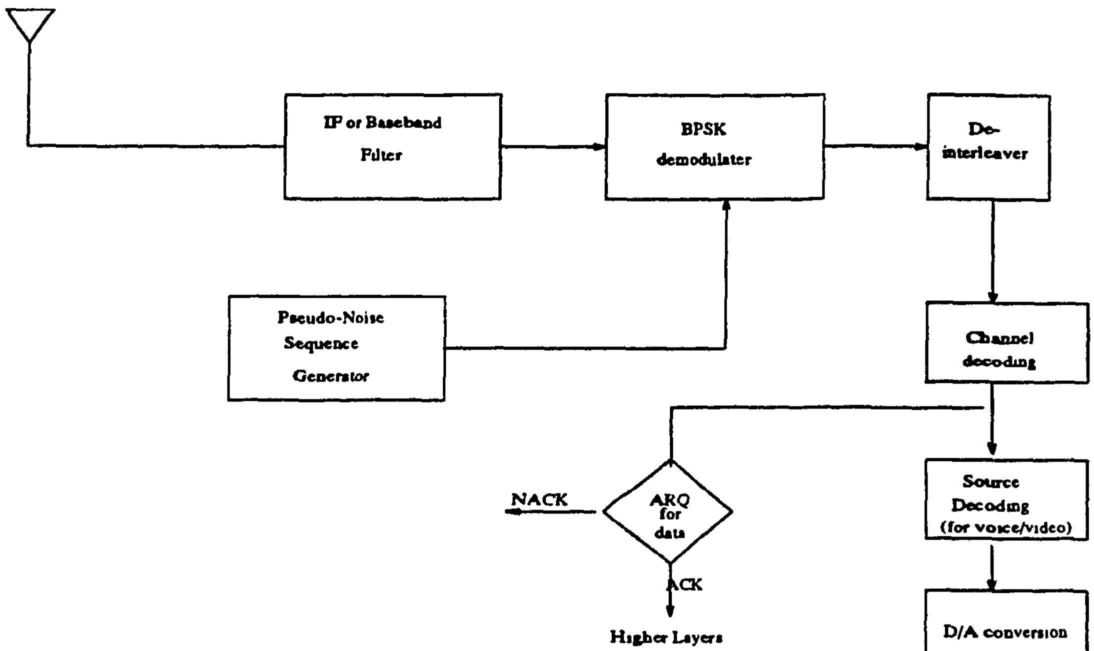

Figure 2.2: An example of a DS-CDMA receiver

signal while the other undesired carriers are present. Figure 2.2 shows an example of a receiver. Ignoring for the time being all noise, intermodulation, and multipath effects, the received signal, x(t), is of the following form:

$$x(t) = \sum_{i=1}^{U} A_i \sin\left(\omega_c t + \frac{\pi}{2} m_i(t-\tau_i) + \psi_i\right) \tag{2.2}$$

$$= \sum_{i=1}^{U} A_i m_i(t-\tau_i)\cos(\omega_c t + \psi_i) \tag{2.3}$$

$$= \sum_{i=1}^{U} A_i d_i(t-\tau_i) q_i(t-\tau_i)\cos(\omega_c t + \psi_i) \tag{2.4}$$

where ψ_i and τ_i are the different phase and time shifts of U different users. To recover the j'th signal, a receiver needs to correlate x(t) with a reference signal

$$r_j(t) = 2q_j(t)\cos(\omega_c t + \psi_j), \tag{2.5}$$

resulting in an output $y(t)$ [12],

$$y(t) = d_j(t)A_j + C_j. \tag{2.6}$$

Our desire is to reduce C_j as much as possible to end up with the desired signal d_j. C_j is actually the sum of the correlations of the the reference signal with all other BPSK signals other than the j'th one. Since this value is a random variable, it was shown [12] that its mean value is given by the following equation:

$$\bar{C}_j^2 = \sum_{i \neq j}^{U} \frac{1}{2} A_i^2 \chi_{ij}^2 \tag{2.7}$$

where χ_{ij}^2 is given by

$$\chi_{ij}^2 = \frac{1}{T_b} \int_0^{T_b} \left(R_{ij}^2(t) + \hat{R}_{ij}^2(t) \right) dt \tag{2.8}$$

and $R_{ij}^2(t)$ and $\hat{R}_{ij}^2(t)$ are the partial correlations given by the following equations:

$$\begin{aligned} R_{ij}(t) &= \frac{1}{T_b} \int_0^t q_i(\tau - t) q_i(\tau) d\tau && (2.9) \\ \hat{R}_{ij}(t) &= \frac{1}{T_b} \int_t^{T_b} q_i(\tau - t) q_i(\tau) d\tau && (2.10) \end{aligned}$$

χ_{ij}^2 could not be calculated exactly, but many useful bounds have been evaluated including the lower bound of the maximum of χ_{ij}^2 which was derived by Welsh [13] who showed that

$$\max \chi_{ij}^2 \geq \sqrt{\left(\frac{U-1}{kU-1} \right)} \tag{2.11}$$

where U is the number of codes and k is the number of code chips per bit. For large U, this lower bound is approximately $\sqrt{1/k}$.

Pseudo-random codes can be generated as maximal length code via shift registers but this regular *m-sequence* suffers from very high cross correlation and a very limited number of users compared to the length of the code sequences. Two other families of codes were defined by Gold [14] and Kasami and Lin and are known as the Gold code and the Kasami codes respectively. Gold developed his class of codes by combining a PN code with every shifted version of another PN code to get a total of U+2, where U is the maximum number of maximal length codes. In addition, the original two codes used to get the Gold code are properly selected to get a maximum pairwise crosscorrelation that is shown to be less than $\sqrt{U+2}$. This value when normalized to k, we get

$$\frac{\chi_{ij}}{k} \leq \frac{\max \chi_{ij}}{k} \tag{2.12}$$

$$\leq \sqrt{\frac{2}{k}} \tag{2.13}$$

This value is only a factor of $\sqrt{2}$ bigger than the Welsh bound. Gold codes are going to be the codes we will be assuming in the next chapters.

2.3.2 Mobile Channels

There has been much interest lately in Mobile Satellite and/or Cellular communications as a domain that might benefit greatly from CDMA ([15, 16, 17, 18, 19], also [20, 21, 22, 23]. The development of VSAT's (Very Small Aperture Antennas) was one factor that made the prospect of Satellite-Switched personal systems a possibility in the near future as power requirements are drastically reduced. Another development that also motivates research in this area is LEO (Low Earth Orbit) satellites. These are satellites that have a very close orbit (800 to 1000 miles). The significantly shorter distance between earth stations and the satellite hub results in much lower path loss (around 26.9 dB less than that of geo-stationary ones for an LEO at an altitude of 1000 miles) and also a much lower propagation delay (25 times less for the same example mentioned above). This is a significant factor in ARQ.

A mobile channel presents an additional challenge as degradation in performance results from time-varying multipaths which cause fading and frequency shifting. This degradation can often be combatted or at least reduced if the channel is modeled properly. Figure 2.3 shows for example three different ways a transmitted electromagnetic field is received by a moving VSAT (car antenna this case). The three different components shown are

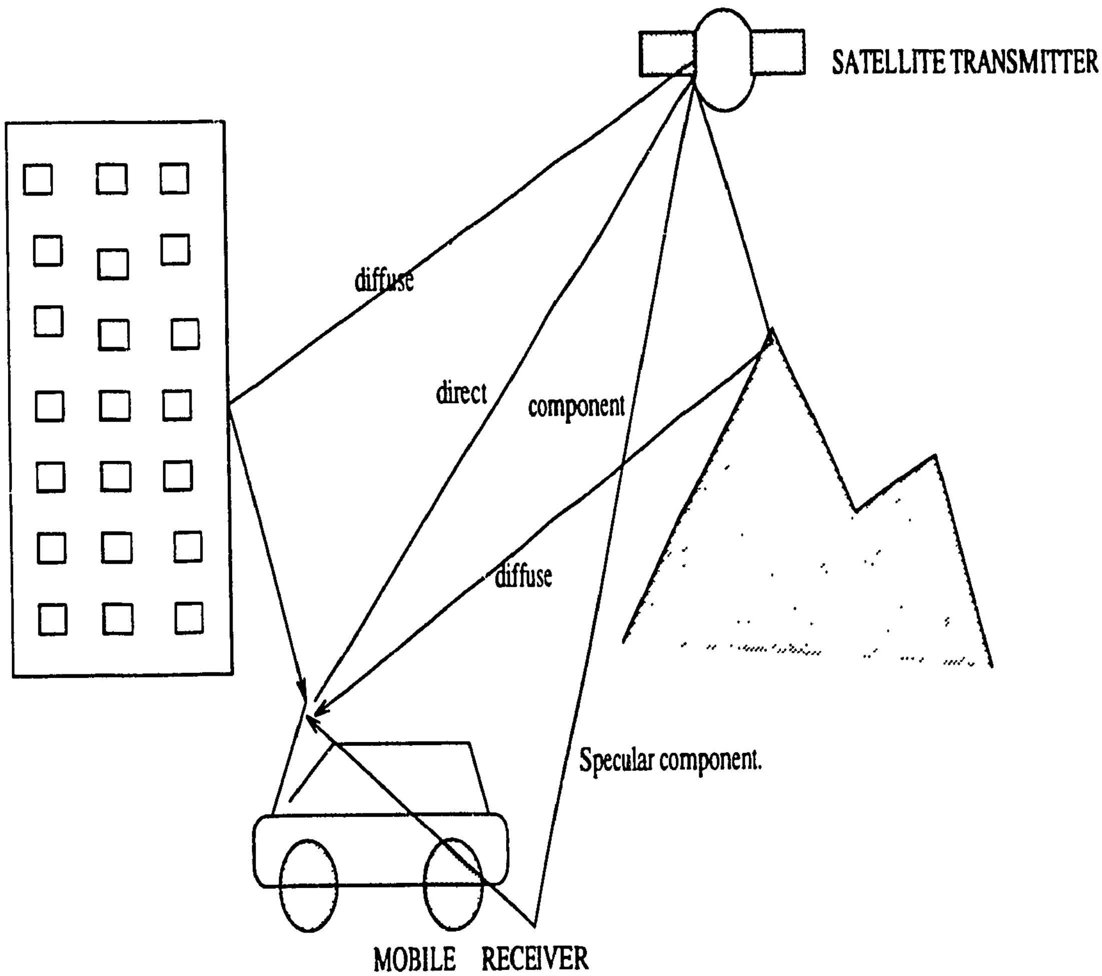

Figure 2.3: The three different components of the faded signal received by the mobile

1) **A direct component** or *line of sight,* which is the signals that reaches the receiver via a direct path.

2) **A specular component,** which results from a single reflection and is received as a delayed version of the direct component. This component is assumed small compared to the first one in many instances in the literature [12].

3) **A diffuse component,** being the sum of all other reflections.

This third component is assumed to be a random variable whose effect in the receiver is similar to additinal noise in the channel, which makes the amplitude at the receiver a random variable α with the following distribution

$$p(\alpha) = (1+r)e^{-s(1+r)-r} I_0 \left(2\sqrt{sr(1+r)}\right) \tag{2.14}$$

where r is the ratio of the direct component over the diffuse component and s is the normalized total power of the receiver (total power divided by power in direct component). When r goes to zero (meaning that the diffuse component overwhelms the direct one), the distribution is Rayleigh, otherwise it is Rician. More theoretical developments about the performance of DS-CDMA in fading will be presented in the next section.

There have been quite a few contributions about CDMA that changed our view of CDMA from being a technique that is warranted only for military purposes to a feasible commercial Multi-access system. Not all the literature can be covered in a limited space, but a brief description of the major results about CDMA under fading follows.

2.3.3 Performance Under Multiple-Access Interference

One of the early works by Pursley [24] analyzed an asynchronous phase-coded spread-spectrum multiple-access communication system. The system model considered was similar to the one introduced earlier in this chapter. To study the performance, an average signal to noise ratio approach was taken as opposed to worstcase performance which was discounted as meaningless. The phase ψ_i above, the time shifts and the value of the bit sequence were assumed independent random variables and the interference from $M-1$ other users was assumed to be an additive Gaussian noise (AWGN). In this case, the average signal to noise ratio at the receiver was approximated as

$$SNR = \frac{1}{\sqrt{\frac{M-1}{3N} + \frac{N_0}{2E_b}}} \tag{2.15}$$

where M is the number of simultaneous users, N is the period of the spreading sequence, and E_b/N_0 is the bit energy to noise ratio.

Another way to measure performance was developed by Geraniotis and Pursley in [25] where they obtained approximations for the average probability of error for both binary PSK and QPSK modulated DS-CDMA systems. The approximation was based on the integration of the characteristic function of the multiaccess interference.

2.3.4 Performance under fading

Fading issues have been studied extensively in the literature. Borth and Pursley [26] studied the performance of DS-CDMA in a general class of fading channels. The channels considered were those for which the channel output consists of a strong stable specular signal plus a faded version of this signal. These channels (whose fading model is referred to as Rician fading model) arise when the multipath faded signals are weaker than the main *line-of-sight* signal. Equations were derived assuming general wide-sense-stationary uncorrelated scattering channels. The average signal to noise ratio at the receiver output was computed in terms of the spread-spectrum signature sequences and the covariance function of the fading process. The general expression (which will not be included here for the sake of brevity) was then applied to two kinds of fading: frequency selective and time selective fading, In a later work [27], Geraniotis and Pursley investigated the performance of coherent DS-CDMA over specular multipath fading channels and derived the average probability of error of the correlation receiver assuming either determinitic or random gain coefficients for an arbitrary number of paths. The gain coefficients, delays, and phase angles of all distinct paths were modeled as mutually independent random variables.

More recently, Direct-Sequence Code Division Multiple Access (DS-CDMA)) networks have gained momentum [28] due to such characteristics as multipath fading resistance, some inherent security,[1] low power cost per station (compared for example to TDMA networks), inherent voice silence utilization, automatic cellular frequency reuse (in contrast to cellular FDMA networks) [29], etc. CDMA networks were presented more and more as potential candidates for indoor communications, personal and consumer communications [29], Satellite communications, factory automation, cellular communications, cordless Local Area Networks, etc. to name but a few.

The literature is rich with signal to noise ratio comparisons between CDMA and non-CDMA networks with voice applications being the main motivation. In [30] for example, the capacity of direct sequence CDMA was shown to be much higher than that of FDMA systems for mobile satellite communications especially for voice transmission. On the data transmission side, some CDMA queueing models have been analyzed [31], either without evaluating the exact bit or packet error probabilities in the various fading environments, or else these probabilities were evaluated without attention to the networking aspects such as traffic characterization, buffering, etc. [24, 27]. Networking and modulation aspects where both emphasized in the

[1]Conventional digital communication systems use encryption techniques such as the Data Encryption Standard (DES) to achieve security, thus adding complexity and requiring more bandwidth. In spread spectrum systems, each user is identified by a certain code , which results in a certain measure of inherent security.

SUGAR system [16], where a 2 dB improvement in the SNR needed for QoS requirements was obtained compared to classical CDMA networks by splitting the users into groups using orthogonal codes. Also recently [17] field tests were conducted to demonstrate the feasibility of overlaying CDMA signals on top of the existing narrowband microwave traffic and, in [18], a cellular system using CDMA on both mobile to station links (where user signals fade independently) and station to mobile links (where signals fade simultaneously) was investigated. The work that has been added on CDMA since 1993 has been enormous. Most of the results of the papers published in the last two years have been crucial in pushing for the implementation of the CDMA standard. Some of the work published recently will be compared to our work in the last chapter.

2.4 Frequency Hopping

Frequency Hopping took a boost in the early eighties as it has been presented as an alternative multiaccess method that is immune to jamming and resistant to doppler shifts [32, 33, 34, 35, 36]. The analysis of feasible Frequency Hopping systems as in the Direct Sequence case very rarely dealt with issues such as delay and throughput of a mixed taffic network in a fading environment, especially for bursty traffic. One noteable exception is a recent paper by Yang and Stuber [37] which discussed the maxium throughput achievable for a slotted Frequency Hopped CDMA network from a cutoff rate perspective (assuming the best FEC coding possible).

In the following pages, the recent literature about Frequency Hopping will be surveyed starting with the fundamentals and the first works that derived the probability of bit error for Frequency Hopping.

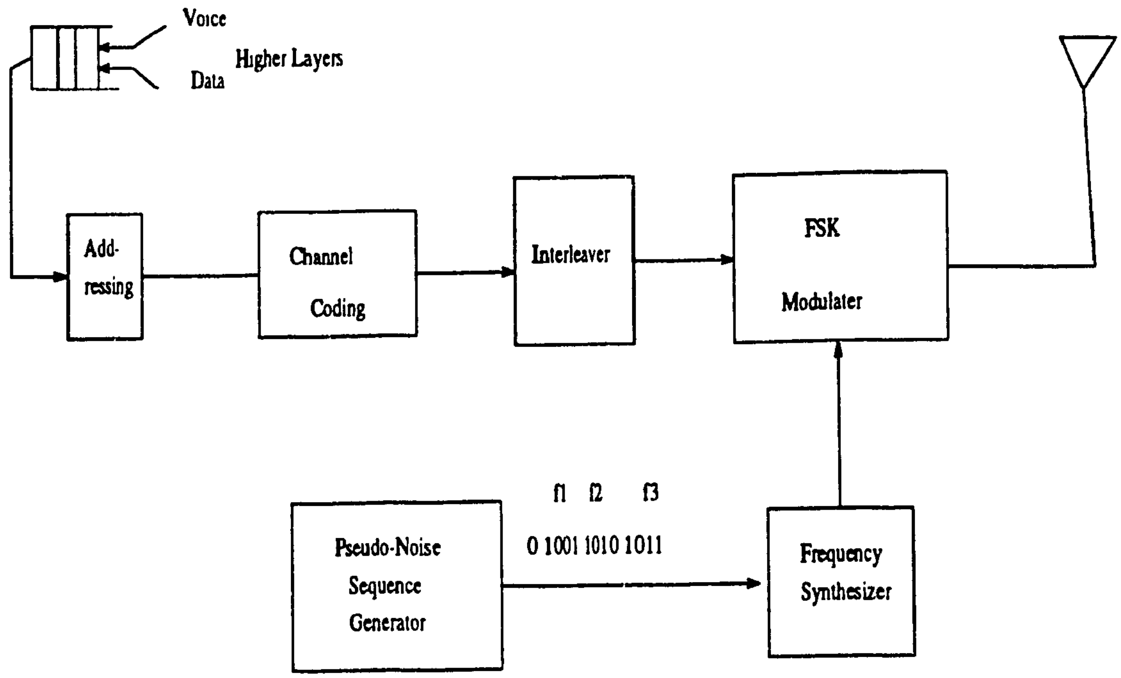

Figure 2.4: A simplified Frequency hopping transmitter and an example of how a frequency sequence is generated from a PN sequence

2.4.1 Fundamentals

Frequency hopping multiple access assigns a sequence of frequencies to every user who "hops" according to the assigned sequence. Just like in the case of DS-CDMA, an ideal system would assign orthogonal sequences (meaning that the resulting radio-transmitted signals are orthogonal and thus do not interfere with each other), but with real systems, there are overlaps. A frequency sequence is generated as shown in Figure 2.4 and the transmitter hops between the assigned frequencies at a *hopping rate* of $1/T_H$. If T_H is less than T_s (the period of a symbol), we have Fast Frequency Hopping. Otherwise, we have Slow Frequency hopping. Each of these two FH methods has its merits, but SFH systems have been used more extensively with hybrid systems. We will assume in this thesis a slow frequency hopping system with a hopping rate of one hop per bit, an assumption that was used quite often in the literature [37]. In the analysis of FH systems, one of two assumptions is made about the frequency patterns: [25, 38]:

A deterministic pattern where no frequency can be hopped into more than once in one sequence. In most cases, for an N-length deterministic sequence of frequencies $f_0^{(i)}, f_1^{(i)}, ..., f_{N-1}^{(i)}$, where the number of frequencies is U, $N = U - 1$, $f_j^{(i)} \neq f_k^{(i)}$ if $j \neq k$, and any cyclic shift of a sequence is another sequence with the same properties. An example of such sequences is the Reed Solomon hopping pattern [39].

Random sequences An example of such patterns is the memoryless pattern. In this model $f_k^{(i)}$ is a sequence of independent ransdom variables where each is uniformly distributed in the set of U frequencies. In other words hopping patterns for different transmitters are statistically independent. Assuming one hop per symbol, only L frequencies would be used in a packet of length L.

Modulation for Frequency hopping can be either MFSK (BFSK being one special case) or DPSK. In both cases noncoherent modulation is used since phase referencing during demodulation following each hop is difficult. Figure 2.5 gives a block diagram of a FH receiver. The multiaccess method assumed has been always a random access scheme where the users either are slotted and thus interfere completely or not at all as in slotted ALOHA, or unslotted in which case interference between users can be only in a part of their packets. There are also issues concerning how the packets are synchronized with the hopping, if frequencies are changed at the same time for all users, we are dealing with *Hop Synchronous* system, otherwise the system is *Hop-Asynchronous.*

2.4.2 Multiple Access Interference

The probability of error of a Frequency Hopping system is much more complicated to compute than that of a DS-CDMA. This is partly because the modulation scheme

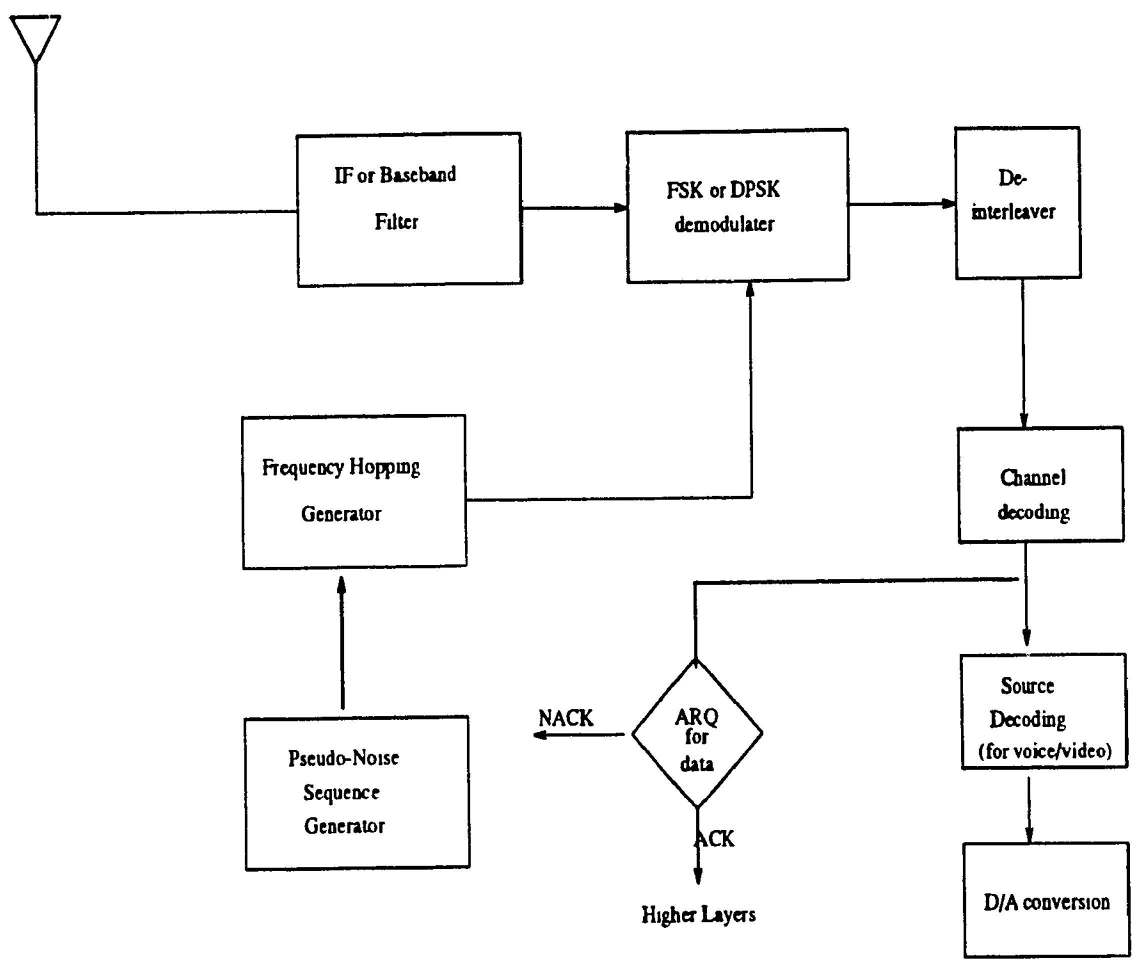

Figure 2.5: A diagram of a Frequency Hopping receiver with either FSK or DPSK demodulation

assumed is MFSK, whose analysis is not easy, and also becasue the effect of interference cannot be estimated by Gaussian noise anymore. In FH systems, the probability of error depends mainly on the probability of a frequency hit (more than one user hopping to the same frequency in one bit). For the case of memoryless random frequency patterns, assuming a slow frequency hopping with a hopping rate of N_H bits per hop, the probability of a hit in any single bit is easy to compute [25]:

$$p_H = \frac{1}{U}\left[1 + \frac{1}{N_H}\left(1 - \frac{1}{U}\right)\right] \tag{2.16}$$

which for large U can be reduced to

$$p_H \approx \frac{1}{U}\left[1 + \frac{1}{N_H}\right] \tag{2.17}$$

The above analysis does not actually consider the effect of a variable number of interferers, which was calculated later in more recent works [37]. The probability of having k simultaneous hits from $M-1$ simultaneous interferers for memoryless hopping patterns is given by the binomial distribution. Assuming one hop per bit, the probability of symbol error for a multiaccess system with M simultaneous users is then given in terms of the conditional probabilities:

$$p(s/M) = \sum_{k=0}^{M-1} \binom{M-1}{k} \left(\frac{1}{U}\right)^k \left(1 - \frac{1}{U}\right)^{M-1-k} \mathrm{Prob}(s/k) \tag{2.18}$$

where $\mathrm{Prob}(s/k)$ is the probability of symbol error given k hits. This latter conditional probability is quite complicated and hard to compute, and there are a few results given in the literature with some interesting approximations. This subject will be discussed more in detail in Chapter 4.

2.4.3 Recent Advancements in FH Research

A multitude of issues concerning FH systems are being discussed and analyzed [40, 41, 42, 43], also [44, 45, 46, 47]. One issue that is usually overlooked in the literature is the spatial distribution of stations and its effect on the different power levels at the receiver. An interesting statistical distribution is discussed in [34] for the power levels as a function of spatial distributions of interfering users. This issue, although important, may not be of particular importance in the case where the differences between the power levels of all received signals is negligible, such as in the case of Satellite-switched systems. FH systems have already been deployed in wireless LAN's as means of wireless random access,

In our work, the emphasis is more towards integrated systems that can have periods of intensive traffic and variable bit rate transmissions. FH systems will be shown to be a less attractive option in that case.

Chapter 3

Networking Issues in the analysis of Integrated Wireless Communication Systems

3.1 Introduction

The "Quality Of Service" (QOS) figures of merit for a communication network vary according to the application: voice quality for example is assured by a low voice packet loss and real-time transmission, while data files need to have low message/packet delay and high throughput in a virtually errorless transmission.

The large amounts of bandwidths promised for future wireless networks offer the possibility of integrating real-time applications (such as voice and video) together with data-oriented services (such as file transfer, electronic mail, and remote log-in) within a single common network. The characteristics of real-time applications, however, differ significantly from those of non-real-time-ones. Typically, the desired delivery time for a real-time message across the network is bounded by a specific maximum delay or latency, resulting in a deadline being associated with each message. This delay bound is an application layer end-to-end timing constraint [48].

Messages that exceed this bound are considered lost. The study of an integrated wireless network will highly depend on the relationship between real-time traffic and non-real-time traffic, particularly on the effect of each type of traffic on the other type's QOS parameters. One example that will be studied is the effect of data traffic intensity on voice loss and the effect of total voice traffic load on data delay and throughput.

One particular characteristic of CDMA networks is the effect of interference from other users on the bit-error rate. This effect actually makes a data-link layer analysis essential in the study of the performance of CDMA networks as there is a relationship between the data load of all interfering users and the probability of error at the hub's receiver.

In the literature, while many papers concerned with data ARQ schemes assume a given probability of correct packet transmission, we cannot afford to assume that the probability of error in CDMA channel will be invariable to data or voice traffic load. The data-link layer analysis of CDMA networks that will be presented in the following chapters will take this factor into account. In this chapter, general higher-layer issues that are essential to any wireless network performance analysis will be discussed. The main layers that are relevant to wireless communications are similar to the ones adopted for ATM networks: a physical layer concerned with coding and modulation schemes that achieve a successful transmission, a link layer concerned with ARQ schemes for data transmission as well as voice jitter control or voice packet discarding, and an adaptation layer that is concerned with access control issues. Having discussed physical layer issues in Chapter 2, this chapter will only discuss issues in the higher layers that will affect the analysis of Chapter 4, 5, and 6.

3.2 Integration Issues in Wireless CDMA Systems

A typical wireless CDMA or TDMA system will have to adopt a layered protocol that must be able to support a wide range of services, many of which can be Asynchronous and who might have different constraints [49], [50], [51] and [52]. An example [52] is given in Table 3.1.

Table 3.1: An example of QoS requirements in an integrated environment:

Traffic Type	Design Constraint	Performance Targets
Voice packets	Delay<30ms	Packet error rate < 2%,
real-time data	Delay < 30 ms	bit error rate < 10^{-6}
non-real time data	Packet loss < 10^{-6}	delay: Average < 50 ms, 90%<100ms

Depending on the type of wireless system used (a satellite macrocellular system, a personal micro- or pico-cellular radio frequency network, or an indoor mobile network), the network architectures and the protocol models assumed might vary. For a typical personal communication service such as UMTS (universal Mobile Telecommunications Service), a protocol consisting of a Radio Physical Layer (RPL), a Radio Link Layer (RLL), and a Network Adaptation Layer can be adpated and used as shown in Fig. 3.1 . For a CDMA system, the multiplexing is done also partly in the link-layer. In our models in Chapters 5 and 6, the multiplexing is actually done in the mobiles in the uplink, while spreading is done in the physical layer for multiple access between different mobiles.

One issue that emerges in the analysis of an integrated services system is circuit switched vs. packet switched service. In a circuit switched service a channel

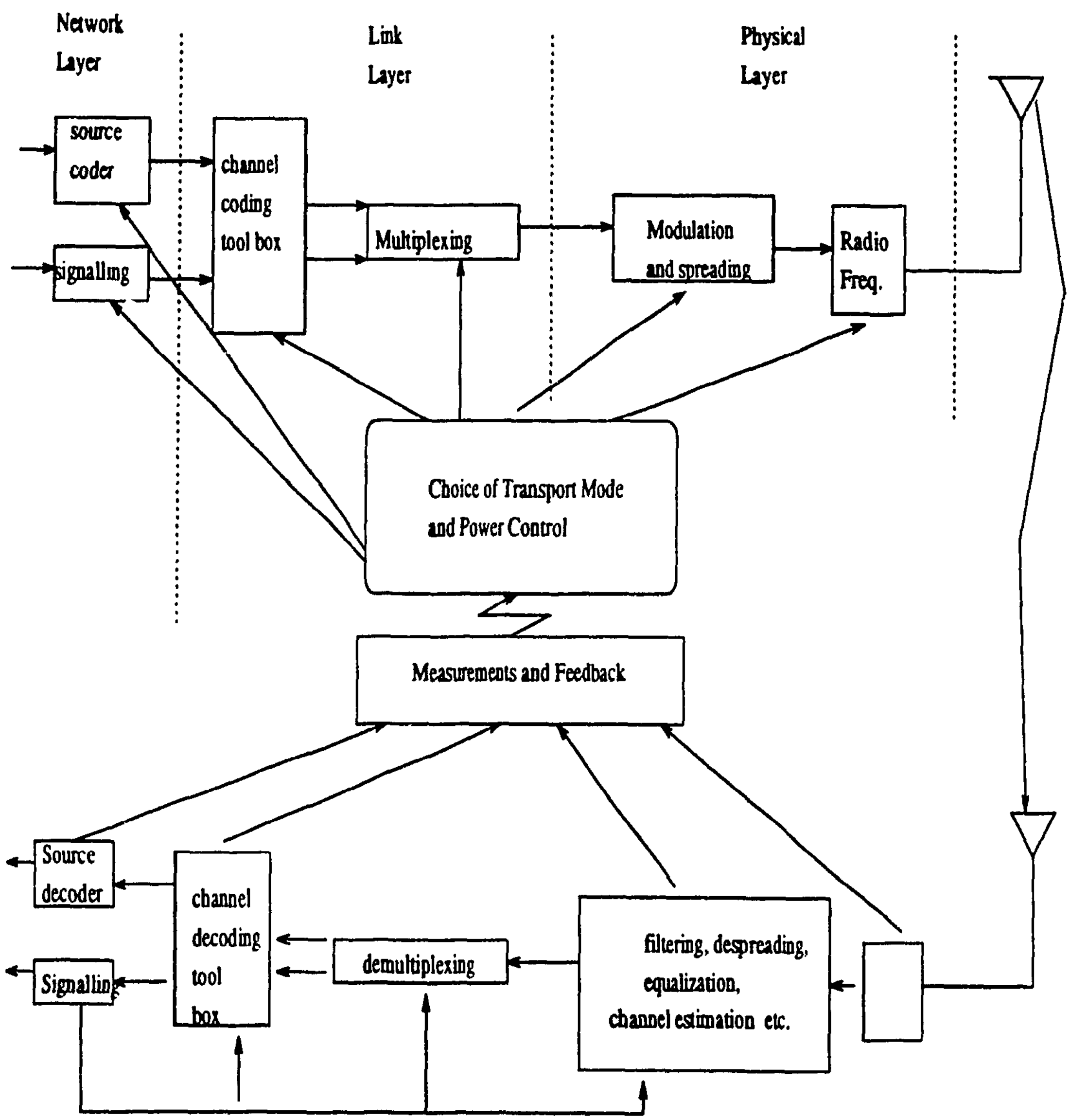

Figure 3.1: Example of an adaptive downlink traffic channel illustrating the layered protocol.

is reserved for the length of the call, while for packet-switched service, calls compete for the link on a packet per packet basis.

A data call typically can afford to be packet-switched, while voice calls are traditionally circuit-switched. Table 3.2 gives an example on the switching of different classes of calls in an ISDN environment

Table 3.2: An example of switching type and prorities for some classes of traffic

Traffic Type Transmitted	Switching Mode	Priority (respectively)
Voice packets	Circuit/Packet	highest/highest
real-time data	Circuit/Packet	high/highest
File transfer	Packet	low/high

3.3 Queueing Concepts for Multiple Access Systems

In any mutiaccess system, users share the same channel but are separated by some protocol in time, frequency, code, or a combination thereof. This implies that users do not actually use the same queue, i.e they are ideally served separately and have no direct effect on each other's queues. An indirect effect is multiple access interference in the case of CDMA, which increases the probability of error and thus increases the average service time for data ARQ, and decreases the data throughput and voice quality. Note that the queue in our case of an uplink (reverse link) is the buffer of the transmitter while the server is the channel itself.

In the queueing model adopted throughout the analysis in this thesis, we will

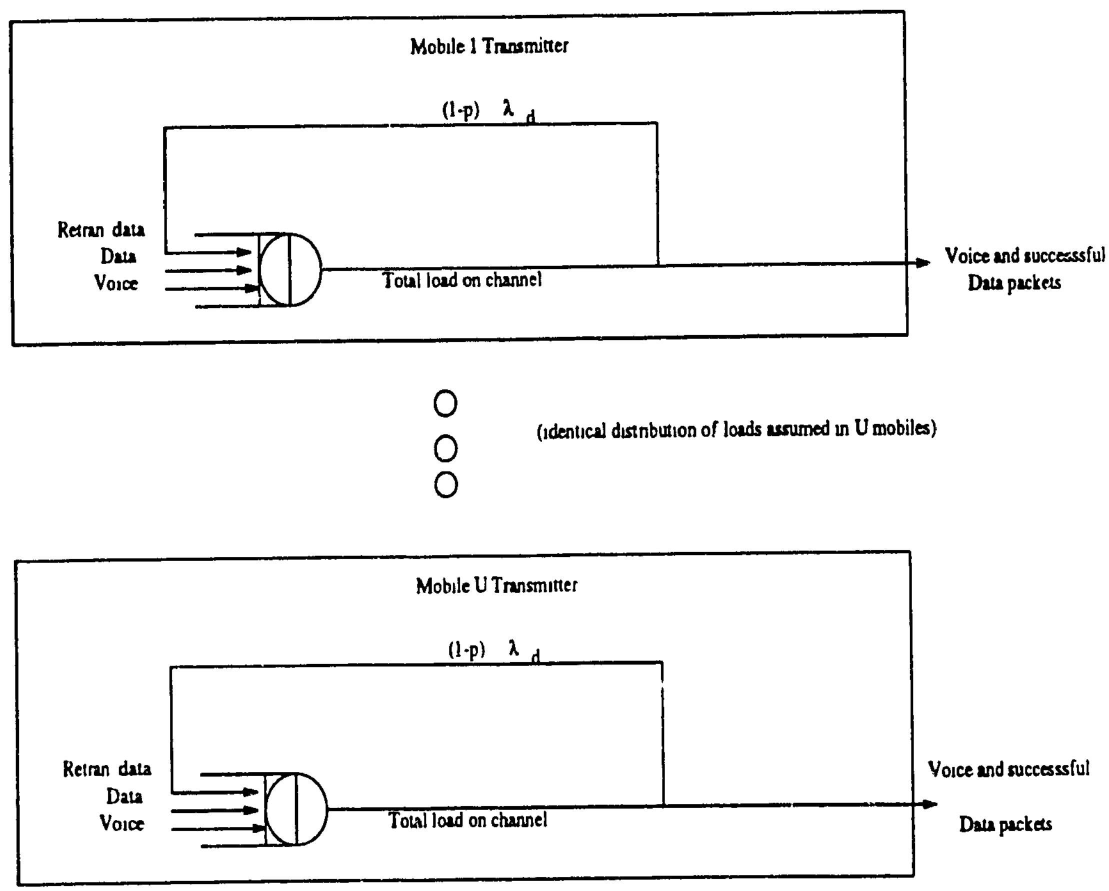

Figure 3.2: A diagram of a node of interest in the overall multiaccess system

assume a single queue in the buffer of the transmitter of every station, with messages having random lengths arriving at a poisson rate to the transmitter. Voice messages are transmitted in a time proportional to their lengths, while data messages are served when all packets are transmitted and/or retransmitted , which makes the average service time per message ($\bar{X}$) depend on the probability of error. Figure 2.5 shows an example of the queue considered. It is to be noted that since the service times of data messages depend on the probability of error, and the probability of error depends on the amount of multiaccess interference, which in turns depends on the amount of traffic per user, the service rate depends on the arrival rate itself. This , however, should not be an obstacle in analyzing the system since both service and new arrival rates (excluding retransmitted traffic which actually depends on the traffic load for CDMA) are independent of the number of messages in the buffer.

A transmitter receives a message with a random length E (packets) that could have any general distribution and is to transmit it according to the schedule allowed by the multiaccess scheme. The rate of arrival of messages, λ, depends on the source bit rate among other things and is assumed to be a Poisson process for data traffic, meaning that the probability of having k messages arrive in any interval of length T time units is

$$p(k,T) = \frac{(\lambda T)^k}{k!}\epsilon^{-\lambda T}. \tag{3.1}$$

Note that this arrival rate is for messages not packets, so it does not depend on retransmitted traffic. An entity of interest which we will use mostly in the analysis is the utilization factor, or traffic intensity. This is defined as

$$\rho = \lambda \bar{X} \tag{3.2}$$

where $\bar{X}$ is the average service time per message, which obviously depends on the number of retransmissions needed per packet for data. This traffic intensity must be less than one to insure the stability of the queue in the buffer of the transmitter. It is not equivalent to the traffic on the channel. This is because in some instances service time for messages includes timeouts between retransmissions where no transmissions might take place in the channel itself.

Another entity of interest is the useful throughput, meaning the actual data rate that is transmitted. For data traffic, this useful throughput depends on the probability of retransmission and on the ARQ scheme. Formulas for delay and throughput for different ARQ schemes will be presented in detail in the following chapters as needed in the analysis. The ARQ schemes that will be used will be Go-Back-N (GBN), Stop and Wait (S/W), and Selective Repeat (SR).

-For GBN, a buffer is needed in the transmitter as it has to go back s frames to retransmit s packets whenever an error occurs and a NACK is received from the destination. s is chosen such that the transmission of . packets is greater than or equal to the total time needed between retransmissions.

-For Stop and Wait, no buffer is needed in the transmitter at the physical level as it stops transmitting until it receives either an ACK or a NACK. This results in a reduction of throughput and an increase in delay as the time-out is wasted even if there is no error.

-For SR, the transmitter transmits continuously, retransmitting only the packets for which it receives a NACK. In this case, both the receiver and the transmitter need to have a buffer: to be able to go back to the specific packet that was not successfully received For the transmitter, and to reorder the packets for the receiver. Delay at the receiving end might not be better than GBN, but throughput is definitely better as only one packet per error is retransmitted, thus reducing traffic.

Voice traffic may be modeled using Poisson statistics under certain conditions (see Chapter 5), and in those cases a classic M/G/1 queue with two classes can be assumed. Chapter 5 presents a more generalized modeling of voice.

The average service time, $\bar{X}$, is the time it takes to transmit successfully a message. For voice it is equal to the time needed to transmit a packet times the average number of voice packets per message, to reduce packetizing delay, the number of packets per message is assumed to be unity for voice. For data, it is equal to the average time it takes to transmit successfully a data packet (an entity that was calculated in the literature [53] for various ARQ schemes) times the average number of packets per message. More detailed expressions will be listed in the following two chapters.

Chapter 4

Delay and Throughput Characteristics of TDMA, CDMA, and TDMA/CDMA Hybrid Networks for Multipath-Faded Channels

4.1 Introduction

In this chapter, we analyze the delay and throughput performance of five multiaccess techniques- namely, TDMA, CDMA, CDMA/TDMA, adaptive Time Hopping CDMA (CDMA/TH) and variable frame CDMA/TDMA for either voice or data transmission.

In this initial analysis, a Markovian arrival process is assumed for data and the propagation delay is neglected. This allows us to ignore ARQ issues for now and to compare various CDMA schemes to TDMA to get initial insights on their performance in either data or voice transmission in mobile fading channels. The propagation delay in a microcellular system is actually a very small fraction of the frame length and its inclusion affects the analysis very minimally. Satellite systems, however, suffer from large propagation delays, and will be studied in the next chapter.

The probabilities of bit and packet errors are evaluated for the five systems in Rayleigh and Rician multipath fading environments assuming that user signals fade independently. The channel model is suitable for instances when the bandwidth of the spread spectrum signals is larger than the inverse of the time spread, meaning that interference from adjacent bits occurs because of the multipath.

In all systems, we use the same channel bandwidth $W \approx 1/T_C$(Hz), where T_C is the chip width of the DS code, the same source bit rate $R_b^{(S)}$, the same fading channel multipath spread T_m, the same number of bits per packet L, the same mean square power in the fading signal components, and the same number of users, U. The following section gives a list of the symbols used in this chapter. A few necessary assumptions and simplifications are assumed and will be explained in Subsection 4.3.1.

4.2 List of The Symbols Used in This Chapter

E: Message length in packets, a random number.

E_b/N_0: Signal to channel noise ratio.

(E_b)': Bit energy in the line of sight component.

G_i: Total traffic load on System i.

L: Number of bits per packet.

L_m: number of multipath images interfering in every bit/chip.

N: Number of slots the users are split into in Systems 3 and 4.

N_5: Number of slots thes users are split into in System 5.

M_i: Number of interfering users for System i.

PG: Processing gain.

P: Power of the received signal.

$P_b^{(i)}$: Probability of bit detection error for System i.

$P_c^{(i)}$: Probability of packet detection error for System i.

$R_b^{(i)}$: Transmission bit rate for System i.

S_i: Net throughput for System i.

SNR: Signal to total noise/interference ratio.

T_F: Frame length.

T_S: Slot Length.

T_C: Chip length.

T_b: Bit length.

T_i^d: Mean delay for data in System i.

T_i^v: Mean delay for voice in System i.

T_m: Delay spread in the multipath.

U: Number of connected users sharing the channel.

W: Channel bandwidth.

$c_{m,n}$: Code chip identity of the n'th path of the m'th user.

$g_{n,m}$: Complex gain of the n'th multipath scatterer of the m'th user signal.

$n(t)$: Received AGWN waveform.

$r(t)$: Received signal waveform.

x: Processing gain for System 5.

α: Voice activity ratio.

$\beta_{m,n}$: Real part of the complex gain, line of sight component.

γ: An adaptability factor for System 5.

λ: Arrival rate of messages per unit time.

μ_i: Service rate (in slots per frame) for System i.

ω_c: The carrier frequency.

ρ: Traffic load equivalent to that of System 1.

ρ_i: Actual traffic load on System i.

$(\rho_i)'$: Net throughput for System i (same as S_i).

$\tau_{m,n}$: A random number modeling the delay in the n'th path of the m'th user.

$\theta_{m,n}$: A random number modeling the phase in the n'th path of the m'th user.

$(\bar{\xi}^2{}_n)'$: The average power in the n'th faded multipath.

$(\bar{\xi}^2{}_0)'$: The average power in the Line of sight component.

ζ: An adaptability factor for System 4.

4.3 Descriptions of the Five Multiple Access Systems Considered

4.3.1 System Assumptions

All five Systems are assumed to have the same initial message arrival rate per user, λ, in messages per time unit, the same allowable transmission power on the channel P, the same average number of packets per message $\bar{E}$, and same mean square value $\bar{E}^2$. As was mentioned above, the total number of users, the source bit rate, and the channel fading characteristics are also the same for all systems. A frame length, T_F, is defined to be the time it takes for all users to transmit a packet each for TDMA, and will be used as unit of measurement for delay. For each of the five systems, let λ_i be the total user arrival rate in packets per user per frame, and μ_i the service rate in slots per user per frame. Then the effective (or equivalent) frame length is $T_F^{(i)} = T_F/\mu_i$. The definition of such an effective frame is necessary since μ_i is different among the systems introduced.
The single-user utilization factor (or traffic intensity) is given by

$$\rho_i = \lambda_i/\mu_i. \tag{4.1}$$

For TDMA, this is given by

$$\rho_1 = \lambda T_F \bar{E} \tag{4.2}$$

this entity will be defined as the traffic load on all systems (for fair comparison) and will be noted by ρ and assumed to be less than 1.

Fading and Additive White Gaussian Noise (AWGN) result in bit detection errors at the receiver which lead to occasional packet errors. If a packet is incorrectly demodulated (an event with probability (1-$P_c^{(i)}$) to be found shortly), it has to be retransmitted until success occurs. This can be accounted for in the analysis by increasing the mean message length from $\bar{E}$ to $(\bar{E}_i)'$, where

$$(\bar{E}_i)' = \sum_{k=1}^{\infty} k(1 - P_c^{(i)})^{k-1} P_c^{(i)}.\bar{E} = \frac{\bar{E}}{P_c^{(i)}}. \tag{4.3}$$

Equation (4.3) assumes a geometric distribution for the packet successful trial on the channel and $1/P_c^{(i)}$ is the average number of those trials. The underlying assumption here is that no timeout between retransmissions is needed or it is so small that it can be neglected. The NACK (negative acknowledgment) is received instantly and thus retransmission occurs instantly after the end of the packet. This allows us to ignore (in this chapter) the effect of different ARQ strategies on the queueing, and to assume that the only effect of retransmissions is an increase in the message length.

It follows from (4.3) that the useful throughput per user is

$$(\rho_i)' = \rho_i P_c^{(i)}. \tag{4.4}$$

The network throughput S_i and total traffic (input plus retransmission traffic) G_i are given by

$$G_i = \rho_i \tag{4.5}$$

and

$$S_i = \rho_i P_c^{(i)} = (\rho_i)'. \tag{4.6}$$

The mean square effective message length $(\bar{E^2}_i)'$ is also related to the corresponding original value as

$$(\bar{E^2}_i)' = \sum_{k=1}^{\infty} k^2(1 - P_c^{(i)})^{k-1} P_c^{(i)} \bar{E^2} = \frac{2 - P_c^{(i)}}{(P_c^{(i)})^2} \bar{E^2}. \tag{4.7}$$

Assuming independent bit errors, we obtain

$$P_c^{(i)} = (1 - P_b^{(i)})^L \tag{4.8}$$

where $P_b^{(i)}$ is the probability of bit error of a typical user of System(i) using Direct Sequence Phase Shift Keying modulation (DS/PSK) in the different fading environments involved with M_i overlapping users in each slot. The value of M_i will also be

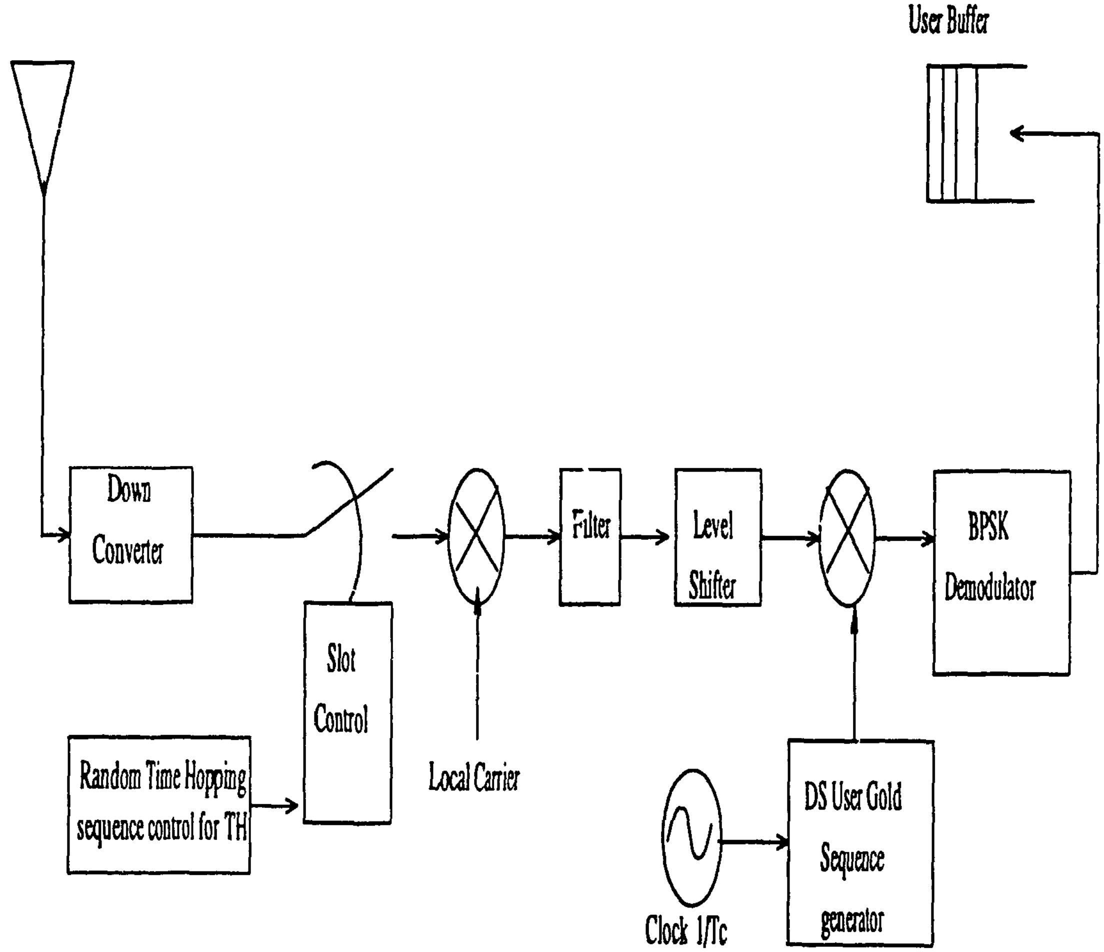

Figure 4.1: A Block Diagram of a Receiver in all Five Systems

determined below as a function of traffic intensity for all systems.

At the receiver, shown in Fig. 4.1, which is representative of the five systems, the received multiaccess signal plus noise in one of the five systems is received, down-converted, switched back according to the type of system (CDMA, TDMA,...), then carrier recovery and demodulation take place. Forward error correction (FEC) follows this step but this aspect is not covered in this chapter as all coding is used for error detection for retransmission of data packets while no error correction takes place even for voice which cannot be retransmitted. This might imply an inherent advantage to data traffic as we do not take into account the loss of efficiency (compared to voice) due to decreased coding rate while we assume perfect error detection (something that cannot happen theoretically without infinite system complexity). Furthermore, we can use the same coding rate for voice to achieve a very good error correction rate. This discrepancy will be accounted for in the next chapter, where we will assume the same coding rate for voice and data and use FEC for voice and retransmissions for data.

Another issue of importance is whether the bit errors are actually independent. Many studies in the literature dealt with this problem and showed that the bit errors are highly dependent when fading occurs except when perfect interleaving is used. In our systems we assume that interleaving is good enough to allow us to assume independent error bits.

4.3.2 Performance Parameters of the Five Systems

In the TDMA case (System(1), Fig. 4.2), each user is assigned one slot of width $T_S^{(1)}$ ($T_S^{(1)} = T_F^{(1)}/U$) to transmit a packet. For TDMA, by definition, $T_S^{(1)} = T_S$ and $T_F^{(1)} = T_F$. The transmission bit rate is then

$$R_b^{(1)} = L/T_s = (UL)/T_F, \tag{4.9}$$

where L is the number of bits in a packet.

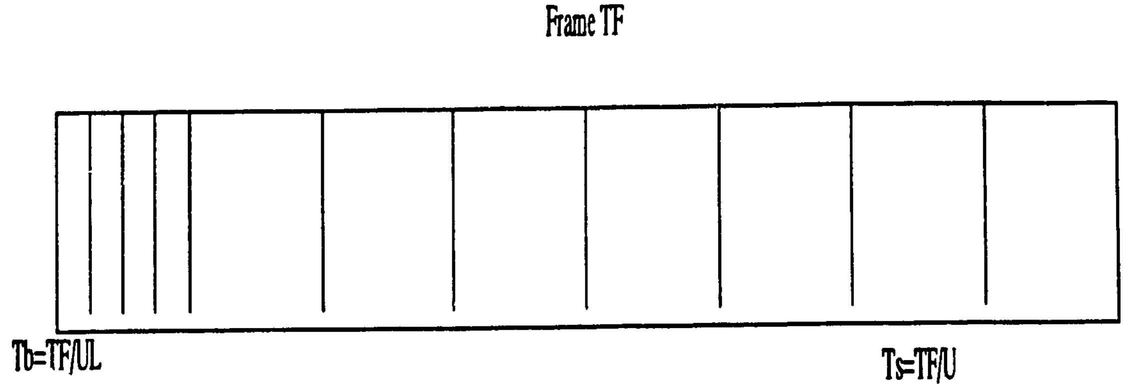

Figure 4.2: Transmission Diagram of U Users in a TDMA Frame (L=4, U=8).

The user total arrival rate (in packets per time unit per user) is given by

$$\lambda_1 = \lambda T_F \bar{E} \tag{4.10}$$

while the service rate is

$$\mu_1 = 1. \tag{4.11}$$

$\bar{E}$ is the average number of packets (data units) in each message, and T_F is the frame length in time units.

In many published articles, $P_c^{(1)}$ has been assumed to be 1, which gives TDMA an unfair advantage over CDMA systems. The narrow bits of TDMA systems and the lack of spread spectrum processing gain, gives rise to more severe intersymbol interference problems, even if the users are confined to their own TDMA slots as will be seen shortly.

In the CDMA case (System(2), Fig. 4.3), active users transmit during the whole length of the frame T_F. Each user still transmits one packet per frame to support the same source bit rate, giving rise to a transmission bit rate $R_b^{(2)}$ and slot

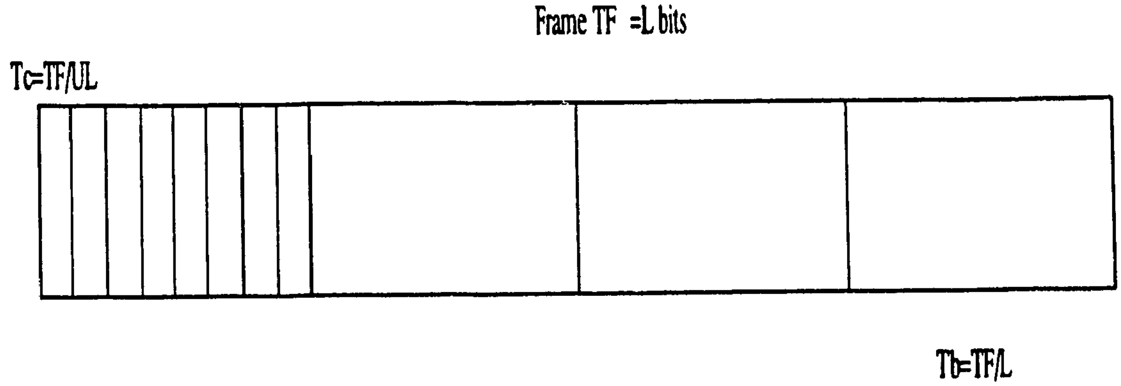

Figure 4.3: Transmission Diagram of U Overlapping Users in a CDMA

duration ($T_S^{(2)} = T_F$)

$$R_b^{(2)} = L/T_F = 1/T_b^{(2)} \tag{4.12}$$

For System(2) (and also 3, 4, and 5), source data bits served at the channel rate $R_b^{(2)}$ above, will be "XOR"ed with the Direct Sequence user signal chips generated by linear maximal length or Gold Codes [54] whose chip duration is T_C, where

$$T_C \approx 1/W. \tag{4.13}$$

The resulting signal modulates the carrier ($\cos \omega_c t$), after which upconversion and transmission take place. We note in System(2) that the transmission switch is ON all the time. The spread spectrum processing gain is defined as

$$PG_2 = T_b^{(2)}/T_C = 2^{n_2} - 1 \approx U \tag{4.14}$$

where n_2 is the length of the user feedback shift register generator and where Gold codes have been assumed [54].[1] Similar to the case of System(1), the per user input

[1]The approximation in Equation (4.14) is due to the fact that the maximum number of Gold sequences with period $2^n - 1$ is $2^n + 1$.

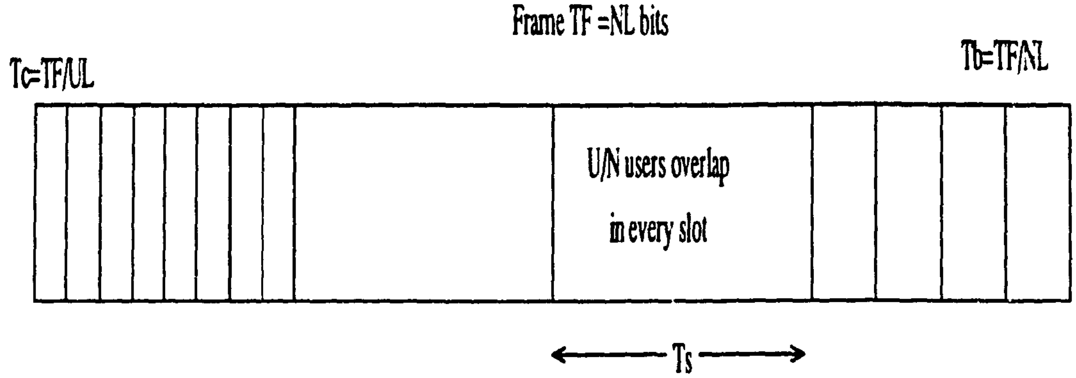

Figure 4.4: Transmission Diagram of U Users in a Hybrid CDMA/TDMA (U/N users share the same slot if active, N=4)

traffic, service rate, utilization, etc.. are defined as

$$\lambda_2 = \lambda T_F \bar{E}, \tag{4.15}$$

$$\mu_2 = 1, \tag{4.16}$$

$$\rho_2 = \frac{\lambda_2}{\mu_2} = \lambda T_F \bar{E} = \rho, \tag{4.17}$$

$$(\rho_2)' = \rho P_c^{(2)}, \tag{4.18}$$

and

$$S_2 = G_2 P_c^{(2)} = (\rho_2)' \tag{4.19}$$

where the number of simultaneous users in this case is[2]

$$M_2 = \lceil \rho U \rceil. \tag{4.20}$$

In the Hybrid CDMA/TDMA case (System(3), Fig. 4.4), the frame is divided into N slots, where $N \leq U$. Each of the (U/N) users is assigned the same slot ($T_S^{(3)} =$

[2]Note that $\lceil x \rceil$ denotes the smallest integer greater than or equal to x.

T_F/N), and each user transmits L bits in his assigned slot (using a DS/PSK signal). Similarly to Systems (1) and (2), it is easy to see that

$$R_b^{(3)} = L/T_S = NL/T_F = 1/T_b^{(3)}, \tag{4.21}$$

$$PG_3 = T_b^{(3)}/T_C = U/N = 2^{n_3} - 1, \tag{4.22}$$

$$\rho_3 = \lambda_3/\mu_3 = \rho, \tag{4.23}$$

$$(\rho_3)' = \rho P_c^{(3)} = \lambda T_F \bar{E} P_c^{(3)}, \tag{4.24}$$

$$\mu_3 = 1, \tag{4.25}$$

and

$$M_3 = \lceil \rho U/N \rceil. \tag{4.26}$$

The transmitter is the same as in previous cases save for the change in transmission bit rate ($R_b^{(3)}$), and the switch control (each user transmits in only one of N slots as above).

At this point, we recall that the source bit rate for the first three systems is not necessarily the same as the transmission bit rate in each case ($R_b^{(i)}$). However, this source rate reflects itself in λ_i and, consequently, ρ_i, and so is solely represented by these entities.

In the hybrid adaptive CDMA/TH system (System(4), Fig. 4.5), each CDMA user randomly selects one out of N' frame slots for transmission where N' depends on the status of network traffic. When ρU is low, N'=1 and each active user transmits exactly N packets per N-slot frame in this case. T_F and N are as given for System 3, yielding a slot size of $T_S^{(4)} = T_F/N$. Furthermore parameters such as $R_b^{(4)}$,T_C, U, L and synchronization of users to the frame are the same as those of System(3), but the networking decision as to how frequently the user transmits within this frame is adaptively controlled by a strong control broadcast message transmitted periodically from one of the user stations.

This arrangement is equivalent to flow control messaging in a separate channel (out of band signalling) typically used in terrestrial networks. This centrally located

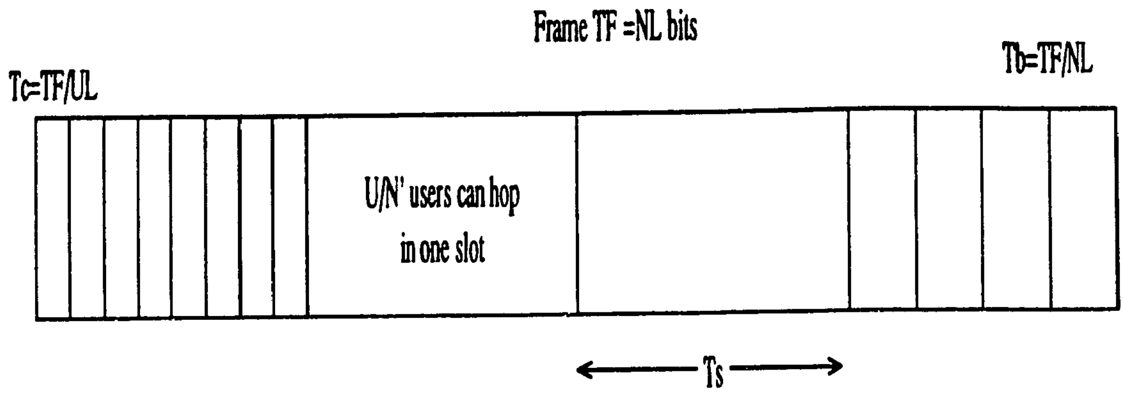

Figure 4.5: Transmission Diagram of U Users in Hybrid CDMA/TH. Parameters T_C , $R_b^{(1)}$, T_F, N, L, and U remain exactly the same as in Fig1.c. However each user transmits 1 out of N' slots, where N' is proportional to traffic (ρU).

station listens to all traffic, estimates the load, and broadcasts the updated value of N' to all stations. At high traffic, N' grows up to a maximum of N'=N, in which case we return back to the previous CDMA/TDMA case. Great adaptability to traffic is well achieved by the adaptive effective frame N' notion above.

As an example, when few users (1 or 2) are active and they have long files, they can access the channel faster by having $N'=1$, i.e by continuously transmitting. The delay is higher in the corresponding TDMA or CDMA/TDMA systems for this level of traffic as TDMA still uses one slot per frame per user and CDMA still sends one packet per frame per users, while CDMA/TH transmits μ_4 packets per frame. In the mean time, $P_c^{(4)}$ is sufficiently high for this low traffic to justify the continuous transmission mode and overlap of all active users in all slots.

The penalty that the user pays in this adaptive CDMA/TH case is more delay jitter (delay variance) compared to the TDMA and CDMA/TDMA systems, because of the randomness of slot selection and service. For example, CDMA/TH might switch from 4 slots/user/frame to 2 slots/user/frame making service for the same number of packets suddenly last twice as long. Also, unlike the CDMA/TDMA case, the source and destination addresses have to be incorporated in each packet, leading to more overhead loss.

The transmitter of the CDMA/TH case is still similar with a difference in the value of $R_b^{(4)}$ which will be found shortly, and the switch control mechanism which was explained in the previous paragraph.

Based on the above description of System(4), the other parameters can be easily derived as follows:

$$R_b^{(4)} = L/T_S^{(4)} = NL/T_F = 1/T_b^{(4)} \tag{4.27}$$

$$PG_4 = T_b^{(4)}/T_C = U/N = 2^{n_4} - 1 \tag{4.28}$$

and

$$N' = \lceil \zeta \rho N \rceil \tag{4.29}$$

where N' is the transmission window size, i.e, the user transmits one slot every N' slots, and ζ is an adaptability parameter $0 < \zeta \leq 1$. The total user traffic per slot is

$$\rho_4 = \frac{\lambda T_F \bar{E}}{(N/N')} = \rho N'/N = \rho^2 \zeta \tag{4.30}$$

where

$$\mu_4 = N/N', \tag{4.31}$$

$$(\rho_4)' = \frac{\rho P_c^{(4)}}{(N/N')} = \rho^2 \, \zeta \, P_c^{(4)} \tag{4.32}$$

and

$$M_4 = \lceil \rho U/N' \rceil = \left\lceil \frac{U}{\zeta N} \right\rceil . \tag{4.33}$$

The source bit rate ($= (L/T_F) \times (N/N')$) seems to be higher than that of the first three systems (which was only L/T_F). However, the final throughput ($(\rho_4)'$ in Equation (4.32) above) will depend on the packet errors which depend on the number of overlapping users per slot (N/N'), thus providing for a self-adjusting

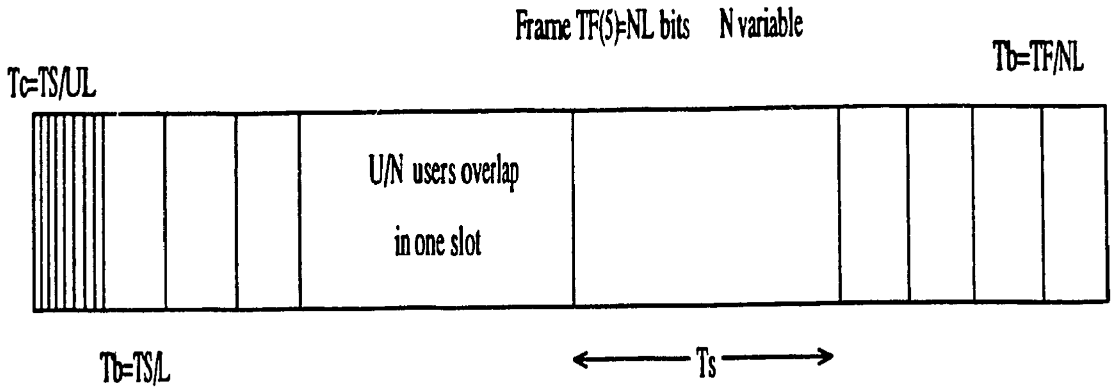

Figure 4.6: Transmission Diagram of U Users in Variable Frame CDMA/TDMA ($T_F'^{(5)} = NT_F$, $Nx\rho U = \gamma\rho U$).

mechanism for fair comparisons of the five cases. The service rate is larger than 1 since each user can transmit more than one packet per frame (on average, N/N' packets per frame per user are transmitted as opposed to one in the previous three cases). This gives rise to an effective frame $T_F^{(4)} = T_F/\mu_4 = T_F/(N/N')$, where each user transmits in only one slot within this equivalent frame. In the delay equation (Equation (4.45), to follow), $T_F^{(4)}$ will be used.

In movable frame CDMA/TDMA systems (System(5), Fig. 4.6), we have the frame expanding from one slot of width $T_S^{(5)} = xT_F/U$ (where x is a parameter to be suitably selected) to a maximum of N_5 such slots. Users listen to a powerful central flow control signal that broadcasts the updated length of the frame to all users. The allocation of users to slots for each traffic level is known *a priori* to all users. As with N' in System(4), let γ is an adaptability parameter $0 < \gamma \leq 1$, then N_5 adapts to the rising traffic according to $N_5 = \lceil \gamma\rho U/x \rceil$. At low traffic, all users share one repeated slot in a CDMA fashion with $PG_5 = x$.[3]

[3]With the source bit rate, the channel bit rate and the number of information bits per slot the same as in System(1), the expansion of the slot from T_S in System(1) to xT_S in System(5)

As the traffic builds up the users are split into a number of groups N_5 proportional to the level of traffic. Defining System(5) parameters in this case, the effective frame (same as the real frame in this case) is given by

$$T_F{}^{(5)} = N_5 T_S^{(5)} = \lceil \gamma \rho U/x \rceil T_S^{(5)} = x \lceil \gamma \rho U/x \rceil T_F/U = \gamma \rho T_F \qquad (4.34)$$

Moreover,

$$PG_5 = x = 2^{n_5} - 1 \qquad (4.35)$$

and

$$R_b^{(5)} = 1/T_b^{(5)} = W/x. \qquad (4.36)$$

We note that the processing gain PG_5 and the data rate $R_b^{(5)}$ are invariant to the load. The rest of the parameters are as follows:

$$\rho_5 = \lambda T_F{}^{(5)} \bar{E}/\mu_5 = \lambda N_5 T_S^{(5)} \bar{E} = \frac{\lambda x N_5 T_F}{U} \bar{E} = \lambda \gamma \rho T_F \bar{E} = \gamma \rho^2 \qquad (4.37)$$

where

$$\begin{aligned} \mu_5 &= 1, & (4.38) \\ (\rho_5)' &= x \rho N_5 P_c^{(5)}/U = \gamma \rho^2 \; P_c^{(5)}, & (4.39) \end{aligned}$$

and

$$M_5 = \lceil \rho U/N_5 \rceil = \lceil x/\gamma \rceil \qquad (4.40)$$

We also note that N_5 is restricted to the values $1, 2, ..., \lceil U/x \rceil$, and that System(5) does not suffer from the source and destination overhead associated with each packet (as is the case for System(4)), nor to delay jitter due to time hopping.

Now having found the traffic and queueing parameters of each of the systems, we note that the five cases, though different, obey a general frame structure of different number of slots, different ρ_i's, $T_S^{(i)'}$s, $T_F^{(i)'}$s, etc. This leads us to try to model all five cases by an effectively equivalent TDMA network as will follow shortly.

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allows a ratio of bit rate to chip rate of $PG_5 = xT_S/T_S = x$ which is the number of DS chips per information bit.

The total message transfer delay can be found by substituting the appropriate traffic service rate, utilization, slot size, and message statistics of each case into the general TDMA delay formula. This is given by [55],[56],

$$T_i = \frac{\rho_i T_F^{(i)} (\bar{E}^2{}_i)'}{2(1-\rho_i)(\bar{E}_i)'} + \frac{2(\bar{E}_i)' - 1}{2\mu_i} T_F^{(i)} + T_S^{(i)} \tag{4.41}$$

where ρ_i is the total traffic intensity (including retransmissions) and is assumed to be variable, and the third term is the channel transmission time (part of service time) of one packet, while the second is the number of frames necessary to transmit a message of $\bar{E}$ data units, with μ_i units (packets) served per user per frame. The first component is the M/G/1 mean waiting delay of all messages in the user buffer while waiting to be served. One has only to substitute the appropriate values of $(\bar{E}^2{}_i)'$, $(\bar{E}_i)', \rho_i, \mu_i$, $T_S^{(i)}$, and $T_F^{(i)}$ for each case in (4.41).

For the TDMA system, substituting (4.3) and (4.7) into (4.41) and noting that $T_S^{(1)} = T_F^{(1)}/U$ and $T_F^{(1)} = T_F/\mu_1$, for the data traffic, we get

$$T_1^d = \frac{\rho T_F (2 - P_c^{(1)})}{2(1-\rho) P_c^{(1)}} \frac{\bar{E}^2}{\bar{E}} + \frac{2\bar{E} - P_c^{(1)}}{2P_c^{(1)}} T_F + T_F/U. \tag{4.42}$$

For System(2) (CDMA), we substitute 4.3 and 4.7 for i=2, substitute (4.16) and (4.17) into (4.41), and note that $T_S^{(2)} = T_F, T_F^{(2)} = T_F$ so that

$$T_2^d = \frac{\rho T_F (2 - P_c^{(2)}) \bar{E}^2}{2(1-\rho) P_c^{(2)} \bar{E}} + \frac{\bar{E} T_F}{P_c^{(2)}}. \tag{4.43}$$

For System(3), (CDMA/TDMA), we substitute (4.3), (4.7), (4.23) and (4.25) into (4.41) and note that $T_S^{(3)} = T_F/N, and T_F^{(3)} = T_F$, to obtain

$$T_3{}^d = \frac{\rho T_F (2 - P_c^{(3)}) \bar{E}^2}{2(1-\rho) P_c^{(3)} \bar{E}} + \frac{(2\bar{E} - P_c^{(3)}) T_F}{2P_c^{(3)}} + T_F/N. \tag{4.44}$$

For System(4) (Adaptive CDMA/TH), we substitute (4.3), (4.7), (4.30) and (4.31) into (4.41) and note that $T_S^{(4)} = T_F/N, T_F^{(4)} = T_F/\mu_4 = \zeta \rho T_F$, obtaining

$$T_4{}^d = \frac{\rho^3 \zeta^2 T_F (2 - P_c^{(4)}) \bar{E}^2}{2(1-\rho^2\zeta) P_c^{(4)} \bar{E}} + \frac{(2\bar{E} - P_c^{(4)}) \zeta \rho T_F}{2P_c^{(4)}} + T_F/N. \tag{4.45}$$

For System(5), (variable frame CDMA/TDMA), we substitute (4.3), (4.7), (4.37) and (4.38) into (4.41) and note that $T_S^{(5)} = xT_F/U, T_F^{(5)} = \gamma\rho T_F$ so that

$$T_5{}^d = \frac{\gamma^2\rho^3 T_F(2 - P_c^{(5)})\bar{E}^2}{2(1-\gamma\rho^2)P_c^{(5)}\bar{E}} + \frac{(2\bar{E} - P_c^{(5)})\gamma\rho T_F}{2P_c^{(5)}} + xT_F/U. \qquad (4.46)$$

The analysis so far dealt with data transmission only. Since digital voice transmission is not uncommon, and packet retransmission does not take place in voice (faded packets are lost), all $P_c^{(i)}$'s are replaced by one in (4.42) to (4.46), leading to the following voice queueing delays[4]:

$$T_1{}^v = \frac{\rho T_F \bar{E}^2}{2(1-\rho)\bar{E}} + (\bar{E} - 1/2)T_F + T_F/U, \qquad (4.47)$$

$$T_2{}^v = \frac{\rho T_F \bar{E}^2}{2(1-\rho)\bar{E}} + \bar{E} \times T_F, \qquad (4.48)$$

$$T_3{}^v = \frac{\rho T_F \bar{E}^2}{2(1-\rho)\bar{E}} + (\bar{E} - 1/2)T_F + T_F/N, \qquad (4.49)$$

$$T_4{}^v = \frac{\rho^3\zeta^2 T_F \bar{E}^2}{2(1-\rho^2\zeta)\bar{E}} + (\bar{E} - 1/2)\zeta\rho T_F + T_F/N, \qquad (4.50)$$

$$T_5{}^v = \frac{\gamma^2\rho^3 T_F \bar{E}^2}{2(1-\gamma\rho^2)\bar{E}} + (\bar{E} - 1/2)\gamma\rho T_F + xT_F/U \qquad (4.51)$$

In (4.47) to (4.51), the same assumptions for the M/G/1 queue made earlier were assumed. However delay jitter is more annoying for voice communication (CDMA/TH case). Moreover, portions of the voice traffic are lost because of reception errors. This we estimated (in all five systems) by

$$P_\ell^{(i)} = G_i(1 - P_c^{(i)}), \quad i = 1,2,3,4,5. \qquad (4.52)$$

This loss leads to a corresponding service signal to noise ratio degradation at the receiver.

It is worthwhile to recall here (see also Chapter 2) that CDMA has more useful features than were exploited in this chapter. As an example, automatic frequency

[4]we assume that voice packets can be queued and that packets that are delayed by more then a certain amount (determined in higher layers) are considered lost. It will be shown in Chapter 5 that the average delay for voice can be kept at very reasonable levels

assignment in CDMA satellite networks yields an improvement factor by $\alpha = 3$ (for example) in ρ. Also voice traffic is bursty in nature, and the activity ratio, a, is around 0.30 to 0.6 depending on modulation and bandwidth compression details (a=1 for data traffic). The designer may take these issues into consideration in Equations (4.42) to (4.51) simply by substituting a reduced $\rho_r = \rho a/\alpha$ in all equations except those of the TDMA system (Case 1) where a=1 for data traffic.

4.3.3 Evaluation of the Probability of Correct Packet Reception For The Five Multiaccess Systems

In the previous queueing discussion, we have implicitly assumed that the users are synchronized to the frame start (by means of frame headers) in all five systems. Also, we have neglected the throughput loss due to frame and packet overheads. While the frame overhead is of the same order in all five systems, the packet overhead is more elaborate in the CDMA/TH system because of the presence of source and destination addresses that are not needed in TDMA-related networks (Systems 1,2,3), since users are there assigned to slots on a permanent basis (unlike TH where user packets hop randomly everywhere within the frame). For modulation purposes, we assume perfect carrier synchronization. Also perfect code acquisition and fine synchronization are assumed in all CDMA systems (Systems 2,3, and 4). In the following, we treat the problem of evaluating the probability of bit errors in a Direct Sequence PSK multiaccess network under multipath frequency selective fading [27]. Later we derive the various probabilities of correct bit and packet reception, previously introduced but not yet evaluated. We also notice that from the receiver analysis point of view, it does not matter if the system is CDMA, CDMA/TDMA, or CDMA/TH as long as we substitute the right number of active users per slot $(1, [\rho U], [\rho U/N], [\rho U/N'], [x/\gamma])$ and use the data rates $R_b^{(1)} = W$, $R_b^{(2)} = W/U$, $R_b^{(3)} = WN/U$, $R_b^{(4)} = WN/U$, and $R_b^{(5)} = W/x$ and the processing gains $PG_1 = 1$, $PG_2 = U$, $PG_3 = U/N$, $PG_4 = U/N$, and $PG_5 = x$. The thermal bit SNR = $(2E_b^{(i)}/N_0)$, the number of multipath components L, the energy in each multipath component $(\xi_n^2)E_b^{(i)}$, the traffic S = ρ , the number of users, U, the frame length T_F, the total channel bandwidth W, and the multipath spread T_m, are all assumed to be the same for the five systems discussed.

With this introduction, we next evaluate $P_b^{(i)}$, and $P_c^{(i)}$, for the various systems. The received faded CDMA signal is given by

$$r(t) = n(t) + \sum_{m=1}^{M_i} \sum_{n=0}^{L_m} \sqrt{2P} d_{m,n}(t) c_{m,n}(t)\, \Re \left(g_{m,n} U(t - \tau_{m,n}) e^{j\omega_c(t-\tau_{m,n})} \right) \quad (4.53)$$

where $(j-1)T_b^{(i)} \leq t \leq jT_b^{(i)}$ is the information bit interval of a typical user in System(i) (i=1→5), and n(t) is AWGN of density $N_0/2$,

M_i is the number of CDMA overlapping users in one slot, as defined earlier for all five systems.

$\sqrt{2P}$ is the signal amplitude of all CDMA users (no near-far problems),

$L_m = (T_m/T_C)$ is the number of DS chips spanned by the given channel multipath spread of T_m.

ω_c is the carrier frequency,

$\tau_{m,n}$ is the random delay of the m'th interfering CDMA user, assumed uniformly distributed in the range of $\{0,T_b^{(i)}\}$, except that $\tau_{j,0} = 0$,[5]

$\theta_{m,n}$ is the random phase of the n'th multipath component of m'th user signal, assumed uniformly distributed in the range $0,2\pi$ except that $\theta_{j,0} = 0$,

$c_{m,n}(t)$ is the code chip identity (-1,1) of duration $T_C = 1/W$ of the n'th path of the m'th user signal overlapping the intended user bit,

$d_{m,n}(t)$ is the n'th path of the m'th user input data bits overlapping the intended received data bit, in the general j'th time interval, (j-1)$T_b^{(i)} \leq t < jT_b^{(i)}$, j=1,2...,∞, i=1,..,5, and

$g_{n,m}$ is the complex gain of the n'th multipath scatterer of the m'th user signal, which could be either Rayleigh or Rician distributed with an average power of E(ξ_n^2).

[5]The intended user line of sight signal component is assumed to be well tracked in time and carrier phase.

For example, for the main line of sight component of the m'th user interfering signal,

$$g_{j,0} = \beta_{j,0} + \xi_0 e^{j\theta_{m,0}} \tag{4.54}$$

while for the n'th scatterer

$$g_{m,n} = \xi_n e^{j\theta_{m,0}} \quad n = 1, ..., L_m \tag{4.55}$$

and E(ξ_n^2) is the average power of the n'th scatterer referenced to that of the line of sight component which has a mean square value E(β^2)=1 (in the Rician case) and does not exist for the Rayleigh case ($\beta_{0,0} = 0$).. The random variables $\xi_n, \theta_{m,n}$, and $\tau_{m,n}$ are assumed to be independent and are assumed to be slowly varying to justify a constant value for each bit duration.

At the j'th receiver (see Fig. 4.1 and Fig. 4.7), upon carrier recovery, correlation, and perfect despreading by the j'th user code (in spite of fading) one obtains [24]

$$Z = \int_0^{T_i} r_j(t) \cos(\omega_c t) C_{j,0}(t) dt. \tag{4.56}$$

$$Z = n_0 + \sqrt{P/2} \left\{ \beta_{j,0} d_{j,0} T_i + \sum_{m=1}^{M_i} \sum_{n=0}^{L_m} \gamma_{m,n} \cos(\phi_{m,n}) \left[d_{-1,m,j} R_{m,n}(\tau_{m,n}) + d_{0,m,n} \hat{R}_{m,j}(\tau_{m,n}) \right] \right\} \tag{4.57}$$

and

$$\phi_{m,n} = \theta_{m,n} - \omega_c \tau_{m,n} \tag{4.58}$$

where $d_{1,m,j}$ and $d_{0,m,j}$ are the previous and current bits of the n'th path of the m'th user signal and R(.), $\hat{R}$(.) are defined as the continuous partial crosscorrelations of the users' codes, i.e.,

$$R_{m,j}(\tau_{m,n}) = \int_0^{\tau_{m,n}} c_m(t - \tau_{m,n}) c_j(t) dt \tag{4.59}$$

and

$$\hat{R}_{m,j}(\tau_{m,n}) = \int_{\tau_{m,n}}^{T_i} c_m(t - \tau_{m,n}) c_j(t) dt. \tag{4.60}$$

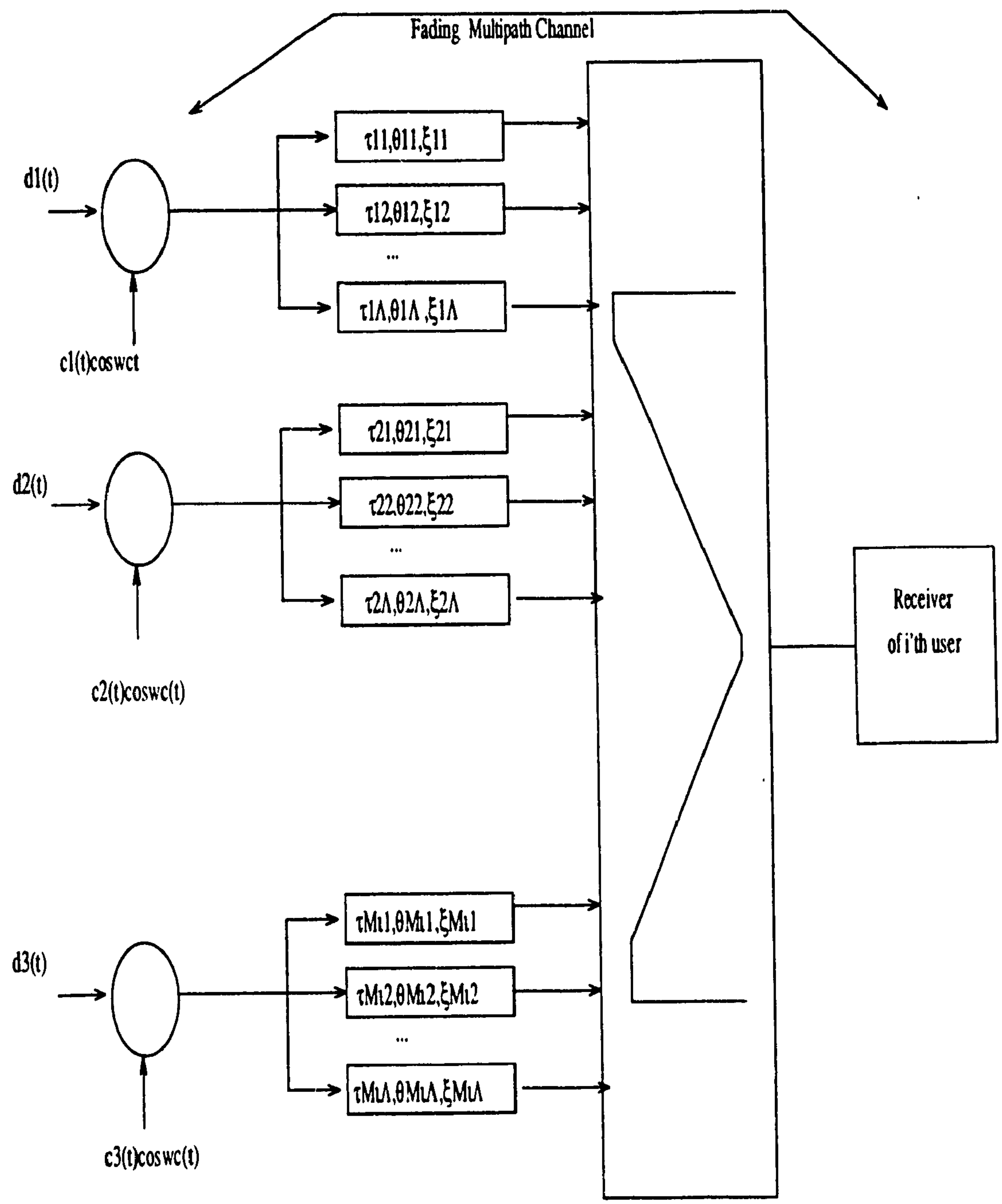

Figure 4.7: CDMA signals received in multipath fading Multiple Access Channel.

Evaluating the mean and variance of Z [27] and invoking the Gaussian approximation, one obtains the probabilities of bit error in the various fading cases and networks. For the CDMA-related Rician fading case, $\beta_{j,0} = 1$, one has

$$P_b^{(i)}(1) = (1/2)\mathrm{erfc}\sqrt{\mathrm{SNR}_i(1)} \tag{4.61}$$

where

$$\begin{aligned}\mathrm{SNR}_i \ \approx\ & \Big\{\Big(2(E_b^{(i)})'/N_0)^{-1} + (\bar{\xi^2}_0)'/2 + (1/2)(T_i^{\,2}(T_b^{(i)} - 2T_C)^{-1}\Big) \times \\ & \sum_{m=0}^{M_i} \sum_{n=0}^{L_m} (\bar{\xi^2}_n)' \int_{T_C}^{T_b^{(i)}-T_C} [R_{m,j}^2(\tau_{m,n}) + \hat{R}_{m,j}^2(\tau_{m,n}) d\tau\Big\}^{-1}\end{aligned} \tag{4.62}$$

Where $(E_b^{(i)})'$ is the bit energy of the line of sight component for System(i) and $(\bar{\xi^2}_n)'$ is the power in the n'th multipath component relative to the intended component $((E_b^{(i)})'/T_b^{(i)})$. For Gold codes, [24], [19], the square of the crosscorrelation can be expressed as

$$(R_{m,j}^2(\tau_{m,n}) + \hat{R}_{m,j}^2(\tau_{m,n})) = \frac{2(PG_i)^2 T_C{}^2}{3PG_i} = (2/3)PG_i(T_C)^2. \tag{4.63}$$

Substituting (4.63) into (4.62), we get

$$\mathrm{SNR}_i \approx \left\{(2(E_b^{(i)})'/N_0)^{-1} + (1/2)(\bar{\xi^2}_0)' + \frac{T_C{}^3 PG_i^2}{2(T_b^{(i)^2}(T_b^{(i)} - 2T_C))} \sum_{m=1}^{M_i} \sum_{n=0}^{L_m} (\bar{\xi^2}_n)'\right\}^{-1}. \tag{4.64}$$

Assuming equal powers for all multipath components and that $T_b^{(i)} \gg T_C$, this reduces to

$$\mathrm{SNR}_i(1) \approx \left\{(2(E_b^{(i)})'/N_0)^{-1} + (1/2)(\bar{\xi^2}_0)' + \frac{(L_m+1)M_i(\bar{\xi^2}_n)'}{3PG_i}\right\}^{-1} \quad i = 2,3,4,5. \tag{4.65}$$

In this case $(\bar{\xi^2}_0)'$ is the average power in the faded zero component of the intended signal and $(\bar{\xi^2}_n)'$ is the average power in the faded n'th component of the intended and interfering signal.

In the CDMA-related Rayleigh fading case, $\beta = 0$, $\bar{\xi^2}_n$ is the power of the n'th multipath component relative to the main intended fading signal power, $\bar{\xi^2}_0$=1, and

$E_b^{(i)}/N_0$ is the total thermal SNR in the absence of any multipath fading. Thus,

$$\mathrm{SNR}_i(2) = \left\{ \frac{L_m(M_i - 1)\bar{\xi^2}_n}{3PG_i} + (2E_b^{(i)}/N_0)^{-1} \right\}^{-1} . \; i = 2,3,4,5 \tag{4.66}$$

Equations (4.67) and (4.68) below relate the parameters of the Rayleigh and Rician fading cases.

$$E_b^{(i)} = (E_b^{(i)})' + (\bar{\xi^2}_0)'(E_b^{(i)})'/2 + L_m(\bar{\xi^2}_0)'(E_b^{(i)})'/2 \tag{4.67}$$

and

$$\bar{\xi^2}_n = \frac{(\bar{\xi^2}_n)'}{\left(1 + [(\bar{\xi^2}_0)' + L_m(\bar{\xi^2}_n)']/2\right)}. \tag{4.68}$$

The probability of bit error in this case is given by

$$P_b^{(i)}(2) = (1/2)\left(1 - \sqrt{\left[\frac{\mathrm{SNR}_i(2)}{(1 + \mathrm{SNR}_i(2))}\right]}\right). \tag{4.69}$$

In the fading free multiaccess case, we obtain [57] (also from Equation (2.15))

$$\mathrm{SNR}_i(3) = \left\{ (2E_b^{(i)}/N_0)^{-1} + \frac{M_i - 1}{3PG_i} \right\}^{-1} \quad i = 1,2,3,4,5 \tag{4.70}$$

$$P_b^{(i)}(3) = (1/2)\mathrm{erfc}\sqrt{\mathrm{SNR}_i(3)}. \tag{4.71}$$

As an example, for plain TDMA systems and Rician fading,

$$\mathrm{SNR}_1(1) \approx \left\{ (2(E_b^{(i)})'/N_0)^{-1} + (\bar{\xi^2}_0)'/2 + L_m(\bar{\xi^2}_0)'/2 \right\}^{-1} \tag{4.72}$$

and

$$P_b^{(1)}(1) = (1/2)\mathrm{erfc}\sqrt{\mathrm{SNR}_1(1)} \tag{4.73}$$

whereas for plain TDMA systems in Rayleigh fading,

$$\mathrm{SNR}_1(2) = \left\{ (2E_b^{(i)}/N_0) \right\} \tag{4.74}$$

and

$$P_b^{(1)}(2) = (1/2)\left[1 - \sqrt{\left[\frac{\mathrm{SNR}_1(2)}{1 + \mathrm{SNR}_1(2)}\right]}\right]. \tag{4.75}$$

It should be noted that we will use the same $E_b^{(1)}$ *or* $(E_b^{(1)})'$ to the CDMA and TDMA cases although this puts the CDMA related systems at a cost disadvantage. For example, in asynchronous TDMA satellite networks, each user uses the full channel power since his bit duration $T_b^{(1)}$ is less than that of the equivalent energy in CDMA networks (which have a wider $T_b^{(2)}$) Thus, a typical CDMA user uses much less power than its TDMA counterpart to keep E_b the same.

4.4 Numerical Results and Discussion of System Performance

4.4.1 An Initial Comparison Based on Arbitrary Values of γ and ζ

The packet error, delay, throughput, and voice packet loss were obtained assuming the same parameters for the five systems presented, namely, U=256 users, L=256 bits, W=1.28MHz, $T_m = 10^{-5}s$, $\bar{E} = 2$, $\bar{E^2} = 5$, and $T_F = 1s$. The signal to thermal noise ratio and fading parameters were $2E_b/N_0$=30dB, $\bar{\xi^2}_0 = 0.4$, and $\bar{\xi^2} = 0.02$. A slightly optimistic value for SNR is used (practical values are around 20dB), to stress the effect of multiple access interference, given yet another advantage to TDMA.

The delay and throughput as a function of traffic intensity was computed for all five systems (using Equations (4.42) to (4.51)) in the three fading cases (using Equations (4.65) to (4.71)) and assuming some initial arbitrary values for the adaptability parameters ζ and γ and for the processing gain of System(5): $\zeta = \gamma = 0.8$, $x = 20$, and $N = 8$ for the respective systems.

Fig. 4.8 shows the delay for data traffic for a fading-free environment while Fig. 4.9 shows the throughput. We notice that System(1) outperforms all other systems at medium to high traffic (ρ), both in terms of delay and useful throughput. This result is largely due to the high received signal to noise ratio and the lack of

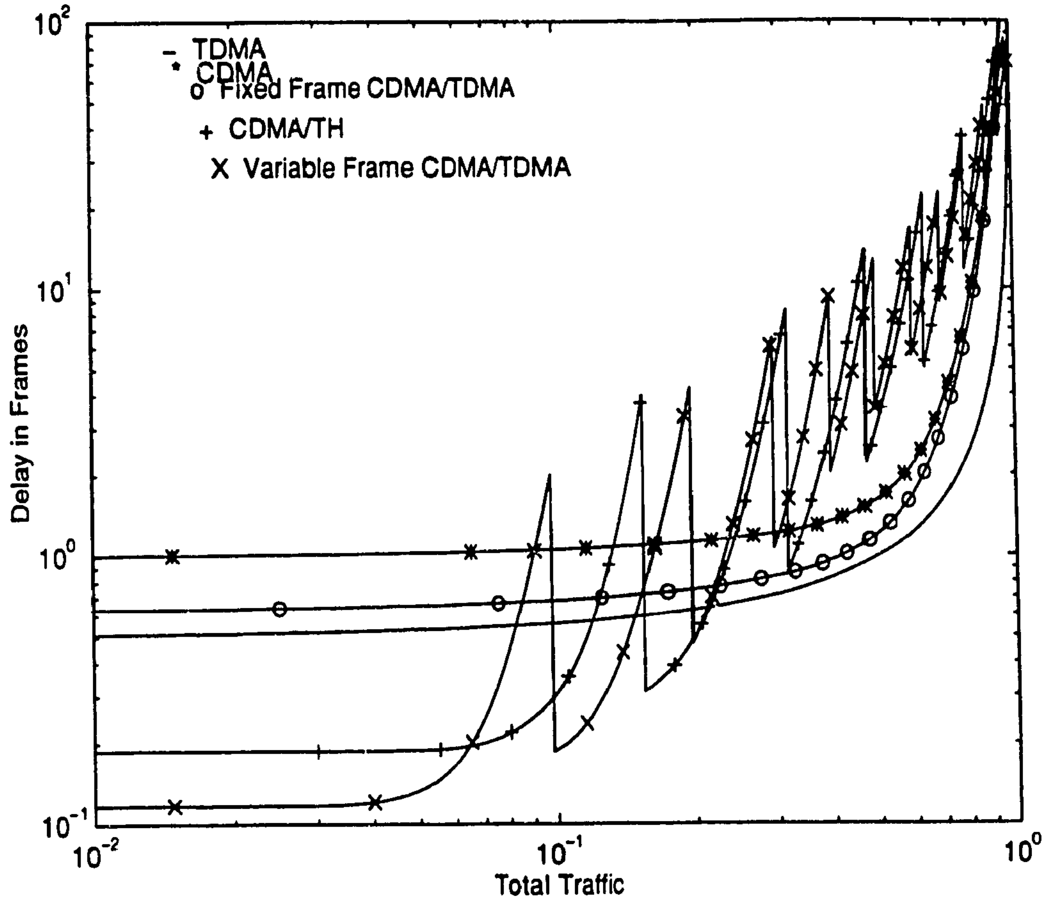

Figure 4.8: Delay for data Traffic, fading-free channel.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E}^2 = 5$, $\bar{E} = 2$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$.

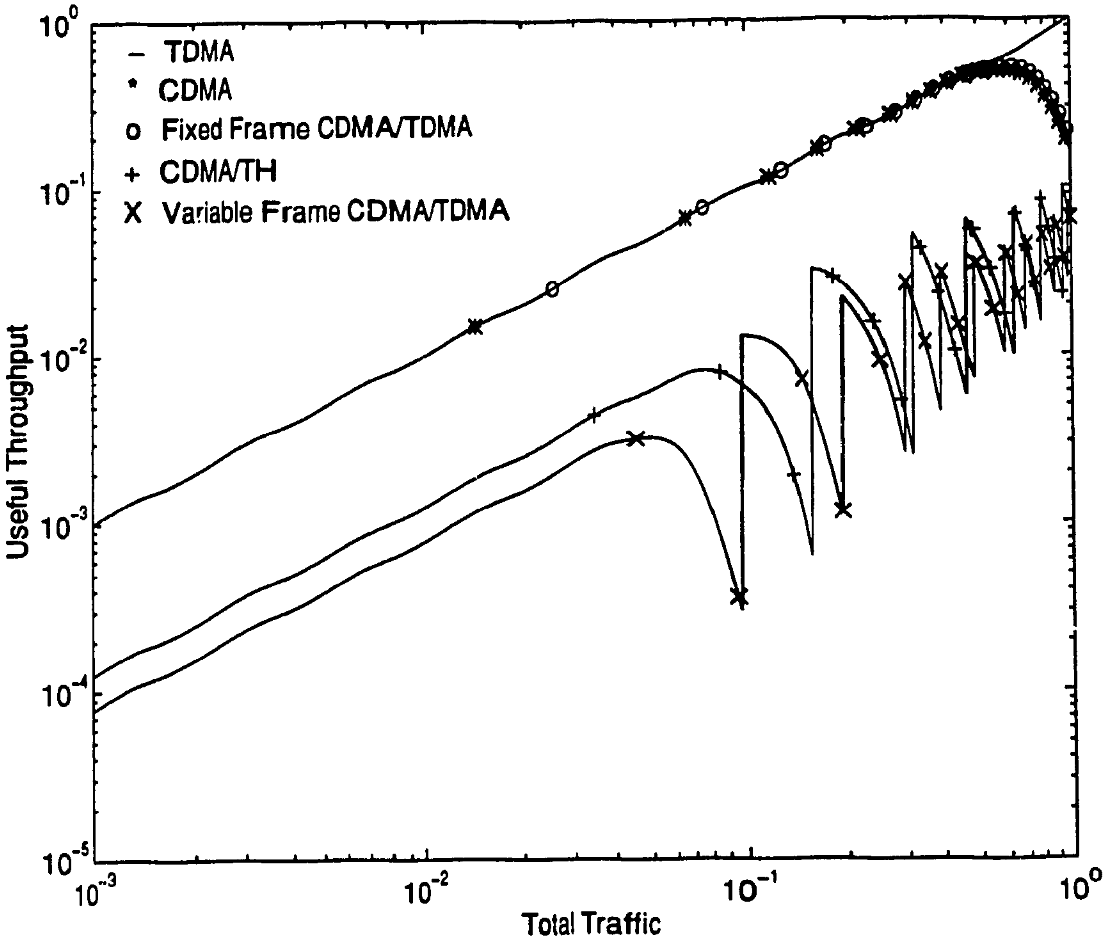

Figure 4.9: Throughput for data traffic, fading-free channel. $U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$.

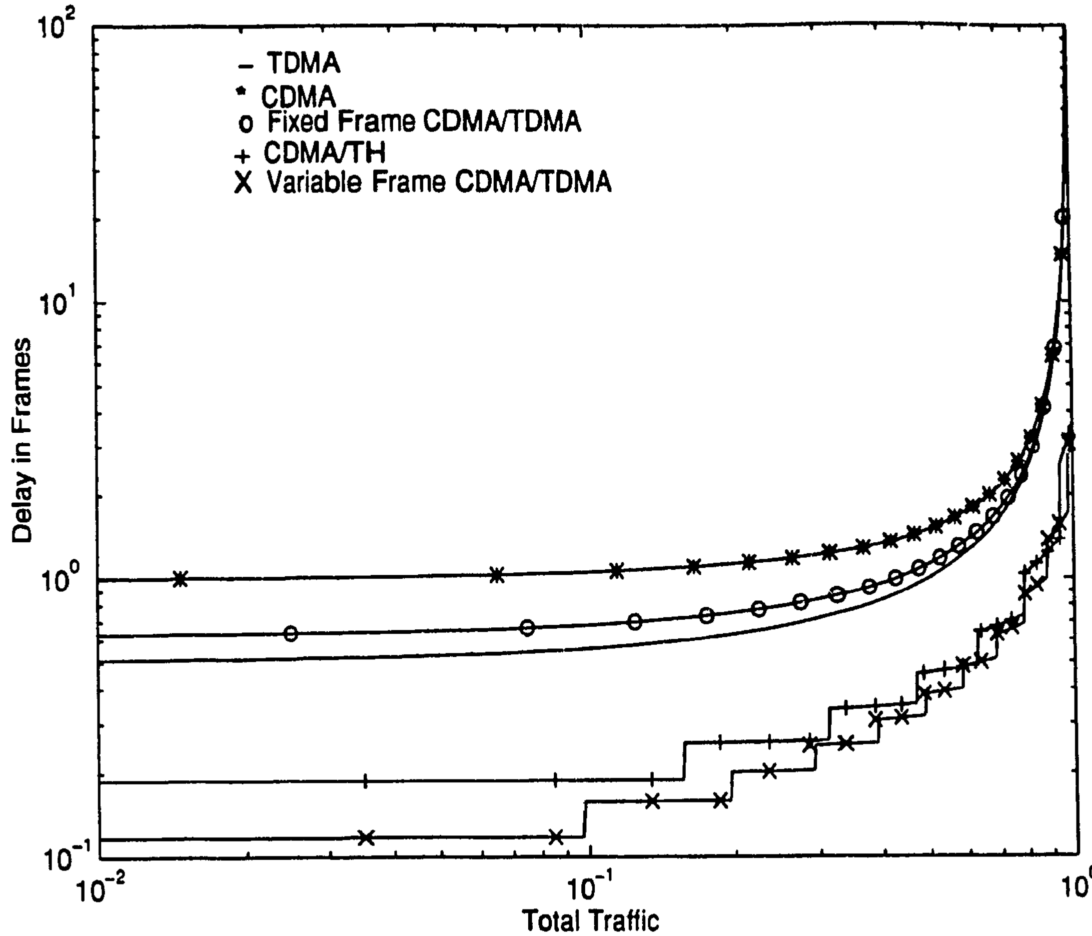

Figure 4.10: Delay for voice Traffic, All fading environments.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$.

overlapping users, which results in a very high packet success probability (close to 1 for all ρ).

One thing that is unusual about the above figures, is the "oscillating" behavior of Systems 4 and 5. This is actually not surprising and will be explained in Section 4.4.2 to follow. Fig. 4.10 shows the voice delay and Fig. 4.11 the voice packet loss for the same fading-free environment, although the delay is also identical in the other environments because there are no retransmissions.

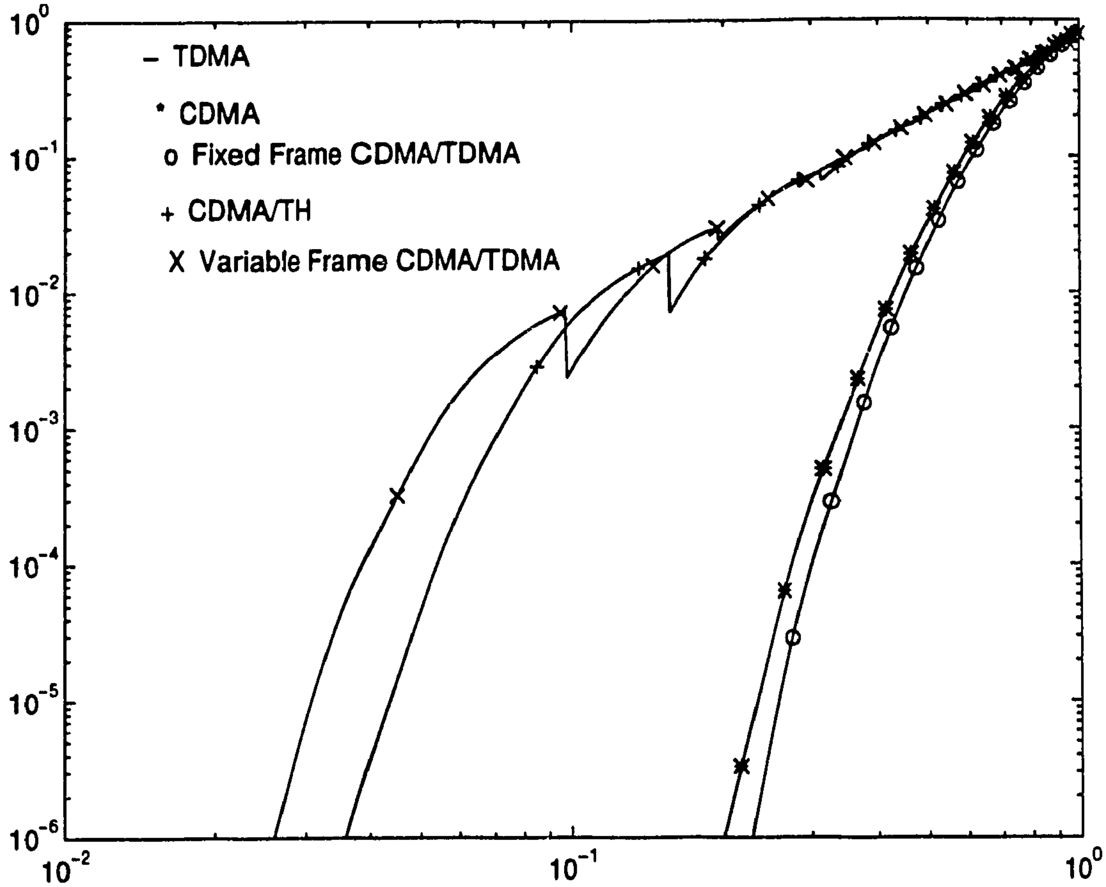

Figure 4.11: Loss of voice traffic, fading-free channel.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$.

Systems (4) and (5) (CDMA/TH and variable frame CDMA/TDMA respectively) offer the best delay performance at low traffic ($\rho < 0.2$). Figs. 4.8 and 4.9 also show clearly the adaptability of these CDMA systems. Before the delay becomes excessively high, the system adapts, changing N' for System(4), and N_5 for System (5). On the negative side however, Systems (4) and (5) suffer from low throughput as a result of the high probability of bit error experienced at most traffic levels. At medium traffic, Systems (2) and (3) (CDMA and fixed frame CDMA/TDMA) give comparable and favorable delay and throughput performances. However, at the extreme traffic ranges the voice packet loss of Systems (4) and (5) is comparable to that of the other systems.

Figs. 4.12 to 4.14 show the results when a Rayleigh fading environment is assumed. For data traffic, we again notice the superior performance of TDMA for both delay and throughput. These TDMA users have a high AWGN signal to noise ratio and do not experience the multiuser interference that affects the CDMA-type systems (i.e Systems (2) to (5)). In terms of delay and throughput, Systems (4) and (5) yield reasonable values only at very low traffic levels (See Figs. 4.12 and 4.13). The distinguishing factor between the three environments is the voice packet loss. We notice from Fig. 4.14 that the best system in terms of voice packet loss is TDMA. In fact, CDMA and fixed frame CDMA/TDMA have the worst packet loss performance.

Although TDMA systems do not suffer from multiuser images (since users have a slot all to themselves), the single user self-images are not discriminated against since TDMA has no processing gain, resulting in intersymbol interference. We see that for data traffic (Fig. 4.15), Systems (4) and (5) offer the best delay performance for all traffic ranges, while the second and third systems provide the best useful throughput (Fig. 4.16). For voice traffic (Fig. 4.17), Systems (4) and (5) not only have good delay characteristics, but also outperform the other systems in

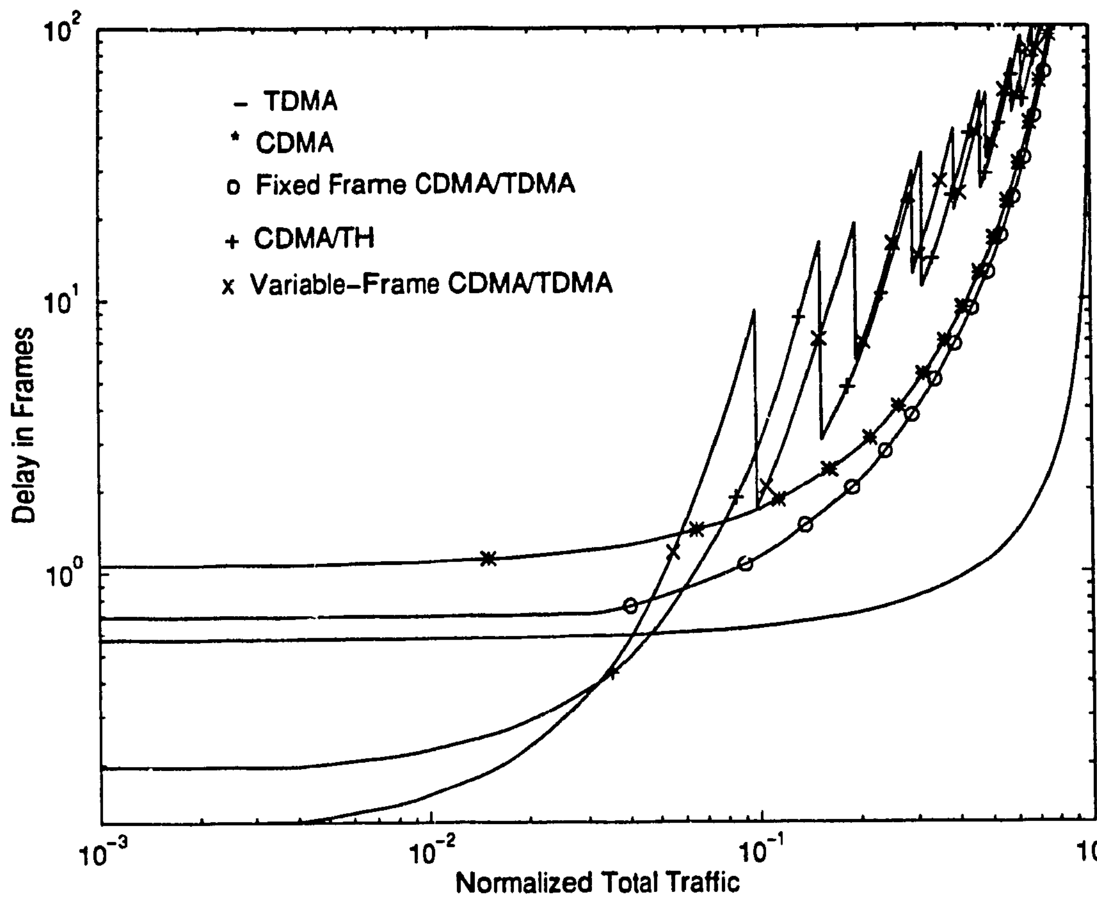

Figure 4.12: Delay for data Traffic under Rayleigh fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

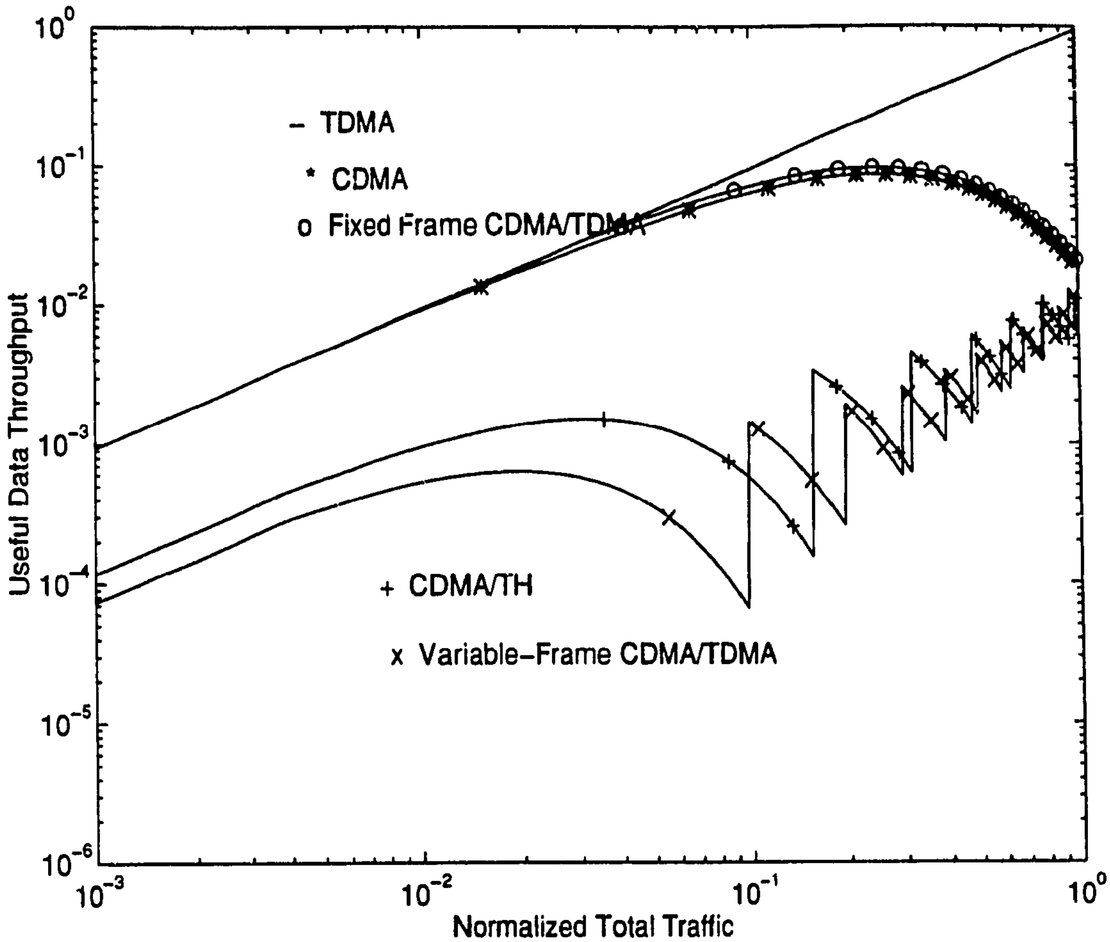

Figure 4.13: Throughput for data traffic under Rayleigh fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

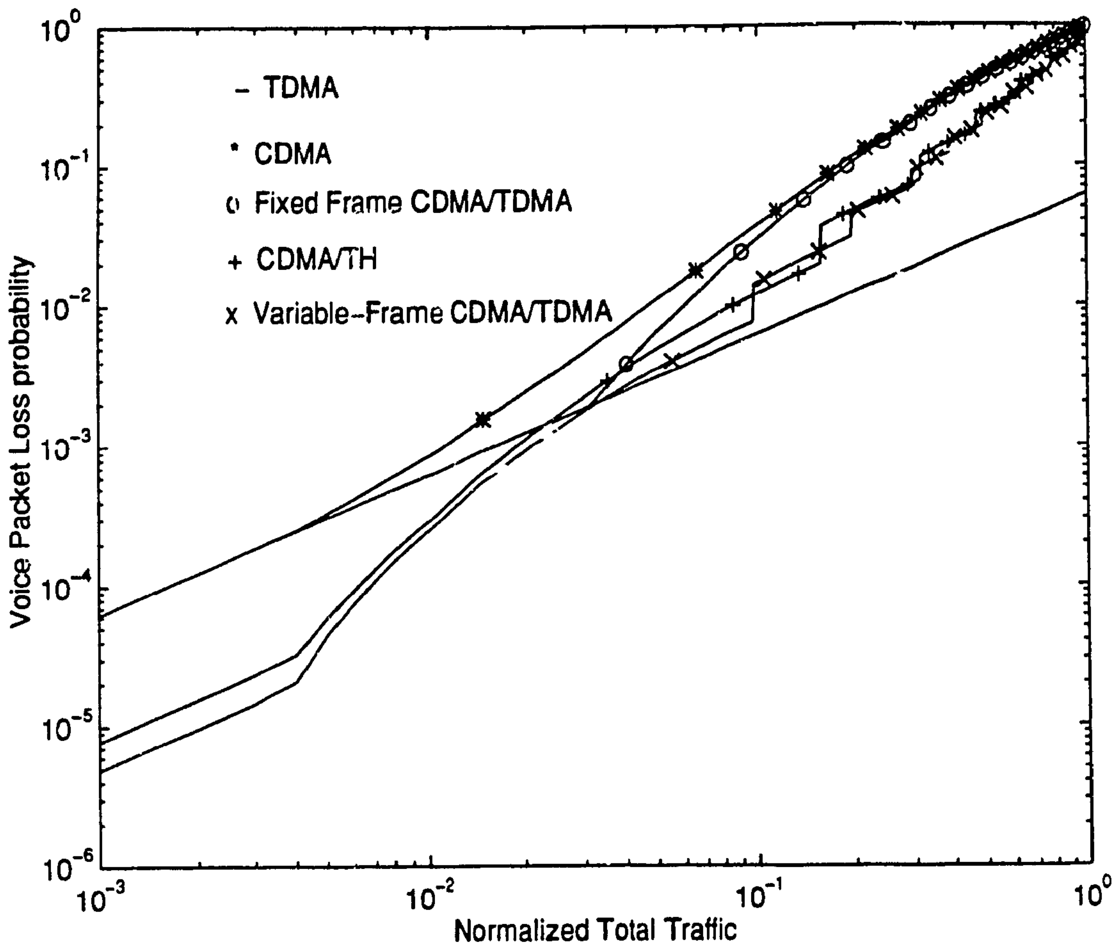

Figure 4.14: Loss of voice traffic under Rayleigh fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

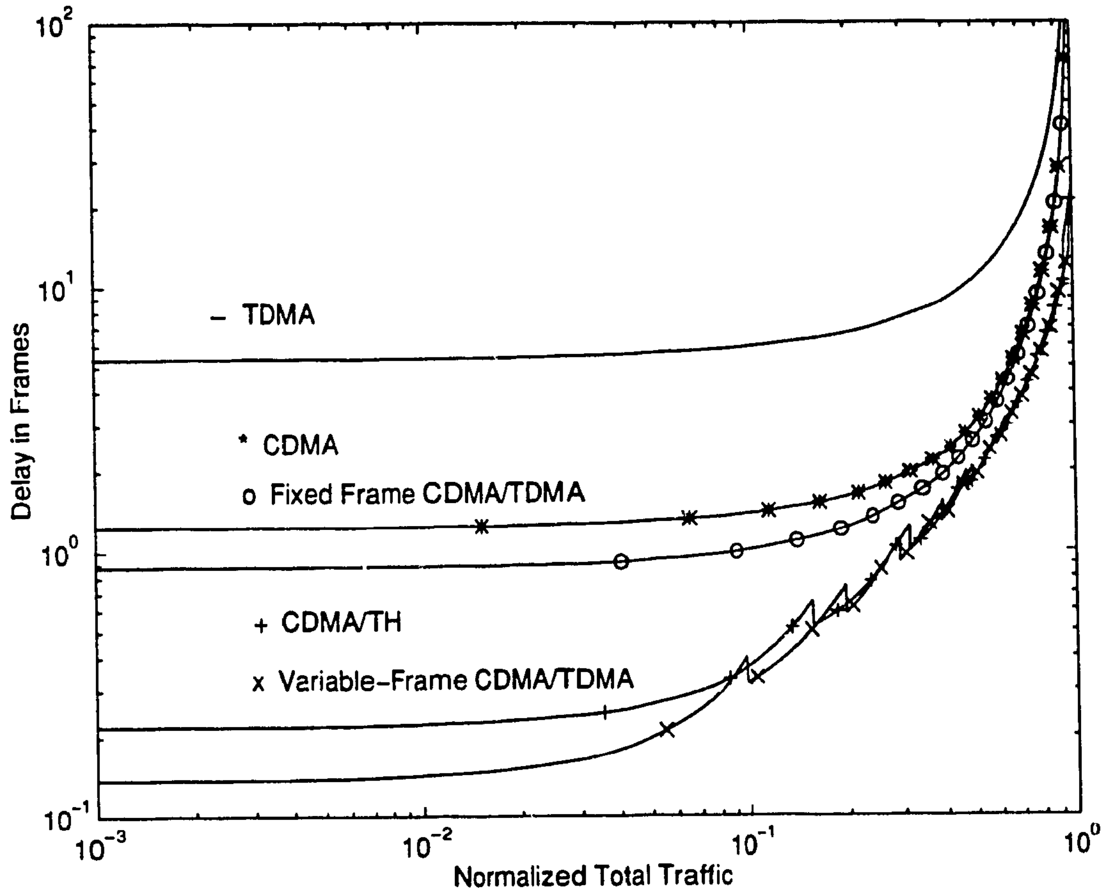

Figure 4.15: Delay for data Traffic under Rician fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

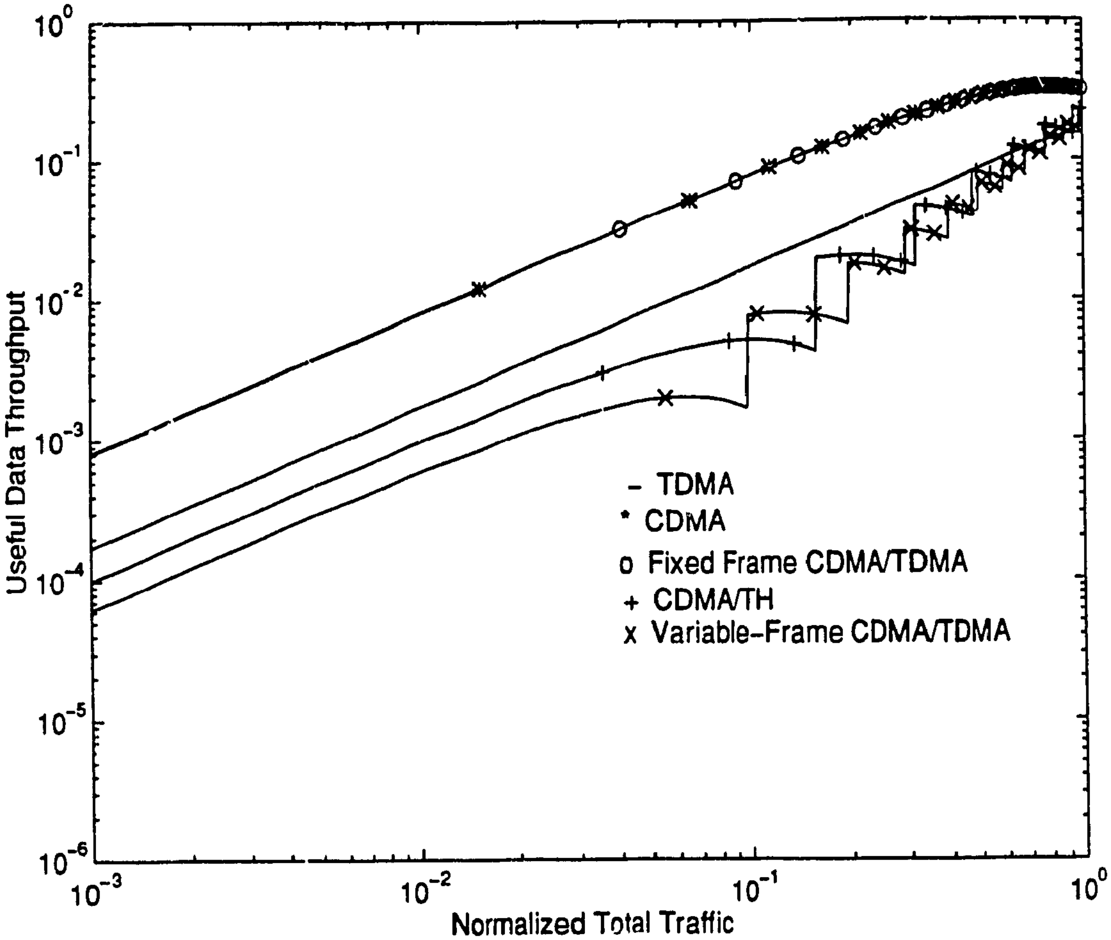

Figure 4.16: Throughput for data traffic under Rician fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $x = 20$, $\gamma = \zeta = 0.8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

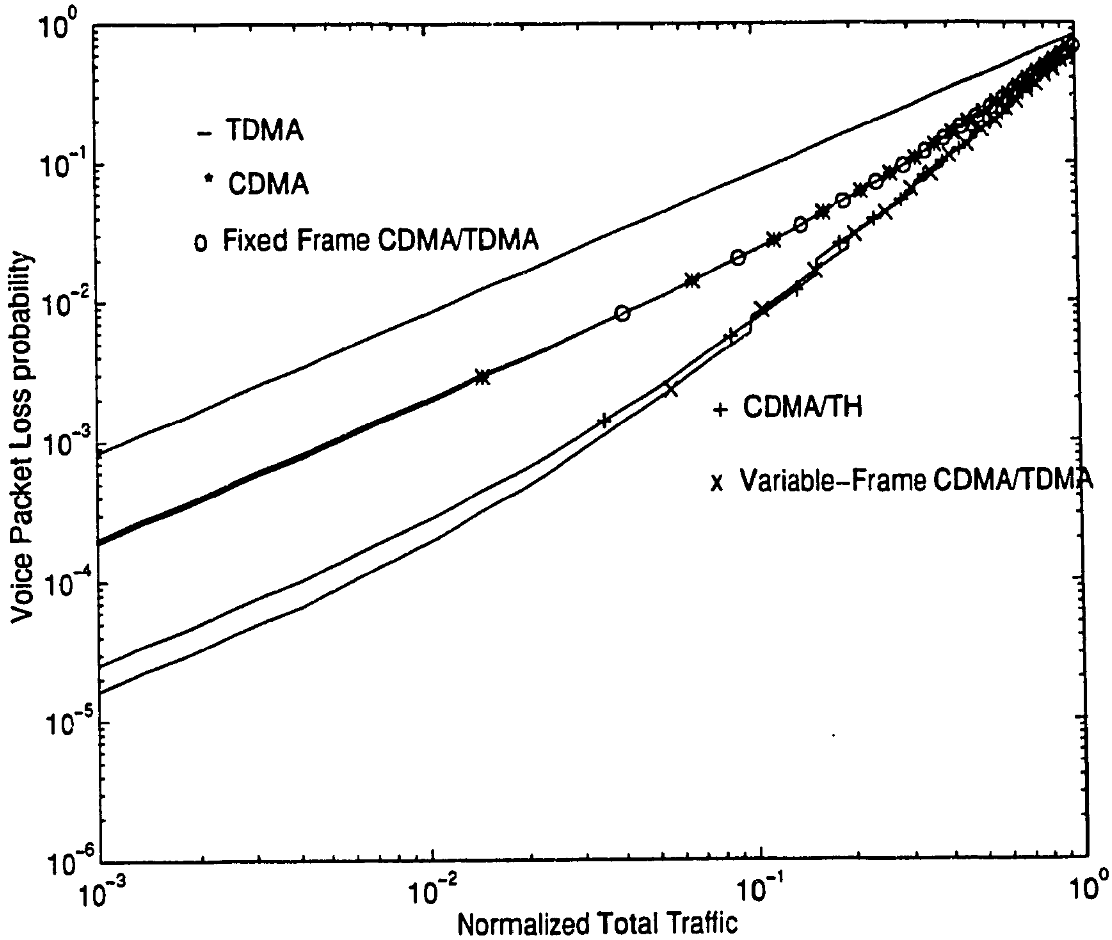

Figure 4.17: Loss of voice traffic under Rician fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$. $x = 20$, $\gamma = \zeta = 0.8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

terms of packet loss. Two other points not apparent from the graphs should also be noted. Firstly, the performance of Systems (3) and (4) converge at high traffic. This convergence can be justified from Equation(28) by defining N' as $[\zeta \rho N]$. At high traffic and for $\zeta \approx 1$, N' becomes N, and each user transmits in only 1 out of every N slots. If one neglects the randomness of the slot selection, this system is identical to fixed frame CDMA/TDMA. Secondly, there seems to be a crossover point in the value of x where System(5) becomes better than System(4) in terms of delay. This crossover point occurs when the processing gains of the two systems are equal, that is when x=U/N=8. With this value of x, the slot sizes for the two systems, the number of slots per effective frame, and the number of overlapping users are all equal to each other. As a result, the delay expressions also become identical. When x is chosen such that $x < U/N$, System (5) has better characteristics while the situation reverses itself for $x > U/N$.

4.4.2 Comments About the Steady State Behavior in Systems IV and V

It is clear from the graphs that for systems (4) and (5), the average delay and useful throughputs could undergo some erratic behavior as jumps occur as systems change service rates. The sudden change causes the apparent "singularities" in average delay in Fig. 4.8 and jumps in net throughput in Fig. 4.9. Actually a transient response is expected as the system switches from a one -slot frame to a two-frame slot for example (in the case of System(5)), but we are dealing only with steady state.

It is important to note that the graphs of delay and throughput for Systems 4 and 5 are actually the equivalent to the superposition of portions of multiple graphs, each graph corresponding to a service rate. What is apparent as a singularity is actually a sudden transition from one graph to another as the system switches from one number of slots to another. This results from the fact that the number of slots (or service rate) is discrete while the traffic load is a continous entity. For example, in Equation (4.33) if $\gamma\rho U/x = 4$, then the number of slots $N_5 = 4$, while if $\gamma\rho U/x = 4.01$, $N_5 = 5$ slots. This results in a phenomenon that is actually equivalent to the one observed in supermarket queues with variable number of servers. Some of these queues insure that no more than 3 people for example are waiting in the queue and keep adding servers up to a maximum number. What is observed in these queues is that a slight increase in the number of customers (equivalent to traffic load in our case) might be beneficial to the customers in terms of delay! As an example: suppose we have 3 customers in one queue, then each of these customers expects an average waiting time of $3\bar{X}$. Now for 4 customers, a new queue is created making the average waiting time of the new customer $2\bar{X}$.

Transient analysis of the system is very difficult as it involves an $M/G/m$ queue where m is a function of load. The adaptability parameters, as well as other higher layer controls could be used to insure that the average delay does not overshoot before the switch happens. Packets could be lost in the process, and an analysis of the packet loss due to any control method would be interesting although it is at this point out of the scope of the thesis. However, the steady state analysis is illuminating in showing the sudden jumps that are predicted in average traffic when higher layer controls are ignored. These surges do not have to correspond to actual network behavior, which is in reality time dependent. If we assumed a time varying uniformly increasing $\rho(t)$, the delay response, if graphed as a function of time, would not be equivalent to that of Fig. 4.8, but assuming slowly increasing load, there would be smooth transitions instead of jumps after the switch to a higher service rate is made.

4.4.3 Search for the Best Adaptability Parameters

Having used some arbitrarily chosen values of ζ, γ, and x for Systems (4) and (5), we now note that best values can be found for these parameters which will maximize useful throughput and minimize delay and voice loss. since we are dealing with so many parameters and we do not have a closed-form relationship between all of the above criteria (equations having Error functions among other things), analytical optimization is rather impossible. Graphical inspection however can be effective in many instances. Two methods will be used in this chapter. The first one is to fix the value of traffic intensity (a good choice would be a medium value) and plot the value of delay, throughput and voice loss as a function of ζ (for System (4)), γ, and x (for System (5)). The second method is to plot the delay, throughput, and voice loss versus traffic intensity for several selected values of ζ and γ.

Only graphs for the adaptive CDMA/TDMA will be presented in the sequel for brievety. For further graphs of all the systems involved, please refer to [10].

For System (5), the parameter optimization procedure becomes a two-dimensional problem. The delay and throughput for variable (x,γ), $\rho = 0.7$, and for data traffic transmission in Rician fading are shown in Fig. 4.18 and Fig. 4.19 for Rician fading. In this case a value of γ around 1 seemed to be best for processing gains around 100. However, depending on the permissible packet loss, one may be obliged to select an alternate (x,γ) pair, especially that we need to use the same parameters for voice and data.

Fig. 4.20 shows the data delay as a function of data load for four values of γ, while Fig.4.21 shows the throughput vs load. It is clear that the erratic behavior is less prevalent when γ is higher while a higher throughput can be achieved for all traffic levels.

Fig. 4.22 and Fig. 4.23, on the other hand, shows that smaller values of γ are more suitable for voice. This causes the designer to make a tradeoff when the same adaptability parameter has to be used for both data and voice.

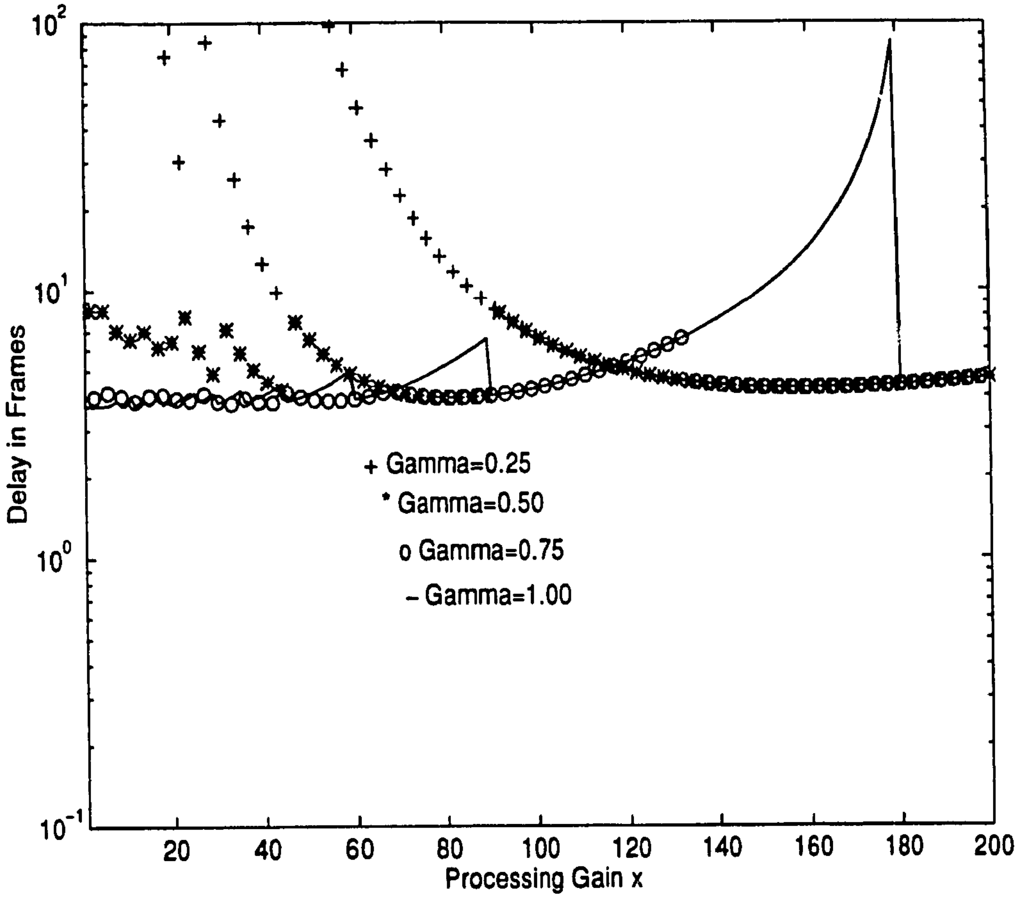

Figure 4.18: Delay vs x for selected values of γ for data traffic in System(5), $\rho = 0.7$, Rician fading.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$, $(\bar{\xi^2})'_0 = 0.4$, $(\bar{\xi^2})'_n = 0.02$.

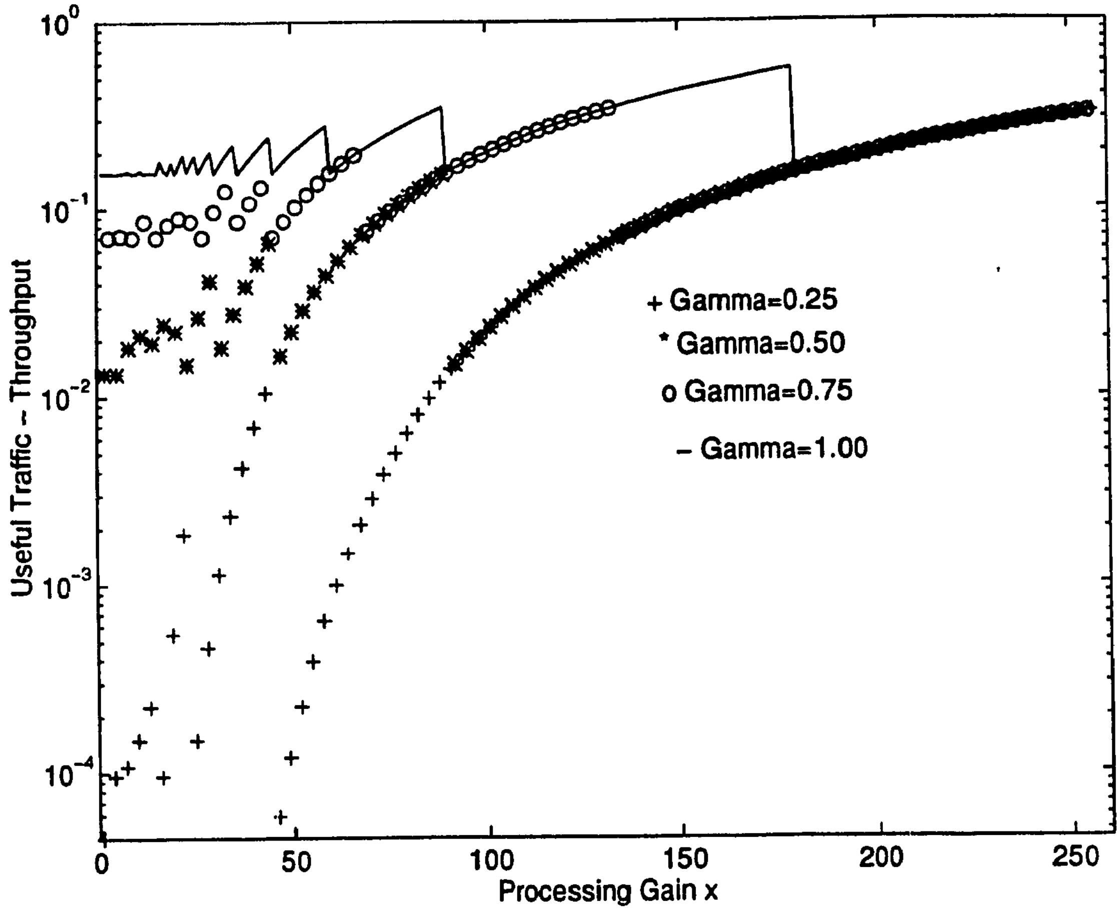

Figure 4.19: Throughput vs x for selected values of γ for voice traffic in System(5), $\rho = 0.7$, All fading environments.
$U = L = 256$, $W = 1.28$ MHz, $T_m = 10^{-5}s$, $2E_b/N_o = 1000$, $\bar{E} = 2$, $\bar{E^2} = 5$, $T_F = 1s$, $N = 8$.

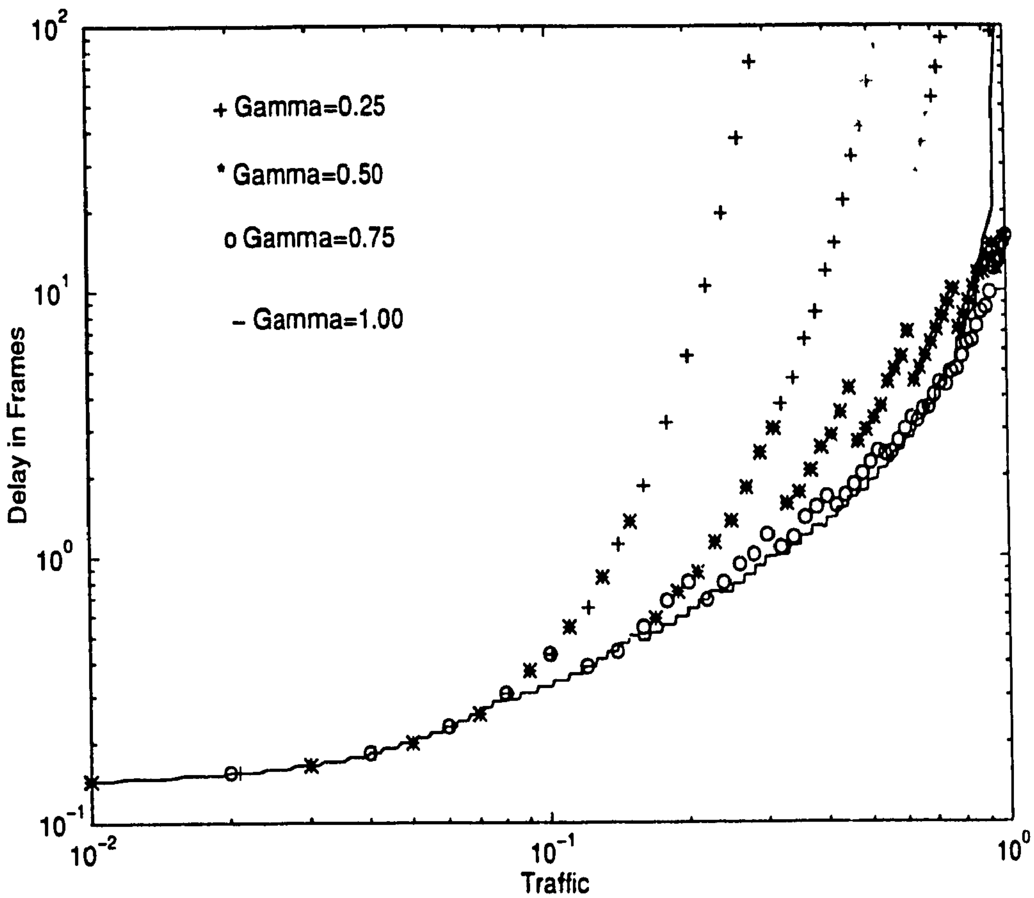

Figure 4.20: Data delay vs ρ for selected values of γ- Rician fading, x=20, The rest of the parameters are as before

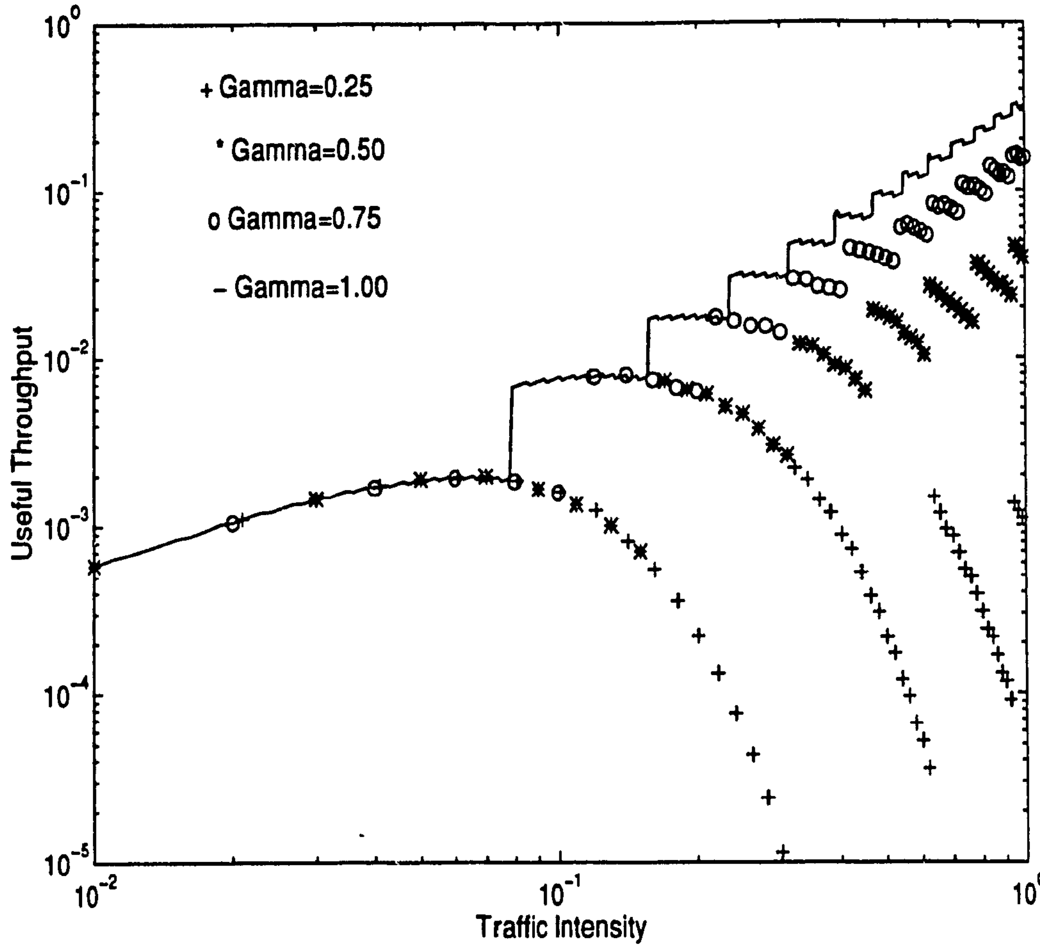

Figure 4.21: Data throughput delay vs ρ for selected values of γ- Rician fading, x =20, The rest of the parameters are as before

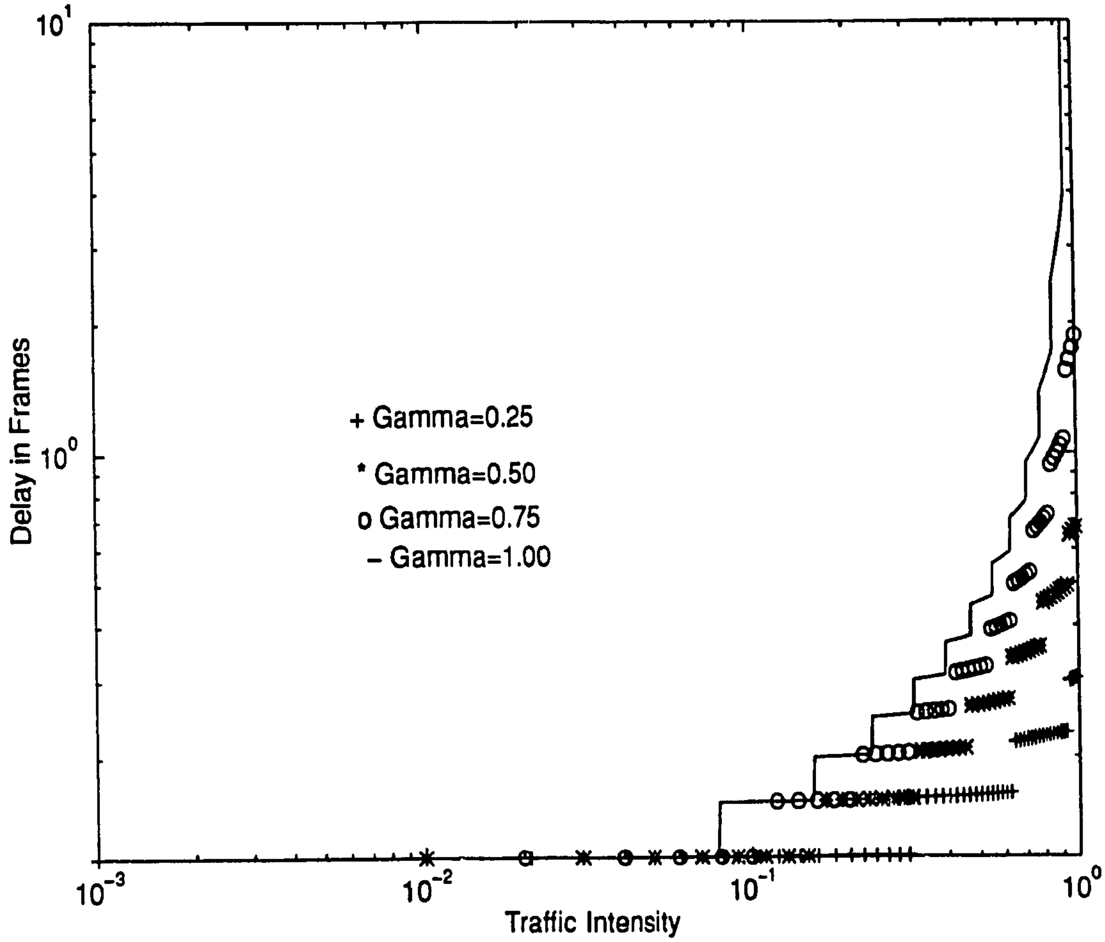

Figure 4.22: Voice delay vs ρ for selected values of γ- All fading env., x=20, The rest of the parameters are as before

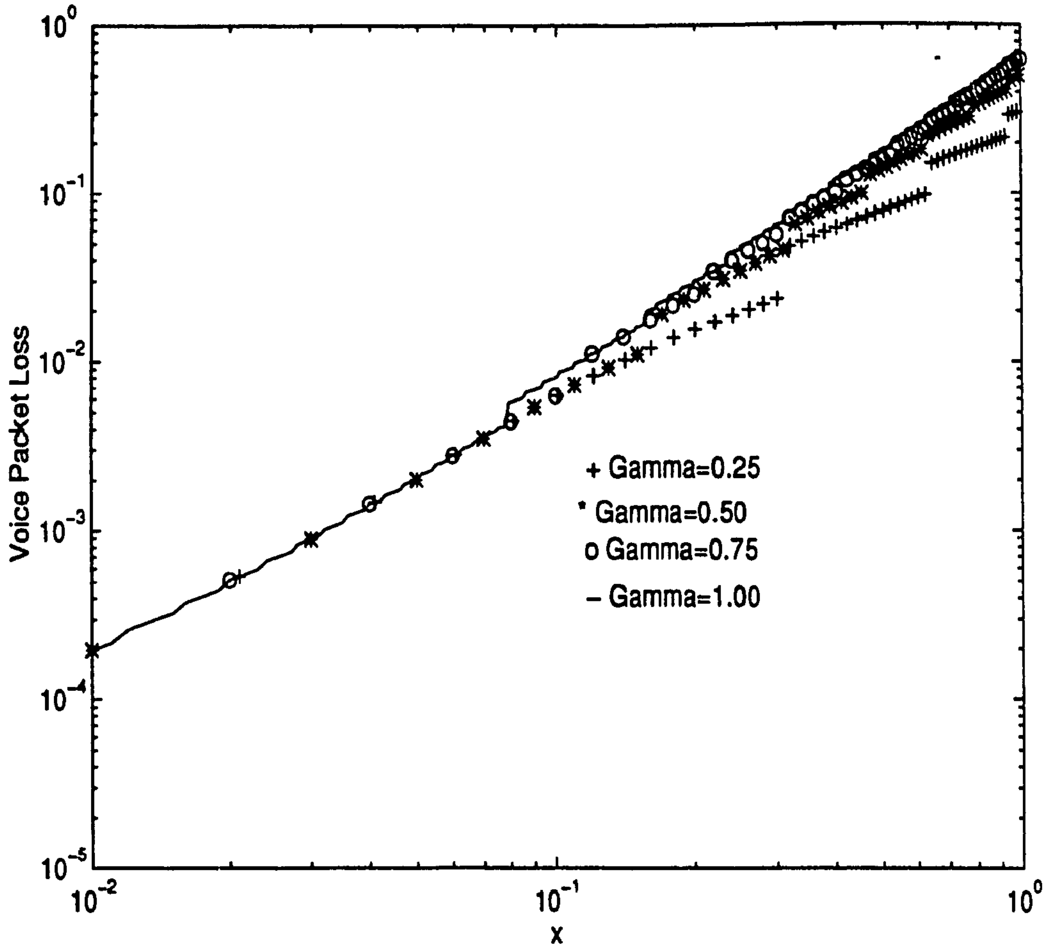

Figure 4.23: Voice loss vs ρ for selected values of γ- Rician fading, x=20, The rest of the parameters are as before

4.5 Conclusions

The delay and throughput characteristics of five CDMA related access techniques were investigated in various fading situations. The hybrid fixed frame CDMA/TDMA, CDMA/TH, and variable frame CDMA/TDMA techniques and the associated flow control mechanisms that have been introduced provided robustness against fading and rising traffic conditions. The results clearly indicate that Systems (4) and (5) provide the best delay results for voice traffic in nonfading channels and in Rayleigh and Rician fading channels. Moreover, these systems generally outperform TDMA systems in Rician fading for all levels of traffic.

The new systems introduced have a variable service rate (in bits per second per user) as the number of slots allocated per user is varied depending on traffic condition. This caused the performance to be generally better than CDMA in high traffic and better then TDMA in low traffic. One way to implement the systems while minimizing jittery response is to make the number of slots depend on delay and packet loss rather than traffic load. this is done by a protocol that makes the base stations send a signal to all mobiles to switch to a lower service rate as delay and packet loss reach a threshold in the transport level.

Further work is needed to compare the new systems presented here to systems involving Frequency Hopping, which will be discussed in the next chapter. The model for an integrated voice/data transmission system was not used here yet and will be introduced in the next chapters and will be the main contribution for the rest of the thesis.

Chapter 5

Analysis of the Performance of ARQ Schemes for DS and SFH CDMA Integrated Networks

5.1 Introduction

The delay and throughput performance of satellite-switched Slow Frequency Hopping CDMA network for simultaneous voice and data transmission is analyzed and compared to that of a DS-CDMA system. Two ARQ schemes are suggested for data while Forward Error Correction using the same encoder is used for voice packets. The queueing analysis assumes priority for voice and two models for voice traffic are used (Markovian and IPP). The probability of successful packet transmission is derived for all systems as a function of traffic load allowing us to evaluate the systems using delay, throughput, and voice packet loss as figures of merit. In the previous chapter, we compared the delay and throughput characteristics for data transmission of newly introduced hybrid Direct Sequence Code Division Multiple Access (DS-CDMA) networks that used TDMA and Time Hopping adaptively with the level of traffic neglecting the effect of different ARQ schemes and round trip propagation delay. The effect of ARQ for data on the simultaneous transmission of data and voice was not addressed, though. In this chapter, the performance of CDMA in an integrated voice/data satellite-switched network is analyzed assuming that both voice and data are being transmitted in the CDMA channel. Comparisons are made between DS and SFH systems when operating under similar conditions. The relevant application is a network of multiuser mobile transceivers such as an airplane with tens of users sharing the same transmitter. The aggregated arrival of voice packets coming from the users who share the same concentrator will be modeled in two ways, a Markovian Process and an Interrupted Poisson Processes.

The next section will discuss the system model along with the assumptions made for the parameters of the system. The second section will present the queueing equations for SFH and DS systems assuming a given probability of successful packet transmission. The equations used for the probability of error before and after error correction will be conditioned on the number of interfering users. Two type I Hybrid Automatic Repeat Request schemes (ARQ's) will be compared for data packets while FEC will be used for voice. This will allow us to use the same code for data and voice assuming that the decoder will switch to error detection mode when it receives a data packet and to error correction mode when it receives a voice packet.

The following is a list of symbols used in this chapter:

B: Buffer size for the voice queue.

E: Message length in packets, a random number.

E_b/N_0: Signal to channel noise ratio.

L: Number of bits per packet.

N: Number of voice sources sharing a mobile transmitter.

M: Number of interfering users.

PG: Processing gain.

P_d: Probability of error detection

P_c: Probability of successful packet correction.

Q: Voice queue size, a random number.

R_C: Chip rate for CDMA.

R_S: Source bit rate, before overhead and channel coding.

S_d: Net data throughput.

S_v: Net voice throughput.

T_d: Delay for data in System i.

T_v: Delay for voice in System i.

X_d: data message service time, a random number.

X_v: Voice message service time.

U: Number of connected users sharing the channel.

W: Channel bandwidth.

W_d: Data packet waiting time, a random number.

W_v: Voice packet waiting time, a random number.

$p_{\textbf{block}}(k)$: Probability of voice packet blocking given k.

$p_{\textbf{block}}$: Mean probability of voice packet blocking.

p_k: Probability distribution of the number of active voice sources.

$q_k(n)$: Probability of having n packets generated by k active voice sources.

s: Time-out between retransmissions for ARQ.

α, β: Voice activity transition parameters.

η: Transmission efficiency before queueing losses.

λ: Arrival rate for voice packets when a source is ON.

λ_d: Total arrival rate for data packets.

λ_v: Total arrival rate for voice packets.

π_{m}, n: Probability of transition from $Q = m$ to $Q = n$.

π_k: Steady state distribution of the queue size given k.

ρ_d: Data traffic load per transmitter.

ρ_v: Voice traffic load per transmitter.

ρ_t': Total traffic load per transmitter not accounting for voice activity.

ρ_t: Actual traffic load per transmitter.

ρ_{ch}: Total traffic load on the channel.

5.2 System Model

The analysis in this section assumes a general star network topology where the hub is a satellite link. Every transmitter could get voice and data calls from a number of terminals, and the aggregated arrival at each terminal is considered as a single integrated voice and data traffic source with one transmitter and two separate queues for voice and data. Voice calls may be blocked if voice loss becomes extensive as a result of multiaccess interference, but this is done at higher layers and is not dealt with in this chapter. The arrival rate of voice packets results from voice transmission rate and the number of active voice calls sharing the channel. The following paragraphs discuss the main assumptions about the common parameters of the systems to be studied.

Packetizing: A fixed packet size, L, will be assumed for both data and voice. L includes parity bits, and message bits, M. M itself includes overhead bits H and data bits D (the term "data bits" here refers to both voice and data). It is to be noted that while L, M and D are invariable to the type of traffic, H depends on the system used and how much information is needed in the header. For example, a pure CDMA system that assigns a unique code to every user/station does not need to include source information in the uplink and destination information in the downlink. In the case when a number of users share a code or a dynamic assignment of codes is used, user identification is needed in both directions.

System Capacity: The available total bandwidth, W, as well as the maximum number of users, U, will be fixed. The net throughput and delay is compared among systems and the source bit rate is represented by the traffic intensity. Actual system capacity will therefore be determined by the maximum per-user throughput given U and W.

Uplink vs downlink: A complete study of source-to-destination performance would ideally involve combining the results of both the up-link and the down-link. Fig. 1 illustrates the paths of a packet during this process. In the downlink, some other multiaccess method (such as Asynchronous Time Division Multiplexing) could be used, and even a connection to another network might be made, which makes the study of the downlink beyond the scope of this study. A study of the uplink should be sufficient, however, to compare the CDMA systems considered.

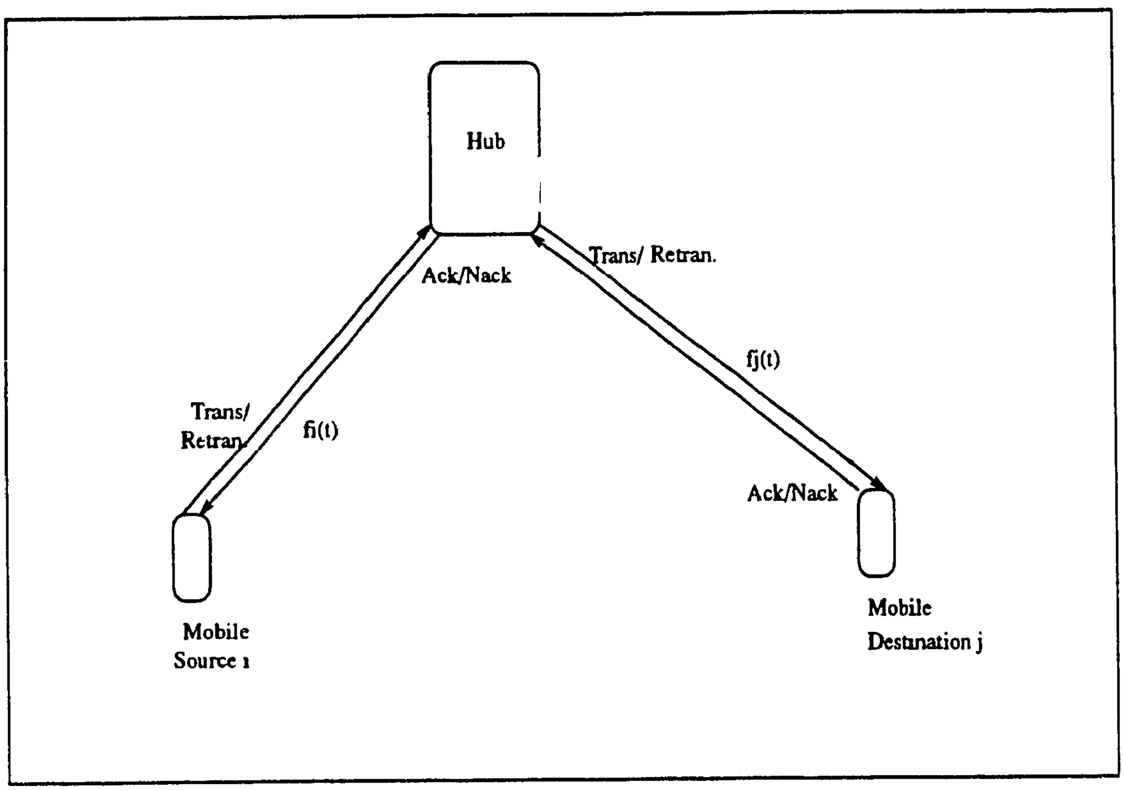

Figure 5.1: Data Transmission from source to destination. In case of voice, there would be no retransmissions. The possibility of error comes from both the uplink and downlink. The hub transmits only successfully received packets using the code of the destination receiver (fj(t)).

Voice loss: The percentage of lost voice packets is usually assumed to be equal to the probability of failed packet correction but this is a pessimistic approach as a lost packet usually has correctly decoded bits and voice will most likely be less corrupted than predicted if source coding is not ideal. One could extend the analysis to cases where a percentage of voice loss is allowed as we will show later.

Traffic and channel model: Throughout the analysis, each channel is considered to be binary symmetric channel where the only effect of multi-access is an increase in the probability of error. Thus, from a queueing point of view, the transmitter buffer of every station will be treated as a separate queue as every SFH or DS sequence is considered as a separate channel that receives on average the same level of traffic as other users. In other words, all stations will be assumed to have the same probability of being idle at any point in time. The following section will discuss the queueing aspects of the problem in more detail.

5.3 Queueing Analysis of The ARQ Schemes

At every mobile station, we assume that N users compete for the channel to make voice calls, send network control commands or acknowledgments (Ack's),send short data files, or make long data file transfers. The priority among the activities is determined by whether they can afford any delay. An analysis involving all of these classes is possible, but for the purpose of analysis convenience we will restrict ourselves to two types of traffic: voice packets from connected voice calls and data. Voice will have preemptive priority over data in service, meaning that a voice message is able to interrupt the transmission of a data message at the start of the next time slot. One major factor that will greatly affect the complexity of the analysis is the nature of voice traffic. It has been shown recently [58, 59] that voice packet arrivals do not generally obey Markovian statistics due to the interdependence of packets within each burst. Models using a general GI/G/1 queue and its statistics [60] (involving approximations) are becoming popular, although their application to voice traffic and priority service is still difficult analytically. On the other hand, it would be illuminating to study the performance of the system using Markovian statistics as a first simple case. Actually many studies [61, 59] have shown that an exponential interarrival time can be assumed when traffic is light, buffers are limited, and/or the number of multiplexed calls is large[1].

Two cases of analysis follow, in the first we assume Markovian statistics for both data and voice and in the second we assume a more general model based on an IPP (Interrupted Poisson Process) model [58, 62]. In the first model, the combined load of multiplexed voice packets of the different users will be considered as a single Poisson-distributed traffic source. In the second case, the effect of the number of multiplexed voice sources will be accounted for analyzing the voice packet loss resulting from blocking due to finite transmitter buffer sizes.

5.3.1 Markovian Model

Let λ_v and λ_d be the message arrival rate per user for voice and data, respectively, in messages per time unit. Noting that λ_d is independent of service time (and also of the number of retransmissions needed per packet in an ARQ scheme), we can proceed to derive the total traffic intensity which depends on the average service time per message. Having assumed an M/G/1 queue (exponential inter-arrival times, and general service times), we note that the average service time per message is equal to the average number of packets times the average service time per packet. Let E_v and E_d be the number of packets per voice and data message, respectively. The distribution ofE_v and E_d is assumed to be geometric, with the first three moments readily derived from its Probability Generating Function (PGF) [63].

[1]albeit this latter condition will be shown not to affect the analysis by much later

$$G_E(z) = \frac{\sigma}{1 - \sigma z} \tag{5.1}$$

where σ is the transition probability of the geometric random variable. Let n be a random variable representing the number of retransmissions per data packet. Then n has a geometric distribution that depends on P_d, the probability of successful packet reception. For type I hybrid ARQ systems, P_d is the probability that FEC fails, which will be computed in the next section. E_d is assumed to be independent of n. Let $G_n(z)$ be the probability generating function of n, and let m be a random variable equal to the number of frames needed for the eventual successful transmission of a packet, including the holding time, s (in frames), between retransmissions. This means that m depends on the ARQ scheme. For *Stop-and-Wait (S/W)* protocols, it is $m = (1 + s)n$. This follows from the fact that there is a time-out of s frames between packet transmissions whether or not the transmission is successful. For *Go-Back-N (GBN)* protocols, it is $m = (1 + s)n - s$. The probability generating function of m, $G_m(z)$, and the first two moments are:

For (S/W) protocols,

$$G_m(z) = G_n(z^{1+s}) = \frac{P_d z^{1+s}}{(1 - (1 - P_d)z^{1+s})} \tag{5.2}$$

$$\bar{m} = \frac{1 + s}{P_d}, \tag{5.3}$$

$$\bar{m^2} = \frac{2 - P_d}{P_d^2}(1 + s)^2. \tag{5.4}$$

For (GBN) protocols,

$$G_m(z) = G_n(z^{1+s})z^{-s} = \frac{P_d z}{(1 - (1 - P_d)z^{1+s})} \tag{5.5}$$

$$\bar{m} = \frac{1 + s(1 - P_d)}{P_d}, \tag{5.6}$$

$$\bar{m^2} = \frac{(1 - P_d)(1 + s)^2 + 1 + s(1 - P_d)^2}{P_d^2}. \tag{5.7}$$

In the above expressions, s is dependent on both the frame length T_F, the time needed to send one packet, and the roundtrip propagation delay τ. Although s

could be a random number, in our analysis we assume it is a fixed number equal to the maximum time needed for a time-out.

Voice and data messages will be treated as two different classes of customers sharing a common queue with voice having priority. Whether or not the priority is preemptive in actual implementation depends on how long the data service time is and whether some data files can afford to be repetitively interrupted. Voice silence utilization is one advantage of CDMA over fixed-assignment TDMA that will be taken into account. We will assume that the actual traffic intensity of voice will be α times the original traffic intensity[2]. A typical value for α is 0.33.
Let $\bar{X_v}$ and $\bar{X_d}$ the total service times for voice and data. The total throughput for voice is then

$$\rho_v = \alpha \lambda_v \bar{X}_v \tag{5.8}$$

and for data, it is

$$\rho_d = \lambda_d \bar{X}_d \ , \tag{5.9}$$

The total utilization not accounting for voice silence periods is

$$\rho_t' = \lambda_v \bar{X}_v + \lambda_d \bar{X}_d \tag{5.10}$$

and ithe actual traffic intensity at the queue (which must be less than one) is given by

$$\rho_t = \alpha \lambda_v \bar{X}_v + \lambda_d \bar{X}_d \tag{5.11}$$

Issues such as overhead and frequency guard bands affect the efficiency and should be accounted for in the analysis. An important issue when comparing systems that may or may not need source and/or destination information is the effect of address

[2]The average voice traffic intensity is equal to the probability that a source is active times the probability that an active source generates a packet in the slot (which is equal to the original traffic intensity calculated assuming the voice sources are on all of the time), this arises from the independence of the above probabilities

overhead. Let

$$\eta = r\frac{D}{D+H} = r\frac{D}{L} \tag{5.12}$$

be the initial efficiency before queueing losses, where r is the code rate. The total throughput efficiency for either voice and data will be equal to η times the respective values of the useful throughput. For voice, this is

$$S_v = \rho_v P_c \ , \tag{5.13}$$

where P_c is the probability of successful voice packet correction.

The throughput for data, S_d, depends on the ARQ scheme and is readily available in the literature. It is equal to the inverse of $\bar{m}$ times the traffic load [53]. For S/W, it is

$$S_d^{(S/W)} = \rho_d \frac{P_d}{1+s}, \tag{5.14}$$

while for GBN, it is

$$S_d^{(GBN)} = \rho_d \frac{P_d}{1+s(1-P_d)} \tag{5.15}$$

For mixed traffic, the total useful throughput is the sum of the useful throughputs for voice and data.

To compute delay for both data and voice, the mean square service times per voice or data message, $\bar{X_v^2}$ and $\bar{X_d^2}$, are required. Let $\bar{E_v^2}$ and $\bar{E_d^2}$ be the mean square of the number of packets per voice and data message, respectively. Let $\bar{T}_v$ and $\bar{T}_d$ be the mean delays for voice and data. For M/G/1 queues, three cases are of interest in case of mixed traffic, *Preemptive/Resume (P/R) priority.* non-preemptive priority, and no priority. In P/R priority, data transmissions could only occur if the voice queue is empty and data message transmission are interrupted at the next slot as soon a voice packet arrives. In non-preemptive priority, once a data message starts transmission, it could not be interrupted, Otherwise voice packets have priority. Our analysis will restrict itself to P/R priority but it can straightforwardly be extended to the other two cases. Delay for voice and data are in this case [63]:

$$\bar{T}_v = \bar{X}_v + \frac{\lambda_v \bar{X_v^2}}{2(1-\rho_v)} \quad (5.16)$$

$$\bar{T}_d = \frac{\bar{X}_d}{(1-\rho_v)} + \frac{\lambda_d \bar{X_d^2} + \lambda_v \bar{X_v^2}}{2(1-\rho_t)(1-\rho_v)}. \quad (5.17)$$

To determine the above delay values, we need to derive $\bar{X}_v$, $\bar{X_v^2}$, $\bar{X}_d$, and $\bar{X_d^2}$ in terms of the probability of correct packet transmission, P_d, and the given message characteristics.

For voice, assuming an unslotted FH system,

$$\bar{X}_v = \bar{E}_v T_F, \quad (5.18)$$

$$\bar{X_v^2} = \bar{E_v^2} T_F^2 \quad (5.19)$$

since there are no retransmissions for voice packets.

For data packets, let m be number of frames needed for successful packet transmission, with a mean $\bar{m}$ and a mean square $\bar{m^2}$. $\bar{X}_d$ and $\bar{X_d^2}$ are then

$$\bar{X}_d = \bar{E}_d \bar{m} T_F, \quad (5.20)$$

$$\bar{X_d^2} = \bar{E_d^2} \bar{m^2} T_F^2. \quad (5.21)$$

Using (5.3), (5.4), (5.6), and (5.7) results in

$$\bar{X}_d = \frac{1+s}{P_d} \bar{E}_d T_F, \quad (5.22)$$

$$\bar{X_d^2} = \frac{2-P_d}{P_d^2}(1+s)^2 \bar{E_d^2} T_F^2, \quad (5.23)$$

for S/W, and

$$\bar{X}_d = \frac{1+s(1-P_d)}{P_d} \bar{E}_d T_F, \quad (5.24)$$

$$\bar{X_d^2} = \frac{(1-P_d)(1+s)^2 + (1+s(1-P_d)^2}{P_d^2} \bar{E_d^2} T_F^2. \quad (5.25)$$

for GBN. Now the delay equations can be written solely as functions of $\bar{E}_v$, $\bar{E}_d$, T_F, $\bar{E_v^2}$, $\bar{E_d^2}$, ρ_v, ρ_d, ρ_t, and P_d in both ARQ cases). To simplify the equations, it is assumed that the data and voice message lengths are identically distributed and have the moments of unified random variable E. The resulting data delay for P/R Priority and S/W ARQ is

$$\bar{T}_v = \bar{E}T_F + \frac{\rho_v}{2(1-\rho_v)}\frac{\bar{E^2}}{\bar{E}}T_F \tag{5.26}$$

$$\bar{T}_d = \frac{(1+s)\bar{E}T_F}{P_d(1-\rho_v)} + \frac{\rho_d(2-P_d)(1+s)+\rho_v P_d}{2P_d(1-\rho_t)(1-\rho_v)}\frac{\bar{E^2}}{\bar{E}}T_F \tag{5.27}$$

while for P/R Priority and GBN ARQ, only data delay is modified from above as follows:

$$\bar{T}_d = \frac{1+s(1-P_d)\bar{E}T_F}{P_d(1-\rho_v)} + \frac{\rho_d\left[(1-P_d)(1+s)^2+1+s(1-P_d)^2\right]+\rho_v P_d}{2P_d(1-\rho_t)(1-\rho_v)}\frac{\bar{E^2}}{\bar{E}}T_F \tag{5.28}$$

Voice delay jitters can be studied using the variance of the delay of an M/G/1 queue with priority. The following equation gives the mean square waiting time of a voice message for a preemptive priority. It is derived as a special case from a more general equation in [64].

$$\bar{W_v^2} = \frac{\lambda_v \bar{X_v^3} + (1-\rho_v)T_F^2}{3(1-\rho_v)} + \lambda_v \bar{X_v^2}\left(\frac{\lambda_v \bar{X_v^2} + (1-\rho_v)T_F}{2(1-\rho_v)^2}\right). \tag{5.29}$$

Noting that $\bar{X_v^3} = \bar{E_v^3} \times T_F^3$, $\lambda_v = \rho_v/(\bar{E}_v T_F)$, and substituting the value of $\bar{X_v^2}$ and $\bar{X}_v$,

$$\begin{aligned} \bar{W_v^2} &= \frac{\lambda_v \bar{E^3}T_F^3 + (1-\rho_v)T_F^2}{3(1-\rho_v)} + \lambda_v \bar{E^2}T_F^2\left(\frac{\lambda_v \bar{E^2}T_F^2 + (1-\rho_v)T_F}{2(1-\rho_v)^2}\right) \\ &= \left[\frac{1}{3} + \frac{\bar{E_v^3}}{\bar{E}_v}\frac{\rho_v}{3(1-\rho_v)} + c + 2c^2\right]T_F^2, \end{aligned} \tag{5.30}$$

where

$$c = \frac{\rho_v}{2(1-\rho_v)}\frac{\bar{E_v^2}}{\bar{E}_v} \tag{5.31}$$

Using (5.30), the variance of the voice packet waiting time can be written as a function of both traffic intensity and voice message length. The analysis, however, depends on the definition of a voice message and thus the arrival rate of messages.

One can assume single packet messages and thus a deterministic service time per message, but this assumption is realistic only if the arrival process of voice packets is approximately Poisson.

It should also be noted that ρ_v represents the utilization rate assuming that the aggregated arrival of the voice sources sharing the mobile terminal have an exponential inter-arrival time. The total traffic load is the probability that a packet is generated at any slot due to this aggregated traffic. As was mentioned previously, while long-term averages can be modeled in this way, the un-Poisson-like characteristics of voice sources *in general* result in a less accurate prediction of the short-term (or instantaneous) average delay, and in the case of a finite buffer, the probability of voice packet blocking. The following section will use an entirely different method for modeling voice that can also be used for a single voice source and high traffic load. The Markovian model is still valid when the conditions mentioned in the beginning of this section are satisfied, however.

5.3.2 Interrupted Poisson Process Model

A voice call is modeled as a two state continuous-time Markov chain with talking (ON state) and silence (OFF state) having an exponentially distributed duration. When active, a voice source generates voice packets at a rate proportional to the source bit rate. Let α be the transition rate from the OFF state to the ON state and β the transition rate from the ON state to the OFF state (Fig. 2.a). The probability that a call is ON would be equal to $\frac{\alpha}{\alpha+\beta}$. Fig. 3 shows the queueing model used for a single mobile transmitter. Assuming N voice sources are allowed to share the queue, the probability p_k that k out of N are active at any given moment can be found from the state transition diagram shown in Fig. 2.b. This probability is a binomial distribution,

$$p_k = \binom{N}{k} (\frac{\alpha}{\alpha+\beta})^k (1 - \frac{\alpha}{\alpha+\beta})^{N-k}, 0 \leq k \leq N \tag{5.32}$$

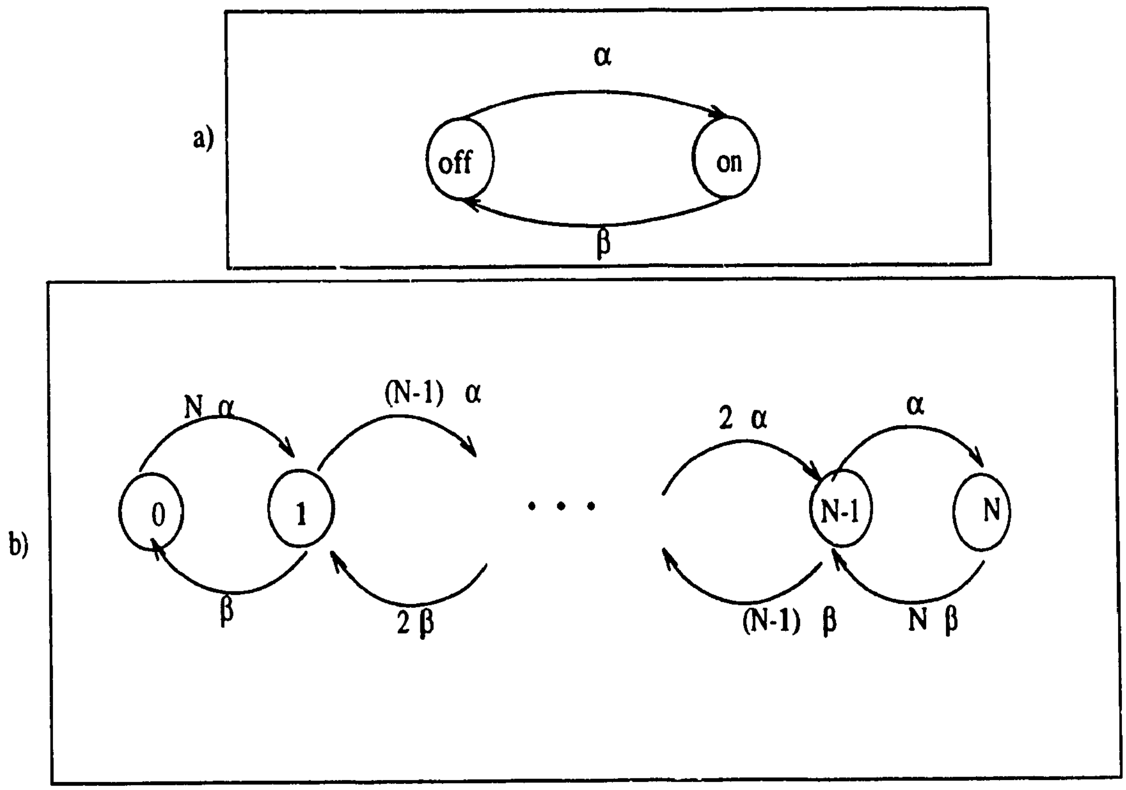

Figure 5.2: a) Markov chain of the state of a call. b) State Transition Diagram of the number of active calls. α and β are the transition rates of the continuous-time two-state Markov chain

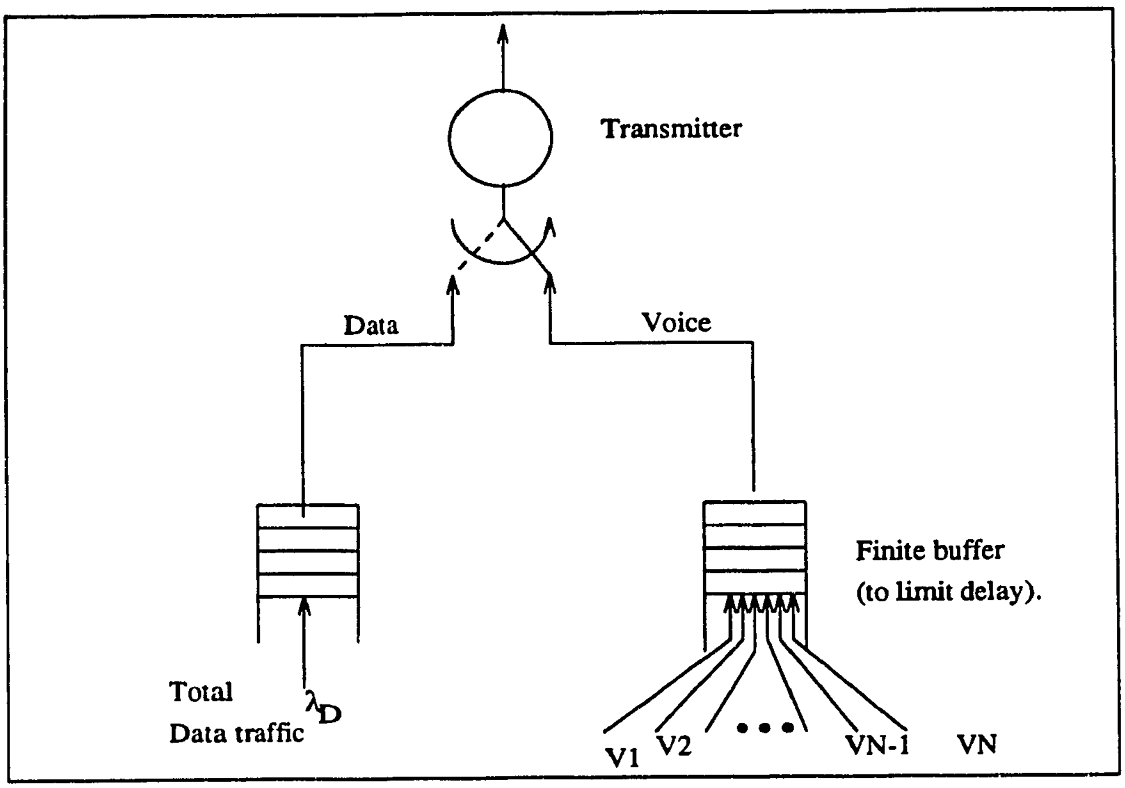

Figure 5.3: Queueing Model for the combined voice-data traffic. The switch is on data only when the voice buffer is empty or while the current packet is transmitted.

Now let λ (a constant) be the voice utilization rate of a single voice source while it is ON. This represents the number of packets per slot that a voice source generates when active and depends on the total transmission bit rate and the voice source bit rate. If R_C is the transmission bit rate, T_F is the frame length (equal to the time needed for transmission of a packet), L and η are as given before, and R_S is the source bit rate, λ is then given by the following equation:

$$\lambda = \frac{R_S \times \mathrm{PG}}{\eta R_C}. \tag{5.33}$$

As an example, for PCM-coded voice at a rate of R_S = 64kbps, and an available channel bandwidth (before spreading) of R_C/PG=640kbps, a coding rate of 0.8, and an overhead of 10%, resulting in an efficiency $\eta = 0.72$, the utilization rate would be $0.1/0.72 = 0.139$. Let B be the size of buffer for voice packets. To find the steady state distribution of the probability distribution of the number of packets in the queue generated by N voice sources, one assumption is made (from [62] and [65]): *The number of active calls changes slowly enough to allow the queue to reach a steady state in between changes.* This assumption is not unreasonable: suppose the transmission bit rate is 640 kbps, with a packet length of 256 bits, or 0.4 ms. Noting that a typical change of a single state from ON to OFF is in the order of one second, This means that with a number of active voice sources in the tens, it takes tens or hundreds of packets to have another change in the number of active sources. A more detailed analysis can be found in [62]. Assuming a given number of active voice sources, k, all generating truncated-poisson-distributed voice traffic at a rate of λ, and a combined traffic at a rate of $k\lambda$, the number of packets generated in one slot is a random number, n, with a steady state distribution of $q_k(n)$. For $k = 0$, $q_k(n)$ is zero except at $n = 0$. For $n > k$, it is assumed to be zero (thus the term "truncated" Poisson Process). For other values of n, one notes that the steady state probability would be equal to the probability that a Poisson process with rate $k\lambda$ generates n packets in a slot divided by the probability that n is less or equal k, which is the sum of the probabilities of having any number of packets less than or equal to k. In other words,

$$\text{Prob}(n/k/n \leq k) = \frac{\text{Prob}(n/k)}{\sum_{i=0}^{k} \text{Prob}(n = i/k)}. \tag{5.34}$$

The steady state probability of the number of packets generated by k active voice calls is then given by

$$q_k(n) = \begin{cases} \frac{\frac{(k\lambda)^n}{n!}}{\sum_{i=0}^{k} \frac{(k\lambda)^i}{i!}}, & 0 < k \leq N, 0 \leq n \leq k, \\ 1, & k = n = 0 \\ 0, & \text{otherwise.} \end{cases} \tag{5.35}$$

As in the previous subsection, voice has preemptive priority over data and its queue is not affected by data traffic. The average waiting time of a voice packet will depend on the average queue length. A finite voice buffer of B packets will be assumed, and voice packets that find the buffer full are blocked and discarded. To derive the average queue length, one has to write the state transition diagram of the number of packets in the buffer at the end of each slot. The state of the buffer can be described by a $(B+1)$-state, discrete-time Markov chain (Fig. 4 shows a state transition diagram of this chain). Let Q be the voice queue size, and $\pi_{m,n}(k)$ be the transition probability from $Q = m$ to $Q = n$ given k active calls. These probabilities are given by [62, 65]

$$\pi_{m,n}(k) = \begin{cases} q_k(n - m + 1) & 0 < m \leq n + 1 < B + 1 \\ \sum_{i=B-m+1}^{k} q_k(i) & n = B \\ 0 & n < m - 1 \end{cases} \tag{5.36}$$

with the following boundary conditions:

$$\pi_{m,n}(k) = \begin{cases} 1 & n = m - 1,\ 1 \leq m \leq B,\ k = 0 \\ 0 & \text{otherwise for k=0} \\ q_k(0) + q_k(1) & m = n = 0,\ k \neq 0 \end{cases}, \tag{5.37}$$

where a summation is equal to zero if its upper limit becomes smaller than its lower limit. The above $\pi_{m,n}(k)$ are the elements of a $(B+1) \times (B+1)$ matrix, $P(k)$, which is actually the limit of the Markov transition probability matrix $P(k,t)$.

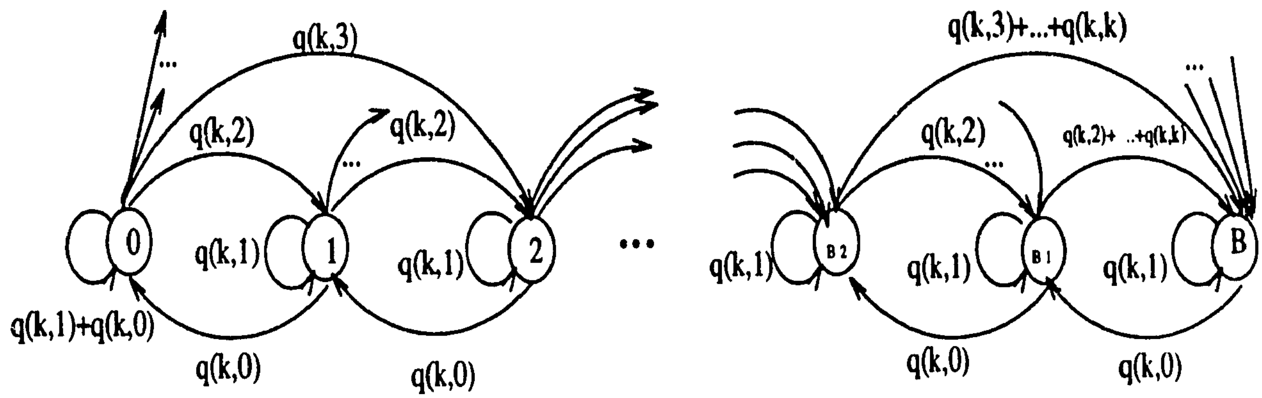

Figure 5.4: Transition probabilities of the voice queue size given k active voice sources. A finite buffer B is assumed and equilibrium is assumed to be reached before any change in k happens.

The steady state probabilities of the queue length $\pi_k(n)$, for k active voice calls $1 \leq k \leq N$, and $0 \leq n \leq B$, can be computed recursively from this matrix using the eigen-vector equation $\Pi(k)P(k) = \Pi(k)$ (see Appendix A). The resulting steady-state probabilities can be used to compute the mean queue length:

$$\bar{Q} = \sum_{n=1}^{B} n \sum_{k=0}^{N} \pi_k(n) p_k. \tag{5.38}$$

The voice packets are blocked and lost if they find the buffer full, giving rise to the following probability of voice packet blocking:

$$p_{\text{block}} = \sum_{k=1}^{N} p_{\text{block}}(k) p_k, \tag{5.39}$$

where $p_{\text{block}}(k)$ for k active voice calls is the probability of packet blocking given k active voice sources.

The probability of blocking for any given voice packet is the probability that it arrives to the transmitter buffer and finds it full. Next we Note that this probability times the average number of packets generated in any slot is equal to the average probability of blocking in any given slot, and that this latter entity is equal to the probability that the voice queue size in the slot in which the packet arrives plus the number of packets generated during that slot is larger than the buffer size. Given k sources active, this probability can be found by averaging over all possible queue sizes:

$$p_{\text{block}}(k)\bar{q}_k = \sum_{n=0}^{B} \pi(n) \sum_{i=B+1-n}^{k} q_k(i), \tag{5.40}$$

provided that the lower limit of the second summation is less or equal to the upper limit. This is equivalent of summing over all i, $0 \leq i \leq k$ while multiplying the argument by $\max(n+i-B-1, 0)$, this results in the following equation:

$$p_{\text{block}}(k) = \frac{\sum_{n=0}^{B} \pi(n) \sum_{i=1}^{k} \max[(n+i-1-B, 0]q_k(i)}{\bar{q}_k} \quad 1 \leq k \leq N \tag{5.41}$$

Given the above quantities, the voice traffic intensity is given by:

$$\rho_v = \sum_{k=1}^{N} (1 - p_{\text{block}}(k))\bar{q}_k p_k, \tag{5.42}$$

The mean voice waiting delay can then be computed using Little's formula,

$$W_v = \frac{\bar{Q}}{\rho_v} + \rho_v/2. \tag{5.43}$$

Data traffic, although Markovian, is affected by the nature of voice traffic since the latter has priority (we assume only preemptive priority here). Computation of the residual time is the first step in determining the mean waiting delay of a data message.

Assuming the voice parameters ($\bar{Q}, \rho_v$, etc.) to be given, let $\bar{W}_0$ be the average residual service time [66, 67] in slot 0 (the slot in which our tagged data message arrives). This is equal to the sum of the products of arrival rates and the second moment of the service time of arrivals from all traffic divided by two. In a slotted system (our case) one has to add the residual time left in the slot which is equal to one half the sum of the products of arrival rates of each class times the mean square of its service time [67], in other words:

$$\bar{W}_0 = \frac{1}{2}\left(\lambda_d \bar{X_d^2} + \rho_v\right). \tag{5.44}$$

The total mean waiting time for a message $\bar{W}_m$ is then given by

$$\bar{W}_m = \bar{W}_0 + \bar{Q} + \bar{Q}_d \bar{X}_d + \bar{N}_v, \tag{5.45}$$

where $\bar{Q}_d$ is the average data queue size at the moment the data packet arrives, $\bar{N}_v$ is the average number of voice packets which arrive after the data packet and are served ahead of it. This equation can be solved (see Appendix B) for $\bar{W}_m$ and used to find the total waiting delay for successful packet transmission of a single packet $\bar{W}_d$:

Writing (B.12) in Appendix B in terms of P_d and s, we get for S/W and GBN protocols respectively:

$$\bar{W}_d^{S/W} = \frac{\rho_d(2-P_d)(1+s)\bar{E}^2}{(1-\rho_t)\bar{E}} + \frac{\bar{E}-P_d}{2P_d(1-\rho_d)} + \frac{\rho_v+\bar{Q}}{1-\rho_t}, \tag{5.46}$$

and

$$\bar{W}_d^{GBN} = \frac{\rho_d((1-P_d)(1+s)^2 + (1+s(1-P_d)^2))\bar{E}^2}{(1-\rho_t)(P_d+sP_d(1-P_d))\bar{E}} + \frac{\bar{E}-P_d}{2P_d(1-\rho_d)} + \frac{\rho_v+\bar{Q}}{1-\rho_t}. \tag{5.47}$$

All we need now is to determine P_d in terms of the multiaccess interference which depends on the the average load on the channel. This issue will be addressed in the next section assuming a given traffic intensity on the channel.

5.4 Probability of Error Analysis

Up to this point, we assumed the probability of correct packet detection and that of successful packet correction to be given although they depend on multi-access interference and thus on the traffic intensity. In this section, the probability of error for a BFSK-FHMA will be discussed assuming a given number of simultaneous users. The relationship between the number of simultaneous users and traffic intensity will be derived according to the type of system and how users are split. This will depend on the systems that will be considered. For a regular Frequency Hopping system, all users interfere when they are active, but interference in time results in a possibility of error only if a frequency hit occurs. One might use an average number of interfering users,

$$M_{FH} = \lceil \rho U \rceil, \tag{5.48}$$

but a more rigorous treatment would evaluate the probability of error given the number of interfering packets and then use the theorem of total probability to compute the expected value of the probability of bit error. Let ρ_t be the total integrated traffic (as defined above) for any of the U nodes (assumed to have identical traffic statistics), then the probability that a packet is generated at any moment of time is equal to ρ_t, and the probability that M packets are interfering with our designated packet at the receiver is

$$P(M) = \binom{U-1}{k} \rho_t^k (1-\rho_t)^{U-1-k}, k = 1, ..., U-1, \tag{5.49}$$

and the average packet success probability is:

$$P_d = \sum_{M=0}^{U-1} P_d(M) P(M), \tag{5.50}$$

where $P_d(M)$ is the probability of packet error detection given M simultaneous users interfering during the packet detection. This is still not exactly right however, as the traffic on the channel is not exactly equal to the traffic intensity at the queue of the transmitter. For a S/W strategy for example, service time includes holding time where the transmitter waits for ACK/NACK without actually transmitting. This issue is important but not hard to solve. Let $\bar{m}'$ be the average number of transmissions per packet (until success), For a geometric random variable with a probability P_d, this is equal to $1/P_d$. For S/W systems, the transmitter transmits only during

$T_F \times \bar{m}'$ out of the service time of a packet. This means a total transmission time of $T_F \times \bar{m}' \times \bar{E}_d$ per message, while the queueing service time, $\bar{X}_d$, is given above ($\bar{X}_d = ((1+s)/P_d)\bar{E}_d T_F$). The average data traffic load that actually affects the channel (let us call it ρ_{ch}) can be related to the regular data traffic load felt by the queue (ρ_d, above) as follows:

$$\rho_{ch} = \rho_d/(1+s) \tag{5.51}$$

This equation is especially important for the case where s is much larger than one as is the case for satellite systems, especially geostationary ones.

The following analysis will assume a given number of simultaneous users, M who use a Binary Frequency Shift keying modulation and Frequency Hopping at a rate of one hop per bit. The total number of Frequency bands that a particular user is hopping in is assumed to be equal to the processing gain and to the total number of users:

$$PG = U \tag{5.52}$$

Young and Stuber derived an equation for the exact probability of error by summing up the conditional probabilities of error given k number of hits from zero to $M-1$. Geraniotis derived an approximation and a lower bound for this same probability while Young/Stuber gave some upper bounds to the same equations.

Given a thermal signal to noise ratio $\sigma = r \times E_b/N_0$, The lower bound for the probability of bit error is [37]:

$$P_b(M) \geq \frac{1}{2} \sum_{k=0}^{M-1} \binom{M-1}{k} \left(\frac{1}{U}\right)^k \left(1 - \frac{1}{U}\right)^{M-1-k} \exp\left\{-\frac{\sigma}{2+k\sigma}\right\} \tag{5.53}$$

while the upper bound is

$$P_b(M) \leq \frac{1}{2} \sum_{k=0}^{M-1} \binom{M-1}{k} \left(\frac{1}{U}\right)^k \left(1 - \frac{1}{U}\right)^{M-1-k} \exp\left\{-\frac{\sigma}{2+1.5k\sigma}\right\} \tag{5.54}$$

The above two bounds are fairly tight as Stuber and Geraniotis have shown in their two works and any one of the two could be used. The tightness of the bounds was also verified throughout our computations while the exact formula is quite elaborate and involves a lot of computations. One has to note though that what we actually need is the probability of packet error which is not necessarily straightforwardly related to the probability of bit error [68, 69, 70]. It is only when the probabilities of bit errors are independent from bit to bit that we have the following expression for the probability of successful packet transmission, P_d:

$$P_d(M) = (1 - P_b(M))^L \tag{5.55}$$

This can happen if we assume slotted Frequency hopping which insures that the interference stays constant within the packet. This assumption was not actually used in the delay formulas. This leads us to either slightly modify the delay formulas to account for slotting, or to do a more rigorous analysis as in [70] to account for varying interference within the packet. For this initial analysis, we will assume slotted frequency hopping and increment the delay equations by $T_F/2$ which is the third term needed for slotted systems.

Now for voice, FEC is used and a decoder corrects t errors, while it can detect twice as many errors. The probability of detection failure and thus loss of data packets will for now be considered negligible, while since the number of errors unrecovered by FEC will be calculated assuming a linear block code of the same size as the packet size, the probability of voice packet loss is not negligible and is equal to the probability that there are more errors than t, the maximum number of correctable errors for the code. In case of bit-to-bit independence, This is equal to

$$P_c(M) = \sum_{k=t+1}^{L} \binom{L}{k} P_b(M)^k (1 - P_b(M))^{L-k} \tag{5.56}$$

The above value of $P_d(M)$ is expected to converge rapidly to zero for large values of L if $P_b(M)$ is not negligibally small. For Type I Hybrid ARQ, if we use a BCH code for example, let (n,k) be a code that can correct t errors, then it can be used to correct t_1 errors and simultaneously detect $2t - t_1$ errors. $P_d(M)$ will then be given by the same equation as $P_c(M)$, substituting the smaller value t_1 instead of t. For voice, packet loss occurs when correction fails, so one can estimate packet loss rate by

$$v_l(M) = 1 - P_c(M). \tag{5.57}$$

However, a voice packet is not automatically lost if correction fails, as voice can tolerate a certain degree of packet loss. Neglecting for now packet loss due to blocking in the transmitter, let ι be the permissible loss percentage, beyond which one has *total packet loss*, then the probability of total packet loss is assumed to be the probability of having at least one error in the overhead or having more than t_2 errors in the message part of the packet, where

$$t_2 = \lfloor \iota D \rfloor. \tag{5.58}$$

Let $P_{bc}(M)$ be the average bit error probability of a voice packet after correction, the probability of total loss of packets is:

$$v_l * (M) = 1 - \left[(1 - (P_{bc}))^H \left[\sum_{k=0}^{t_2} \binom{L}{k} P_{bc}^{L-k} (1 - P_{bc})^k \right] \right]. \tag{5.59}$$

P_{bc} can be approximated for a binary BCH coder with minimum distance d_{min} and a correction capability of t as follows [33] :

$$\begin{aligned} P_{bc}(M) &\approx \frac{d_{min}}{L} \sum_{i=t+1}^{d_{min}} \binom{L}{i} P_b(M)^i (1 - P_b(M))^{L-i} \\ &+ \frac{1}{L} \sum_{d_{min}+1}^{L} i \binom{L}{i} P_b(M)^i (1 - P_b(M))^{L-i}. \end{aligned} \tag{5.60}$$

The expected value of the average total loss probability can then be found from the average over all possible values of M as explained earlier.

Given the above equations, we can proceed to evaluate the delay, throughput, and voice loss as functions of traffic intensity for a Slow Frequency Hopping Multiaccess system. To compare this system with a direct sequence system, the only change in all equations introduced earlier would be in the evaluation of the probability of bit error given M, the number of simultaneous interferers. This probability has been discussed widely in the literature, the equations that will be used are from [10].

5.5 Numerical Results

5.5.1 Results Using the Markovian Model

The following parameters were assumed for the Markovian model for both voice and data: U=256 users, L=256 bits, an overhead of 24 bits, W=10MHz, $\bar{E} = 1$ packet, and $E_b/N_0 = 15\ dB$, Results were computed for a P/R priority queueing strategy, An extended (256,164) BCH code was used for pure error correction for voice and simultaneous correction and detection for data (Hybrid type I ARQ). The length of the frame being 6.5536 ms, the roundtrip delay is approximated at s= 100 frames (for a geostationary orbit). Comparisons were made with direct sequence systems based on the analysis in [10] assuming negligible fading.

As a first case, the performance of the systems was analyzed assuming voice and data traffic are on average proportional to each other, meaning ρ_v/ρ_t is assumed to be constant. Figures 5.5 to 5.7 show plots of the data delay, data throughput and voice loss vs total traffic per transmitter for DS and SFH systems using SW or GBN schemes for data. these were obtained using Equations (5.14), (5.15), (5.27), (5.28), (5.50), (5.51) and (5.53) for SFH. For DS systems, Equation (4.70) was used instead of (5.53). It is to be noted that the data load value used in the X axis in all figures is the normalized value ρ_d, times the maximum bit rate *per transmitter* for TDMA, 39.0625 kbps. All other normalized values of throughput were also multiplied by this value.

One observation that can be made is that while Fig. 5.5 shows that DS systems are much better than SFH systems in all cases, Fig. 5.6 shows that, in the high traffic range, the difference between DS systems and SFH systems is more apparent when GBN schemes are used. It is to be noted that voice delay is very minimal. From Equation (5.26), the average voice delay when ρ_v=0.5 is 1.5 T_F= 10.2 ms. Voice delay variance for this case can be found using (5.30) and is equal to $0.167 * T_F^2$. The standard deviation for voice delay would be 2.68 ms.

It is assumed, as a second case, that a fixed voice traffic load is reserved to a fixed number of connected voice calls and delay, throughput, and voice loss are studied as a function of data traffic load. The voice load is assumed to be equivalent to two connected calls per user (mobile) needing 4.8kb/s each and combining for an average load of $\rho_v = 0.1$ (taking voice silence, coding rate and overhead into account).

Although the Markovian assumption might not be valid for a small number of voice calls, this was deliberately assumed to compare with the IPP model later using the same number of voice calls. If the Markovian model predicts similar results with a small number of voice sources, it would be expected to have even more similar results with a higher number of sources.

Figures 5.8 to 5.10 show the data delay, useful throughput and voice loss for DS and SFH systems as functions of the data load assuming a constant voice traffic. It is observed that in this case S/W systems appear to be the better choice for SFH systems as it even achieves a better maximum throughput than GBN (around 200b/s, still a very low figure). Using a S/W scheme for data in the SFH case might actually be advantageous as the interference is much lower than GBN and results in a much lower probability of error. The down side is that the traffic intensity for S/W corresponds to a very low useful throughput. Nonetheless, it is interesting to note that GBN systems deteriorate very fast as data traffic increases, while S/W systems might even help reduce voice loss while not reducing data useful throughput by much (Figures 9 and 10). One has to note however that DS systems clearly outperform SFH systems by a large margin in all cases and that GBN can be used efficiently with them. As an example, Fig. 10 shows that for DS systems, voice traffic is not affected much by an increasing data load, while in the case of SFH, voice loss is already significant at low data loads and reaches more than 10% at data loads in the order of 10% of the maximum.

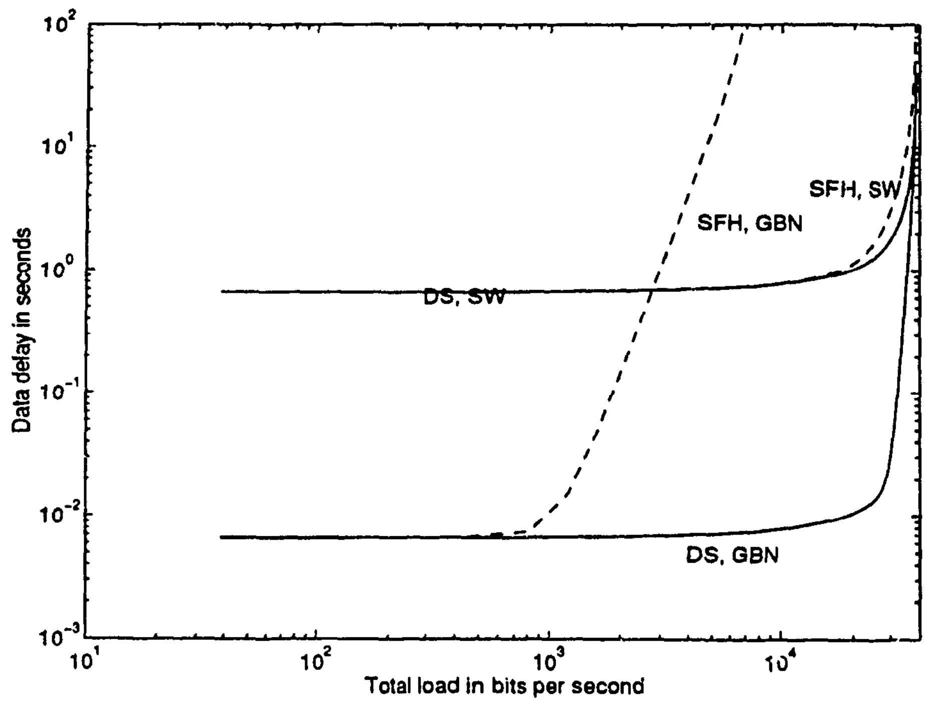

Figure 5.5: Data Delay vs combined voice and data load for DS and SFH CDMA using S/W or GBN ARQ. Markovian model assumes voice makes up 10% of the traffic. See text for the other parameters which are constant throughout the analysis.

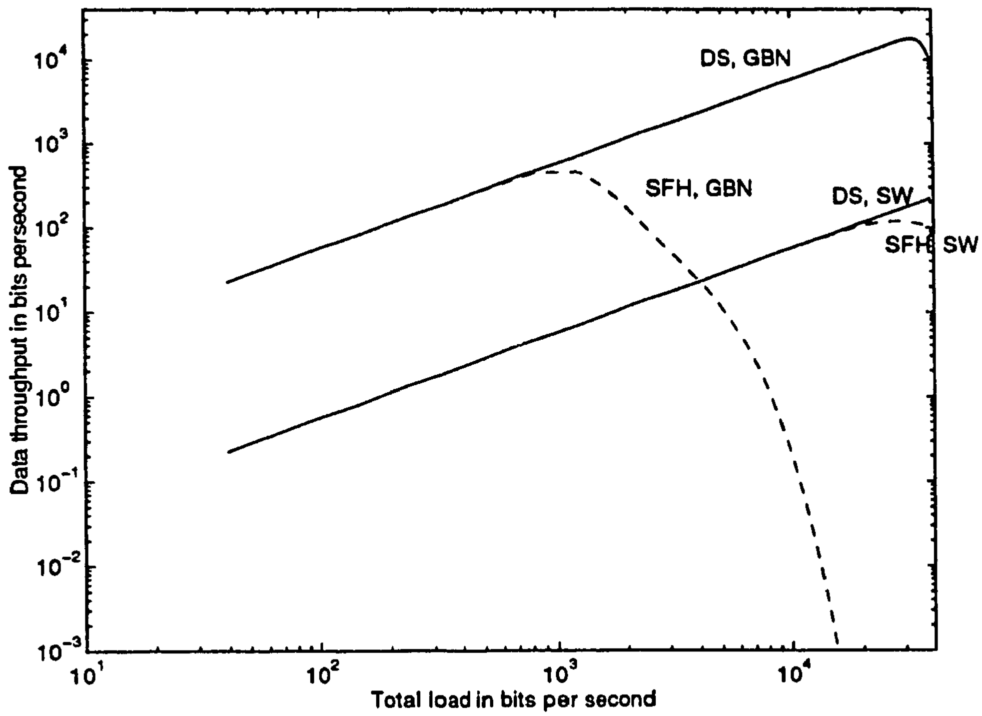

Figure 5.6: Data throughput vs combined voice and data load for DS and SFH CDMA using S/W or GBN ARQ. Markovian model assumes voice makes up 10% of the traffic.

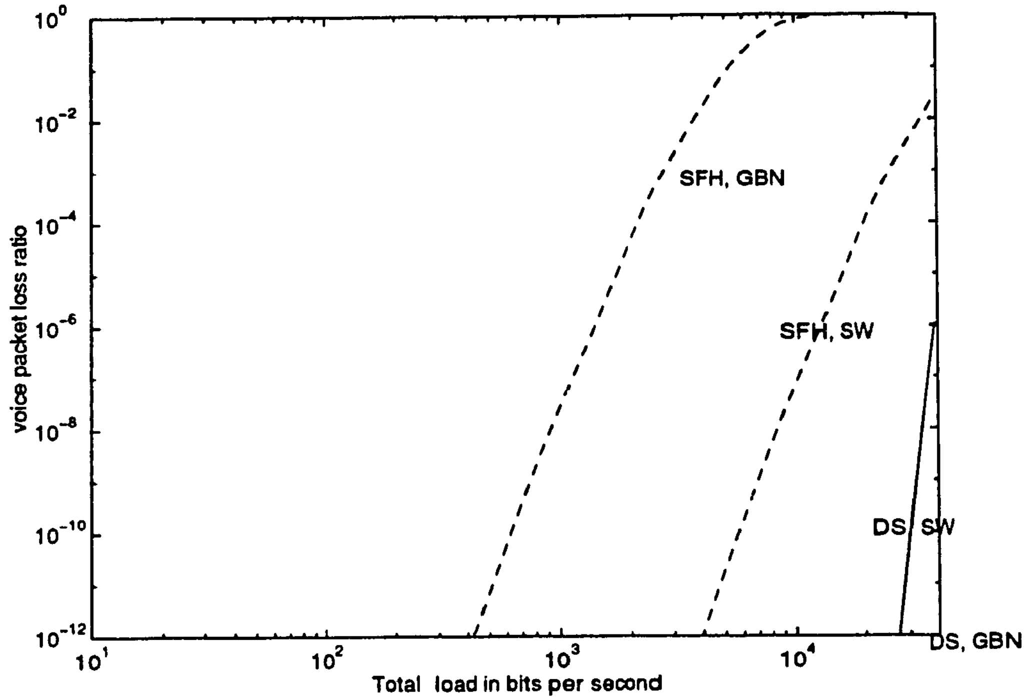

Figure 5.7: Voice loss vs combined voice and data load for DS SFH CDMA using S/W or GBN ARQ. Markovian model assumes voice makes up 10% of the traffic. The result for DS and GBN is too small to show in the figure.

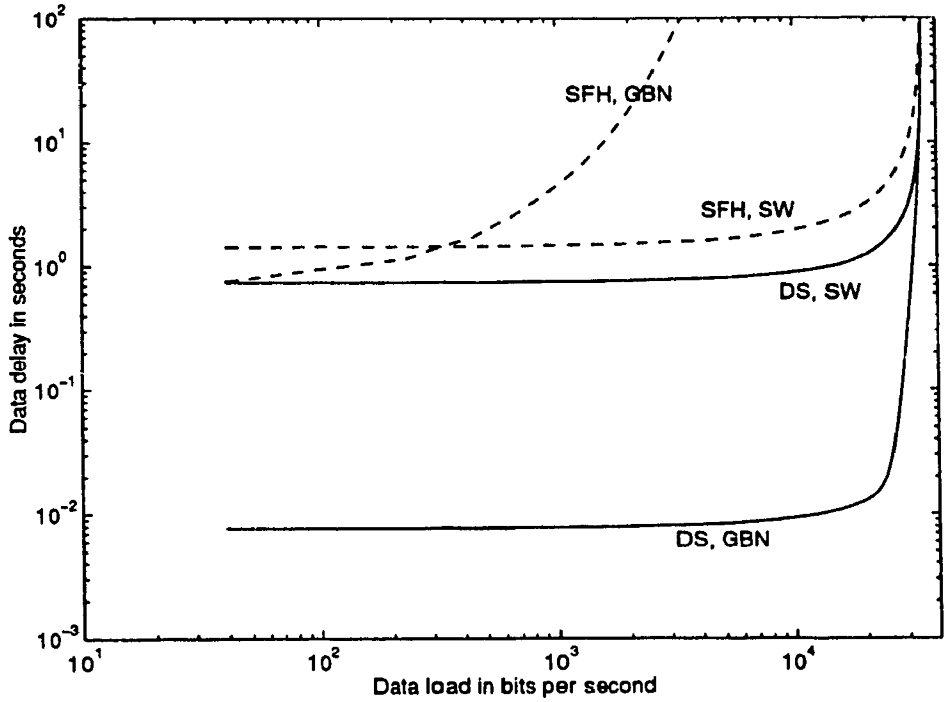

Figure 5.8: Data delay vs data load for DS and SFH CDMA using S/W or GBN ARQ. Markovian model assumes a constant average voice load $\rho_v = 0.1$.

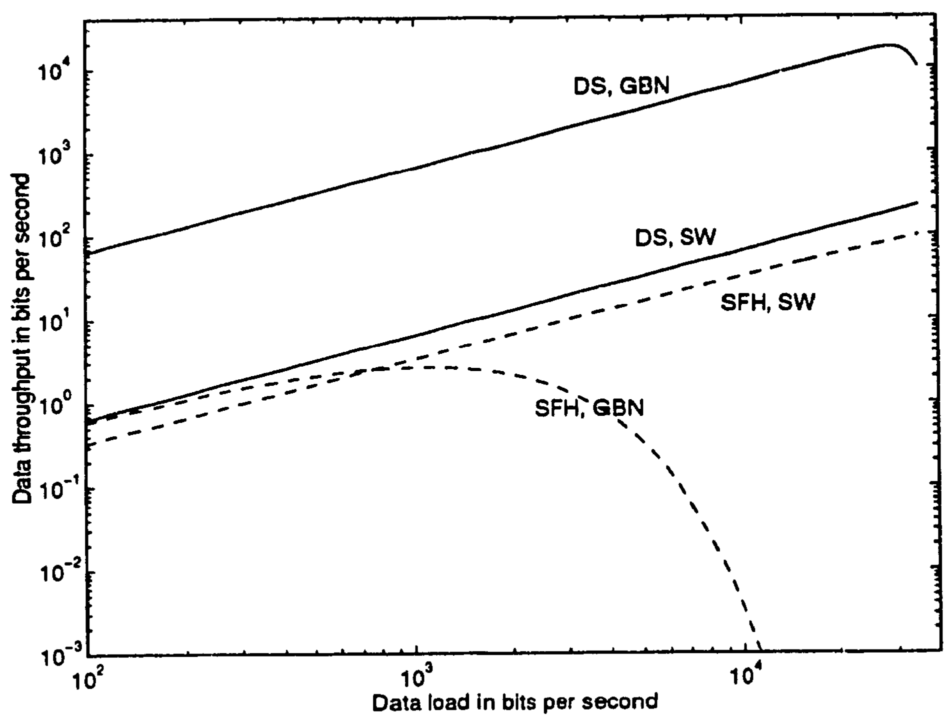

Figure 5.9: Data throughput vs data load for DS and SFH CDMA using S/W or GBN ARQ. Markovian model assumes a constant average voice load $\rho_v = 0.1$.

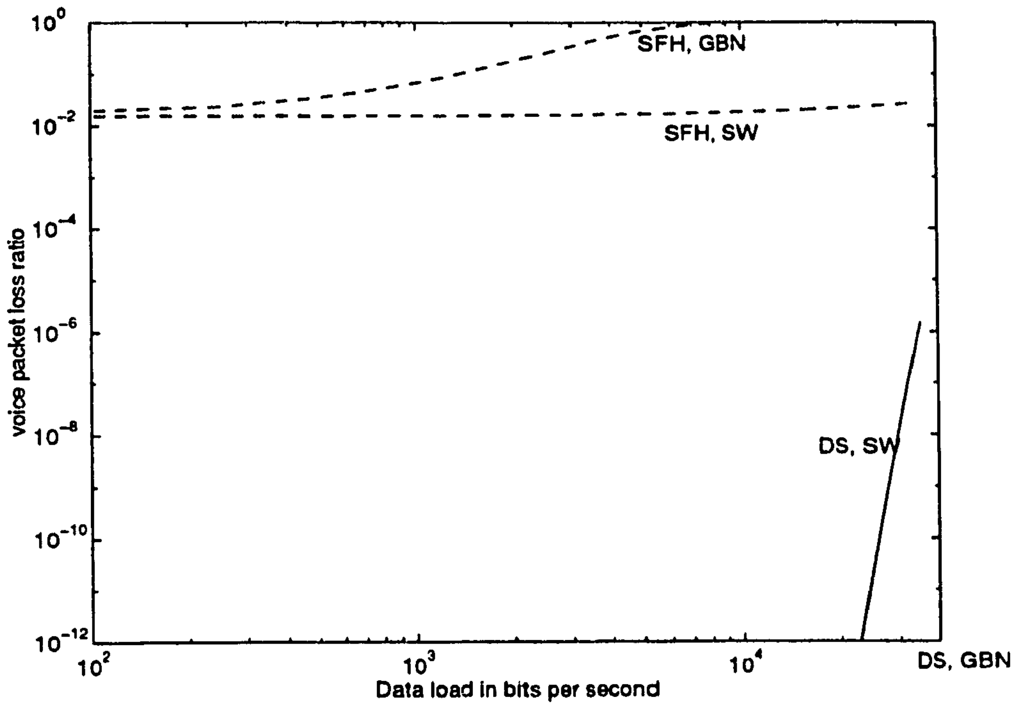

Figure 5.10: Voice loss vs data load for DS and SFH CDMA using S/W or GBN ARQ. Markovian model assumes a constant average voice load $\rho_v = 0.1$.

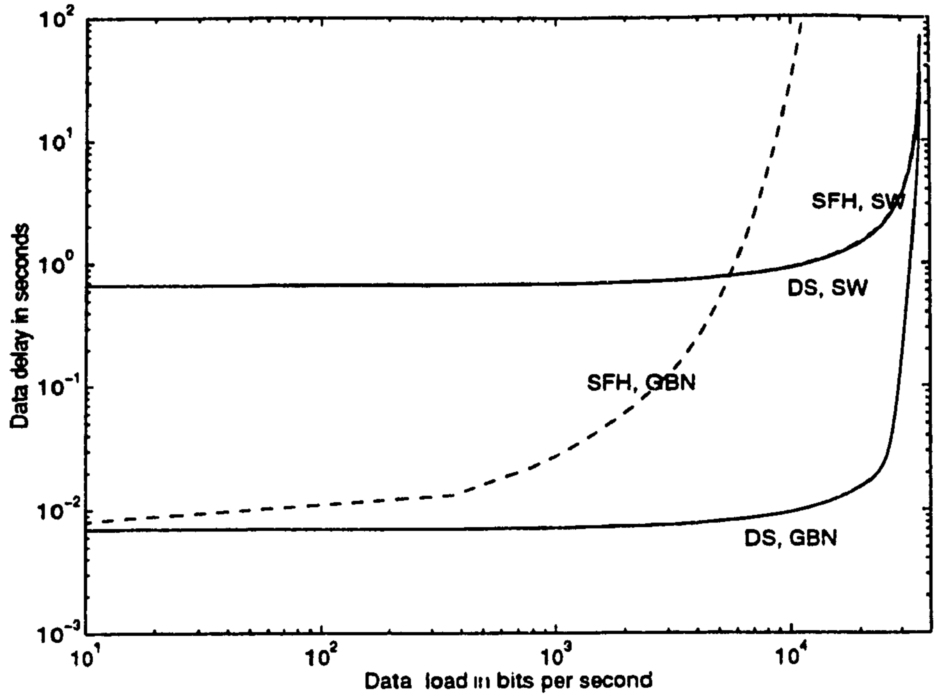

Figure 5.11: Data delay vs data load for DS and SFH CDMA using S/W or GBN ARQ. IPP model for voice assumes 2 voice calls connected transmitting at 4.8 kbps when ON

5.5.2 Results Using the IPP model

Using the IPP model, we assume voice sources transmitting at a 4.8 kb/s before channel coding and overhead (which are assumed to be the same, along with the other parameters, as Example 1. The probability of a source being active when connected is assumed to be p=0.33 and a voice buffer of B=40 packets (corresponding to a maximum delay of 40x6.5536 = 262.144 ms) is adopted. The data delay, voice loss (including losses due to packet blocking) and data throughput are plotted for increasing values of N, the number of connected voice calls per user.

figures 5.11 to 5.13 show these plots for N=2. The results obtained reinforce the previous conclusion that DS systems outperform SFH systems, and that S/W schemes seem to be more suitable than GBN for SFH. Actually Figures 5.11 and

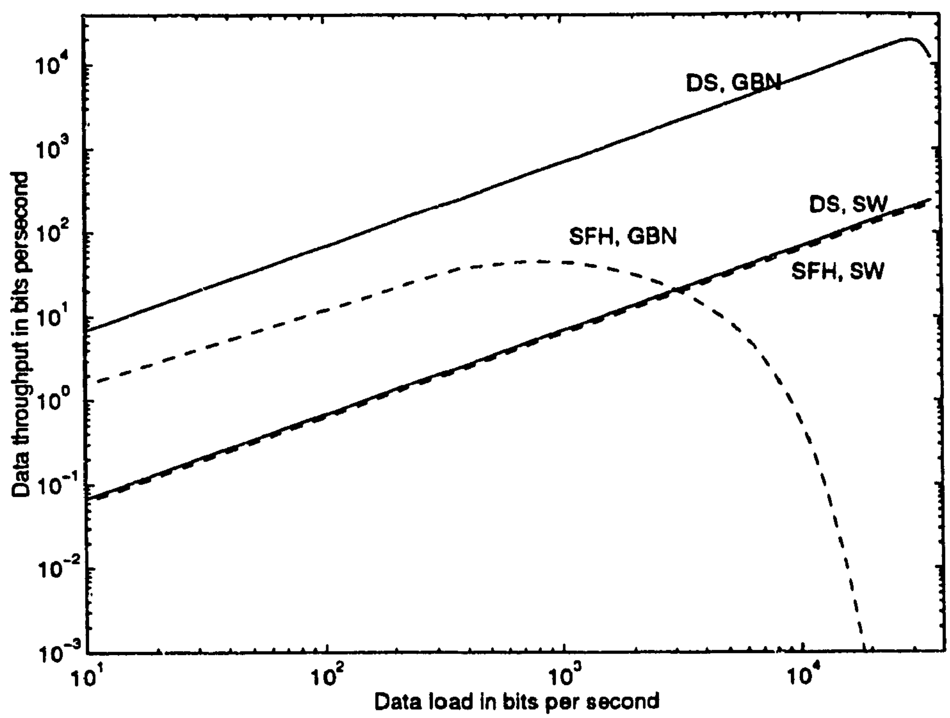

Figure 5.12: Data throughput vs data load for DS and SFH CDMA using S/W or GBN ARQ. IPP model for voice assumes 2 voice calls connected transmitting at 4.8 kbps when ON

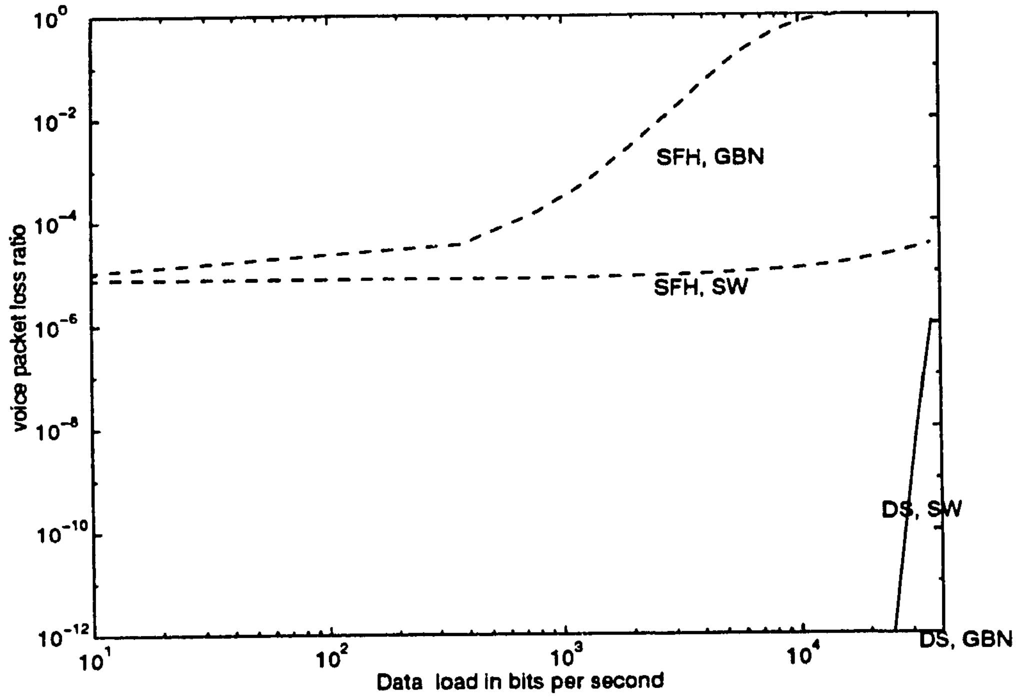

Figure 5.13: Voice loss vs data load for DS and SFH CDMA using S/W or GBN ARQ. IPP model for voice assumes 2 voice calls connected transmitting at 4.8 kbps when ON

5.12 on one hand and 5.13 on the other, show that SFH systems achieve comparable performance with DS systems when S/W scheme are used, albeit with more (but reasonable) voice loss.

Finally, It is interesting to compare Figures 5.8-5.10 to 5.11-5.13 as they correspond to the same case except that the model is different (Markovian vs IPP) and a voice buffer is assumed (generating more voice loss and limiting voice delay). One can see that the graphs are similar, although the Markovian model predicts more pessimistic results for voice loss and data delay in the case of SFH systems.

5.6 Conclusions

An analysis of the performance of satellite-switched SFH and DS CDMA systems in an integrated environment has been made assuming one of two ARQ schemes for data and two models for packetized voice. It was observed that while voice delay and delay jitter can be minimized in all cases using finite voice buffers and preemptive priority for voice packets, DS systems appear to be much more effective than SFH in all cases and for both voice and data in an environment where fading effects could be neglected. From the current literature about the physical layer performance of both systems in fading, it is not expected that fading will reverse this favorable comparison for DS systems.

Adopting a preemptive priority queueing policy favoring voice packets has shown that DS sequence systems can be very effective with voice transmissions even at loads close to 100%.

Another interesting result is that SFH systems perform relatively well when S/W rather than GBN schemes are employed for data. This is because multiple access interference is more destructive in SFH systems, and follows from the fact that S/W systems do not clogg the channel with redundant transmissions as is the case of GBN methods.

It is to be noted that the inclusion of SFH in our analysis in this chapter has the sole purpose of choosing one CDMA system to compare it with TDMA systems in realistic fading environments for satellite-switched networks. It appears from the comparison in this chapter that a DS system would be the best bet to actually compete with TDMA system, thus it will be chosen for the comparison in the next chapter.

Next, one needs to incorporate the effect of shadowing and fading on performance. Future work will also include comparisons of CDMA systems and hybrid CDMA/TDMA schemes (such as those introduced in [10]) with TDMA systems in these environments using realistic traffic models for voice.

The next chapter will discuss several analytical shadowing and fading models and compare CDMA with TDMA in those environments.

Chapter 6

Comparison of CDMA, TDMA and Hybrid Integrated Voice/Data Networks for Satellite Mobile Fading Channels

6.1 Introduction

This chapter presents an analysis of DS CDMA, TDMA and a variable frame TDMA/CDMA and compares their performance in an integrated voice/data transmission environment assuming similar protocols to the previous chapter while also taking into account the effect of fading and shadowing on the BER performance. The next section will present two channel models that are valid for two different satellite links (LEO and GEO mobile links) in the L-band and the BER performance prediction based on those models, while the rest of the chapter will deal with issues such as packet transmission protocols and analytical prediction of data ARQ performance for all systems involved.

The intended application of the protocols analyzed is in the broadband transmission of multimedia calls, but since analytical models for the Ka band (which is the most eligible for broadband satellite networks), and since the L-band's available bandwidth is limited, the analysis will be confined at this stage to narrowband voice/data transmission performance, assuming that similar results would be expected for broadband systems if the fading models are equally valid for the Ka band.

6.2 List of Symbols

All symbols and equations used in Chapter 5 still apply. The following list defines the symbols that are used only in this chapter:

I_0: Bessel function of the first kind with zero order.

P_{cl}: Probability distribution of the number of voice calls per mobile.

T_b: Bit period.

T_m: Time delay spread in the multipath components.

b_0: Average total power in the multipath components.

b_i: Average power in the i'th component of the multipath.

erfc: Complimentary error function.

r: Signal amplitude in the receiver, a random number.

μ_0: Mean value of the Lognormal distribution.

α: Fraction of the time the channel suffers from shadowing.

6.3 Channel Models for CDMA and TDMA Systems

6.3.1 Lognormal Shadowed Rician Model (LSRM)

This model has been tested experimentally in [71] and derived analytically in [72]. A more recent article by the same authors [73, 74] uses the results in a study of the throughput performance that was done in parallel with our work in [10]. The model depends on whether or not we are dealing with flat fading, meaning that the chip duration T_c is less than the delay spread T_m.

The equations quoted below are equally valid for narrow-band TDMA applications as well as broadband CDMA systems.

Let $p(r)$ be the probability density function of a narrowband received signal envelope r (as given in Chapter 4) of one specific user signal affected by multipath fading (as shown in Fig. 5.7) plus Log Normal shadowing. P(r) is assumed to have the following form:

$$p(r) = \frac{r}{b_0\sqrt{2\pi d_0}} \int_0^{\infty} \exp\left[-\frac{(\ln(z)-\mu_0)^2}{2d_0} - \frac{r^2+z^2}{2b_0}\right] \times \frac{I_0(rz/b_0)}{z} dz, \tag{6.1}$$

where μ_0 and d_0 are, respectively, the mean value and the variance of the signal after shadowing, b_0 is the average total power in the multipath, and I_0 is the Bessel function of the first kind with order zero. Now, adopting a more realistic distribution for the power in the multiple paths than in Chap. 4), let b_i be the average power in path i, then it can be approximated by [75]

$$b_i = b_0\left[1 - \exp(-\frac{T_C}{T_m})\right] \exp\left[-(i-1)\frac{T_C}{T_m}\right], \tag{6.2}$$

assuming a DS CDMA with a chip rate T_C and BPSK modulation. Furthermore, the receiver is assumed to be able to track the code and carrier phase, resulting in the following probability of error for narrowband signals (which applies for TDMA for example):

$$P_e = \frac{1}{2}\int_0^\infty \mathrm{erfc}\,(SNR_1(r)))\,p_r(r)dr, \tag{6.3}$$

where

$$SNR_1(r) = \frac{r}{\sqrt{2N_0/T_b}}. \tag{6.4}$$

Now for DS CDMA systems, we will consider two cases, the case when $T_m > T_C$ and the case when $T_m < T_c$. In the second case (flat fading), the probability of error is

$$P_e = \frac{1}{2}\int_0^\infty \mathrm{erfc}\,(SNR_2(r)))\,p_r'(r)dr, \tag{6.5}$$

where [74]

$$SNR_2(r) = \frac{r}{\sqrt{2\left[\frac{N_0}{T_b} + \frac{2(M-1)A^2}{3U}\left(b_0 + \frac{\exp(2\mu_0+d_0)}{2}\right)\right]}}, \tag{6.6}$$

and $p_r'(r)$ is obtained by replacing b_0 by b_1 (the power in first reflected path) in the expression for $p\ (r)$.

In the case where the chip time is less than the time spread, diversity can be used (as in the RAKE receiver and one can use equations similar to the ones introduced in [75], this is an easy extension of the current work and will not be discussed in this thesis).

6.3.2 Rayleigh Shadowing Model

Another model that was proposed for LEO systems and verified in [76] adopts a probabilistic approach towards the shadowing phenomenon by assuming the probability density of the received signal to be Rician part of the time and Rayleigh part of the time.

It is then given by the following equation:

$$p_r(r) = \alpha\frac{2r}{b_0}\exp(r/b_0) + (1-\alpha)\frac{2r}{b_0}\exp\left[-\frac{r^2+A^2}{b_0}\right] I_0\left(\frac{rA}{b}\right) \tag{6.7}$$

where α is a parameter determined by experimentally. This probability distribution can be used to calculate the average probability of bit error as in the above section. It is interesting to note that a closed form expression for this probability of error will not be attempted and numerical integration will be used. Another alternative was to use the already available expressions and approximations used for Rayleigh and Rician PDF's and combine them using α assuming independence between the probability of shadowing and the shape of shadowing. The probability of correct packet reception assuming a BCH code and a type I hybrid ARQ can be then calculated using the same equations introduced in the previous chapter.

6.4 Queueing Analysis of CDMA and TDMA in the Link Layer

In this chapter, we will adopt a similar approach to Chapter 5 but the analysis will be generalized to include applications where the number of users per mobile is small, while the number of mobiles is much larger than the number of channels. In this case, we could have U mobiles connected to the network each having either a TDMA slot or DS code reserved. Within each mobile, any combination of data traffic and from 0 to N_{max} voice calls are being packet switched as explained in Chapter 5. N_{max} could be set to 1, in which case we would be dealing with single-user mobiles which transmit voice or data or both, or it could be any set number to allow for an application where a number of voice terminals are sharing the mobile transmitter.

Ideal ISDN systems assume integration schemes would allow for three classes of calls [77]: circuit switched voice, circuit or packet switched data and packet switched data. The second class would use the available circuit switched channels and/or the silence periods of the voice calls, and as a last resort use the packet switched channel. The third lowest priority class would use only packet switching (see Fig. 6.1).

In our case, only packet switching is considered all traffic with two priority

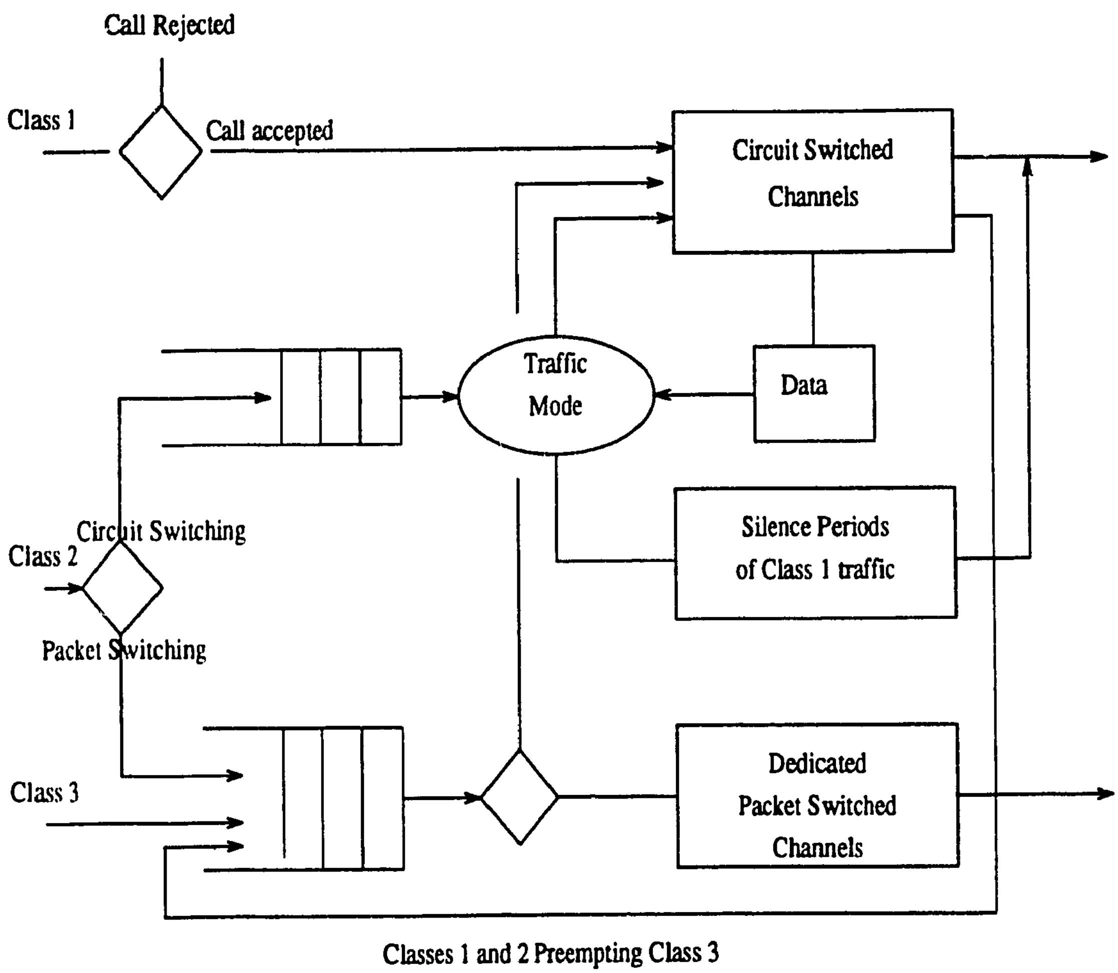

Figure 6.1: A schematic diagram on how three classes are multiplexed in an integrated services transmitter

classes voice and data. The maximum number of simultaneous calls is still assumed to be equal to the spread spectrum gain, and packets from voice calls are assumed to be asynchronously multiplexed with data packets originating from the same mobile. Time or code division is used to multiplex the aggregated traffic from different mobiles..

The analysis in Chapter 5 assumed for simplicity that N, the number of connected voice sources per connected transmitter, is a given constant. However, this is not always the case: A connected mobile, as explained earlier, might have only data traffic, or have any number of voice sources up to the maximum number allowed, N_{max}.

The number of connected voice sources at any instant is a random number that is given by birth-death process with a maximum number of customers equal to N_{max}, The number of phone lines available in each mobile. Assuming a given required

Although we are not dealing with circuit switching, as none of the customers reserves the channel for the duration of the call, Blocking of calls is used to insured a minimum measure of congestion control. Assuming a given population of voice terminals per cell, and given a required probability of voice call blocking one can determine the maximum number of voice calls that is allowed to be connected *per mobile* from Erlang's B formula, as this is a classic $M/M/N_{max}/N_{max}$ queueing case. Assuming a voice call arrival rate of 5 Erlang per mobile[1], the approximation given in [67] is used. the probability distribution, $P_{cl}(N)$, of the number of voice calls connected per mobile is then computed and the average voice traffic load per mobile is computed as follows:

$$\rho_v = \sum_{N=0}^{N_{max}} \rho_v(N) P_{cl}(N) \tag{6.8}$$

[1]Although this seems to be a high number, one notes that the model assumes a large number of calls within each mobile. Assuming 100 potential users, each with an average utilization rate of 0.05 Erlangs (3mn/hour), 5 Erlangs per mobile becomes very reasonable in this case

The average voice traffic is then computed based on the equilibrium probabilities of the number of active voice calls.

The *potential capacity* of the systems is therefore set to 5x256 = 1280 Erlangs for voice traffic, while the capacity for data will be determined from the results of the analysis of the effect of the data traffic on the voice and data calls.

The reason voice capacity is called a potential one is because the calls are actually packet switched and *no guarantee* of capacity is made. " A connected voice call" is not a voice call with a dedicated line as in the case of circuit switching. ACtually, the analysis in this chapter has an aim of actually determining the feasibility of this potential capacity in an integrated environment. As was mentioned earlier, the voice call blocking is done to reduce congestion, and blocking (or discarding) at the packet level is done to insure delay does not reach a threshold value. despite these two restrictions, it will be shown that a good performance for voice can be insured.

It is assumed that voice and data calls are independent, and that while a data call is connected, any number of voice calls can be connected within the same mobile up to N_{max}. The distribution of the number of connected voice calls is given by the Erlang distribution [67]:

$$p_{cl}(N) = \frac{A^N/N!}{\sum_{i=0}^{N_{max}} \Lambda^i/i!} \tag{6.9}$$

where Λ is the total voice traffic load (in our case 5). The probability of voice call blocking is then given by $p_{cl}(0)$. For a probability of blocking of 0.01, N_{max} turns out to be 11.

6.5 Numerical Results

Three cases of shadowing will be considered and compared for both CDMA and TDMA, negligible showing with Rician flat and frequency selective fading, medium shadowing with flat fading and low shadowing with flat fading. In all cases data packets are competing with voice packets coming from a random number of voice calls who have preemptive priority.

6.5.1 Case I : Multipath Fading with Negligible Shadowing

The following parameters were assumed for the Markovian model for both voice and data: U=256 connected users, L=256 bits, an overhead of 24 bits, W=10MHz, $\bar{E} = 1$ packet/data message, and $E_b/N_0 = 15\ dB$. The Time delay spread was assumed to be equal to An extended (256,164) BCH code was used for pure error correction for voice and simultaneous correction and detection for data (Hybrid type I ARQ) The length of the frame being 6.5536 ms, the roundtrip delay is approximated at $s = 2$ frames. Fig. 6.2 shows graphs of data delay as a function of average total traffic load assuming a maximum of 11 voice calls connected per mobile corresponding to a voice traffic load of 5 Erlang per mobile and a probability of blocked calls of 0.01. The fading multipath spread is assumed to be $1\mu s$. Fig. 6.3 and 6.4 show the graphs of data throughput per mobile, and voice packet loss (respectively). Fig. 6.5, 6.6, and 6.7 show the same parameters evaluated at a more severe fading case $T_m = 5\mu s$.

It is clear from the graphs that CDMA is much more feasible in fading environments. Fig. 6.5 and Fig. 6.6 show how TDMA becomes inefficient at large delay spreads, while CDMA still delivers acceptable performance. It is to be noted that the extreme traffic conditions chosen above where designed to favor TDMA as it performs best when traffic is close to capacity. The results however assume a fixed-frame TDMA, which limits the multiple access capacity of TDMA in a variable traffic environment. while a dynamic allocation of slots is preferable, it is very difficult to use to transmit in other mobiles's silence periods for example. Furthermore, it is to be noted that the extreme traffic conditions chosen above where designed to favor TDMA as it performs best when traffic is close to capacity.

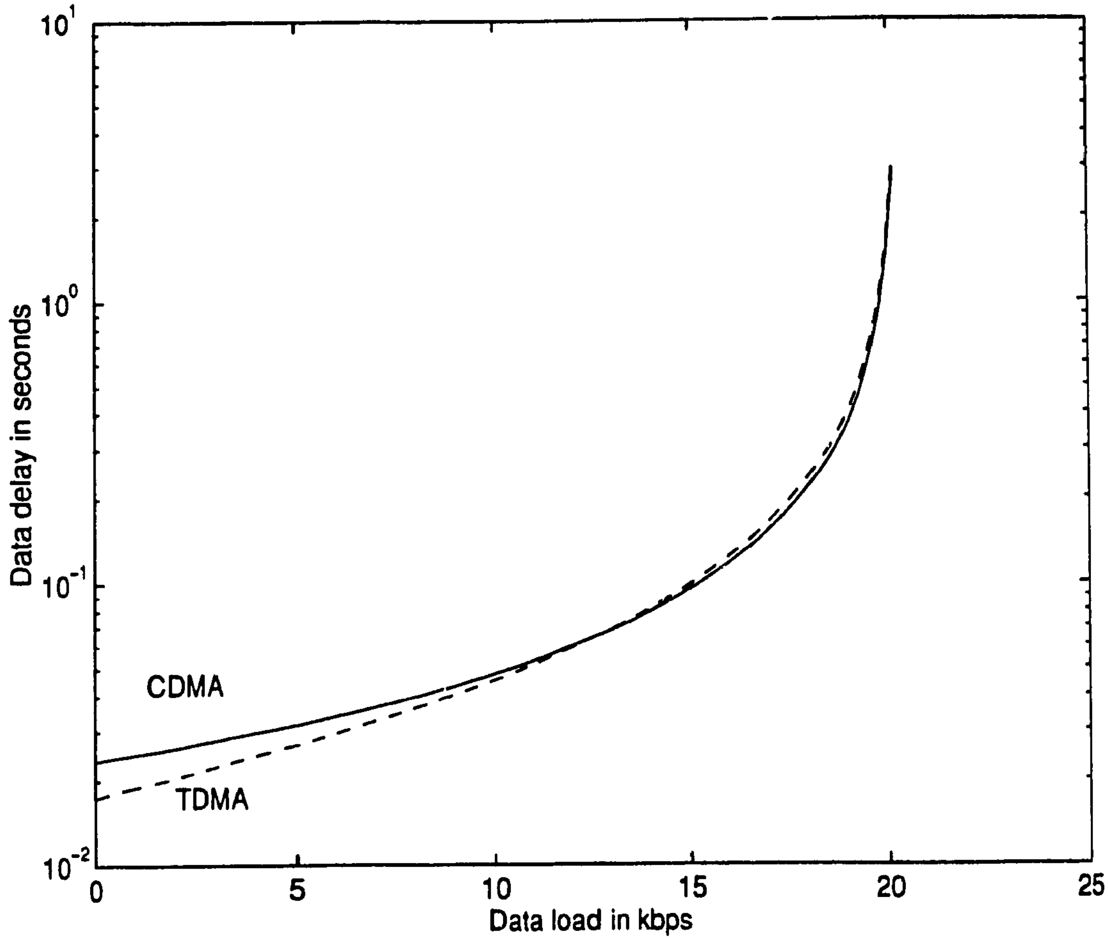

Figure 6.2: Delay vs Data load in Rician Fading, TDMA vs CDMA, $T_m = 1\mu s$.

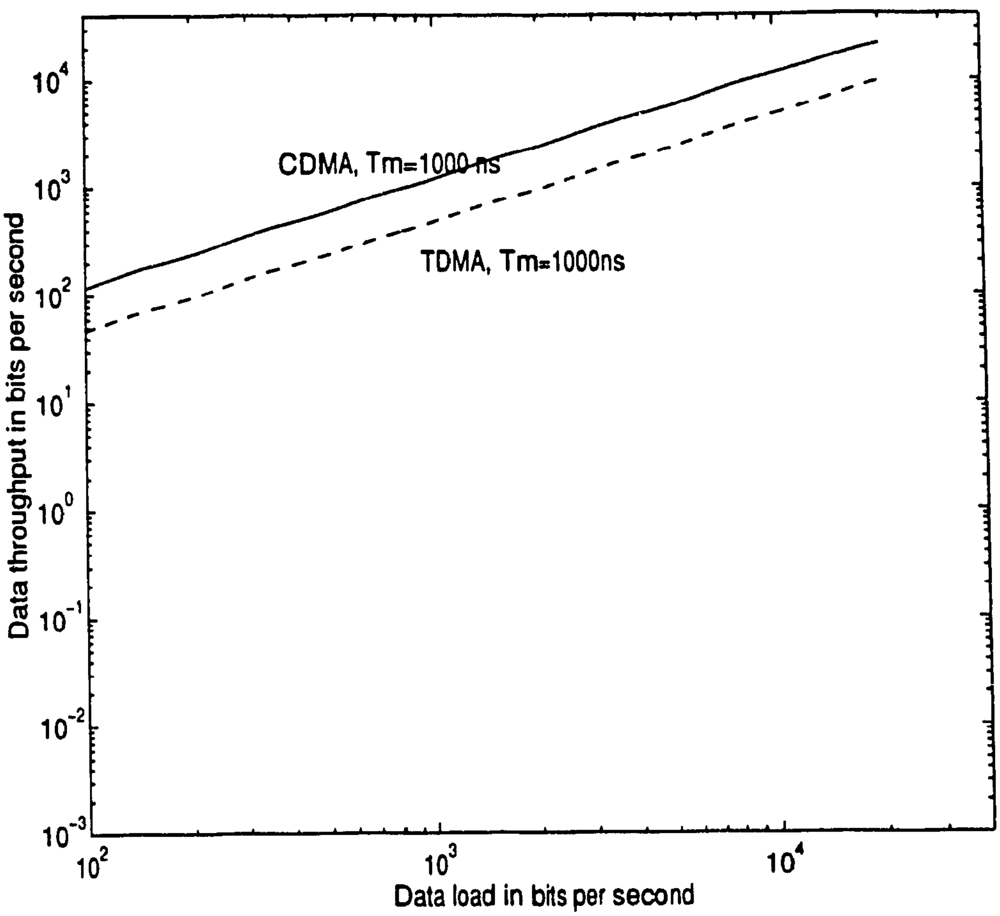

Figure 6.3: Data throughput vs Data load in Rician Fading, TDMA vs CDMA, $T_m = 1\mu s$

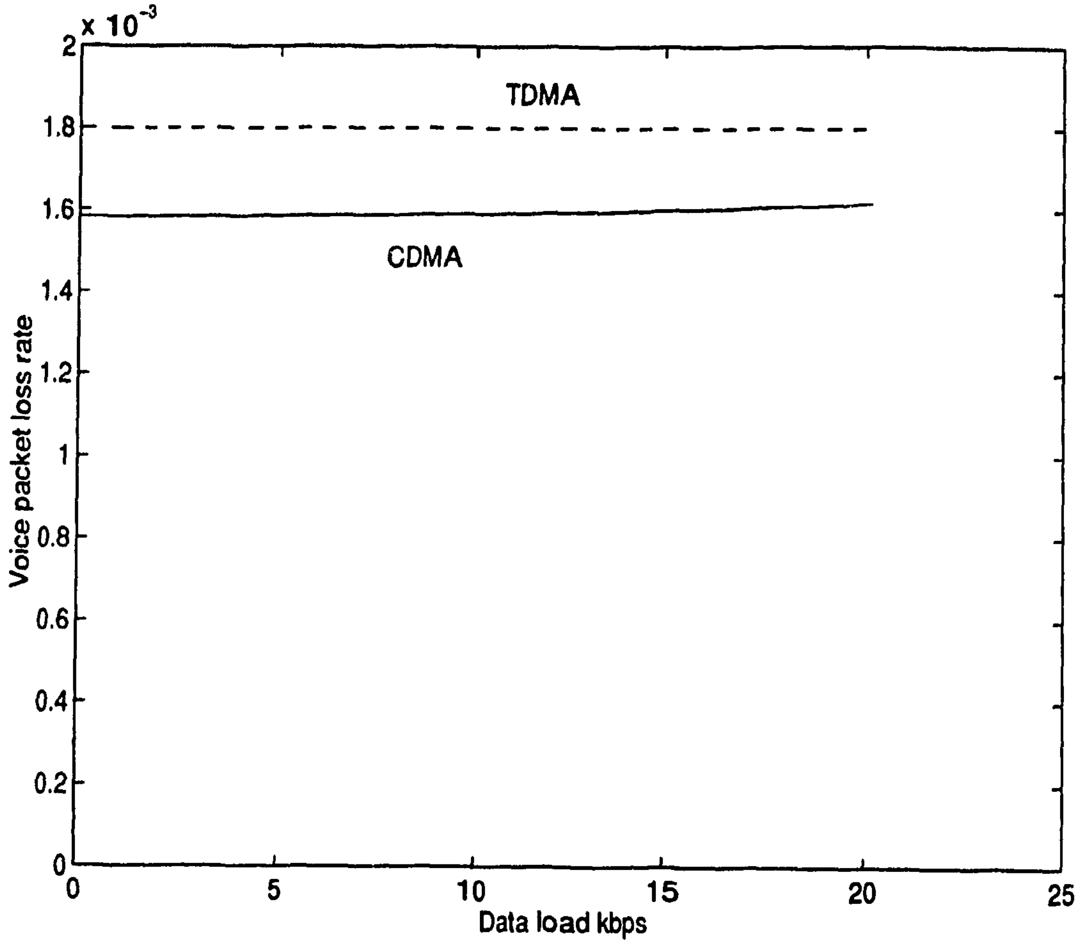

Figure 6.4: Voice packet loss vs Data load in Rician Fading, TDMA vs CDMA, $T_m = 1\mu s$

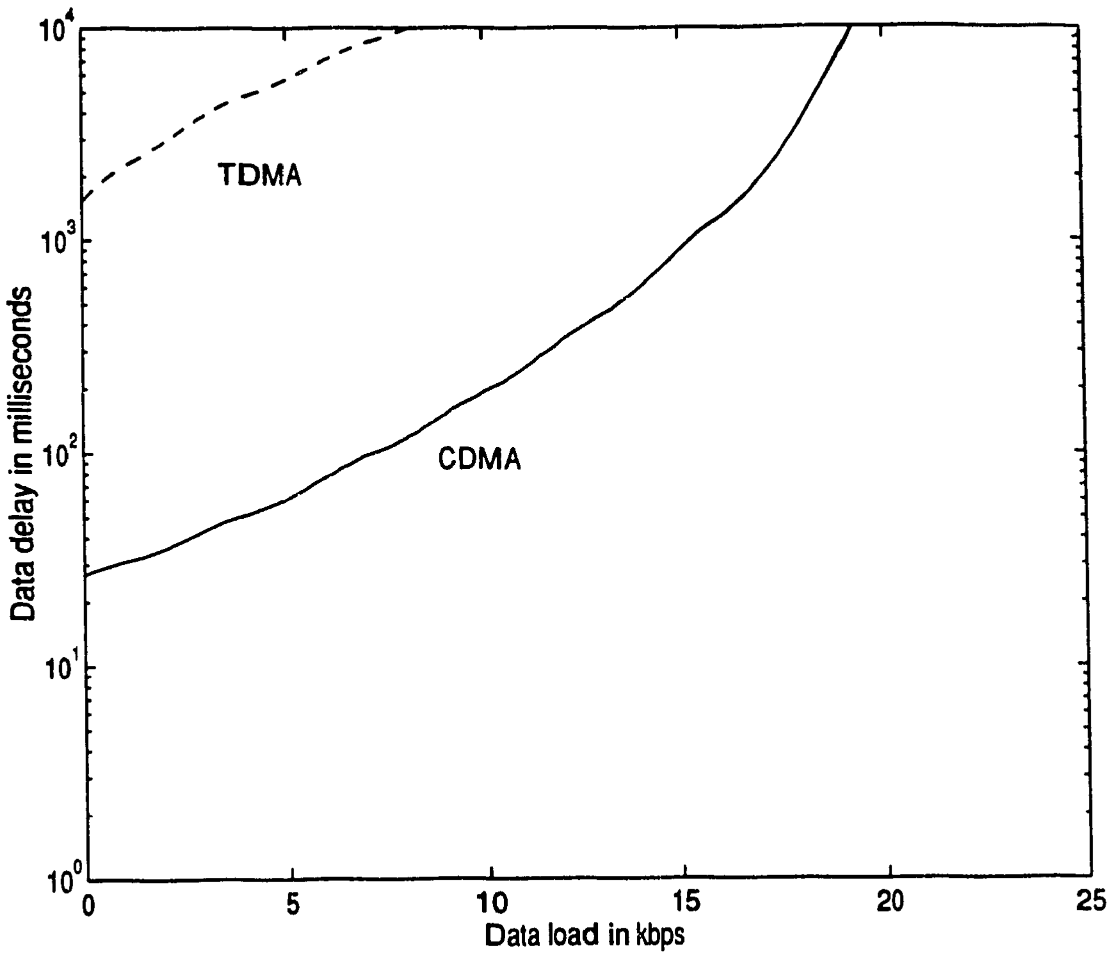

Figure 6.5: Delay vs Data load in Rician Fading, TDMA vs CDMA, $T_m = 5\mu s$.

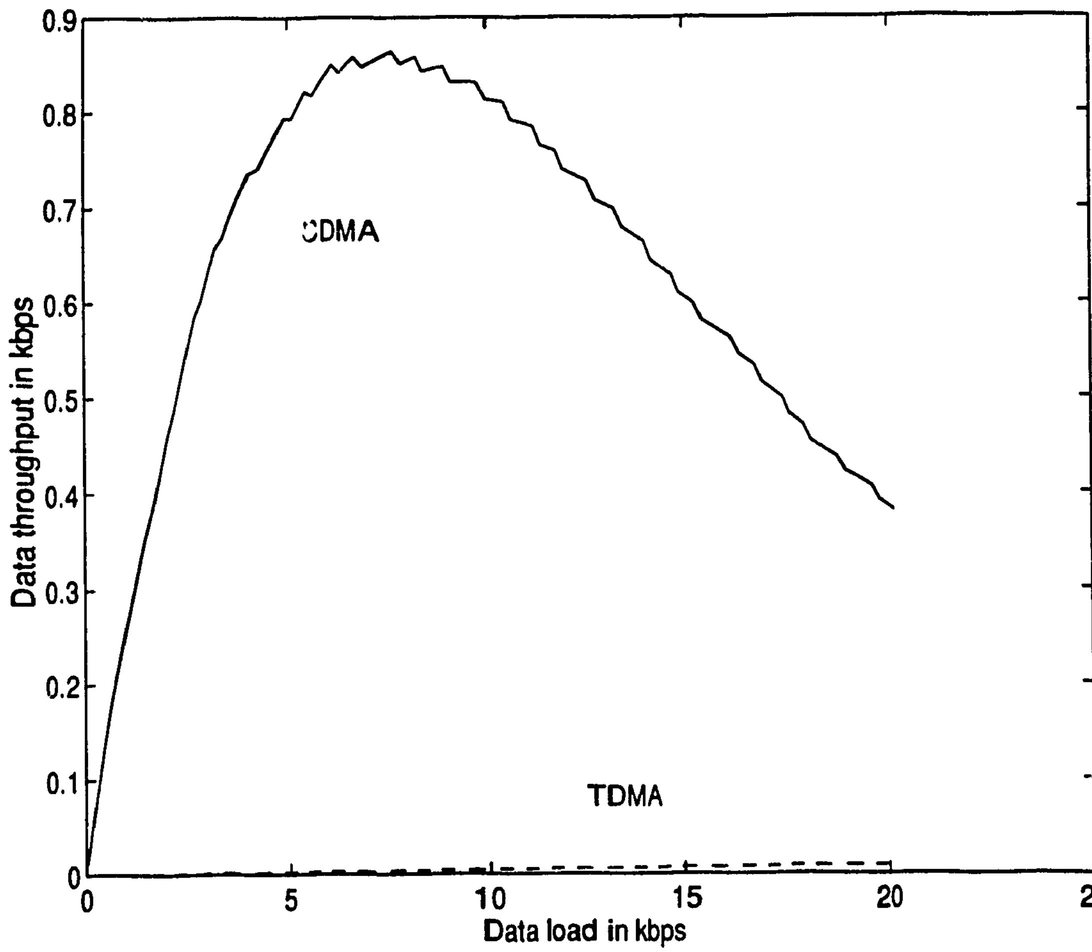

Figure 6.6: Data throughput vs Data load in Rician Fading, TDMA vs CDMA, $T_m = 5\mu s$

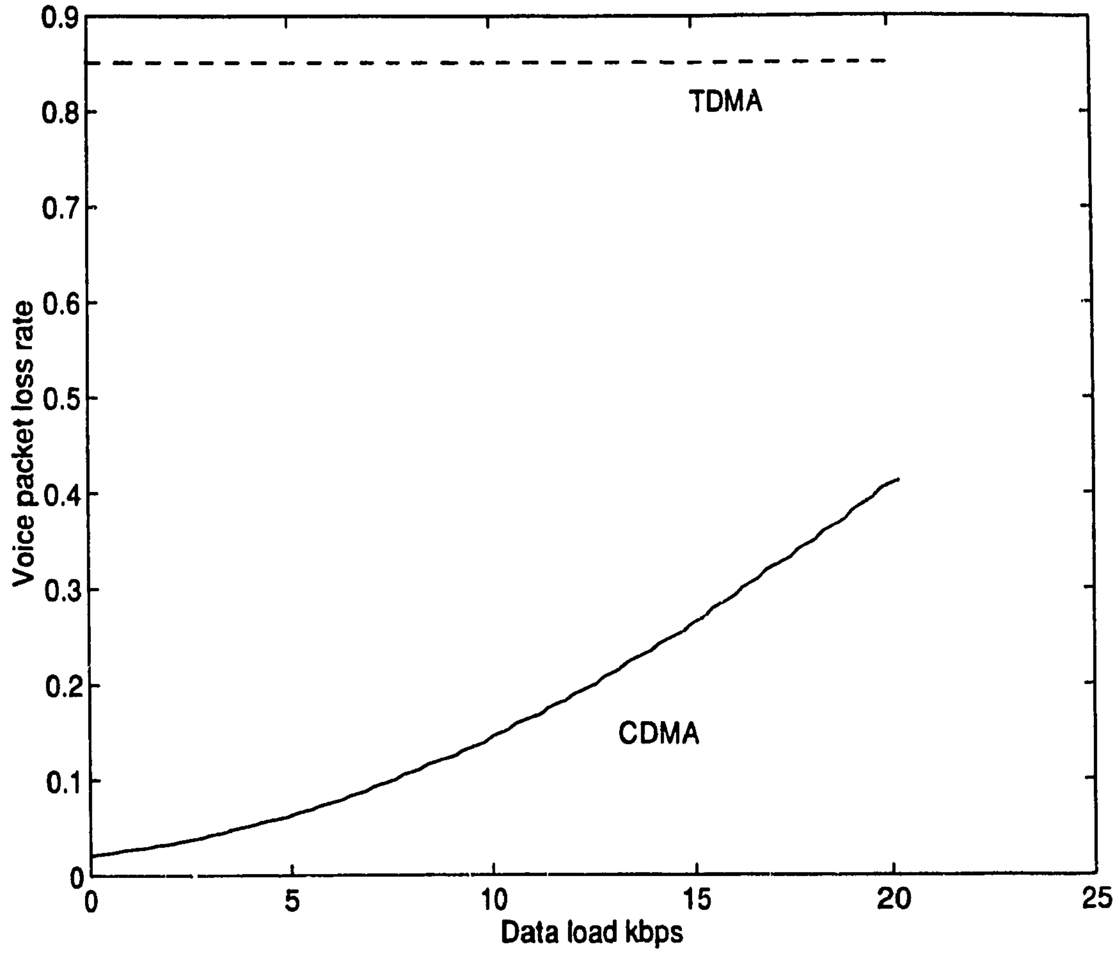

Figure 6.7: Voice packet loss vs Data load in Rician Fading, TDMA vs CDMA, $T_m = 5\mu s$

6.5.2 Results with shadowing and Rician flat fading

Numerical results for data delay, throughput, and voice loss rate were computed assuming the same assumptions above, except that the channel is assumed to be shadowed with probability of shadowing $\alpha = 0.5$. Two values of multipath time spread were assumed: $T_m = 10^{-7}$ seconds and $T_m = 10^{-6}$ seconds.

Figs. 6.8, 6.9, and 6.10 show the values of delay, throughput and voice loss respectively. It is not clear from these graphs if any of the two system outperform the other, and since TDMA is simpler, this might give an edge to TDMA systems. The value of multipath spread seems to be the key value here.

Now, Figs 6.11, 6.12, and 6.13 show the same graphs when the multipath spread is higher and while the fading is still flat, the corruption of the TDMA bit is higher now. The results with shadowing seem to be in high contrast with those in the previous subsection.

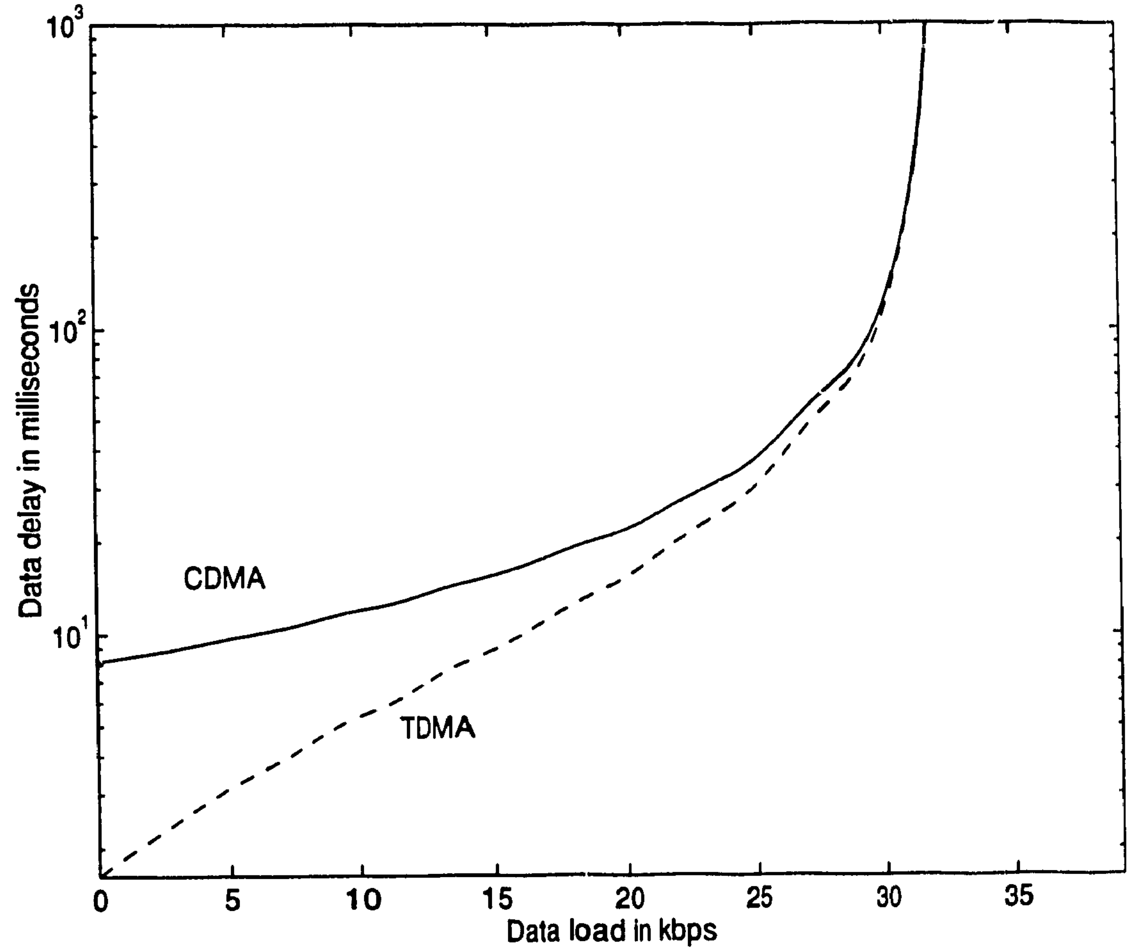

Figure 6.8: Data delay vs data load in shadowed Rayleigh/Rician Fading channels , T DMA vs CDMA, $T_m = 0.1\mu s$, $\alpha = 0.5$

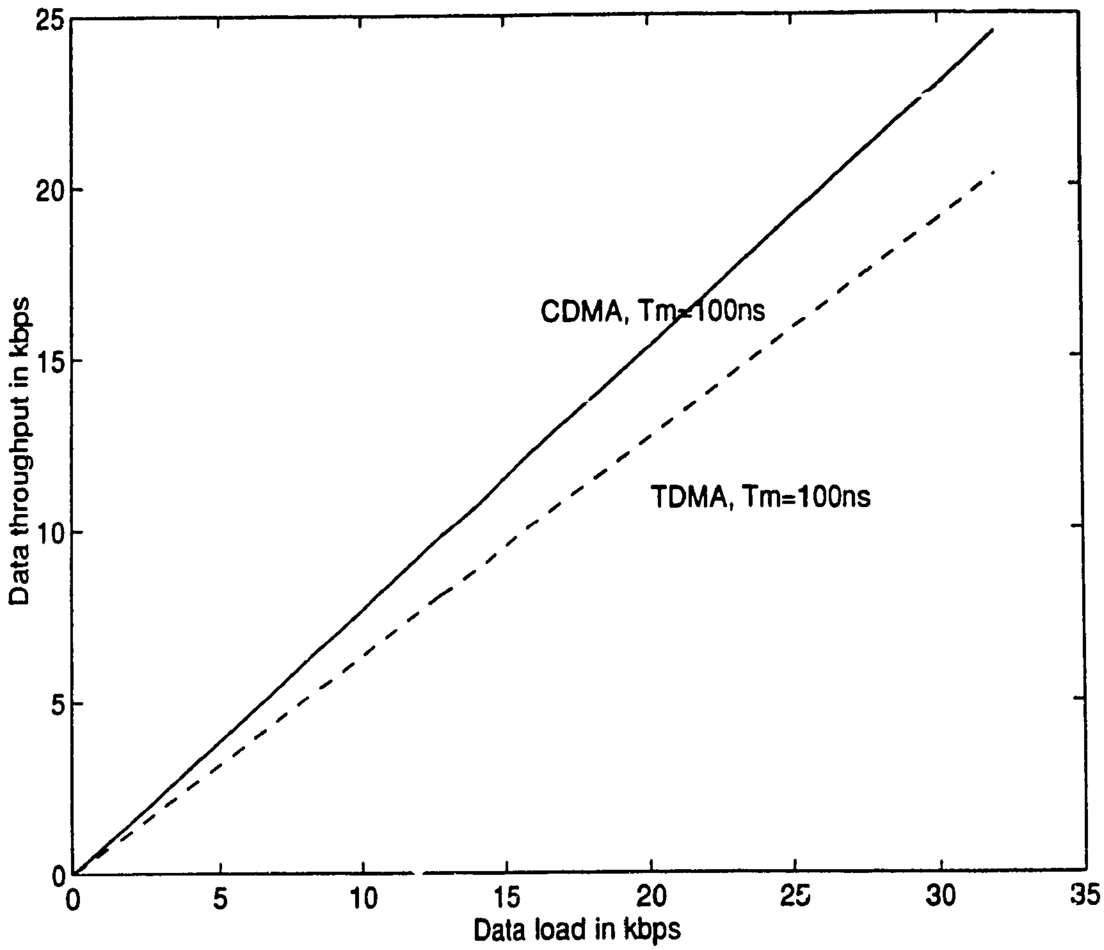

Figure 6.9: Data throughput vs data load in shadowed Rayleigh/Rician Fading channels, TDMA vs CDMA, $T_m = 0.1\mu s$, $\alpha = 0.5$

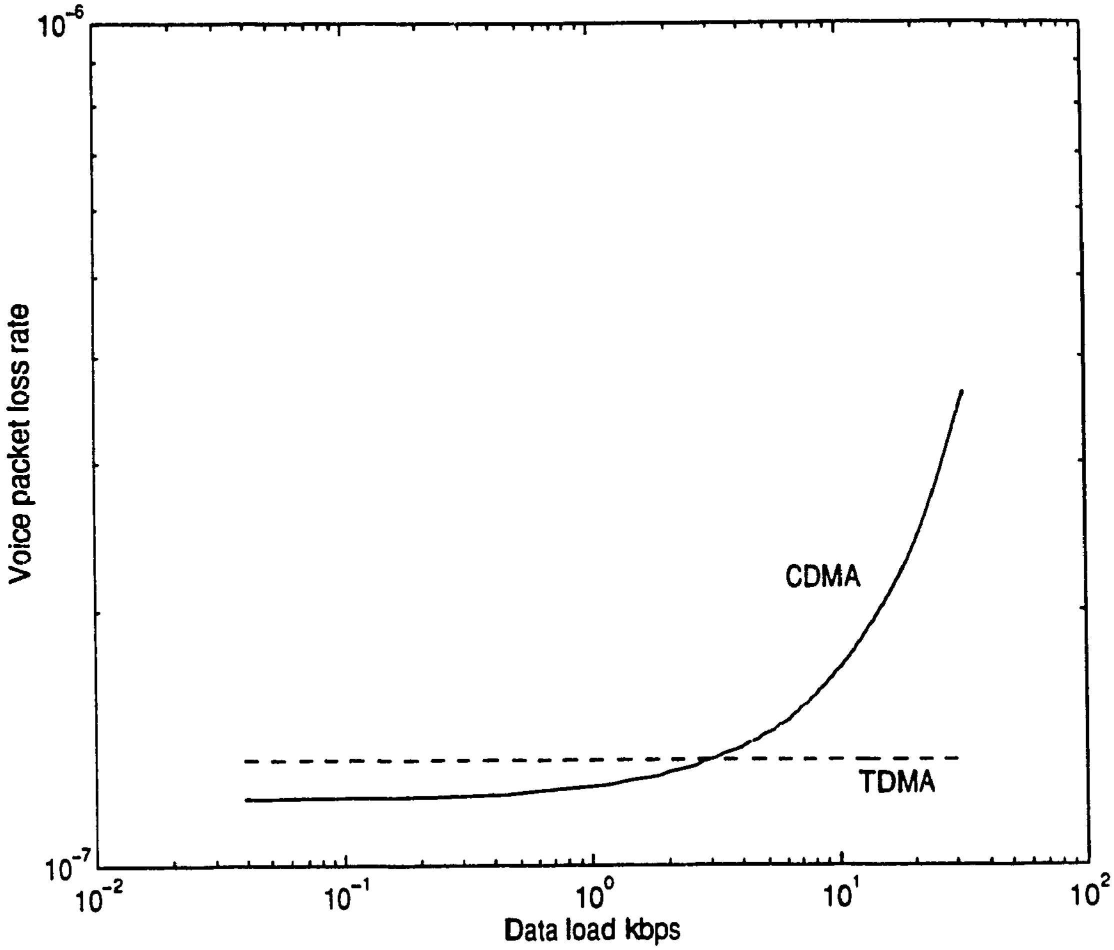

Figure 6.10: Voice loss vs data load in shadowed Rayleigh/Rician Fading channels, TDMA vs CDMA, $T_m = 0.1\mu s$, $\alpha = 0.5$

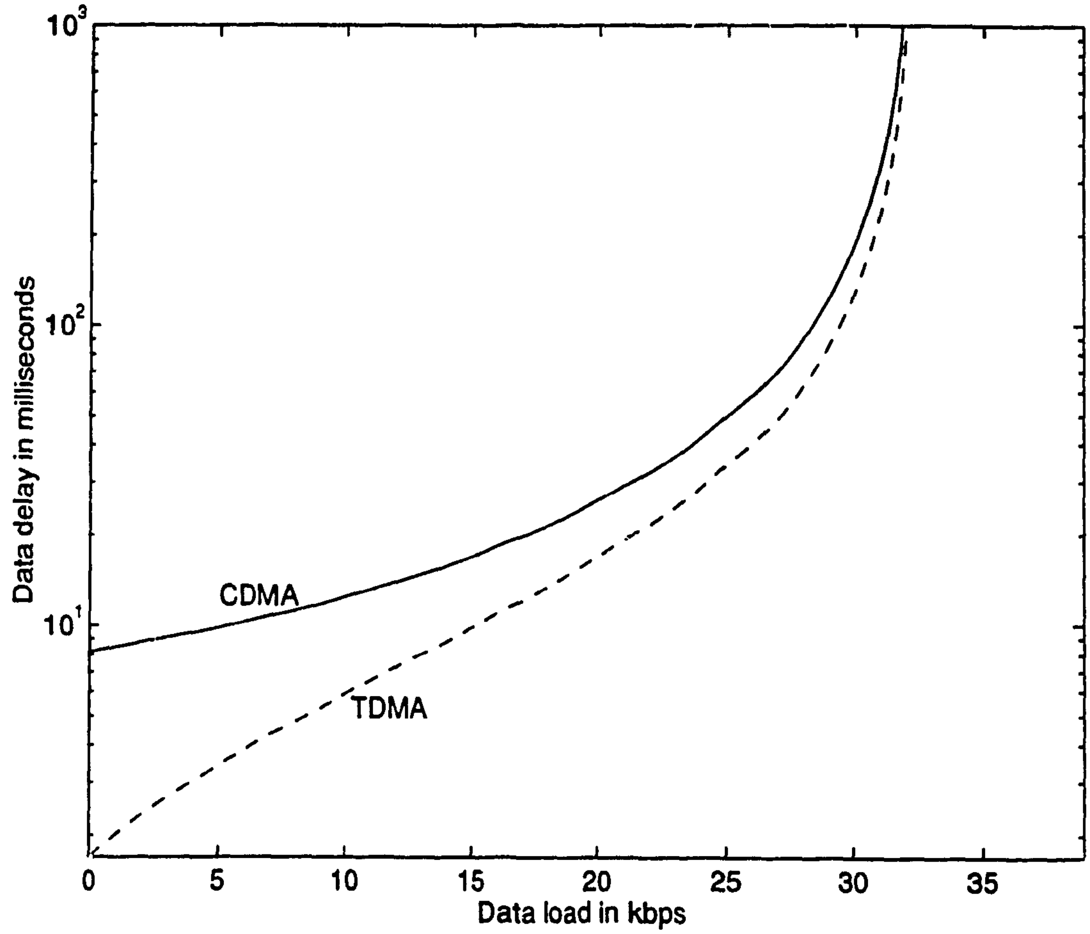

Figure 6.11: Data delay vs data load in shadowed Rayleigh/Rician Fading channels, TDMA vs CDMA, $T_m = 1\mu s$, $\alpha = 0.5$

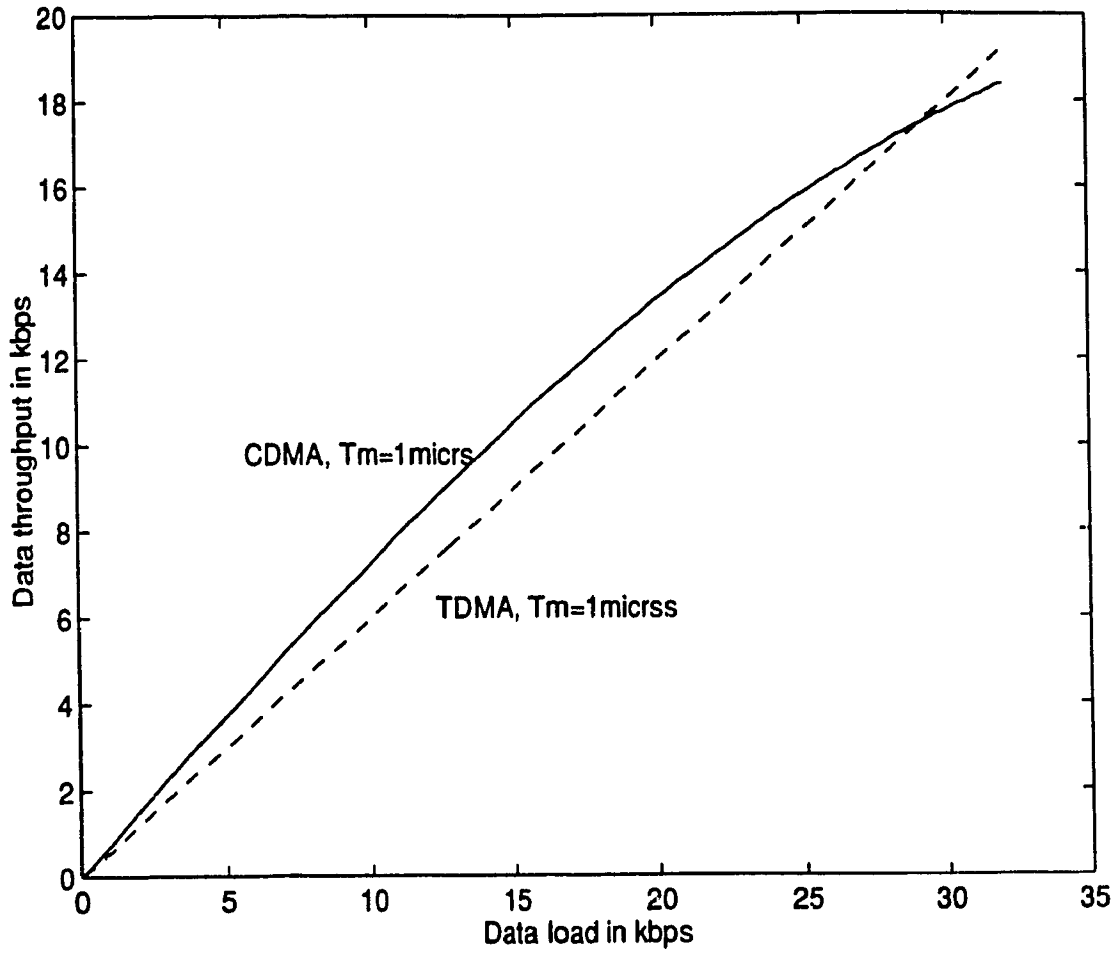

Figure 6.12: Data throughput vs data load in shadowed Rayleigh/Rician Fading channels, TDMA vs CDMA, $T_m = 1\mu s$, $\alpha = 0.5$

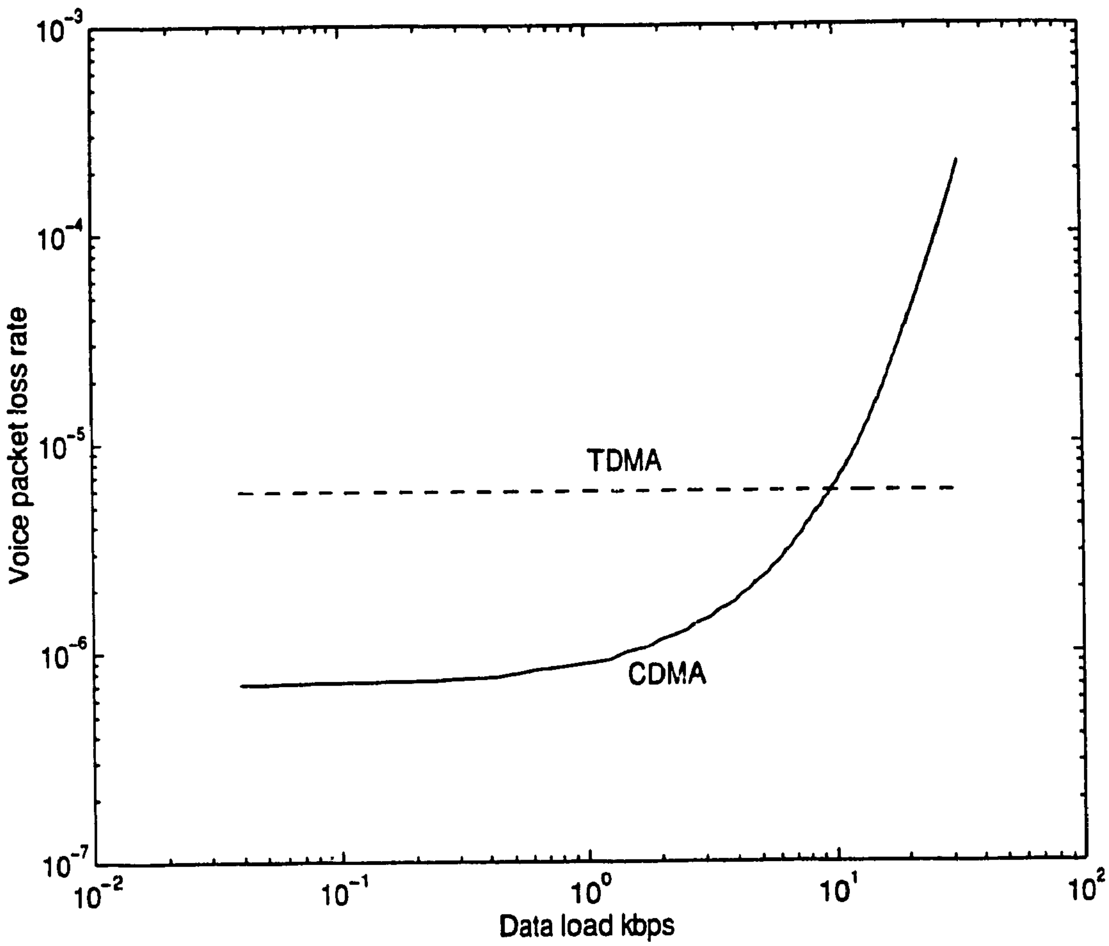

Figure 6.13: Voice loss vs data load in shadowed Rayleigh/Rician Fading channels, TD MA vs CDMA, $T_m = 1\mu s$, $\alpha = 0.5$

6.6 Conclusions

An analysis of the performance of mobile CDMA and TDMA systems in an integrated environment is made assuming a type-I hybrid GBN ARQ scheme for data and buffered packetized voice. It is observed that while voice delay jitters are minimized in all cases using finite voice buffers and preemptive priority for voice packets, CDMA systems appear to be more effective than TDMA even when traffic loads are kept to maximum levels in an available bit rate basis. The analysis results are highly dependent on the severity of multipath and on the Rician model for multipath. Shadowing seems to be a handicap for CDMA, but for satellite systems Near-Far problems are not an issue, and power control may be more successful.

Chapter 7

Conclusions and Suggestions for Future Research

In this thesis, a comparative analytical study of some link-layer protocol performance of direct sequence and slow frequency hopping CDMA was performed. This study included comparisons with TDMA and introduced new hybrid traffic-adaptive systems. Two main contributions were made as a result of this work: 1) ARQ methods were shown to influence the comparisons of CDMA to TDMA. It was shown that in an integrated environment, CDMA still performed better then TDMA in spite of the increased load caused by ARQ, but only in severe fading and unshadowed environments. 2) Hybrid systems were shown to be better alternatives to both pure CDMA and pure TDMA in fading environments.

The multiple access systems assumed the same total bandwidth and maximum total number of users (population). The same error-correcting codes were used and a mixed traffic of both voice and data will be assumed. The delay and throughput characteristics were then expressed in terms of average traffic intensity from all users assuming various fading environment types. The network was assumed fully integrated where any terminal can send data or voice at any time and where two queues were assumed in every transmitter, one for voice and one for data. Numerical results were computed assuming some realistic values of traffic characteristics and bandwidth and optimization issues were investigated numerically.

In Chapter 4, where we showed that two hybrid systems outperformed regular CDMA and TDMA for low to medium average traffic intensity in Rician fading for both voice and data traffic, ARQ issues were not discussed and no integration protocol was analyzed. It however provided an interesting result: hybrid adaptive CDMA/TDMA systems may provide the best performance in cases were the traffic on the channel varies widely but slowly from very low loads where CDMA does much better than TDMA in fading, to loads that are more than 50% where TDMA emerges as the better alternative and CDMA self destructs like an ALOHA system.

Chapter 5 provided an analysis of the performance of ARQ schemes for data in integrated-services DS-CDMA networks, and a comparison was made to with SFH systems. Fading was assumed negligible at this stage. In this chapter, a more realistic model was also assumed for voice (IPP model) and a hybrid system was assumed where voice and data is asynchronously multiplexed locally in mobiles and than spread spectrum is used to provide multi-access in the satellite channel. Numerical results showed that DS-CDMA system are much more feasible than SFH systems in such an environment and that GBN ARQ, although superior to SW ARQ, does not achieve higher maximum throughputs in all cases. It was also shown that moderate data traffic does not affect voice performance very much if preemptive priority is given to voice packets, but voice loss becomes extensive with very high data traffic load, and some higher layer control should done to limit data traffic when many voice calls are connected.

Chapter 6 applied three models for land-mobile satellite fading channels in the L-band, and extended the analysis of Chapter 5 to the general case of varying level of shadowing and varying multipath spread levels in a comparison of the performance of CDMA and TDMA. It was shown that CDMA is a more feasible alternative in the cases where fading is frequency selective (which is probable if the results are applied to broadband systems), and that TDMA is not necessarily worse than CDMA for cases where channels for which flat fading and moderate levels of shadowing could be assumed. The analysis assumed a full-capacity network which tends to disadvantage CDMA. L-band models where used although the future application that we have in mind would probably be a broadband network that is integrated with the terrestrial broadband (probably ATM) network. Ka-bands however, are not fully modeled yet, and our use of fast fading models is warranted only by an intuitive prediction that those models may apply to Ka-band transmissions that would have bandwidths around 30-50 MHz.

Future research projects are listed below;

1) Simulations to corroborate and/or validate some of the findings.

2) An investigation of broadband CDMA transmission using numerical approximations of the impulse response of the fading channels in Ka-band, or any analytical models that may arise in the literature (none is available to our knowledge yet).

3) Use of video and multimedia and higher layer issues involved and their effect on the performance of the system. Inter-working with ATM/Internet networks also needs to be investigated and compared to interworking between a UMTS system (European 3rd generation Universal Mobile Telecom. Service) for example and ATM/Internet.

4) Adaptive systems should be investigated in terms of their transient behavior (transient data delay, transient voice loss etc..), as this may also be a QoS figure of merit.

APPENDICES

Appendix A

Recursion for the Steady State Probabilities of the Voice Queue Size

From Equations (44) and (45), $P(k)$ for a number of active users larger then zero is given by the following matrix:

$$P(k) = [\pi(m,n)] \tag{A.1}$$

$$= \begin{bmatrix} q_k(0)+q_k(1) & q_k(2) & q_k(3) & \cdots & \sum_{i=B+1}^{k} q_k(i) \\ q_k(0) & q_k(1) & q_k(2) & \cdots & \sum_{i=B}^{k} q_k(i) \\ 0 & q_k(0) & q_k(1) & \cdots & \sum_{i=B-1}^{k} q_k(i) \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & 0 & \cdots & \sum_{i=1}^{k} q_k(i) \end{bmatrix} \tag{A.2}$$

solving for $\Pi(k) = [\pi_0(k), \pi_1(k), \ldots, \pi_i(k), \ldots, \pi_B(k)]$, such that $\Pi(k)P(k) = \Pi(k)$ for any given k, gives the following equations:

$$\begin{aligned} (q_k(0)+q_k(1))\pi_0(k) + q_k(0)\pi_1(k) &= \pi_0(k) \qquad (A.3) \\ q_k(2)\pi_0(k) + q_k(1)\pi_1(k) + q_k(0)\pi_2(k) &= \pi_1(k) \\ &\vdots \end{aligned}$$

$$\sum_{i=0}^{n-1} q_k(n-i)\pi_i(k) + q_k(0)\pi_n(k) = \pi_{n-1}(k) \tag{A.4}$$

The last equation should be redundant as the matrix should have 1 as an eigenvalue. One simple way to compute $\pi_i(k)$ is to normalize in terms of $\pi_0(k)$ and then use the fact that the probabilities must sum up to one to find the actual values [65]. Let $\gamma_i = \pi_i(k)/\pi_o(k)$ (deleting the dependence on k just for convenience in this step), then $\gamma_0 = 1$,

$$\gamma_1 = \frac{1 - q_k(0) - q_k(1)}{q_k(0)}, \tag{A.5}$$

$$\gamma_n = \frac{(1 - q_k(1))\gamma_{n-1} - \sum_{i=0}^{n-2} q_k(n-i)\gamma_i(k)}{q_k(0)}$$
$$\text{for } 2 \leq n \leq B. \tag{A.6}$$

Now since $\sum_{n=0}^{B} \gamma_n = 1/(\pi_0(k))$, all values of $\pi_n(k)$ can be readily computed from the above equations. It is to be noted, however that for k=0, boundary conditions should be used, and for cases where k is small and any summation has its lower limit larger than the upper limit, that summation is reduced to zero, and the above equations are still valid.

Appendix B

Evaluation of the mean waiting time before successful packet transmission

Equation (52), can be solved for using the following equations (from Little's formula):

$$\bar{Q}_d = \bar{W}_m \lambda_d \bar{X}_d \tag{B.1}$$

$$\begin{aligned} \bar{N}_v &= \sum_{n=0}^{\infty} \rho_v \mathrm{Prob}(W_m > n) \\ &= \rho_v \bar{W}_m \end{aligned} \tag{B.2}$$

Equation (52) then becomes:

$$\bar{W}_m(1 - \lambda_d X_d - \rho_v) = \bar{W}_0 + \bar{Q},$$

which implies

$$\bar{W}_m = \frac{\bar{W}_0 + \bar{Q}}{1 - \rho_t} \tag{B.3}$$

which can be written in terms of $\bar{X}_d$ λ_d, and the voice parameters which are for now assumed to be fixed:

$$\bar{W}_m = \frac{\lambda_d \bar{X}_d^2 + \rho_v + \bar{Q}}{1 - \rho_t} \tag{B.4}$$

$$= \frac{\rho_d \bar{X_d^2}}{\bar{X}_d(1-\rho_t)} + \frac{\rho_v + \bar{Q}}{1-\rho_t} \tag{B.5}$$

To find the waiting delay of a single data packet, one has to observe that for a preemptive resume priority a packet at the head of the line can be interrupted and forced to retransmit later if a voice packet arrives during the slot of the failed transmission. To account for that in the analysis by assuming the length of the message to include the retransmitted packets message length equal to $E' = nE_d$ where E_d is the message length. The mean value of this effective message length is equal to

$$\bar{E}' = E_d/P_d. \tag{B.6}$$

Now the mean successful data packet waiting time is the sum of the mean data message waiting time and the time of service of data cells within the message which are served ahead of that cell including earlier unsuccessful retransmissions of the same packet. The mean number of those cells, $\bar{M}$ is computed as follows:

$$\bar{M} = \sum_{i=2}^{\infty} \frac{1}{i} \sum_{j=1}^{i-1} j \times \text{Prob}(E' = i) \tag{B.7}$$

$$= \frac{1}{2} \left(\sum_{i=1}^{\infty} i \times \text{Prob}(E' = i) - \sum_{i=1}^{\infty} \text{Prob}(E' = i) \right) \tag{B.8}$$

$$= \frac{1}{2}(\bar{E}' - 1). \tag{B.9}$$

The service time for those packets obviously depends on the voice traffic as follows:

$$\bar{X}_M = \frac{\bar{M}}{1-\rho_v} \tag{B.10}$$

Substituting for the values of $\bar{E}'$, $\bar{W}_m$, and $\bar{M}$, the mean successful data packet waiting delay is:

$$\bar{W}_d = \bar{W}_m + \bar{X}_M \tag{B.11}$$

$$= \frac{\rho_d \bar{X_d^2}}{\bar{X}_d(1-\rho_t)} + \frac{\rho_v + \bar{Q}}{1-\rho_t} + \frac{1}{2(1-\rho_v)}(\bar{E}' - 1). \tag{B.12}$$

The values of $\bar{X}_d$ and $\bar{X_d^2}$ depend on the ARQ scheme and are as given in Equations (5.22) to (5.25).

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