PERFORMANCE ANALYSIS OF STAR ARCHITECTURE PACKET-SWITCHED VSAT NETWORKS USING RANDOM CODE DIVISION MULTIPLE ACCESS

by

MONCEF BADRI

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APPROVED:

TRI T. HA, CHAIRMAN	M.A.WORTMAN
J.C.MCKEEMAN	S.BINGULAC

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(ABSTRACT)

The intent of this research is to provide a performance analysis of multiple access protocols in packet-switched Very Small Aperture Terminal (VSAT) satellite communication networks. This research consisted of three major thrusts. First, we analyzed the average time delay of the broadcast Time Division Multiplexing (TDM) outbound channel (hub to VSAT). Second, a throughput performance analysis of an asynchronous Direct-Sequence Code Division Multiple Access (DS-CDMA) communication system is carried out for the inbound line (VSAT to hub). Each channel was characterized by its bit error probability, and transmits fixed-length packets generated according to a Poisson process in an unslotted environment. Third, we presented a delay analysis of the ALOHA DS-CDMA/TDM channel to determine the total service time of a packet originating from either the VSAT or the hub station. In addition to its multiple access capability, this thesis is concerned with the use of direct-sequence spread-spectrum signaling primarily because of its ability to combat interference. Emphasis is placed on average throughput performance, and on the average packet delay after solving for the steady state probability generating function of the station queue size. Then, a discussion of the effect of finite buffer size, and an analysis relating the probability of buffer overflow to packet statistics and buffer size is presented. Because of the bursty nature of a

traffic originating from the VSAT's, the Automatic Repeat Request (ARQ) technique used for error control is the Stop-and-Wait (SW) protocol. It is used as a retransmission strategy in both the Asynchronous Time Division Multiplexing (ASTDM) and the ALOHA DS-CDMA channels.

Acknowledgements

First and foremost I would like to express my appreciation to my advisors DRS. TRI T. HA and MARTIN A. WORTMAN for the many valuable suggestions and advice throughout the course of this study. I also wish to thank DRs.S. BINGULAC and J.C. MCKEEMAN for serving as members on my examining committee and for their helpful review of this thesis.

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Research Description

A Very Small Aperture Terminal (VSAT) network with star architecture consists of a hub station (master station) that communicates with many remote VSAT's (slave stations) via satellite channels. This station is connected to host computers via terrestrial lines. Each remote VSAT is also connected to user's terminal devices (PC's,Printers,...),and there are Q VSAT groups, each of which consists of N VSAT's. Each group transmits packets to the hub station via a distinct digital satellite channel with a known bit error rate P_b and a known bit rate R_b. This channel is a Direct-Sequence Code Division Multiple Access (DS-CDMA) channel. There are a total of Q CDMA channels which we refer to as the inbound lines and used to convey any communication between a particular VSAT in the network to the hub. The outbound line, hub to VSAT, is a broadcast Time Division Multiplexing channel (TDM) with a known bit error rate P'_b and a known bit rate R'_b. A network model is given in figure 1.

Both techniques of time division multiplexing will be explained, but only for the case of Asynchronous Time Division Multiplexing (ASTDM) is system performance analyzed in depth. The reasons for this become clear when we discuss the techniques. Due to the

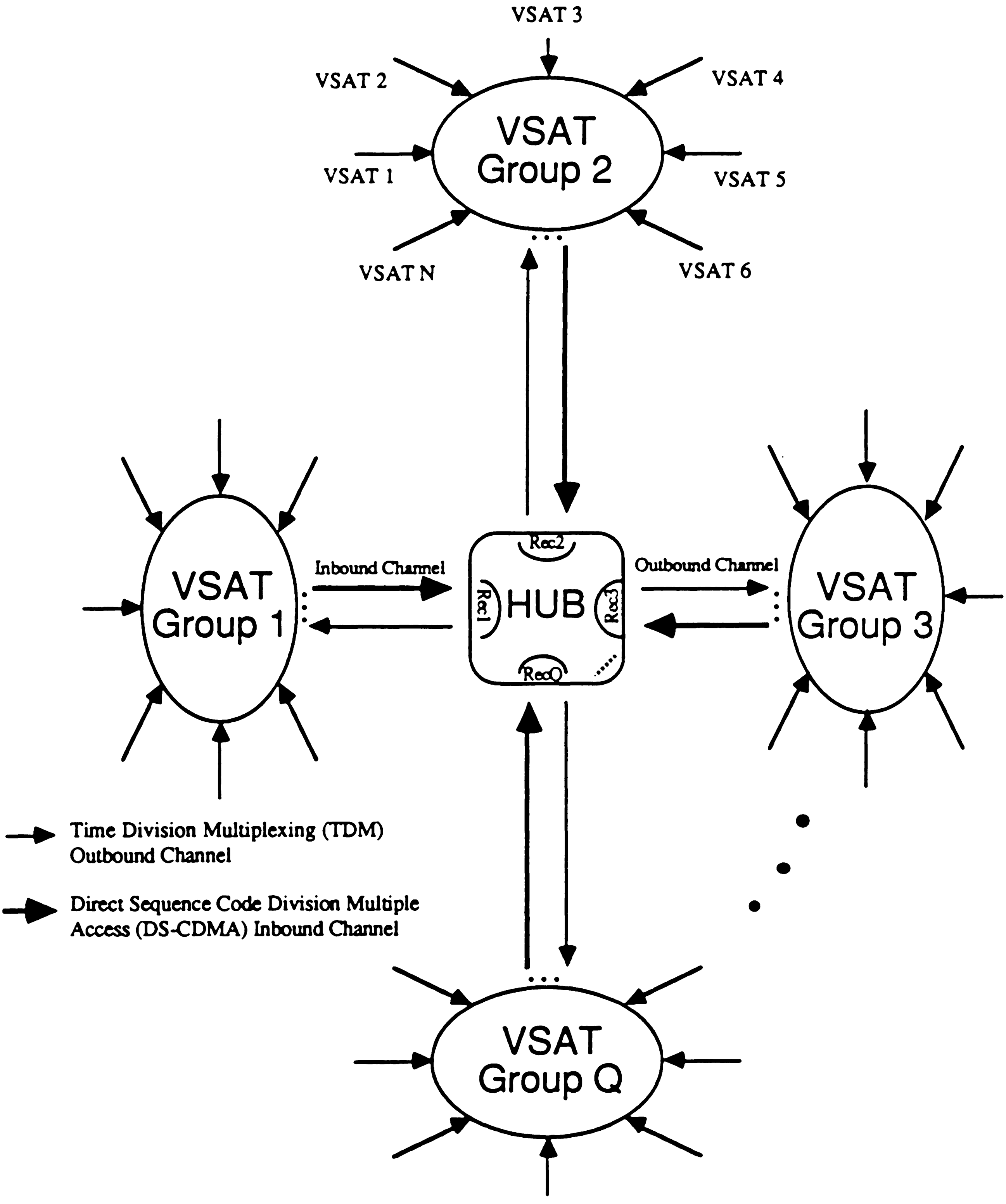

Figure 1. Network Model

bursty nature of traffic, the Stop-and-Wait ARQ is to be considered as the data link control protocol for the inbound line. Both the finite and infinite buffer case for the ASTDM are discussed and explicit formulae for the various queueing delays and buffer size will be derived. In this ASTDM channel, time is slotted so that the channel can carry either information or acknowledgment packets from the hub to any VSAT in the star network. To confine to our assumption of considering only fixed-length packets, it is assumed that dummy information is inserted just prior to the High Data Link Control (HDLC) frame's eight-bit flag indicating an end of a packet. The packet length, L, is assumed to be the optimal size for the stop-and-wait error detection and retransmission strategy that minimizes the time wasted in acknowledgements, retransmissions and frame overhead. It is also considered to include both information and overhead bits.

Each CDMA channel uses a distinct pseudonoise (PN) code selected from a set of quasi-orthogonal codes such as Gold codes. The hub station is equipped with Q receivers each of which is designated to a CDMA channel. The use of CDMA enables Q separate inbound lines to coexist in the same bandwidth at the same time with little interference. Each CDMA channel is shared by a population of VSAT's whose traffic is bursty. In this research, the random access to the CDMA channel considered is ALOHA [26].

For this limiting satellite communication system, in which all messages from ground stations must pass through a common satellite repeater, one way to achieve both multiple access and randomness and also make efficient use of the satellite repeater is to have all users simultaneously use the entire repeater bandwidth by means of spread spectrum. Each user, a VSAT group, is assigned a distinct pseudonoise (PN) carrier. Each active user then modulates his message onto his PN carrier and transmits it through the satellite repeater to the receiving terminals. Each receiving station employs a phase coherent correlator capable of locking onto any one of the transmitted signals while rejecting the others. Once the receiving station is locked onto one of the PN carriers, the message can be recovered by a correlation detector. In this research, it is assumed that the out-of-phase auto-correlations of the PN codes are so small so that the receiver will lock on to the first arrival and reject the others. Because the VSAT population is much larger than Q, the number of PN codes, some of the codes must be reused by the VSAT's. However, the probability that two or more packets using the same PN code will overlap completely and that they will be destroyed is ignored. In other words, no packet collision is detected at any time in this totally asynchronous network. As a result of this assumption, if more than Q packets arrive at the satellite transponder at the same time, those that use different PN codes will experience a degradation in the link error rate and may or may not be received (graceful degradation). The link error probability of CDMA is therefore given as in reference [6]. With this probability, we proceed to present an analytical technique for the throughput performance evaluation of an asynchronous random access packet switching network using code division as its multiple access technique. In the case of slotted ALOHA, the effect of having ACK traffic on the same channel will be discussed and the channel's time delay will be evaluated.

We shall first review the theory of some basic concepts in spread-spectrum communications. Having done just that, we hope the reader is provided with a more or less comprehensive view of spread-spectrum capabilities and is with enough background enabling him or her to relate the theory to the concept of using Code Division Multiple Access (CDMA) for the inbound line.

1.0 SPREAD-SPECTRUM MULTIPLE ACCESS COMMUNICATIONS

1.1 Introduction

Because of the continuing demand for more telecommunication capacity, there is a continuing need for more efficient ways of sharing the radio spectrum. The conventional method of using the spectrum is by frequency division. However, for many kinds of services this is inefficient [1] and it seems desirable to examine alternative procedures to assure that service meets demand. Spread spectrum techniques, which are based on principles antithetic to those currently used in spectrum allocation schemes, seem to offer benefits for spectrum sharing, for some applications, superior to those of frequency division. In fact, spread spectrum communications has over the years become an increasingly popular technique for use in many different systems [1],[31].

Spread-spectrum systems have developed since about the mid-1950's. The initial applications have been to military antijamming tactical communications, to guidance systems, and to other applications [1]. A definition of spread-spectrum that adequately reflects the characteristics of this technique is as follows [2].

> ***spread-specrum is a means of transmission in which the signal occupies a bandwidth in excess of the minimum necessary to send the information; the bandwidth spread is accomplished by means of a code which is independent of the data, and a synchronized reception with the code at the receiver is used for despreading and subsequent data recovery.***

There are many reasons for spreading the spectrum, and if done properly, a multiplicity of benefits can accrue simultaneously. Some of the advantages that the spread-spectrum techniques offer are [3]:

1. Multiple access capability.
2. Low power spectral density for low detectability.
3. Message privacy, a by-product of the pseudorandom coding.
4. Interference rejection.

1.2 Modulation Techniques

The means by which the spectrum is spread is crucial. Three general types of spread spectrum techniques are used, however only one will be discussed in detail in this thesis because of its desirable capabilities. These general modulation techniques are :

1. Direct-sequence modulation.
2. Frequency hopping.
3. Time hopping.

Different combinations of these three methods also exist, and are used when various combinations of advantages are necessary to combat certain undesirable conditions [32].

Pulsed FM or chirp implementations of spread spectrum systems also exist but not generally for information transferral.

Direct-sequence modulation implies data modulation by a pseudorandom code sequence where the term 'pseudorandom' specifically means random in appearance but is reproducible by deterministic means. This realization of Spread Spectrum Multiple Access (SSMA) systems will be further discussed in the main body of this chapter. In frequency hopping systems, the carrier is frequency shifted in discrete increments in a pattern determined by a digital code sequence while time hopping systems use pseudorandom code sequences to select an alloted time slot for data transmission. This time slot is a small fraction of the total available transmission time, and thus results in increased bandwidth.

1.3 Direct-Sequence Modulation

The process of spreading a signal's bandwidth and then collapsing it through coherent correlation with a stored reference signal in the receiver offers a unique combination of advantages such as : selective addressing, code division multiplexing, message privacy, interference rejection, and low density transmission signal.

1.3.1 Selective addressing

Assignment of particular code to a given receiver would allow it to be contacted only by a transmitter which is using the same code to modulate its signal. With different codes assigned to different receivers in a network, a transmitter can select any one receiver for communication by simply transmitting that receiver's code; then, only that receiver will receive the message.

1.3.2 Code Division Multiplexing

A number of transmitters and receivers can operate on the same frequency at the same time by employing different codes. For low duty cycle operation such as mobile systems, for a given band of frequencies the spread spectrum system may make more channels available to users than conventional allocation of a single channel to a single user [4].

1.3.3 Message Privacy

Message privacy is inherent in spread spectrum signals because of their coded transmission format. Of course, the degree of privacy, or security, is a function of the codes used.

1.3.4 Interference Rejection

The correlation and the spreading process at the receiver and transmitter, respectively, give very good rejection capability for interfering signals which can not be matched in any other system [32]. Both deliberate and unintentional interference are rejected by a spread-spectrum receiver, up to a maximum which is known as the 'jamming margin' for that receiver.

1.3.5 Low Density Transmission Signal

The power spectral density of the transmission signal is relatively low because of the bandwidth expansion. Spread spectrum systems can be used to prevent interference to other systems and to provide a low probability of interception.

1.4 Binary Direct-Sequence SSMA Communications

In a direct-sequence SSMA communication system, several asynchronous signals simultaneously occupy the same channel. Each signal employs a signature sequence which was selected because it had certain desirable correlation properties. For multiple access communications, the primary goal is to separate the spread spectrum signals at the receiver eventhough they occupy the same bandwidth at the same time. This problem is now considered for two different forms of direct-sequence spread-spectrum modulation; namely binary phase-shift keying (BPSK) and minimum-shift keying (MSK). First, the emphasis is mainly devoted to the analysis of system performance; selection of signature sequences will be discussed afterwards.

In the binary direct-sequence form of spread-spectrum modulation, a baseband signal X (t) can be expressed as

$$X(t) = \sum_{j=-\infty}^{\infty} X_j \psi(t - jT_c) \qquad [1.4.1]$$

where $\{X_j\}$ is a periodic sequence of elements of $\{+1,-1\}$ and ψ is a time limited signal. The most common choice for the signal ψ is

$$\psi(t) = p_{T_c}(t) \equiv \begin{cases} 1 & ,0 \le t < T_c \\ 0 & , \text{otherwise} \end{cases} \qquad [1.4.2]$$

This is the rectangular pulse of duration T_c which starts at $t = 0$, and T_c is the time of a PN binary symbol referred to as a 'chip'. ψ (t) satisfies the following condition :

$$\frac{1}{T_c}\int_0^{T_c}\psi^2(t)dt = 1 \qquad [1.4.3]$$

Another choice for ψ (t) is the sine pulse

$$\psi(t) = \sqrt{2}\,\sin(\frac{\pi t}{T_c})p_{T_c}(t) \qquad [1.4.4]$$

The rectangular pulse is employed in a phase-shifted key (PSK) system, and the sine pulse is the basic waveform for a minimum-shift-key (MSK) system. It is common to refer to ψ (t), $0 \le t < T_c$, as the chip waveform.

The binary data signal b(t) is given by

$$b(t) = \sum_{n=-\infty}^{\infty} b_n p_T(t - nT) \qquad [1.4.5]$$

where $p_T(t)$ is the rectangular pulse of duration T which starts at $t = 0$ and $b = \{b_n\}$ is the binary data sequence(i.e.,$b_n \in \{+1, -1\}$ for each n). The baseband spread-spectrum signal is then $V(t) = X(t)b(t)$. The $X = \{X_j\}$, which is called the signature sequence, is a periodic sequence which satisfies $X_j = X_{j+N}$ for each j, where N is an integer multiple of the signature sequence period. The data pulse duration T is given by $(T = NT_c)$; Therefore, the bandwidth of V(t) is on the order of N times the bandwidth of b(t).

The actual transmitted signal in a binary direct-sequence spread-spectrum communication system is

$$S(t) = AV(t)\cos(\omega_c t + \theta)$$
$$= AX(t)b(t)\cos(\omega_c t + \theta) \qquad [1.4.6]$$

where ω_c is the carrier frequency and θ is an arbitrary phase angle. In a spread-spectrum multiple-access (SSMA) communications system, there are K such signals which are simultaneously transmitted. K represents the total number of VSAT's originating from any of the Q groups that are simultaneously accessing the channel. For binary direct-sequence SSMA, the signals are given by

$$S_k(t) = Aa_k(t)b_k(t)\cos(\omega_c t + \theta_k) \qquad [1.4.7]$$

for $1 \le k \le K$. The signal $a_k(t)$ is of the form (1.4.1); that is

$$a_k(t) = \sum_{j=-\infty}^{\infty} a_j^{(k)}\psi(t - jT_c) \qquad [1.4.8]$$

where we have denoted the k^{th} signature sequence by $a^{(k)} = \{a_j^{(k)}\}$. The k^{th}data signal $b_k(t)$ is of the form (1.4.5) with the k^{th} data sequence denoted by $b^{(k)} = \{b_n^{(k)}\}$. In general, the phase angles θ_k, $1 \le k \le K$, are not the same because in practice the transmitters are not phase synchronous. Furthermore, the transmitters are not time synchronous, and the propagation delays for the various signals may differ. Thus, as illustrated by figure 2, the received signal is given by

$$r(t) = n(t) + \sum_{k=1}^{K} S_k(t - \tau_k) \qquad [1.4.9]$$

where n(t) is additive white Gaussian noise (AWGN) with two-sided spectral density $(N_0/2)$ Watts/Hz, and τ_k is the time delay associated with the k^{th} signal.

Combining equations (1.4.7) and (1.4.9) we have

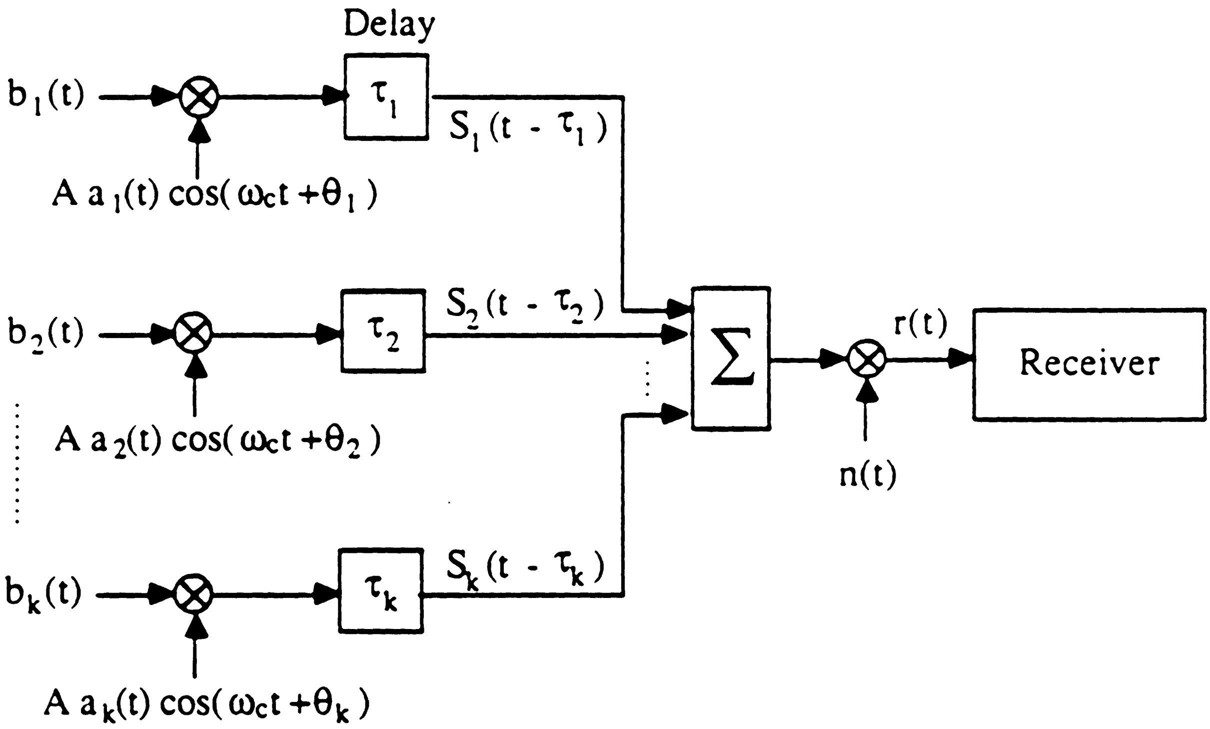

Figure 2. Binary Direct-Sequence SSMA Communications System Model

$$r(t) = n(t) + \sum_{k=1}^{K} A a_k(t - \tau_k) b_k(t - \tau_k) \cos(\omega_c t + \varphi_k) \qquad [1.4.10]$$

where $\varphi_k = \theta_k - \omega_c \tau_k$. Without loss of generality, we can assume $\varphi_i = 0$ and $\tau_i = 0$ in the analysis of the receiver which is matched to the i^{th} signal. In addition, there is no loss in generality in assuming that $0 \le \varphi_k < 2\pi$ and $0 \le \tau_k < T$, where $1 \le k \le K$, since we are only concerned with time delays modulo T and phase shifts modulo 2π.

The i^{th} receiver is assumed to be a correlation receiver (or matched filter) which is matched to the i^{th} signal. Thus, the output of the i^{th} receiver is

$$Z_i = \int_0^T r(t) a_i(t) \cos \omega_c t \; dt \qquad [1.4.11]$$

since $\varphi_i = \tau_i = 0$. From equations [1.4.9] and [1.4.11] it is seen that

$$Z_i = \eta_i + \sum_{k=1}^{K} \int_0^T S_k(t - \tau_k) a_i(t) \cos \omega_c t \; dt \qquad [1.4.12]$$

where η_i is the random variable

$$\eta_i = \int_0^T n(t) a_i(t) \cos \omega_c t \; dt \qquad [1.4.13]$$

If $\omega_c >> (1/T_c)$, then a practical spread-spectrum receiver is such that the double frequency components of the integrand of equation (1.4.11) can be ignored. In all that follows, it is assumed that such components are negligible. Under these assumptions, equations (1.4.12) and (1.4.7) imply

$$Z_i = \eta_i + \frac{1}{2} A \int_0^T b_i(t) dt + \sum_{k=1, k \ne i}^{K} \frac{1}{2} A [f_{k,i}(\tau_k) + \hat{f}_{k,i}(\tau_k)] \cos \varphi_k \qquad [1.4.14]$$

where the functions $f_{k,i}$ and $\hat{f}_{k,i}$ are defined by

$$f_{k,i}(\tau) = \int_0^{\tau} b_k(t-\tau)a_k(t-\tau)a_i(t)dt \qquad [1.4.15]$$

$$\hat{f}_{k,i}(\tau) = \int_{\tau}^{T} b_k(t-\tau)a_k(t-\tau)a_i(t)dt \qquad [1.4.16]$$

since $b_k(t)$ is a signal of the form given in equation (1.4.5), then $b_k(t-\tau) = b_{-1}^{(k)}$ for $0 \le t < \tau$. Thus equation (1.4.15) implies that for $0 \le \tau \le T$

$$f_{k,i}(\tau) = b_{-1}^{(k)}\int_0^{\tau} a_k(t-\tau)a_i(t)dt \qquad [1.4.17]$$

similarly, $b_k(t-\tau) = b_0^{(k)}$ for $\tau \le t \le T$, and therefore equation (1.4.16) implies

$$\hat{f}_{k,i}(\tau) = b_0^{(k)}\int_{\tau}^{T} a_k(t-\tau)a_i(t)dt \qquad [1.4.18]$$

In view of equations (1.4.14),(1.4.17) and (1.4.18) it is of interest to consider the functions

$$R_{k,i}(\tau) = \int_0^{\tau} a_k(t-\tau)a_i(t)dt \quad ,0 \le \tau \le T \qquad [1.4.19]$$

and

$$\hat{R}_{k,i}(\tau) = \int_{\tau}^{T} a_k(t-\tau)a_i(t)dt \quad ,0 \le \tau \le T \qquad [1.4.20]$$

which are known as the continuous-time partial cross-correlation functions. These were introduced in reference [5] and they have been employed in the analysis of binary direct-sequence SSMA communications in several subsequent papers including [6] and [7]. Their effect on the receiver output is via the quantity

$$I_{k,i}(\underline{b}_k, \tau, \varphi) = \frac{1}{T}\left[b_{-1}^{(k)}R_{k,i}(\tau) + b_0^{(k)}\hat{R}_{k,i}(\tau)\right]\cos\varphi \qquad [1.4.21]$$

which is the normalized multiple access interference at the output of the i^{th} receiver due to the k^{th} signal. In equation (1.4.21) the vector $\underline{b}_k$ is the vector $(b_{-1}^{(k)}, b_0^{(k)})$ of two consecutive data bits from the k^{th} data source.

The expression for Z_i given in equation (1.4.14) can be written as

$$Z_i = \eta_i + \frac{1}{2}AT\left\{ b_0^{(i)} + \sum_{k \neq i} I_{k,i}(\underline{b}_k, \tau_k, \varphi_k) \right\} \qquad [1.4.22]$$

where $\sum_{k \neq i}$ denotes the sum over all integers k such that $k \neq i$ and $1 \leq k \leq K$. The first term on the right hand side of equation (1.4.22) is due to the channel noise, the term $\frac{1}{2}ATb_0^{(i)}$ is due to the i^{th} signal, and the final term is the multiple-access interference due to the k-1 signals $\{S_k(t){:}1 \leq k \leq K,\ \ k \neq i\}$. If $k = 1$ the system is not a multiple-access system, the last term is not present, and the analysis of the system is straight forward. The difficulty arises in a direct-sequence SSMA is when multiple-access interference components affect the received signal. Thus, in the quantity

$$\mathrm{I}_i(\underline{b}, \underline{\tau}, \underline{\varphi}) = \sum_{k \neq i} I_{k,i}(\underline{b}_k, \tau_k, \varphi_k) \qquad [1.4.23]$$

where $\underline{b} = (b_{-1}^{(1)}, b_0^{(1)}, \ldots, b_{-1}^{(k)}, b_0^{(k)})$ is a vector of data symbols $(b_n^{(k)} \in \{+1, -1\})$, $\underline{\tau} = (\tau_1, \ldots, \tau_k)$ is the vector of time delays, and $\underline{\varphi} = (\varphi_1, \ldots, \varphi_k)$ is the vector of phase angles is of interest. In view of the definition (1.4.23), the receiver output can now be written as

$$Z_i = \eta_i + \frac{1}{2}AT\left\{ b_0^{(i)} + \mathrm{I}_i(\underline{b}, \underline{\tau}, \underline{\varphi}) \right\} \qquad [1.4.24]$$

.

The decision made by the receiver as to which pulse was transmitted (i.e.,$b_0^{(i)} = +1$ or -1) is based on the observation Z_i. If $Z_i \geq 0$ then the receiver determines a positive pulse was sent, otherwise the decision is that a negative pulse was sent. If the decision is to be based solely on the observation of Z_i, then the above decision rule is the best possible. Furthermore, Z_i is a sufficient statistic if $K = 1$. If $K > 1$ a more complex receiver may provide a statistic which permits a more nearly optimal decision. However, since correlation receivers or matched filters are relatively simple to implement, the vast majority of direct-sequence spread-spectrum systems employ correlation receivers even though they may be suboptimal in a multiple-access environment. This is due to the relatively small performance improvement that can be obtained with the relatively complex optimum receiver. Consequently, this presentation will be confined to performance analysis for direct-sequence SSMA communications systems with correlation receivers.

One of the performance measures that will be considered is the probability of error. From equation (1.4.24) and the fact that η_i is a zero-mean Gaussian random variable with variance $(N_0T/4)$, it follows that the conditional probability of error given that a positive pulse is transmitted (*i.e.*, given $b_0^{(i)} = +1$) is expressed in terms of the standard Gaussian distribution function Φ by [8] :

$$
\begin{aligned}
P_{e,i}(\underline{b}, \underline{\tau}, \underline{\varphi}) &= Pr\left[Z_i < 0 \,|\, b_0^{(i)} = +1\right] \\
&= Pr\left[(2\eta_i/AT) < -1 - \mathrm{I}_i(\underline{b}, \underline{\tau}, \underline{\varphi})\right] \\
&= Q\left(\sigma^{-1}[1 + \mathrm{I}_i(\underline{b}, \underline{\tau}, \underline{\varphi})]\right)
\end{aligned}
\qquad [1.4.25]
$$

for each $\underline{\tau}$ and $\underline{\varphi}$ and for each $\underline{b}$ such that $b_0^{(i)} = +1$. In equation (1.4.25) the function Q is defined by

$$
Q(x) = 1 - \Phi(x) = \int_x^{\infty} \frac{1}{\sqrt{2\pi}} \exp\left(\frac{-y^2}{2}\right) dy \qquad [1.4.26]
$$

and the quantity σ is given by

$$\sigma = 2/AT\{\mathrm{var}\eta_i\}^{1/2} = \left(\frac{2E_b}{N_0}\right)^{-1/2} \qquad [1.4.27]$$

where E_b is the energy per data bit ($E_b = A^2T/2$) .

1.4.1 Average Error Probability

Because of the difficulties encountered in the evaluation of the average probability of error, $P_{e,i}$, from equation (1.4.25), we have used approximations and bounds for $P_{e,i}$. The approximation used in this research was suggested by [6] as

$$P_{e,i} \cong Q(\mathrm{SNR}_i) \qquad [1.4.28]$$

where SNR_i is the signal-to-noise ratio at the output of the i^{th} correlation receiver. In [6] Pursley shows that for an asynchronous system, even ignoring the near-far problem effects, the number of users the system can accommodate is markedly less than G_p, where G_p is the processing gain defined as the number of chips per bit in the system. Indeed, Pursley proved that the peak signal to rms noise voltage ratio, averaged over all phase shifts, time delays, and data symbols of the multiple users, is approximately given by

$$SNR \cong \left[\frac{K-1}{3G_p} + \frac{N_0}{2E_b}\right]^{-1/2} \qquad [1.4.29]$$

where K is the number of interferers.

As mentioned earlier in the chapter, in spread-spectrum systems each user is assigned a particular code or PN sequence. In general, these codes can be classified into four categories: maximal linear or M-sequences, non-maximal linear, maximal nonlinear, and non-maximal nonlinear. M-sequences are explained in section 1.5 of this thesis. They have the following properties [9] :

1.5 Properties of M-Sequences

1.5.1 Correlation Property

In statistical terms, correlation is a measure of the linear relationship between two variables. A basic definition for the correlation between two variables X and Y could be the sum of their cross products as ΣXY. For two valued (+1,-1) variables such as PN codes, the cross-correlation of two different codes can be defined as the sum of agreement minus disagreement between two different codes. Similarly, the auto-correlation relation is the measure of a code and the phase-shifted code replica. For example, if the code sequence X is : <1,-1,-1,1,1,-1> and Y is : <1,1,1,-1,-1,-1>; therefore, the cross-correlation R_{xy} is given by $R_{xy} = \Sigma XY = 1 - 1 - 1 - 1 - 1 + 1 = -2$ or $R_{xy} = \Sigma$ agreements $- \Sigma$ disagreements $= 2 - 4 = -2$. The auto-correlation function of maximal linear codes is a two valued function with maximum value equal to the code length which occurs at zero phase-shift, and negative one elsewhere. The maximal linear sequences were shown to exhibit the best auto-correlation [9], in mathematical terms it is as follows :

$$\rho(t) = \sum_{n=1}^{k} S(t)S(t+\tau) = \begin{cases} 1 & \text{, if } \tau = 0 \\ -1/p & \text{, if } 1 < \tau < p \end{cases} \qquad [1.5.1]$$

where p is the length of the sequences S(t). The superiority of this maximal linear sequence normalized auto-correlation is due to two characteristics of this sequence type, namely, the randomness and the add-and-shift properties. As shown in figure 3, the auto-correlation function repeats regularly with the same periodicity as the pseudorandom code. The power spectrum of a pseudorandom binary waveform with auto-correlation function as just described is

$$S(\omega) = \left(\frac{p+1}{p}\right)^2 \left[\frac{\sin \omega t_0/2}{\omega t_0/2}\right]^2 \sum_{n=-\infty}^{\infty} \delta(\omega - 2\pi n/pt_0) + 1/p^2 \delta(\omega) \qquad [1.5.2]$$

where t_0 is the period of one digit of the binary waveform, and $\delta\ (\ )$ is the impulse function. The power spectrum for the binary waveforms is illustrated in figure 3.

1.5.2 Balance Property

In each period of an M-sequence $\{b_n\}$, the sum of the number of ones N_b for a non-zero symbol b differs from the number of zeros N_0 by at most one. That is $N_b = N_0 + 1$, for all $b \neq 0$. This property, which states that symbols occur as equally often as possible within one period, is referred to as the balance property [32].

1.5.3 Run Property

Among the runs (a run is defined as a consecutive series of zeros and ones) of ones and zeros in each period, one-half of the runs of each kind are of length one, one-fourth of each kind are of length two, one-eighth of length three, and so on. Thus, the statistical distributions are well defined and constant, however the relative position of the runs vary from code to code.

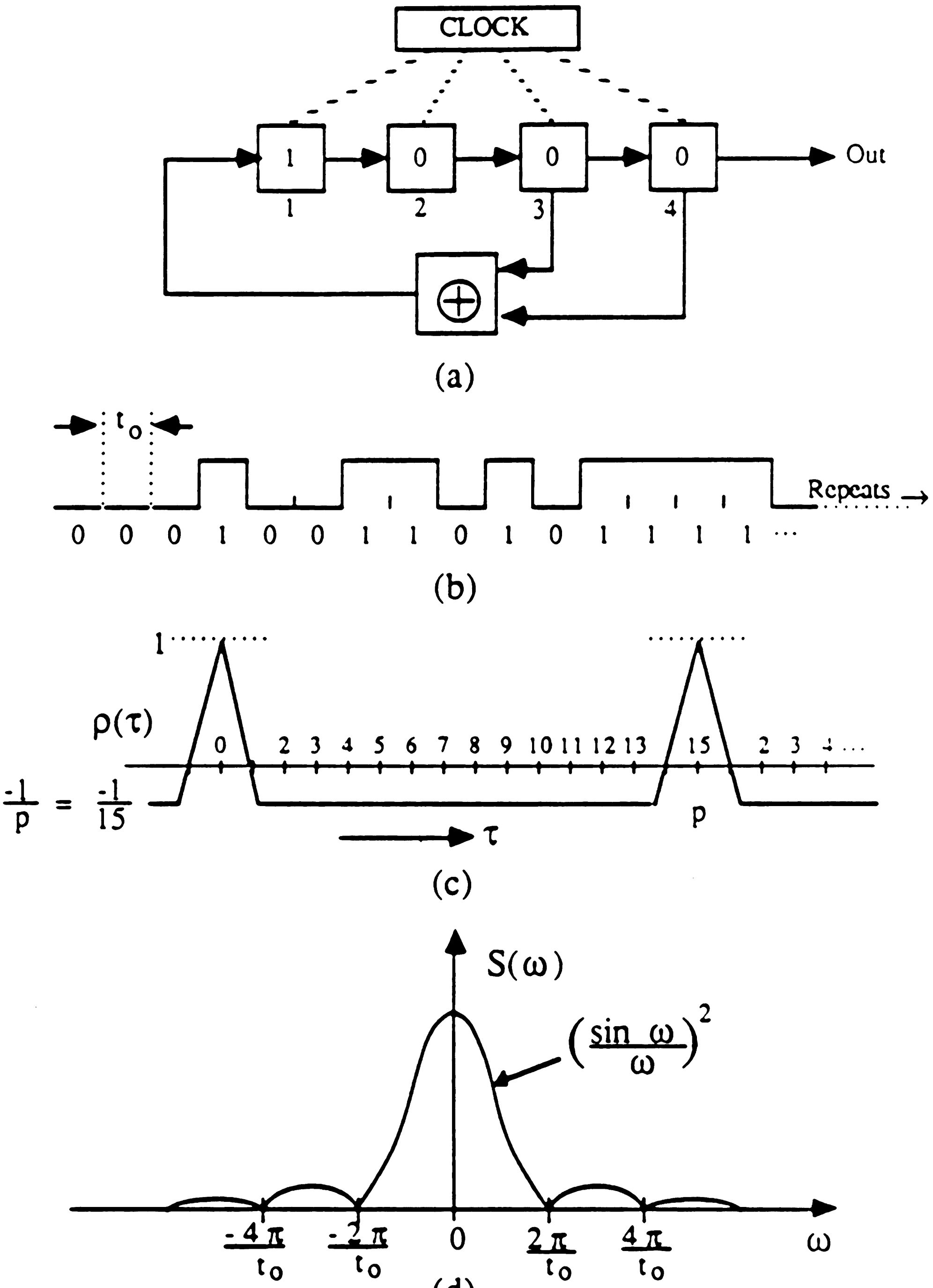

Figure 3. Code Sequence Generator and Spectrum:

(a) Four-Stage Shift Register with Feedback from Stages 3 & 4 (Modulo-2 added)

(b) Pseudorandom Code Sequence Generated which repeats every 15 Code elements

(c) Auto-Correlation Function for 15-element Code

(d) Spectrum of Pseudorandom Code

1.5.4 Add-and-Shift Property

Modulo-two addition of a maximum-length linear code with a shifted replica of itself results in another replica with a phase shift different from either of the original replica.

Linear maximal sequences can be generated by relatively simple electronic implementation of a feedback shift register (see figure 3) known as maximal linear shift register sequence. The application of non-maximal linear codes in spread-spectrum multiplexing was developed by Gold [10] and is known as the Gold code, which is generated by modulo-two addition of a pair of maximal linear codes. Because of the shift-and-add property of maximal linear sequence, the new sequence has the same bit length as those being added, but it is not maximal. Furthermore, every change in phase position between the two sequences causes a new non-maximal sequence; so a large family ($2^n - 1$, where n is the number of stages of a shift register) of non-maximal sequences can be generated and whose cross-correlation Θ satisfies the inequality [10]:

$$|\Theta| = \begin{cases} 2^{(n+1)/2} + 1 & \text{for n odd} \\ 2^{(n+2)/2} + 1 & \text{for n even , } n \neq \text{mod } 4 \end{cases} \qquad [1.5.3]$$

Comparing these two code systems, it can be found that the hardware realization of a maximal code is much easier than a non-maximal code. In considering the maximal linear shift register code, for a fixed stage sequence generator, there are $\Phi(2^N - 1)$ (N is the number of stages and $[\Phi(2^N - 1)]$ is an Euler number which is defined as the number of positive integers that are relatively prime and less than $(2^N - 1)$) possible linear combinations of feedback taps, each feedback tap gives a unique sequence code. Therefore, a large family of codes can be generated through the sequence generator by properly selecting the feedback taps.

In designing the sequence generator,the number of stages, bit rate,and the feedback configurations are the first parameters to be determined, as these parameters are closely related to the multiplexing capability as well as the overall system performance.

The sequence period, L, can be determined by the number of stages of the shift register N, where $L = (2^N - 1)$. The auto-correlation function is determined by the sequence period, and the cross-correlation between the codes with the same period could be roughly predicted by the period. If the length of the code has less prime factors, it will perform lower cross-correlation between each code most of the time [11]. On the other hand, higher cross-correlation could be expected. The existing cross-correlation has two main effects on the spread-spectrum system. First, the unspread signal may cause coherent interference to the wanted signal and hence degraded performance. Second, in case the local receiver is attempting to synchronize with the incoming signal, false correlation may occur delaying or even preventing acquisition of the wanted signal.

The bit rate determines the bandwidth of the spread signal. The ratio of the spread bandwidth to the information bandwidth is known as the processing gain G_p. Usually a system can be evaluated by investigating its process gain.

For the N-stage sequence generator, there are $[\Phi(2^N - 1)]/N$ different sequences that can be generated, each of these has its own feedback configuration. Using a computer simulation program, a subset of codes with low cross-correlation could be chosen.

1.6 Derivation of Channel Throughput

1.6.1 Advantages of CDMA over Conventional Multiple Access Methods

Satellite repeaters provide communication links to users separated by large distances, or in inaccessible locations on sea, land, or in space. Any earth station can listen to any signal including its own and it can detect errors using error correcting codes. But the number of channels in the satellite transponder is limited by the power and bandwidth of the repeater. These channels might be frequency division multiple access (FDMA), time division multiple access (TDMA), or code division multiple access (CDMA). CDMA are those signals to be used to divide the available channel capacity. Unlike the case of TDMA or FDMA where the carriers are separated by time and frequency, in CDMA all active VSAT's in the network transmit in the same allocated satellite transponder bandwidth at the same time. Carrier separation is achieved at the earth terminals by correlation of the received signal with the corresponding addressed PN code. CDMA signals are of the same form as spread-spectrum signals discussed earlier, but designed to have low time cross-correlation and require the use of specific chip sequences to be assigned to the VSAT groups. To better distinguish CDMA technique from SSMA, for SSMA we assume pseudorandom sequences are well modelled as i.i.d sequences and different codes result in independent pseudorandom sequences. We define CDMA signals ,again, as those with low time cross-correlation where the signals are, however, not statistically independent. Generally, CDMA signals with sequences of long periods behave like SSMA signals [12].

The problem of allowing multiple users to simultaneously access a channel without causing an undue amount of degradation in the performance of any individual user is a classical problem in communications. The two most common techniques, TDMA and

FDMA have attempted to solve the problem by separating the signals in time and frequency. However, each of these techniques has certain drawbacks associated with it. For example, in FDMA there is the problem of intermodulation between signals, in TDMA this problem does not exist but users' synchronization becomes of paramount importance in system design. For these reasons [1] that Code Division Multiple Access (CDMA) has become a competetive multiple access scheme in certain situations. In recent years, there has been increased interest in this class of multiple access techniques. CDMA techniques have been considered for a variety of satellite systems including the NASA tracking and data relay system [13], systems to provide communication to aircraft and other mobile users [14], and military satellite communication systems.

The most common form of CDMA is spread-spectrum multiple access in which each user is assigned a particular code sequence which is modulated on the carrier along with the digital data. The SSMA techniques, as explained previously, are characterized by the use of a high-rate code (i.e.,many code symbols per data symbol) which has the effect of spreading the bandwidth of the data signal. The most common form of SSMA is phase-coded SSMA (also known as direct-sequence spread spectrum). As far as this research is concerned, only the direct-sequence form of SSMA that is considered. In this form, the carrier is phase modulated by the digital data sequence and the code sequence. In contrast to the two conventional methods of multiple access, CDMA does not have any sharply defined system capacity. As the number of users increases, the signal-to-interference ratio becomes smaller and there is a gradual degradation in performance until the SNR falls below threshold. Thus, the system can tolerate significant amounts of overhead if the users are willing to tolerate poorer performance. In CDMA, there is an additional privacy feature that is not available in other multiple access techniques. An important consideration in CDMA is the number of users that can be accommodated simultaneously. As it was previously determined, the output signal-to-noise ratio of any receiver is repeated here and given by

$$\mathrm{SNR}_0 = \left[\frac{K-1}{3N} + \frac{N_0}{2E_b}\right]^{-1/2} \qquad [1.6.1]$$

where K is the total number of packets arrived at the satellite transponder at the same time, and N is the PN code length. As mentioned earlier , satellite, with its multiple access capability, can be an attractive means of information distribution in future data networks. Because of bandwidth limitation, however, the available bandwidth of the satellite should be utilized efficiently. The motive behind this research is, then, to adapt the spread-spectrum to random access techniques such as ALOHA or SLOTTED ALOHA.

1.6.2 Throughput of The UA/CDMA Inbound Line

In this research, it is assumed that the low traffic user's information bits are coded with spread-spectrum sequences so that the narrow information band is spread over the entire already allocated satellite bandwidth. The coded information packets then access the satellite on a purely random basis. We call this combination of using CDMA systems in a random access mode of operation, spread-spectrum random multiple-access systems (SS-RMA). Positively speaking, SS-RMA is CDMA in a sense that the mode of access is code division. Also, it is ALOHA type in a sense that there are more codes; therefore, more possible users than the channel access limit. That is, the low duty cycle users have to share the limited access channel on a contention basis. With appropriate choice of the coding technique, there is the potential for improving on the capacity of uncoded ALOHA [15],[16].

Our main interest lies in considering an environment in which there are a relatively large number of users (VSAT's) each of which transmits infrequently. In such an asynchronous environment, the number of transmitters which interfere with the transmission of an arbitrary packet is a stochastic process. Thus, the number of interferers to which a given packet is subjected, and consequently the signal-to-noise ratio, varies throughout the packet's transmission period. We assume that users generate fixed-length packets according to an independent poisson process with parameter λ T where λ is the average packet arrival rate and T is the duration of the packet transmission. All users are assumed to begin transmission as soon as their messages are generated . This is referred to as 'Unslotted Asynchronous CDMA (UA/CDMA).'[17]

Suppose that some user has a packet, call it the 'tagged packet', is ready to transmit at some time, say t_0. The transmission will end at time $t = t_0 + T$. Now, suppose that during the transmission of this tagged packet, a total of j interferers will interfere with its reception. k packets may be assumed to have had their transmission already in progress at time t_0 and an additional n packets will enter the system during the period ($t_0, t_0 + T$) such that ($n + k = j$). This is illustrated in figure 4. Clearly, since packet arrivals are Poisson

diributed, the distribution of the arrival times of the early k interferers and the late n interferers are independent and uniformly distributed over the periods ($t_0 - T, t_0$) and ($t_0, t_0 + T$) respectively [18].

In computing the throughput (β) for the UA/CDMA system, (we refer to this system as the inbound line of the network), we shall define β to be equal to the average number of successful transmissions that occur per packet transmission time. Let S denote the event that the tagged packet is received successfully where success is defined according to some set of criteria. Then,

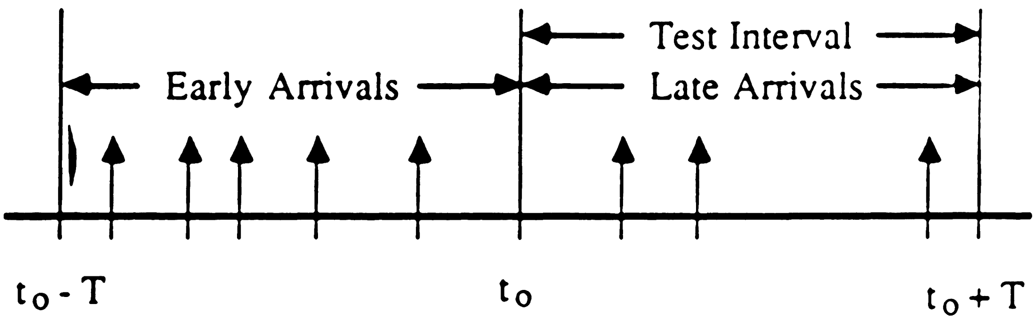

Figure 4. Times at which VSAT's are Ready to Transmit Packets

$$\beta = \lambda T Pr\{S\} = G Pr\{S\} \qquad [1.6.2]$$

where G is the offered traffic, or the average number of attempted packet transmissions per packet transmission time T. For our purposes, we limit ourselves to the special case where the probability of success is taken to be the probability that a particular bit of the packet is received without error, and that this condition depends only on the number of interferers to which it is subjected and not upon whether any other bit in the packet is in error. Then the probability that a bit is received in error is approximated accurately by:

$$P_b = \frac{1}{(1+j)} \sum_{l=0}^{J} Pr\{\text{error } / I_l \text{ interferers}\} \qquad [1.6.3]$$

where $(\frac{1}{1+j})$ is the probability of one interferer present at any time among all the (1+j) possible users from 0 to j. As pointed out at the beginning of this chapter, Pursley provides a simple formula to approximate the probability that a bit is in error given the number of interferers it is subjected to. Specifically, this bit error probability is given by

$$P_{b,l} = 1 - \Phi(\mathrm{SNR}_l) \qquad [1.6.4]$$

where Φ is the standard (i.e.,zero mean, unit variance) Gaussian cummulative distribution function. Substituting back yields

$$P_b = \frac{1}{(1+j)} \sum_{l=0}^{J} P_{b,l} \qquad [1.6.5]$$

then, the probability of a bit error in a packet is equal to

$$P_p = \left\{ 1 - \left(1 - \frac{1}{(1+j)} \sum_{l=0}^{J} (1 - \Phi(\mathrm{SNR}_l)) \right)^L \right\} \qquad [1.6.6]$$

where L is equal to the total number of bits in a packet, and packet success probability is simply $(1 - P_p)$.

It was shown in [16] and [17] that there are exactly 2^j possible orderings of the j independent arrival and departure events, and since the order of their occurrence is arbitrary, it was concluded that each of the 2^j possible realizations is equally likely to occur. Therefore, now we have all we need to state the probability of success as follows

$$Pr\{S\} = \sum_{J=0}^{\infty} P\{S/J = j\} P\{J = j\} \qquad [1.6.7]$$

but

$$Pr\{S/J = j\} = \frac{1}{2^j} \sum_{m=0}^{(2^j - 1)} \left[\frac{1}{(1 + j)} \sum_{J\,fold\,conv} \prod_{l=0}^{j} (1 - P_b)^L \right] \qquad [1.6.8]$$

and we note that $P\{J = j\}$ is simply the probability that there will be a total of j arrival and departure events during the transmission of the tagged packet, and that this quantity is just the probability that there will be j arrivals from the poisson process in a period of length 2T; that is

$$Pr\{J = j\} = \frac{(2G)^j}{j!} \exp(-2G) \qquad [1.6.9]$$

In the above formula, the averaging process was done over the entire period of packet transmission and we assumed that a transmission is unsuccessful if at least one bit in a packet is transmitted in error. Now, combining the two formulas we get the probability of success as

$$Pr\{S\} = \sum_{J=0}^{\infty} \left\{ \frac{1}{2^j} \sum_{m=0}^{(2^j - 1)} \left(\frac{1}{(1 + j)} \sum_{J\,fold\,conv} \prod_{l=0}^{J} (1 - p_b)^L \right) Pr[J = j] \right\} \qquad [1.6.10]$$

Finally, using equation (1.6.2), we get the throughput of the UA/CDMA inbound channel that was dedicated to convey information from any VSAT in the network to the hub. This channel has a known bit error probability P_b specified as given in equation (1.6.5).

As far as the computation for $Pr\{S\}$, in order to evaluate $Pr\{S/J = j\}$ for large j, the execution requires a substantial amount of 'CPU' time, and thus to reduce this computational time to acceptable levels, John N. Daigle in his reference [17] (appendix B) has suggested three possible ways to get around this problem.

2.0 TIME DIVISION MULTIPLEXING OUTBOUND CHANNEL

2.1 Introduction

In order to reduce the communications costs in time sharing systems and multi-computer communication systems, multiplexing techniques have been introduced to increase channel utilization. A commonly used technique is the Synchronous Time Division Multiplexing (STDM). Synchronous time division multiplex [19] carries out the sharing of the transmission medium by making a deterministic, sequential allocation of time intervals, or slots, to each user. The time slots, one for each user plus overhead, are typically organized into frames so that each active user is allocated a time slot in each frame. After one user's time duration has elapsed, the channel is switched to another user. With synchronous operation, buffering is limited to one packet per user line, and addressing is usually not required.

However, as a result of the fixed allocations, channel capacity is often wasted if user demands fluctuate in a random manner. As shown in figure 5, it is inefficient in capacity and cost to permanently assign a segment of bandwidth that is utilized only for a portion of the time. A more flexible system that efficiently uses the transmission facility on an 'instantaneous time-shared' basis could be used instead. The objective would be to switch from one user to another whenever the one user is idle, and to synchronously time multiplex the data. With such an arrangement, each user could be granted access to the channel only when he has a packet to transmit. This is called Asynchronous Time Division Multiplexing (ASTDM) which is tailored to accommodate input that fluctuates randomly. For such input, utilization of the shared channel can be significantly increased. The crutial attributes of such a multiplexing technique are:

1. An address is required for each transmitted packet, and
2. Buffering is required to handle the random packet arrivals.

If the buffer is empty during a transmission interval, the channel will be idle for that interval.

2.2 *Derivation of the Queueing Delay*

In this section, the delay performance of an asynchronous time division multiplexing outbound channel for transmitting data packets is considered. Consider an ASTDM outbound line in which time is slotted and time slots are organized into frames of, say, N slots indexed from 1 to N as shown in figure 6. For our network, we are assuming that the ASTDM 'superframe' is divided into a total of Q frames each of which is reserved to one group of VSAT's in the network, and each of these frames is slotted into N sub-slots for each VSAT within that group.

2.2.1 Infinite Buffer Case

To begin the analysis of the single input model of figure 7, one should resort to the stratagem used in any queueing theory book [20] in discussing M/G/1 queues. We focus on significant time intervals encountered in the queueing process. In the case of the M/G/1 queue, these times were the message completion times and were, of necessity, themselves random variables. In our case we assume a synchronous output line switching one packet every T sec. The time interval is always T sec long. The statistics of operation of the queueing model of figure 7 will be written in terms of the states of the buffer at the end of each time interval T sec long. Except for the emphasis here on time intervals required to transmit one packet, the approach used is essentially the same as that of the M/D/1 queue and the equations developed will be identical: The ones here focusing on the number of packets residing in the buffer, the ones for the M/G/1 queue on messages.

Consider the timing diagram shown in figure 8. In any one time interval T sec long, one packet of data is removed from the buffer if the buffer had at least one packet stored at the beginning of that interval. Any packet entering the buffer during that time interval must wait to at least the next time interval to be served. Let n_i be the number of packets residing in the buffer at the end of the i^{th} time interval just prior to the $(i+1)^{th}$ interval as shown in figure 8.

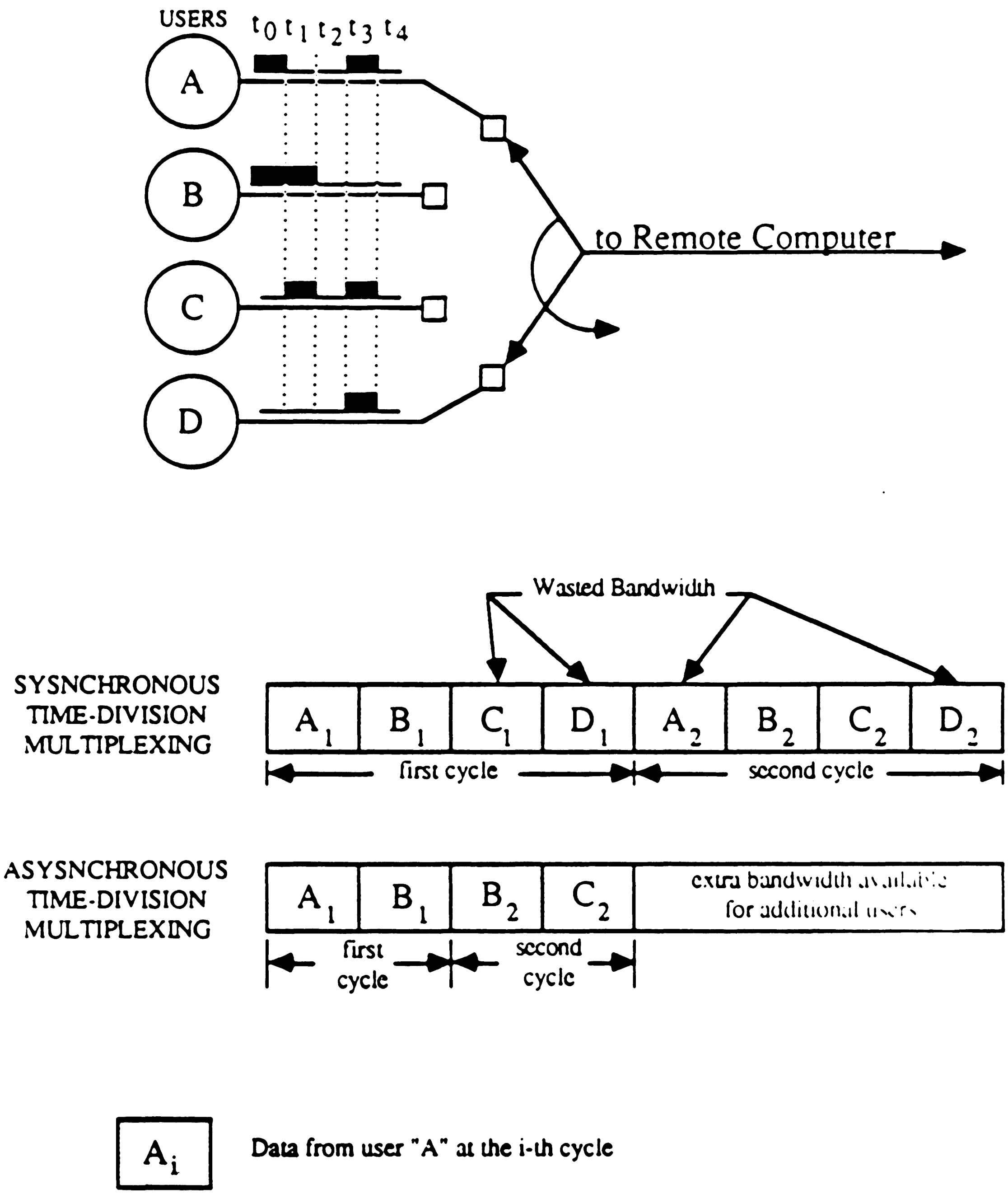

Figure 5. Time Division Multiplexing (STDM and ASTDM)

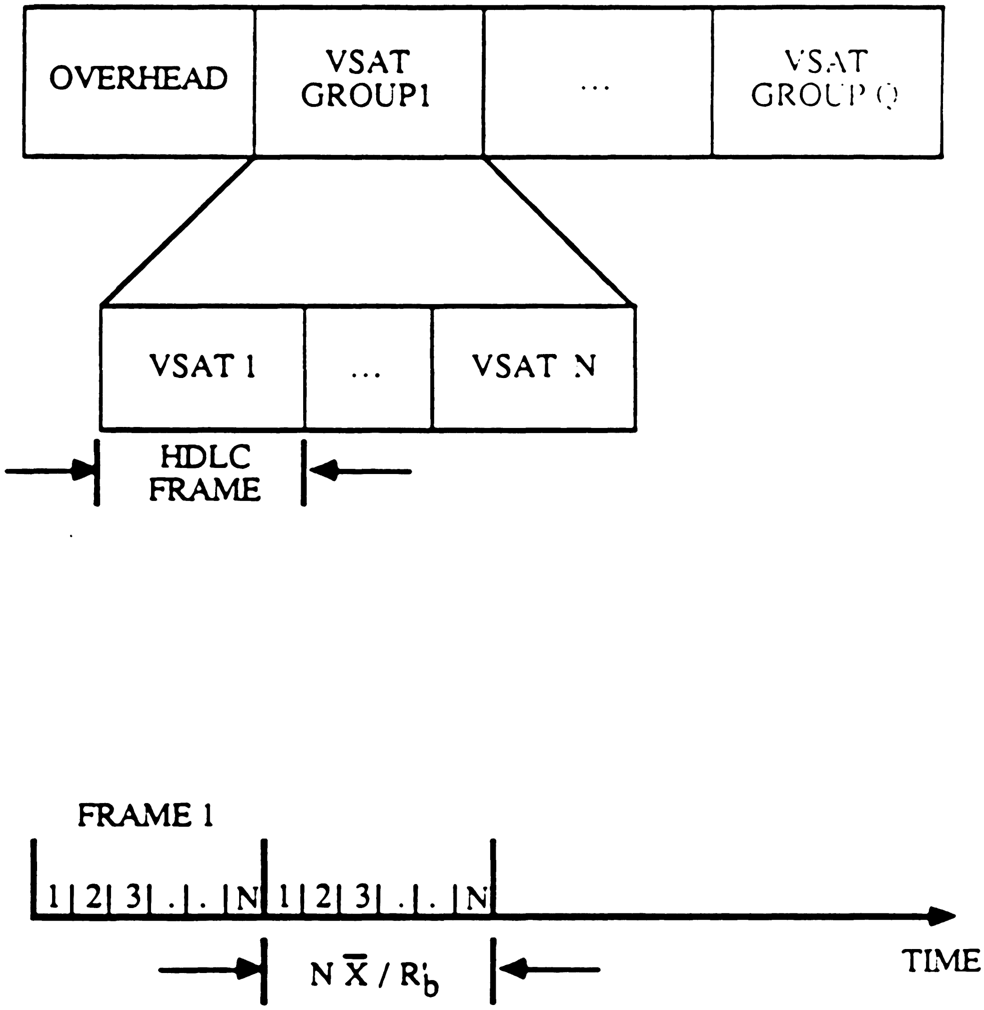

Figure 6. Allocation of ASTDM Channel Time

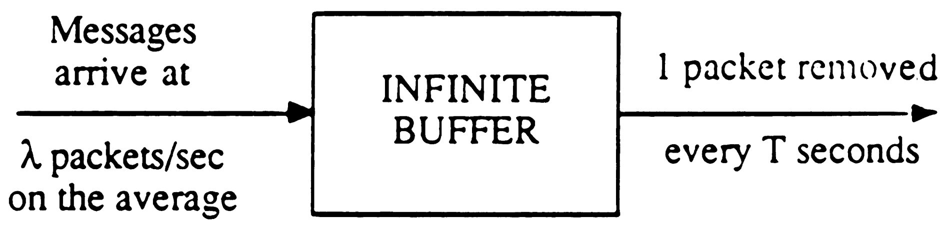

Figure 7. A Queueing Model For Analysis (single input)

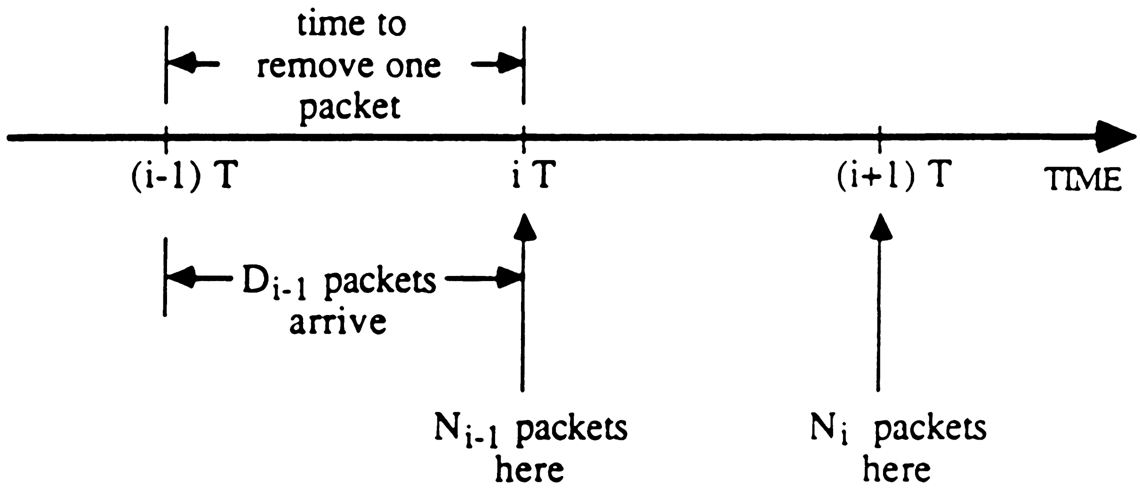

Figure 8. Timing Diagram

Let D_i be the number of data units (packets) arriving on the i^{th} interval. It is apparent that the average of D_i must be less than one to prevent the buffer from building up indefinitly. Then, the relation between the number of units buffered at the end of the i^{th} time interval and those buffered at the $(i-1)^{th}$ must be given by

$$n_i = \begin{cases} n_{i-1} - 1 + D_i & \text{, if } n_{i-1} \geq 1 \\ D_i & \text{, if } n_{i-1} = 0 \end{cases} \qquad [2.2.1]$$

Using this simple relation, one can obtain the essential statistics of operation of the queueing model of figure 7. As noted above, this equation is exactly of the form that shows the variation of message queue length with time in the M/G/1 queue. Equation (2.2.1) can be written in various equivalent forms from which equilibrium statistics can be found. Here, we should particularly focus on another form of (2.2.1) from which average buffer lengths and average time delay are more readily found. Specifically, a little thought will indicate that equation (2.2.1) may be written in the following equivalent form, with U(x) the unit step function :

$$n_i = n_{i-1} - U(n_{i-1}) + D_i \qquad [2.2.2]$$

here

$$U(x) = \begin{cases} 1 & \text{for } x > 0 \\ 0 & \text{for } x \leq 0 \end{cases}$$

If the system has been under operation for some time, statistically stationary conditions prevail. The average buffer occupancy must thus be independent of time and $E[n_i] = E[n_{i-1}] = E[n]$, with E[...] the symbol commonly used to represent statistical average. From equation (2.2.2),then, $E[U(n_{i-1})] = E[D_i] = E[D]$. Assuming messages are of unit length, of T sec in transmission time, and poisson distributed with rate $\lambda' = \lambda/N$ packets/sec and λ is the aggregate packet arrival rate, then

$$
\begin{aligned}
E[D] &= \sum_{k=0}^{\infty} k Pr[D = k] \\
&= \int_0^{\infty} \exp(-\lambda t) \sum_{k=0}^{\infty} k \frac{(\lambda t)^k}{k!} \, dW_s(t) \\
&= \int_0^{\infty} \exp(-\lambda t)\lambda t \sum_{k=0}^{\infty} \frac{(\lambda t)^k}{k!} \, dW_s(t) \\
&= \lambda \int_0^{\infty} t \, dW_s(t) \\
&= \lambda \, E[T_s] \\
&= \lambda T \\
&= \rho
\end{aligned}
\qquad [2.2.3]
$$

where $E[T_s] = 1/\mu = T$

In order to find E[n], the average number of packets in the buffer as a function of ρ, the system load or traffic intensity, we should first determine the steady state probability generating function of the number of packets in the buffer P(Z) :

$$P(Z) = E[Z^n] = \sum_{k=0}^{\infty} Z^k Pr[n = k] \qquad [2.2.4]$$

at steady state

$$P_{l,l\to\infty}(Z) = P_{l-1,l\to\infty}(Z) = P(Z) \qquad [2.2.5]$$

$$P(Z) = E[Z^{(n_{l-1} - U(u_{l-1}) + D_l)}] = E[Z^{(n_{l-1} - U(n_{l-1}))}]E[Z^{D_l}] \qquad [2.2.6]$$

Let $E[Z^{D_l}] = D(Z)$ be the probability generating function for the number of packets in a message arriving in a slot interval [T sec].

Let us define the generating function of the number of data units (packets) in a message as M(Z). In our case, it was assumed that the arrival process is simple Poisson so that each message contains a single data unit, that is $M(Z) = Z$ so

$$D(Z) = E[Z^D] = \sum_{k=0}^{\infty} Pr[D = k] \qquad [2.2.7]$$

D is not a Poisson random variable, but for a Poisson arrivals

$$Pr[D = k \mid T_s = T] = \exp(-\lambda T)\frac{(\lambda T)^k}{k!} \qquad [2.2.8]$$

We, then, calculate the probability generating function of the number of arrivals by first conditioning on the number of arrived packets, then averaging over the number of arrivals.

$$\begin{aligned} D(Z,T) &= E[Z^D] \\ &= E_k\left\{E\ [z^D /\ \text{k packets}]\right\} \\ &= E_k[M^j(Z)] \\ &= \sum_{k=0}^{\infty}\frac{(\lambda T)^k}{k!}\exp(-\lambda T)M^j(Z) \\ &= \exp(-\lambda T\ \ (1 - M(Z))) \\ &= \exp(-\lambda T\ \ (1 - Z)) \end{aligned} \qquad [2.2.9]$$

Recall that the generating function of a sum of independent random variables is the product of the generating functions.

$$\begin{aligned} E[Z^{(n_{i-1} - U(n_{i-1}))}] &= \sum_{k=0}^{\infty} Z^{k-U(k)}Pr[n = k] \\ &= P_0 + \sum_{k=1}^{\infty} Z^{k-1}Pr[n = k] \\ &= P_0 + 1/Z\left\{\sum_{k=0}^{\infty} Pr[n = k] - P_0\right\} \\ &= P_0 + 1/Z\,[P_n(Z) - P_0] \end{aligned} \qquad [2.2.10]$$

so at steady state

$$P(Z) = \left\{P_0 + 1/Z\ [P(Z) - P_0]\right\} D(Z) \qquad [2.2.11]$$

or since $\rho = Pr$[more than zero packets in the buffer] $= (1 - P_0)$

$$P(Z)[D(Z) - Z] = (1 - \rho)(1 - Z)D(Z) \qquad [2.2.12]$$

and

$$P(Z) = \frac{(1 - \rho)(1 - Z)D(Z)}{D(Z) - Z} \qquad [2.2.13]$$

As in the case of M/G/1 queue, in order for a steady state solution to exist we must have $\rho = \lambda T\ \overline{m} = \lambda T < 1$ where $\overline{m}$ is the average number of packets in a message ($\overline{m} = 1$). The average number of packets in the buffer, E[n] is $E[n] = P'(Z)|_{Z=1}$ well

$P(Z)[D(Z) - Z] = (1 - \rho)(1 - Z)D(Z)$

and the first derivative yields

$P'(Z)[D(Z) - Z] + P(Z)[D'(Z) - 1] = (1 - \rho)(-1)D(Z) + (1 - \rho)(1 - Z)D'(Z)$

and the second derivative gives

$P''(Z)[D(Z) - Z] + 2P'(Z)[D'(Z) - 1] + P(Z)D''(Z) = 2(1 - \rho)(-1)D'(Z)$

$+ (1 - \rho)(1 - Z)D''(Z)$

at $Z = 1$, $D(1) = P(1) = 1$ then,

$$E[n] = \frac{(1 - \rho)D'(1,T)}{1 - D'(1,T)} + \frac{D''(1,T)}{2(1 - D'(1,T))} \qquad [2.2.14]$$

and

$$D'(1,T) = \{(\lambda T)M'(Z)\exp(-\lambda T(1 - M(Z)))\}|_{Z=1}$$
$$= \lambda T$$
$$= \rho$$

and

$$D''(1,T) = \{(\lambda T M'(Z))^2 \exp(-\lambda T(1 - M(Z)))\}|_{Z=1}$$
$$= \rho^2$$

substituting into the above equation, we get

$$\begin{aligned} E[n] &= \frac{(1-\rho)\rho}{1-\rho} + \frac{\rho^2}{2(1-\rho)} \\ &= \frac{(2\rho(1-\rho) + \rho^2)}{2(1-\rho)} \\ &= \frac{2\rho - \rho^2}{2(1-\rho)} \\ &= \frac{\rho}{1-\rho}[1 - \frac{\rho}{2}] \end{aligned} \qquad [2.2.15]$$

The expected queue length is, therefore

$$E[n_q] = E[n] - \rho = \frac{\rho^2}{2(1-\rho)} \qquad [2.2.16]$$

Exluding the satellite propagation delay (T_p), The average time delay of packets arriving at the buffer can be found using Little's formula

$$E[t] = \frac{1}{\lambda}E[n] = \frac{T}{1-\rho}(1 - \rho/2) \qquad [2.2.17]$$

Thus, the total queueing delay is given by

$$E[t_q] = \frac{T}{1-\rho}(1 - \rho/2) - T = \frac{T\rho}{2(1-\rho)} \qquad [2.2.18]$$

For this TDM outbound channel, we have N traffic streams from N different VSAT's within a group transmitting equal length packets according to a Poisson process with rate λ/N packets/sec each. As mentioned previously, the time axis in this channel is divided into N-slot frames with one slot dedicated to each traffic stream. Each slot is one time unit long (T sec in duration) and can carry a single packet.If packet transmission from different groups can start only at times NT,2NT,..,i.e.,at the beginning of a frame of NT sec long, then this scheme of dividing the channel into N-slot frames can be modeled as N independent and separate M/D/1 queueing systems with vacations [30] each with a mean arrival rate of λ' packets/sec ; This is the concept of Synchronous Time Division Multiplexing (STDM). When there are no packets in the queue for a given stream at the beginning of a slot, the server (hub) takes a vacation for one frame. Thus, the average vacation time is $\overline{T}_v = NT$ and its mean square value is $\overline{T_v^2} = (NT)^2$. Having taken this vacation time into account, the total queueing delay becomes $E[t_q] = E[t] - 1/C\mu + \frac{\overline{T_v^2}}{2\overline{T}_v}$, and for fixed length packets of duration T ($1/\mu = T$). The factor C denotes the effective bit rate of the channel which is simply R_b'/N bps.

$$\begin{aligned} E[t_q] &= \frac{NT}{1-\rho}(1 - \frac{\rho}{2}) - NT + NT/2 \\ &= NT(\frac{2-\rho}{2(1-\rho)} - 1) + NT/2 \\ &= \frac{NT(-\rho + 2\rho)}{2(1-\rho)} + NT/2 \\ &= NT/2(\frac{\rho}{1-\rho} + 1) \end{aligned} \qquad [2.2.19]$$

Where the utilization factor, ρ, for each queueing system is $\rho = \lambda' NT = \lambda T$. Vacation time will be useful when we later study the delay of the ALOHA DS-CDMA/TDM channel. For data networks, such as the one we are studying, vacations will correspond to the transmission of various control and record keeping packets. In fact, we will use these vacation to send special instruction packets to monitor the DS-CDMA channel

2.2.2 Finite Buffer Case

The analysis of the preceding section is based on the assumption of an infinite buffer. In a practical application, the assumption is considered reasonable if one could expect that the buffer size, although finite, would be large enough so that the probability of overflow would be negligible under normal operating conditions.

When the size of the buffer is such that the probability of overflow is not negligible, the analysis of the preceding section leading to the probability generating function is not applicable. Various papers have appeared indicating the effect of finite buffer size, and provided design curves relating the probability of buffer overflow to message statistics and buffer size. The approaches of both Chu [21] and Medlin [22] consist of actually writing down explicit expressions for the probabilities of buffer occupancy, with a maximum buffer size N, and solving the resultant equations recursively by computer. In this section, we shall adopt their idea but extend it further to obtain an explicit analytical expression for the probability of finding n packets in the buffer, p(n), of length N waiting to be read out in terms of the probability of an empty buffer, p(0). Specifically, let T again be the service time introduced in the last section during which at most one packet is released by the buffer. Assume that the system has been in operation for a long enough time for equilibrium to have set in. A random number of packets r may arrive every T sec interval, and a random number n packets are present in the buffer, this is shown in figure 9.

A packet is removed in its arrival sequence at a constant sampling instant of fixed period T. For a Poisson input rate let $\rho = \lambda T$ where λ is the Poisson arrival rate, then

$$p_1(r) = \begin{cases} \exp(-\rho)\rho^r/r! & , r \neq 0 \\ \exp(-\rho) & , r = 0 \end{cases} \qquad [2.2.20]$$

$p_1(r)$ may be found iteratively by

$$p_1(r) = \frac{\rho}{r} p_1(r-1) \qquad [2.2.21]$$

and $p_1(0) = \exp(-\rho)$

Then

$$p_1(r) = \frac{\rho^r}{r!} p_1(0) \qquad [2.2.22]$$

With the appropriate statistical model chosen, the next step in the solution of the buffer length problem is to determine p(n), where n is the number of packets in the queue immediately after a packet removal attempt. For p(n) to be meaningful, the queue length statistics must be stationary,i.e., $p(n,t+T) = p(n,t)$ for all n and t, where p(n,t) is the probability of there being n packets in the queue at time t, and T is the interval between packet removal attempts. As long as ρ is less than unity, stationarity will exist after the decay of initial transients in the system.

Since the queue can never be longer than the buffer, p(n) must be equal zero for all $n > N$. Also, because the instant of time under consideration has been chosen to be

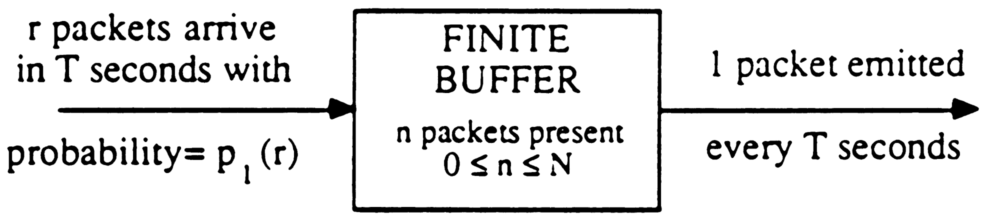

Figure 9. Finite Buffer Model

immediately after an attempt to remove a packet, p(n=N) must also be zero. A little thought will indicate that the probability of L packets in the queue just before sampling instant equals to the probability of (L-1) packets just after,i.e.,p(L-1). So given a buffer of length N

$$\sum_{n=0}^{L} p(n)p_1(L-n) = p(L-1) \quad \text{for} \quad 1 \le L \le (N-1) \qquad [2.2.23]$$

For example,

$L = 1, \qquad p(0) = p(0)p_1(1) + p(1)p_1(0)$

$L = 2, \qquad p(1) = p(0)p_1(2) + p(1)p_1(1) + p(2)p_1(0)$

These equations simply relate the probability of having a particular state present at the end of a T sec interval to the possible states that could have existed at the end of the previous interval, given no more than one packet is removed in T sec. Thus, for example, the first equation says that an empty buffer (with probability p(0)) could have risen if there had been one packet present (this packet was then removed in the T sec interval) and none arrived (the probability of this is $p(1)p_1(0)$) or if the buffer were previously empty and one packet arrived in the T sec interval. (the probability of this event is $p(0)p_1(1)$). Since these are mutually exclusive events, one simply adds the probability of the two events. The second equation equates the probability p(1) of one packet being present in the buffer to the probability of three events that might have contributed to this state :

1. Two packets were present during the previous interval, one was emitted, none arrived $p(2)p_1(0)$.
2. One was present and left, one arrived $p(1)p_1(1)$.
3. Buffer was empty (hence none left) and two arrived but one was emitted $p(0)p_1(2)$.

Then with a little arrangement we get :

$$p(1) = \frac{1}{p_1(0)}[p(0) - p_1(1)p(0)] \qquad [2.2.24]$$

$$p(2) = \frac{1}{p_1(0)}[p(1) - p_1(2)p(0) - p_1(1)p(1)] \qquad [2.2.25]$$

so on until we get a general equation for $n = L$ and that is

$$p(L) = \frac{1}{p_1(0)}\left[p(L-1) - \sum_{r=1}^{L} p_1(0)\frac{\rho^r}{r!}p(L-r)\right] \qquad [2.2.26]$$

If we express p(L-1) in terms of p(L-2) and p(L-2) in terms of p(L-3) ... and so on until we get to p(0), one should get the following expression for p(n) :

$$p(n) = \left\{\frac{1}{p_1^n(0)}p(0) - \sum_{j=0}^{n-1}\sum_{r=1}^{n-j}\frac{1}{p_1^j(0)}\frac{\rho^r}{r!}p(n-j-r)\right\} \qquad [2.2.27]$$

Also, $\sum_{n=0}^{N-1} p(n) = 1$ for $1 \le n \le (N-1)$. Equations (2.2.21) and (2.2.26) can be used to find p(n) in terms of p(0) by iteration.

$$p(1) = \left[\frac{1}{p_1(0)} - \rho\right]p(0) \quad \text{for} \quad n = 1 \qquad [2.2.28]$$

$$p(2) = \left[\frac{1}{p_1^2(0)} - \frac{2\rho}{p_1(0)} + \frac{\rho^2}{2!}\right]p(0) \quad \text{for} \quad n = 2 \qquad [2.2.29]$$

$$p(3) = \left[\frac{1}{p_1^3(0)} - \frac{3\rho}{p_1^2(0)} + \frac{4\rho^2}{2!p_1(0)} - \frac{\rho^3}{3!}\right]p(0) \quad \text{for} \quad n = 3 \qquad [2.2.30]$$

and so for every n $\in$ N

$$p(n) = \left\{ \sum_{k=0}^{n} (-1)^{n-k} \frac{(k + 1)^{n-k} \rho^{n-k}}{p_1^k(0)(n - k)!} \right\} p(0) \qquad [2.2.31]$$

The next step in the analysis is to find $p(n = 0)$. The following equation, $\sum_{n=0}^{\infty} P(n) = 1$, can be used for final scaling and gives a value to p(0) as follows :

$$\sum_{k=0}^{N-1} p(n) = \left\{ \sum_{n=0}^{N-1} \sum_{k=0}^{n} (-1)^{n-k} \frac{(k + 1)^{n-k} \rho^{n-k}}{p_1^k(0)(n - k)!} p(0) \right\} = 1 \qquad [2.2.32]$$

so

$$p(0) = 1 / \left\{ \sum_{n=0}^{N-1} \sum_{k=0}^{n} (-1)^{n-k} \frac{(k + 1)^{n-k} \rho^{n-k}}{p_1^k(0)(n - k)!} \right\} \qquad [2.2.33]$$

Now we note that the buffer input arrival rate is lower than the rate of packet removal attempts by the factor $\rho = \lambda/\mu = \lambda T$. Hence, if the buffer were infinite in length, the probability that on any removal attempt a packet would be removed from the buffer would be ρ. For a finite buffer; however, the loss of a fraction of the packets due to overflow will reduce this probability. Hence,

$$Pr[\text{removing a packet}] = \rho(1 - R_L) \qquad [2.2.34]$$

where R_L is the fractional packet loss due to buffer overflow. Then,

$$p(n = 0)p_1(r = 0) = Pr[\text{not removing a word }] = 1 - \rho(1 - R_L) \qquad [2.2.35]$$

but for a poisson input

$p_1(r = 0) = \exp(- \rho)$

so

$p(n = 0) \exp(- \rho) = (1 - \rho - \rho R_L)$

and

$$R_L = \frac{(1 - p(0)\exp(-\rho))}{\rho} - 1 \qquad [2.2.36]$$

where p(0) is given in equation (2.2.33). However, if one could specify a bound on R_L, we could solve for p(0) from equation (2.2.35) and substitute it back in (2.2.31) to get the size of the buffer length necessary to maintain a prescribed 'negligible' loss of packets caused by buffer overflow. With that, we can proceed to calculate the average queueing delay of a packet before it gets its turn to be serviced by the hub.

2.2.3 Stop-and-Wait Automatic Repeat Request

In studying the asynchronous time division multiplexing, it was assumed that the transmission of data units (packets) was error free in both the forward and feedback channel. However, to achieve reliable communication, various methods of error detection and retransmission (Automatic Repeat Request, ARQ) schemes are used for the control of errors when communication is over noisy channel of certain probability error rate. In such schemes, a message must be stored in a buffer at the sending multiplexer until it is informed (via an ACK/NACK message) that the message has been correctly received by the receiving station. This storage of messages, even after transmission, affects the queue length and waiting time for the multiplexer. The purpose of this section is to analyse an ARQ strategy known as the stop-and-wait.

In the stop-and-wait system, after each transmission of a packet, the transmitter is idle and waits for an ACK/NACK message from the receiver. In this strategy, the receiver checks the received packet and generates an acknowledgment for the sender. If the packet is correctly received, an ACK message is sent via a return channel and the sender is permitted to transmit a new message. If the message is incorrectly received, a NACK message is returned and the sender retransmits the same message. In the following analysis of the stop-and-wait strategy, it is assumed that all channel errors are detected. Furthermore, if errors are detected in the ACK/NACK message the transmitting multiplexer follows the appropriate strategy as if a NACK was received.

When ARQ strategies are employed, errors in transmission of a data unit result in one or more retransmissions of that data unit. For our case, it is sufficient to realize that a data unit may be transmitted several times and we denote by N_i the total number of transmissions (including retransmissions) of data unit i. We assume that the random variables N_i form an i.i.d sequence with probability generating function $G_n(Z)$. Again, for simplicity we shall consider a modified geometric model [23] with parameters p and p'.

As in the case of infinite buffer, we assume simple Poisson arrival of data units (packets). The new element that we shall consider is that the receiver may detect an error in the data unit and the feedback channel is not error free. We shall assume that errors on successive trials are independent events so that the number of retransmissions is geometrically distributed. The generating function is given by : [23]

$$\begin{aligned} G_n(Z) &= \sum_{k=1}^{\infty} Z^k(1-p)p^{k-1} \sum_{k=1}^{\infty} Z^k(1-p')p'^{k-1} \\ &= \frac{(1-p)(1-p')Z}{(1-pZ)(1-p'Z)} \end{aligned} \qquad [2.2.37]$$

This model applies to the situation where once a packet has been received correctly by the receiver, the receiver continues to transmit an ACK for that data unit until the ACK is successfully received by the transmitter. Here p is the probability of error in the forward channel, p' is the probability of error in the feedback channel and errors in the two channels are assumed independent.

As in the previous section, a Markov chain is embedded at the points where a packet is successfully transmitted. However, because of considering the probability of errors in this outbound channel, the embedded points are not separated by single slot times. If (k-1) retransmissions of a message are required, the total number of slots required to transmit a packet is k(R+1), where $R = 2T_p/T$ is the round trip propagation delay in slots assuming negligible processing delays at either the transmitter or receiver. The analysis is the same as before with the only difference that the interdeparture interval is now k(R+1) when there are k transmissions. So

$$Pr[\text{m messages / k trans}] = \frac{(\lambda k(R+1)T)^m}{m!} \exp(-\lambda k(R+1)T) \qquad [2.2.38]$$

and of probability generating function

$$\begin{aligned} G(Z) &= \sum_{n=1}^{\infty} Z^n Pr[\text{n messages / k trans}] \\ &= \sum_{n=1}^{\infty} Z^n \frac{(k\lambda(R+1)T)^n}{n!} \exp(-k\lambda(R+1)T) \qquad [2.2.39] \\ &= \exp(-k\lambda(R+1)T)\exp(Z\,k\lambda\,(R+1)T) \\ &= \exp(-k\lambda(R+1)T(1-Z)) \end{aligned}$$

Under the assumption of simple poisson arrival, the generating function of the number of packets in a message is $M(Z) = Z$. From this it follows that, when averaging over k, the probability generating function of the number of data units in the interdeparture interval is :

$$\begin{aligned} D(Z) &= G_n(G(Z)) \\ &= \frac{(1-p)(1-p')\exp(-k\lambda(R+1)T(1-Z))}{(1-p\exp(-k\lambda(R+1)T(1-Z)))(1-p'\exp(-k\lambda(R+1)T(1-Z)))} \qquad [2.2.40] \end{aligned}$$

so using equation (2.2.14) of this section

$$E[n] = \frac{(1-\rho)D'(1,T_s)}{1-D'(1,T_s)} + \frac{D''(1,T_s)}{2(1-D'(1,T_s))} \qquad [2.2.41]$$

with

$$\begin{aligned} D'(1,T_s) &= (k\lambda\, T(R+1))\left[1 + \frac{p}{1-p} + \frac{p'}{1-p'}\right] \\ &= (k\rho)\left[1 + \frac{p}{1-p} + \frac{p'}{1-p'}\right] \end{aligned} \qquad [2.2.42]$$

and

$$\begin{aligned} D''(1,T_s) &= (k\lambda T(R+1))^2\left[\frac{(1+2p+2p'-3pp')}{(1-p)(1-p')} - \frac{2(p+p'+2pp')(1-pp')}{(1-p)^2(1-p')^2}\right] \\ &= (k\rho)^2\left[\frac{(1+2p+2p'-3pp')}{(1-p)(1-p')} - \frac{2(p+p'+2pp')(1-pp')}{(1-p)^2(1-p')^2}\right] \end{aligned} \qquad [2.2.43]$$

We, then, can get the various queueing delays, except here we are considering the possibility of errors existing in both the inbound and outbound channels. In fact, the probabilities p and p' used above are nothing but the bit error probabilities in the outbound and the inbound channels respectively.

2.3 Channel Performance

Here, we examine the performance of the HDLC in its Asynchronous Response Mode (ARM) of operation. This unbalanced mode is usually used in the case (such as ours) where the primary station (hub) communicates with the secondary station

(VSAT's). A VSAT does not need permission from the hub to initiate a transmission. This is unlike the polling techniques where secondary stations are allowed to initiate transmission only in response to query from the hub. In addition, we only concern ourselves with the reject (Go-Back-N) error recovery mode of the HDLC in this ASTDM communication outbound channel of bit error rate R'_b . When considering the CDMA inbound channel, we will study the performance of HDLC in its balanced operation mode. The Asynchronous Balanced Mode (ABM) is used for point-to-point link between a VSAT (within a group) and the hub station. The stations on both sides of the link are, so-called, combined stations; each station can send and receive both commands and responses [24]. For handling the response to acknowledgments, the HDLC will use the Stop-and-Wait procedure and will operate in its selective reject (SREJ) mode of operation.

Given a noisy feedback channel for the return of acknowledgments, and for a given channel bit error probability (assuming random bit errors) P'_b, the 'superframe' success probability, P_S, can be given by

$$P_S = (1 - P'_b)^L (1 - P_b)^{L'} \qquad [2.3.1]$$

where L is the 'superframe' length in bits including overhead. L' is the size in bits of the frame containing the acknowledgments, and P_b is the bit error probability for the CDMA channel carrying the ACK messages. Knowing this success probability, we can readily calculate the average time for a correct transmission to be received which is

$$\begin{aligned} T_{av} &= T_T + P_S \sum_{i=1}^{\infty} i \; P_E^i \, T_T \\ &= T_T + P_S T_T \frac{P_E}{(1 - P_E)^2} \\ &= \frac{T_T}{P_S} \end{aligned} \qquad [2.3.2]$$

P_E is the probability of a superframe being received in error, and T_T is the minimum time between successive superframes. T_T is greater than the sum of round trip propagation delay 2 T_p, time to transmit a superframe T_{SF}, queueing delay and the receiver processing time (T_q + T_{proc}) and the time to transmit an acknowledgment. The acknowledgment is given by a fixed value equal to the HDLC supervisory frame length t_r. $t_r = L'/R'_b$ where L' is the control field size. $L' = 56$ bits for a modulus $M = 128$ in this satellite link case. If we assume that the hub always has a superframe waiting for transmission, T_{av} represents the average time between transmission of correct superframes. Thus, the maximum throughput of this ASTDM outbound channel under the Stop-and-Wait (SW) ARQ retransmission strategy is $1/T_{av}$ packets/sec.

3.0 DELAY ANALYSIS OF THE ALOHA DS-CDMA/TDM CHANNEL

3.1 Effect of the Number of Simultaneous Users on the CDMA Channel

The communication traffic behavior in the network can be modeled as an M/D/K queueing system i.e. the arrival process is Poisson with mean arrival rate of λ (packets/sec), the access duration in the CDMA channel is of mean duration $1/\mu = T$ (T is the transmission time of a fixed-length packet), and K is the number of available servers or users simultaneously accessing the channel. A server is equivalent to a PN sequence assigned to a group of VSAT's that defines a particular CDMA channel. Further, assume that an access to the channel will be denied and lost, if there are K simultaneous transmissions in the channel at the instant of access attempt. This model then is identical to Erlang's loss system [25] with the probability P_K of K transmissions already present at the instant of access attempt and is given by

$$P_K = \frac{(\lambda T)^K / K!}{\sum_{i=0}^{K} (\lambda T)^i / i!} \qquad [3.1.1]$$

with $\lambda T = G$ Erlangs as the traffic intensity.

With the multi-access scheme for the network specified being TDM/DS-CDMA, the station may access the CDMA channel (i.e. transmit) at random; however, the maximum number of simultaneous transmissions is limited to K_{max} . This is achieved by the HUB station monitoring the CDMA channel and sending a " no access " signal in the TDM channel during the vacation times whenever the number of users $K = K_{max}$. This prevents additional stations from accessing the channel until $K < K_{max}$. Alternatively, the network may be designed for desired values of P_e and P_K for some typically anticipated K, and be permitted "graceful degradation" in performance whenever the number of simultaneous users exceeds (hopefully momentarily) the design value of K. Refer to figure 10. This figure is useful to obtain the permitted simultaneous users in the CDMA channel if one has an idea of what the traffic intensity of any type of communication (voice or data) is equal to given the number of those active VSAT's accessing the channel.

3.2 Study of Slotted ALOHA DS-CDMA

It is known that in unslotted and slotted random access packet switching (ALOHA type) [26] when two or more packets "collide" partially or fully, the packets are consid-

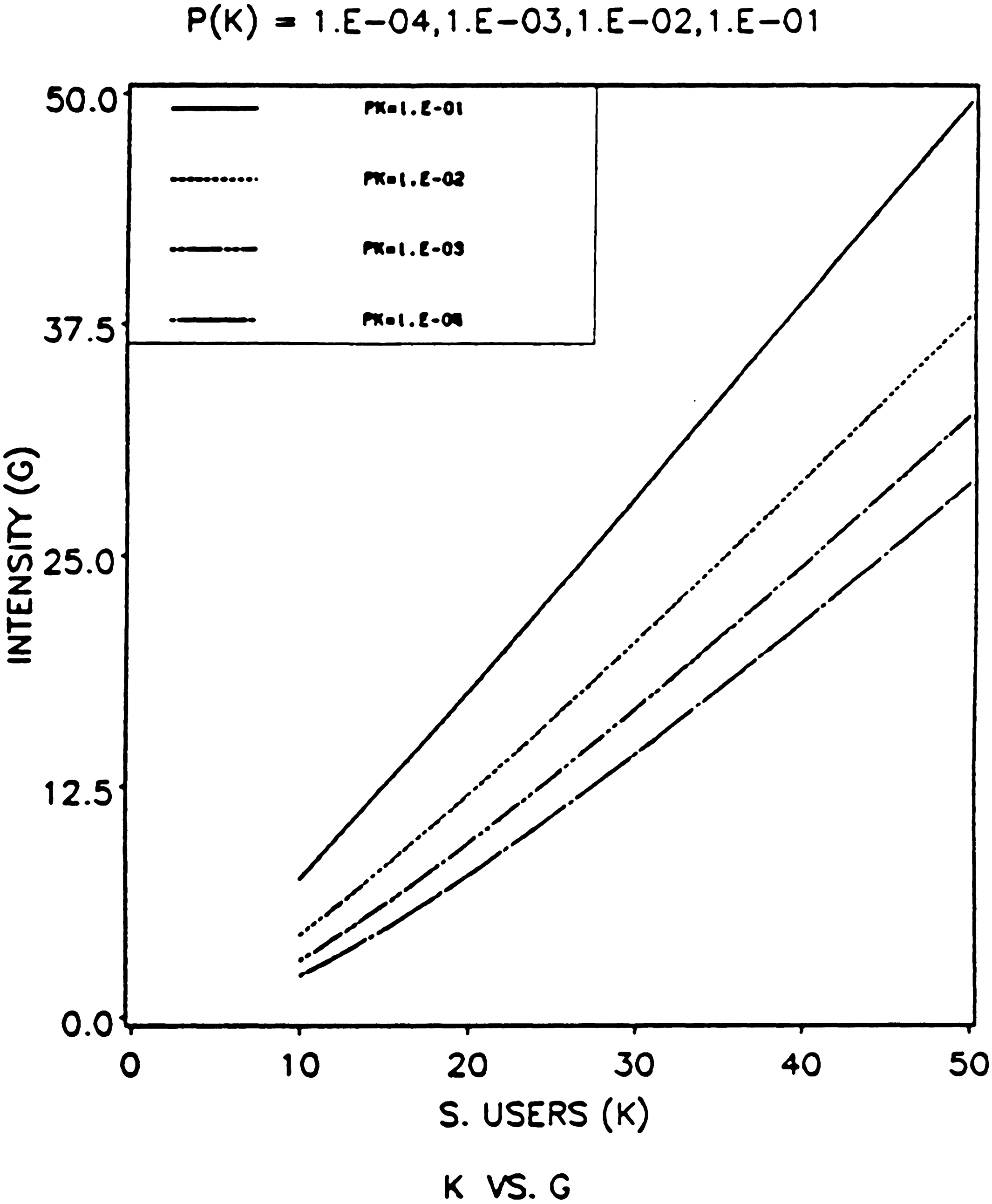

Figure 10. Number of Users vs. Traffic Intensity in the DS-CDMA Model

ered lost. However, when a spread-spectrum technique [6],[27] such as Code-Division Multiple-Access (CDMA) is used in conjunction with the ALOHA techniques, it is possible to recover some of the collided packets and thereby increase the throughput. The throughput can further be increased by correcting errors introduced by the interference due to simultaneous presence of of several spread signals [15]. Most of the studies on random access schemes in the literature do not take into consideration acknowledgments (ACK's) between a receiver and the corresponding transmitter. When ACK's are required (usually positive acknowledgements) for data transmission to indicate correct reception of packets, a recent study on the effect of including ACK traffic also on the same ALOHA channels [28] has shown that the throughput drops significantly. But, one might argue that since recovery of collided packets is possible with the use of spread-spectrum techniques, such channels (CDMA) appear to be attractive for carrying ACK's also, instead of using separate channels. In this section, we attempt to evaluate the effect of positive ACK traffic on the performance of the slotted ALOHA DS-CDMA scheme.

In order to carry out the analysis, we shall need some assumptions. We shall assume that the near-far effects are negligible and that the probability of occurrence of two or more packets addressed to the same terminal in a given slot is also negligible. In addition, since the ACK message is usually quite small compared to the length of a data packet, and to make the analysis simpler, it is safe to assume that the probability of receiving an ACK message correctly can be made nearly equal to one. In this channel, One can construct an ACK packet with several repetitions of the ACK message in that packet. However, this probability will no longer be assumed one when dealing with the TDM outbound channel. An error detecting capability can also be provided and whenever more than 'e' errors occur in a packet, it is recognized by the receiver and can be discarded. Also, the sending VSAT terminal reschedules the packet it just finished transmitting with a random delay chosen much like in a slotted ALOHA [29] if it detects more than 'e' errors in the received signal. In our case, the one way satellite propagation delay has to be incorporated in this random rescheduling delay. When a packet is correctly received, a positive acknowledgment (ACK) is transmitted in the next immediate slot. Since a terminal can only send one packet in a slot (either a data or an ACK packet), the total number of data packets sent in a given slot depends also on the number of successes in the previous slot (or the number of ACK packets in that slot). For simplicity we assume that there is no distinction between the original and the rescheduled packets in the traffic from a VSAT terminal. Also, when a terminal has a data packet ready for transmission but is forced to to send an ACK packet in response to a received packet, this situation is considered to be equivalent to rescheduling the data packet by one slot delay. The effect of this on the Poisson packet arrival distribution is really negligible because the mean rescheduling delay is large due to the large satellite propagation delay. The packet flow in the synchronous (slotted) random access DS-CDMA system is shown in figure 11.

In the absence of ACK packets, the average throughput (expected number of successful packets / slot), β_1, for an infinite number of terminals (Poisson arrival process) is given by [15] as :

$$\beta_1 = \sum_{k=1}^{\infty} Pr[\text{expected \# of successful packets per slot/k simult. trans.}]Pr[\text{k simult. trans.}] \quad [3.2.1]$$
$$= \sum_{k=1}^{\infty} K\, P_c(k) \exp(-G) G^k / k!$$

where G is the offerred traffic ($G = \lambda T$ packets/slot and T is the slot duration).

$$P_c(k) = (1 - P_p(k)) \quad [3.2.2]$$

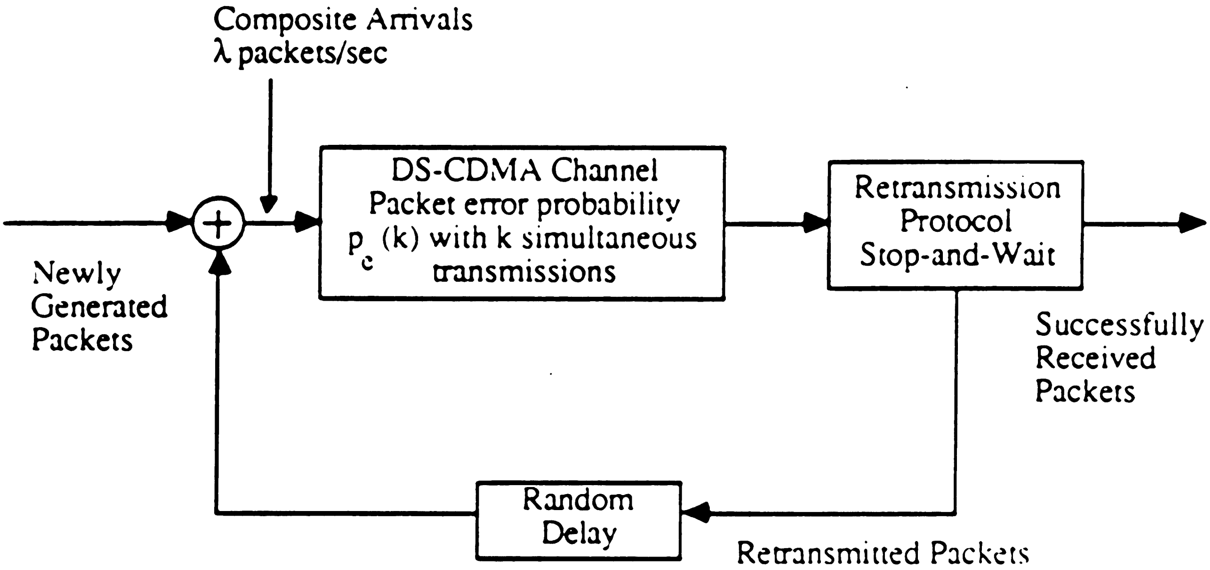

Figure 11. Schematic Representation of Packet Flow in a Slotted Random Access DS-CDMA system

for which $P_p(k)$ is the packet error probability when k packet transmissions are present in a given slot. So the normalized utilization for the ALOHA-CDMA scheme, S_1 is given by

$$S_1 = \frac{\beta_1}{G_p} \qquad [3.2.3]$$

G_p is the processing gain.

As argued in the previous section, in practice $P_c(k)$ decreases with increasing k, then we can treat the summation above as consisting of only k_{max} terms. When ACK packets are also transmitted in each slot, the number of successful data packets i and j in two consecutive slots starting at times t and (t+1), respectively, can be represented by a finite discrete-time Markov chain and is characterized by the state transition matrix $P = [P_{ij}]$ where P_{ij} is the one-step transition probability from state i to state j and is as follows :

$$\begin{aligned} P_{ij} &= Pr[X(t+1) = j/X(t) = i] \\ &= \sum_{n=j}^{\infty} Pr[X(t+1) = j/n \text{ arrivals}]\ Pr[n \text{ arrivals}/X(t) = i] \end{aligned} \qquad [3.2.4]$$

X is a random variable (RV) representing the number of successful data packets in a slot. Note, in this derivation, the throughput takes into account only the successful data packets in a given slot, and the ACK traffic is considred here to be a factor of interference and thus will have an effect on $P_c(K)$. In the (t+1) slot, we have i ACK packets from i terminals (due to i successful data packets in the previous slot) and n data packets arriving from the remaining terminals. If we now restrict the upper limit of the above summation to just $(k_{max} - i)$, then the total number of packet transmissions in the (t+1)th slot is, $k = (i + n), 0 \leq n \leq (k_{max} - i)$ and for $k > k_{max}$, $P_c(k) = 0$. K_{max} can be chosen with reference to figure 9 for a given P_K. Therefore for a large, but finite, number of stations (VSAT's) the composite arrival distribution of the random variabe X at the (t+1) slot becomes a binomial distribution with parameters (j , P_c) for a total of n arrivals. $C_j^n = \dfrac{n!}{(n-j)!j!}$ equals to the number of different groups of j VSAT's that can be chosen from a set of n transmitting VSAT's. So

$$P_{ij} = \sum_{n=j}^{(K_{max}-i)} C_j^n P_c^j(k)(1 - P_c(k))^{n-j} \exp(-G)G^n/n! \qquad [3.2.5]$$

for $j \leq (k_{max} - i)$, and $k = (i + n)$

Also, with a little thought, one can distinguish three more possible cases depending on the values of i and j.

1. if $i = k_{max}$, $j = 0$ $\qquad P_{ij} = 1$
2. if $i \neq k_{max}$, $j = 0$ $\qquad P_{ij} = (1 - \sum_{k=1}^{(K_{max}-i)} P_{ik})$
3. if $j > (k_{max} - i)$, $\qquad P_{ij} = 0$

In the first case, all data packets transmitted in the previous slot were acknowledged at the present slot denoting a successful transmission, next only some of the data packets were acknowledged and the rest were unsuccessfully transmitted, and lastly we assume that none of the data packets would be sucessfully transmitted once we have more than K_{max} simultaneous transmissions.

In order to find the average steady state throughput β_2 (the expected number of successful transmissions per time slot), the long-term state probability of the success vector $\Pi = (\pi_0, \ldots, \pi_{k_{max}})$ must be found. The values of π_i are given by the solution of $\Pi = \Pi P$ under the condition $\sum_{i=0}^{K_{max}} \pi_i = 1$. Finally, the normalized channel utilization is given by $S_2 = \beta_2/G$, where, $\beta_2 = \sum_{i=1}^{K_{max}} i\pi_i$.

For a given channel traffic G and average throughput β_2, the average number of retransmissions per successful packet, r_{av}, is given by [15]

$$r_{av} = \frac{G - \beta_2}{\beta_2} = (G/\beta_2 - 1) \qquad [3.2.6]$$

The average delay $\overline{D}$ in slots can be obtained by multiplying r_{av} with the average rescheduling delay ($\overline{RD}$) in slots per retransmission. The retransmission delay incurred by an unsuccessful packet transmission may be regarded as the sum of a deterministic component (R) and a random component. In the case of a satellite system, the deterministic component corresponds to the round trip satellite propagation delay (one-hop) added to the processing time, T_{proc}, so $R = \dfrac{(2T_p + T_{proc})}{T} = \dfrac{0.50 \sec + T_{proc}}{T \sec / slot}$ where T is the packet transmission time . Then

$$\overline{RD} = (R + 1/2 + (k + 1)/2) \qquad [3.2.7]$$

and the term (1/2 slots) is added to account for packets that arrive after a slot interval has begun, and (1/k) is the probability that a slot is being chosen among the K slots during which each user could retransmit a previously collided packet at random. In the case of pure ALOHA

$$\overline{RD} = (R + (k + 1)/2) \qquad [3.2.8]$$

In the case of pure ALOHA, an arbitrary time interval [1,k] is chosen and a uniformly distributed random retransmission time is selected within that time interval. The time interval covers k packet-unit times of T sec each. Therefore, the average delay in slots is

$$\overline{D} = (R + (k + 2)/2)(G/\beta_2 - 1) \quad \text{slots} \qquad [3.2.9]$$

3.3 *Study of the ALOHA DS-CDMA/TDM Channel*

A star configuration packet switched satellite network with a hub station and N VSAT's is considered in this research. The hub can hear the broadcast from any of the VSAT's, but a VSAT can only hear from the hub through the TDM outbound channel. The hub relays the packets it receives through the satellite transponder. The VSAT's are assumed identical, each with a single buffer for data packets. The hub maintains a queue of packets to be relayed in FCFS order.When its queue is not empty, the hub broadcasts the packet at the front of the queue. We will assume that the hub is acknowledged when it receives an acknowledgment packet from the destination terminal. In the following analysis we proceed to determine the total service time and delay experienced by a packet in this star VSAT packet switched satellite network. We shall use the given bit error probabilities of both the inbound and outbound lines to study the network behavior.

In this analysis, we shall assume that the random errors in the received packets from the ALOHA DS-CDMA channel are solely due to simultaneous presence of the coded packets. The probability that two or more packets using the same PN code will overlap completely and that they will be destroyed is ignored. In other words, no packet collision is detected at any time in this network. In addition, we assume that errors in both channels are independent of each other, and to proceed with the analysis the following commonly used terms are introduced:

T_{VH} = Data packet transmission time from a VSAT to the hub (sec)

T_{HV} = Data packet transmission time from the hub to a VSAT (TDM frame length) (sec)

T_{ack} = ACK packet transmission time when transmitted from a VSAT (sec) (it is the HDLC supervisory frame length carrying the ACK)

T_{out} = Timeout interval (sec)

T_p = One-way satellite propagation delay (0.25 sec) VSAT-satellite-hub or hub-satellite-VSAT

T_q = Queueing delay at the hub station (sec)

P_b = Bit error probability of the DS-CDMA link

P'_b = Bit error probability of the TDM link

R_b = DS-CDMA channel bit rate (bps)

R'_b = TDM channel bit rate (bps), $R'_b = NR_b$

T_{proc} = Processing delay at the receiving VSAT (sec)

T'_{proc} = Processing delay at the receiving hub station (sec)

$E[X_d]$ = Expected number of the down-link retransmissions of a certain data block

$E[X_u]$ = Expected number of the up-link retransmissions of a certain data block

p_d = Packet success probability of a given down-link transmission ($q_d = 1 - p_d$)

p_u = Packet success probability of a given up-link transmission ($q_u = 1 - p_u$)

$\overline{RD}$ = Random retransmission delay on the ALOHA DS-CDMA channel (slots)

L = Total number of bits in a data packet including overhead (depends on the packet transmission time and the corresponding channel bit rate)

L' = Total number of bits in a packet carrying the ACK from the hub (bits)

T_v = Vacation time in the TDM frame (sec)

N = Total number of VSAT's in the network

T_s = Packet service time at the hub or VSAT

Assuming no collisions and that packets arrive at any node are transmitted immediately, a successfull transmission depends only on the channel characteristics. For a message which will be acknowledged, the average time delay, $\overline{TD}$,separating the start of a packet transmission and the receipt of the acknowledgment at the transmitting node depends only on which channel an error has occured. If the ACK is not correctly received at the transmitting node in its first transmission attempt, the corresponding data packet has to be retransmitted until it is correctly acknowledged by the hub before initiating the transmission of the next frame carrying the data packet.

To determine the packet service time T, we distinguish four cases. For instance, let E denote the event where a packet transmission takes place from a VSAT remote terminal to the HUB and define the indicator random variable X by

$$X = \begin{cases} 1 \\ 0 \end{cases} \qquad [3.3.1]$$

A value of 1 is assigned to X if the transmission is successful (error-free channel), and a value of 0 if the data packet is not successfully transmitted and thus has to be retransmitted.

Similarly, let F denote the event where an ACK packet is being transmitted in the TDM channel and the indicator radom variable Y be defined as follows

$$Y = \begin{cases} 1 \\ 0 \end{cases} \qquad [3.3.2]$$

Here, a value of 1 is assigned to Y when the ACK is successfully transmitted, and 0 if the ACK is not correctly received by the VSAT due to random errors in the TDM channel. Then, from the definitions of X and Y it follows that

$$E[X] = P\{E\} \qquad [3.3.3]$$

$$E[Y] = P\{F\} \qquad [3.3.4]$$

Therefore, the conditional expectation of the minimum time delay, given that $X = x_i$ ($Y = y_i$) is given by

$$E[TD/X = 1, Y = 1] = 2T_p + T_{VH} + T'_{proc} + T_{proc} + T_q + \frac{T_{HV}}{N} \qquad [3.3.5]$$

where $\frac{T_{HV}}{N}$ is the infomation frame length (data packet plus overhead) carrying the ACK message from the hub to the transmitting VSAT, and

$$E[TD/Y = 1] = T_p + T'_{proc} + \frac{T_{HV}}{N} \qquad [3.3.6]$$

So, if we assume that the packet is successfully transmitted in its first transmission attempt, then the total transmission time of this packet from a VSAT to the hub T_1 is

$$T_1 = E[TD/X = 1, Y = 1] - E[Y = 1] \qquad [3.3.7]$$

also

$$E[TD/X = 0, Y = 1] = E[TD/X = 1, Y = 1] + E[X_u]T_{VH}\overline{RD} \qquad [3.3.8]$$

and

$$E[TD/Y = 0] = 2T_p + 2T_{proc} + T_{VH} + \frac{T_{HV}}{N} + T'_{proc} \qquad [3.3.9]$$

On the other hand, if the above assumption is not true (the packet is not successfully transmitted), the total transmission time becomes T_2

$$T_2 = E[TD/X = 0, Y = 1] - E[TD/Y = 1] \qquad [3.3.10]$$

When a data block has to be retransmitted, and for the random variable $X_u(X_d)$ denoting the number of the up-link (down-link) retransmissions being geometrically distributed, it is known that

$$E[X_d] = p_d \sum_{n=1}^{\infty} n q_d^n = p_d q_d/(1 - q_d)^2 = \frac{q_d}{p_d} \qquad [3.3.11] .$$

where $p_d = (1 - P_b')^{L'}$ and $L' = R_b' \frac{T_{HV}}{N}$ and

$$E[X_u] = p_u \sum_{n=1}^{\infty} n q_u^n = p_u q_u/(1 - q_u)^2 = \frac{q_u}{p_u} \qquad [3.3.12]$$

where $p_u = (1 - P_b)^L$ and $L = R_b T_{VH}$

For this star network with the hub transmitting an ASTDM superframe with N slots (one slot for each VSAT remote terminal that carries a single data frame of length T_{HV}/N), and when an ACK packet (HDLC supervisory frame) is to be sent from the hub terminal, this acknowledgment packet will be piggybacked in the next TDM frame carrying a data packet to any VSAT in the network. Also, since we can have the hub monitoring the DS-CDMA channel, any special instruction packet to be transmitted by the hub will actually be sent in the same slot that is , in fact, reserved for a frame carrying a data packet. Therefore, for this frame carrying the data packet, and arrived at the hub from a VSAT remote terminal, it must wait a random time [30] T_v called the vacation time for its slot in the STDM superframe. T_v is a ramdom variable uniformly distributed in the interval $[0, T_{HV}]$ where T_{HV} is the TDM frame length. So the average vacation time $\overline{T}_v$ is just $T_{HV}/2$ and its mean square value is $\overline{T_v^2} = T_{HV}^2/3$. Note that T_v should be added in equation [3.3.5] if we consider the TDM outbound channel to be synchronous time division multiplexed.

The random retransmission delay by a VSAT was given in the last section for both the pure and slotted ALOHA techniques, and the average queueing delay for the ASTDM node was derived in a previous section and is given by

$$\overline{T}_q = \frac{\lambda T_{HV}^2}{2(T_{VH} - \lambda T_{HV})} \qquad [3.3.13]$$

if we take λ to denote the overall packet arrival rate expressed in packets/packet transmission time.

For a given packet being transmitted from a VSAT remote terminal to the hub, there exists four different states where this packet has one or more transitions before it is considered successfully received at the receiving station. We distinguish the following transition probabilities P_{ij} where i = 1 or 0 if the forward channel is error-free or not and, likewise, j = 1 or 0 denoting whether or not the acknowledgment in the feedback channel is correctly received at the transmitting node. The inbound channel can be modeled as an M/G/1 queueing system where packets arrive at a rate of λ packets/sec to the system and spend some time for service before they are successfully received at the hub. Upon depature, the packet is considered correctly received at the hub and is ready for transmission by the hub if the packet is originally destined to another VSAT in the network (double-hop communication). Thus, the packet state transition probabilities are defined as follows. For example, P_{00} is the probability that a data packet was not correctly received at the hub and its acknowledgment was not successfully transmitted by the receiving hub either (after it is correctly retransmitted). This and the rest of the transition probabilities are given by

$$P_{00} = q_u q_d \qquad [3.3.14]$$

$$P_{01} = q_u p_d \qquad [3.3.15]$$

$$P_{10} = p_u q_d \qquad [3.3.16]$$

$$P_{11} = p_u p_d \qquad [3.3.17]$$

3.3.1 Derivation of the pdf of the Total Service Time

To calculate the probability density function (pdf), $f_T(t)$, of the total service time of a packet transmitted to the hub on the ALOHA DS-CDMA channel, we need to identify the service time given that we have a successful transmission of the data packet from a VSAT to the hub and that the packet carrying the corresponding ACK packet is correctly received by the transmitting VSAT remote terminal. We know that the probability of this event is exactly determined by the bit error probabilities of both the inbound and outbound channels and is accurately given by

$$p = p_u p_d \qquad [3.4.1]$$

and the service time corresponding to that probability is simply

$$T_s = E[TD/X = 1, Y = 1] \qquad [3.4.2]$$

Therefore, a retransmission could happen with probability

$$q = (1 - p) \qquad [3.4.3]$$

Associated with this retransmission probability is the timeout. T_{out} is used to avoid unnecessary (premature) retransmissions of a data packet. Since there is no NACK issued by the hub, this timeout period must be used for retransmission if the packet is received by the hub with error, or if the packet is correctly received by the hub but its corresponding ACK message is received by the hub with error. This timeout should always be chosen to ensure that the acknowledgment from the receiver arrives at the transmitter under error-free conditions with a probability very close to one. T_{out} must at least be greater or equal to twice the propagation delay ($2T_p$) , plus one packet transmission time T_{VH} , plus a random retransmission delay by the VSAT (in sec), plus queueing delay and the processing delays at either end of the Data Link Control module, and added to all that is T_{HV}/N to account for the transmission of an ACK when sent to the transmitting VSAT via the feedback TDM channel. T_{out} is now repeated here for clarity

$$T_{out} = 2T_p + T_{proc} + T'_{proc} + T_{VH} + T_q + T_{HV}/N + T_{VH}\overline{T}_D \qquad [3.4.4]$$

$\overline{T}_D$ is just the random component of $\overline{RD}$; namely $\overline{T}_D = (k + 1)/2$packet-unit times. After this timeout, if the transmitter has not received an ACK, it assumes that the packet has not been received correctly and it schedules its retransmission in the next available time. In the case, such as ours, when the stop-and-wait ARQ is used as the error control strategy, this procedure is repeated until an ACK is received before a timeout and the VSAT then transmits the next packet.

Let $f_{T_s}(t)$ and $f_{T_{out}}(t)$ denote the probability density functions of T_s and T_{out} respectively. If a packet is acknowledged after n transmissions (which will occur with probability $(1 - q)q^{n-1}$), the total time required for transmission if the original packet is transmitted (n-1) times due to timeouts is given by :

$$T = T_s + (n - 1)T_{out} \qquad [3.4.5]$$

Assuming that the times for successive retransmissions are independent. Then, we have

$$f_T(t) = \sum_{n=1}^{\infty}(1 - q)q^{n-1}f_{T_{out}}^{(n-1)*} * f_{T_s} \qquad [3.4.6]$$

where $f_{T_{out}}^{(n-1)*}$ is the (n-1)-fold convolution of $f_{T_{out}}$ with itself, and * is simply a convolution operator.

Now let $F_T(s)$, $F_{T_{out}}(s)$, and $F_{T_s}(s)$ denote the Laplace transforms of f_T, $f_{T_{out}}$, and f_{T_s} respectively. Then,

$$\begin{aligned} F_T(s) &= \sum_{n=1}^{\infty}(1-q)q^{n-1}F_{T_{out}}^{n-1}.F_{T_s} \\ &= (1-q)F_{T_s}(s)\sum_{n=1}^{\infty}q^{n-1}F_{T_{out}}^{n-1}(s) \\ &= \frac{(1-q)\ F_{T_s}(s)}{(1-q\,F_{T_{out}}(s))} \end{aligned} \qquad [3.4.7]$$

Now that we have the moment generating function $F_T(s)$, the expected value of the total service time of a packet transmitted by a VSAT remote terminal to the hub is evaluated as follows

$$\overline{T} = -\frac{d}{dt}(F_T(s))|_{s=0} \qquad [3.4.8]$$

and its variance

$$\sigma_T^2 = \overline{T^2} - (\overline{T})^2 \qquad [3.4.9]$$

where $\overline{T^2}$ is the mean square value of T and is given by

$$\overline{T^2} = \frac{d^2}{dt^2}(F_T(s))|_{s=0} \qquad [3.4.10]$$

Taking all the derivatives, we get the following formula for the expected value of the packet service time

$$\overline{T} = \overline{T}_s + \frac{q\overline{T}_{out}}{1-q} \qquad [3.4.11]$$

and its mean square value is given by

$$\overline{T^2} = \overline{T_s^2} + \frac{q\overline{T_{out}^2}}{1 - q} + \frac{2q\overline{T}_{out}}{(1 - q)} [\frac{q\overline{T}_{out}}{1 - q} + \overline{T}_s] \qquad [3.4.12]$$

From [25] the mean square value of the queueing delay is given by

$$\overline{T_q^2} = 2(\overline{T}_q)^2 + \frac{\lambda T_{HV}^3}{3(T_{VH} + \lambda T_{HV})} \qquad [3.4.13]$$

so

$$\begin{aligned} \overline{T}_s &= 2T_p + T_{VH} + T'_{proc} + T_{proc} + \frac{T_{HV}}{N} + \overline{T}_q \\ &= T_T + \overline{T}_q \end{aligned} \qquad [3.4.14]$$

and

$$\overline{T_s^2} = T_T^2 + 2T_T\overline{T}_q + \overline{T_q^2} \qquad [3.4.15]$$

also

$$\begin{aligned} \overline{T}_{out} &= T_T + \overline{T}_q + \overline{T}_v + T_{VH}\overline{T}_D \\ &= \overline{T}_s + T_{VH}\overline{T}_D \end{aligned} \qquad [3.4.16]$$

and thus $\overline{T_{out}^2}$ is expressed as follows :

$$\overline{T_{out}^2} = \overline{T_s^2} + T_{VH}^2\overline{T_D^2} + 2T_{VH}\overline{T}_s\overline{T}_D$$

where the first and second monents of the random retransmission delay T_D respectively are $\overline{T}_D = (k + 1)/2$ and $\overline{T_D^2} = \frac{(k + 1)^2}{12}$ for a random retransmission delay that is uniformly distributed between [1,k].

The retransmission probability q is $q = (1 - p_u p_d)$ for $p_u = (1 - P_b)^{R_b T_{VH}}$ and $p_d = (1 - P'_b)^{R'_b T_{HV/N}}$

Further, if due to exessive errors in the ALOHA DS-CDMA channel some of the packets have to be retransmitted, and assume the retransmission traffic can be approximated as a Poisson process with parameter $\lambda_r = k/\overline{T}_D$ where $1/\overline{T}_D$ is the retransmission probability and k is the number of backlogged VSAT remote terminals, then the throughput of the ALOHA DS-CDMA/TDM channel is $S = Gp$ for a channel traffic rate $G = (\lambda + \lambda_r)$ and p is the probability of success of a packet transmitted from a VSAT to the hub through both noisy forward (DS-CDMA) and feedback (ASTDM) channels. This probability is given in equation [3.4.1].

3.4 *Study of the TDM/ALOHA DS-CDMA Channel*

The analysis in this section is very much similar to the one treated in the previous section.In fact, if we interchange the two events E and F and denote E to be the event where a data packet is transmitted from the hub to a VSAT remote terminal, the event F will be the one where an ACK packet is being transmitted to the hub via the ALOHA DS-CDMA channel. Then, the service time corresponding to a successful transmission with probability p is $T_s = Pr[X = 1/Y = 1]$

$$T_s = 2T_p + \frac{T_{HV}}{N} + T'_{proc} + T_{proc} + T_q + T_{ack} \qquad [3.5.1]$$

and $p = p_u p_d$ where $p_d = (1 - P'_b)^{R'_b T_{HV/N}}$ and $p_u = (1 - P_b)^{R_b T_{ack}}$

In this case, the timeout associated with the retransmission probability $q = (1 - p)$, is T_{out}. It should be long enough to include only the round trip propagation delay, the processing delays, the transmission time of a packet from the hub to the VSAT, the queuing delay and the acknowledgment transmission time in the feedback DS/CDMA channel. So

$$T_{out} = 2T_p + T_{proc} + T'_{proc} + \frac{T_{HV}}{N} + T_q + T_{ack} + T_{ack}\overline{T}_D \qquad [3.5.2]$$

Using equation [3.4.7] one can readily find $\overline{T}$, the total service time of a packet transmitted by the hub to a VSAT.

3.5 Numerical Results

Turning our attention to the discussion of numerical results, figure 12 represents curves showing the packet success probability as a function of the number of interferers to which a packet is subjected in the DS-CDMA channel for several values of the information bit energy to noise power density ratio (E_b/N_0. The packet length is fixed to 256 bits and the processing gain is held at 15. From the curve, it is seen that performance degradation is gradual as the number of interferers or (simultaneous users) is increased. However, when the number of interferers is greater than about 9 or 10, the packet will have very little chance to be transmitted successfully. As we decrease $2E_b/N_0$, the curve starts to roll off more quickly at an average of 4 interferers and for an even smaller value of $2E_b/N_0$, the bit in a packet will not have enough energy to survive and be correctly transmitted in this 'hostile' environment of having more users simultaneously accessing the channel. Figures 13 and 14 show different curves of bit error probability and packet success probability as a function of $2E_b/N_0$ for different numbers of simultaneous users. As we increase the number of interferers the bit error probability will start at a higher value for small $2E_b/N_0$ and then falls off gradually after a value of about 9 db. Only for a bit energy to noise power density of approximately 25 db that the bit error probability becomes relatively small, and at about the same point the chances for a successful packet transmission become quite high if we had less than 5 interferers transmitting at the same time.

In figure 15, we plotted the throughput of an uncoded slotted ALOHA and slotted ALOHA DS-CDMA versus offered traffic for sveral values of $2E_b/N_0$. It is observed that the DS-CDMA multiaccess scheme improves channel throughput with respect to slotted ALOHA. In fact, from the curves we notice that for small offered traffic, the troughput β is almost equal to G. As the $2E_b/N_0$ increases, this ideal linear region is extented. For example, for $2E_b/N_0 = 60db$ the linear region extends to about G = 2.5. However, decreasing $2E_b/N_0$ to 40 db the linear region extends to only G = 2.0, and for even smaller values this region will shrink in length until it is non-existing for a value of $E_b/N_0 = 0.0db$. After this initial linear region, all the throughput curves reach a maximum for higer values of G and then start decreasing. Thus, it is clear that random access DS-CDMA systems also exhibit the type of degradation which has been demonstrated for the ALOHA systems. The difference is that when the information bit energy to noise density ratio increases, the shape of the throughput curve changes and the maximum capacity increases. For example, at an $2E_b/N_0 = 60db$ slotted ALOHA DS-CDMA has a capacity of about 3.1 packets /slot, while for $2E_b/N_0 = 40db$ $\beta_{1\,max} \cong 2.9$ packets/slot.

In figures 16 and 17, we plotted the normalized throughput (to a unit bandwith) of uncoded slotted ALOHA and of the ALOHA DS-CDMA with and without ACK traffic as a function of the offered traffic. From the curves we see that for small traffic (to about

G = 2.0), the effect of positive acknowledgment when it is not sent on a separate channel is unnoticible. However, the throughput of the channel carrying the ACK along with the data is about 50% to 30% less than that without the ACK traffic at higher offered traffic. So for random access CDMA networks operating at high traffic loads, the idea of incorporating the ACK traffic on the same channel will significantly drop the channel's throughput. The effect of modelling the number of successful data packets as a finite Markov chain is noticed when k_{max} approaches the chosen number of VSAT remote terminals N. Curves are given for both k_{max} = 5 and 20. We get a better approximation of the throughput at a higer value of k_{max}, that is when we choose a value big enough to approximate infinity. It was also noted that no improvement occurs beyond k_{max} = 20 once we put the restriction that any entry in $[P_{ij}]$ less than 1.E-10 is rounded off to zero.

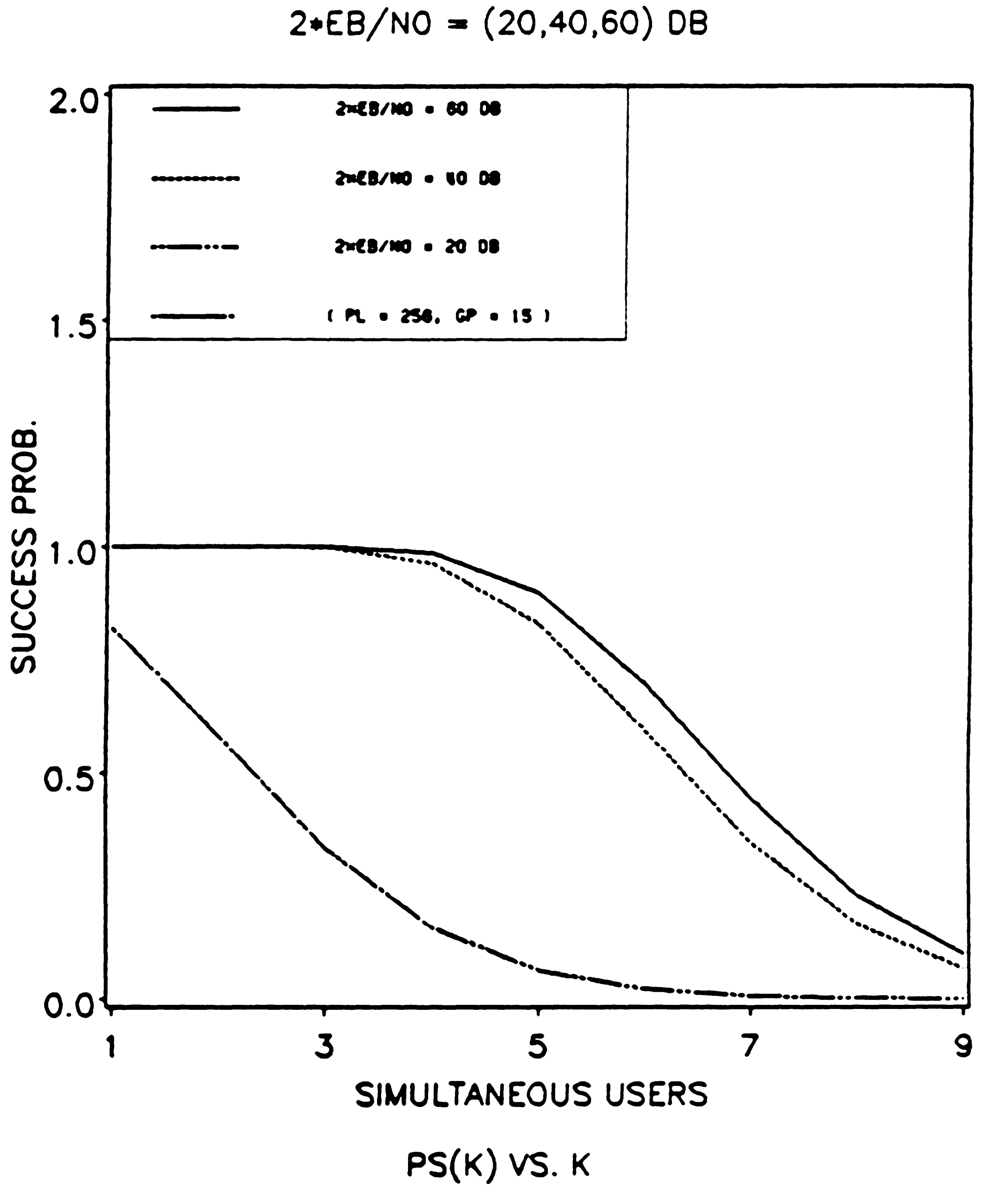

Figure 12. Packet Success Probability versus Number of interferers

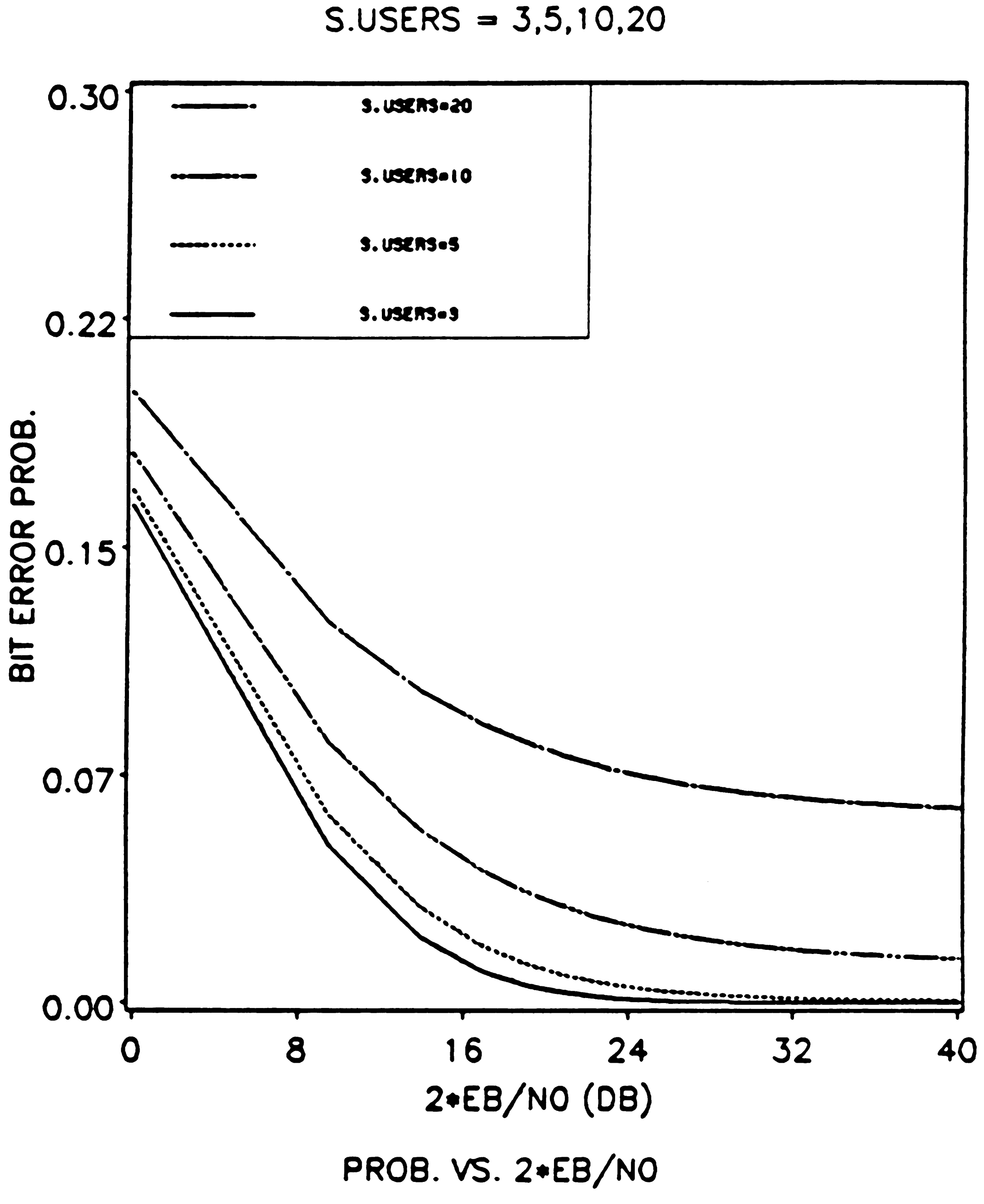

Figure 13. Bit Error Probability versus Bit Energy to Noise Power Density

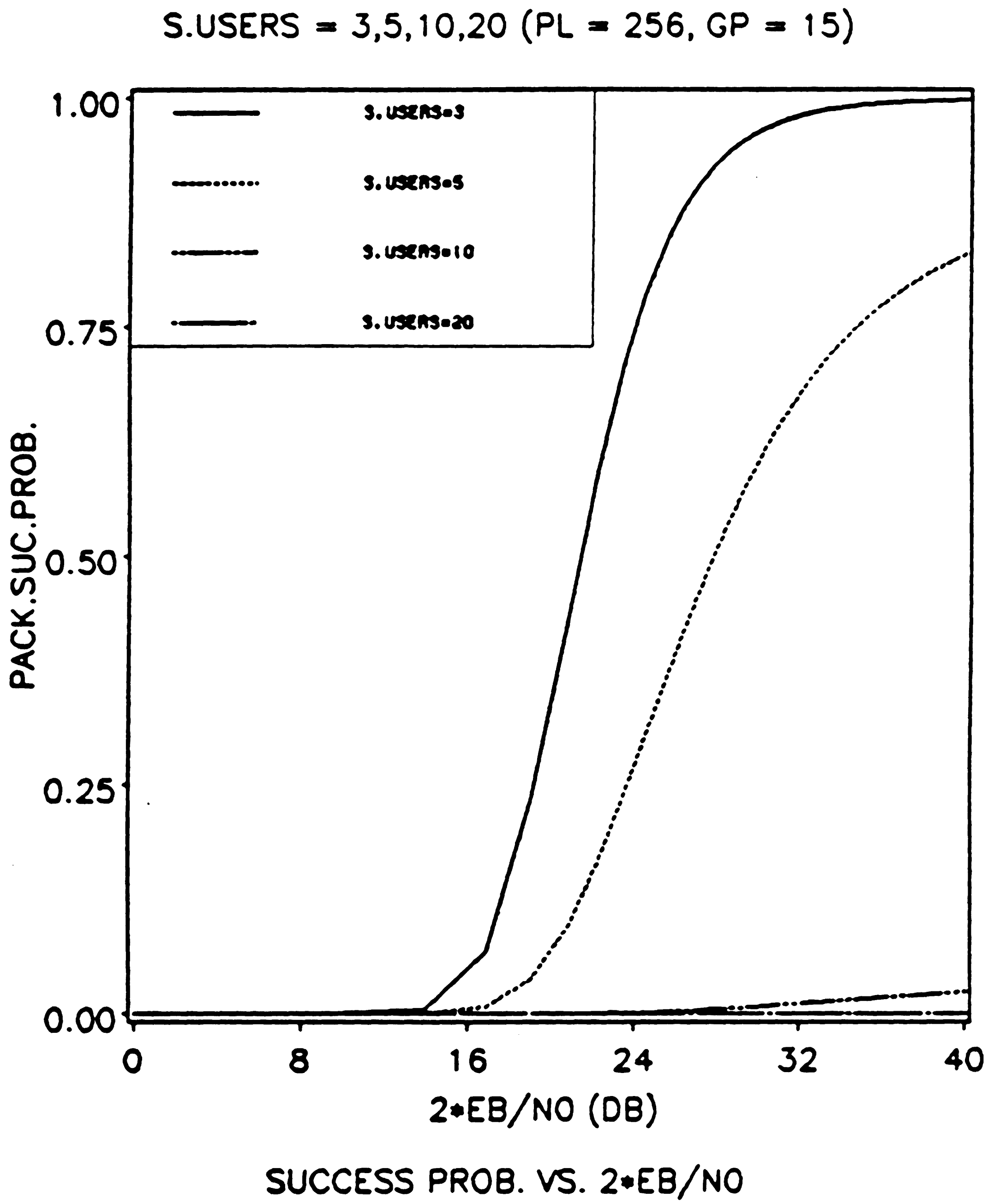

Figure 14. Packet Success Prob. versus Bit Energy to Noise Power Density

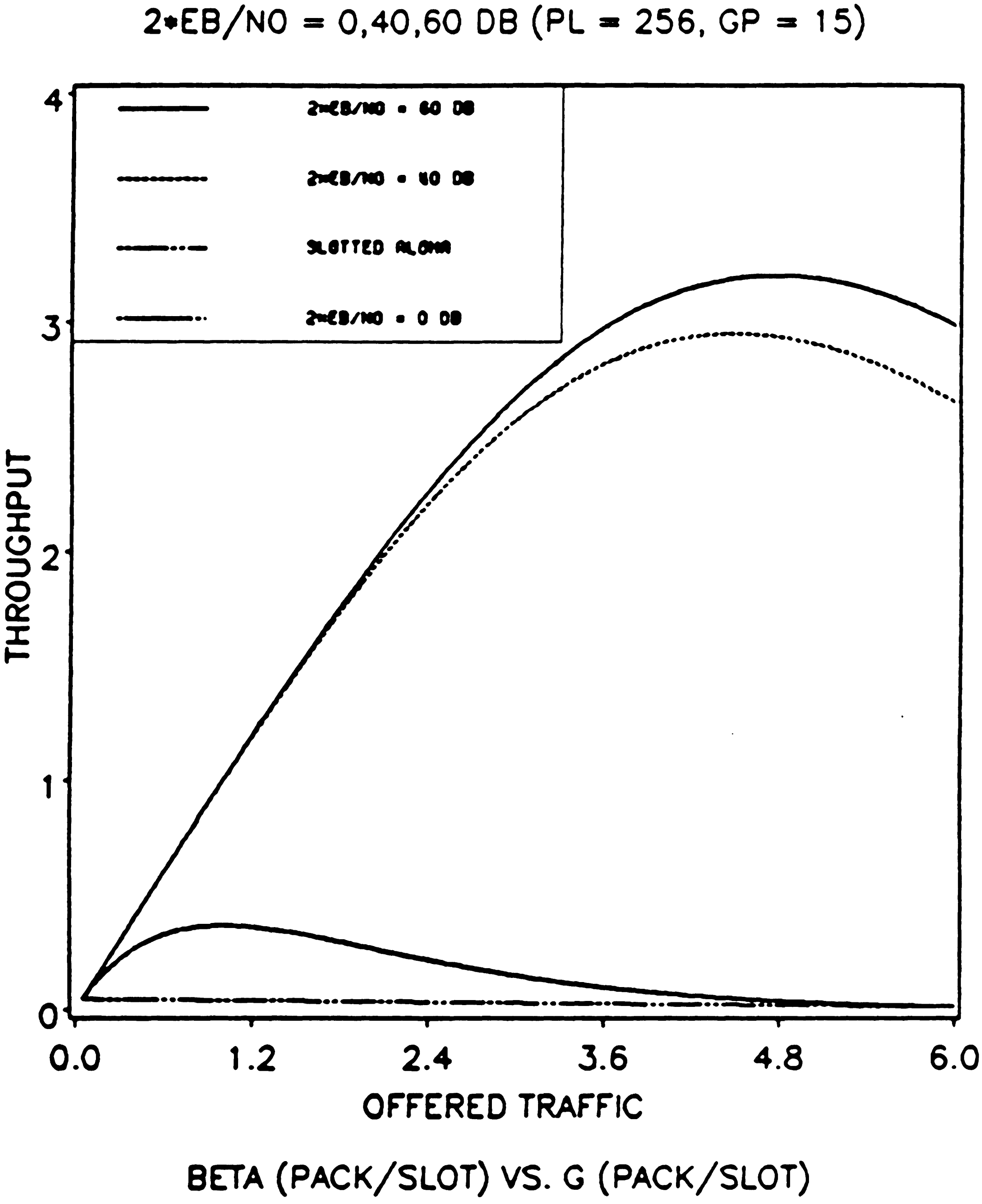

Figure 15. Throughput versus Offered Traffic

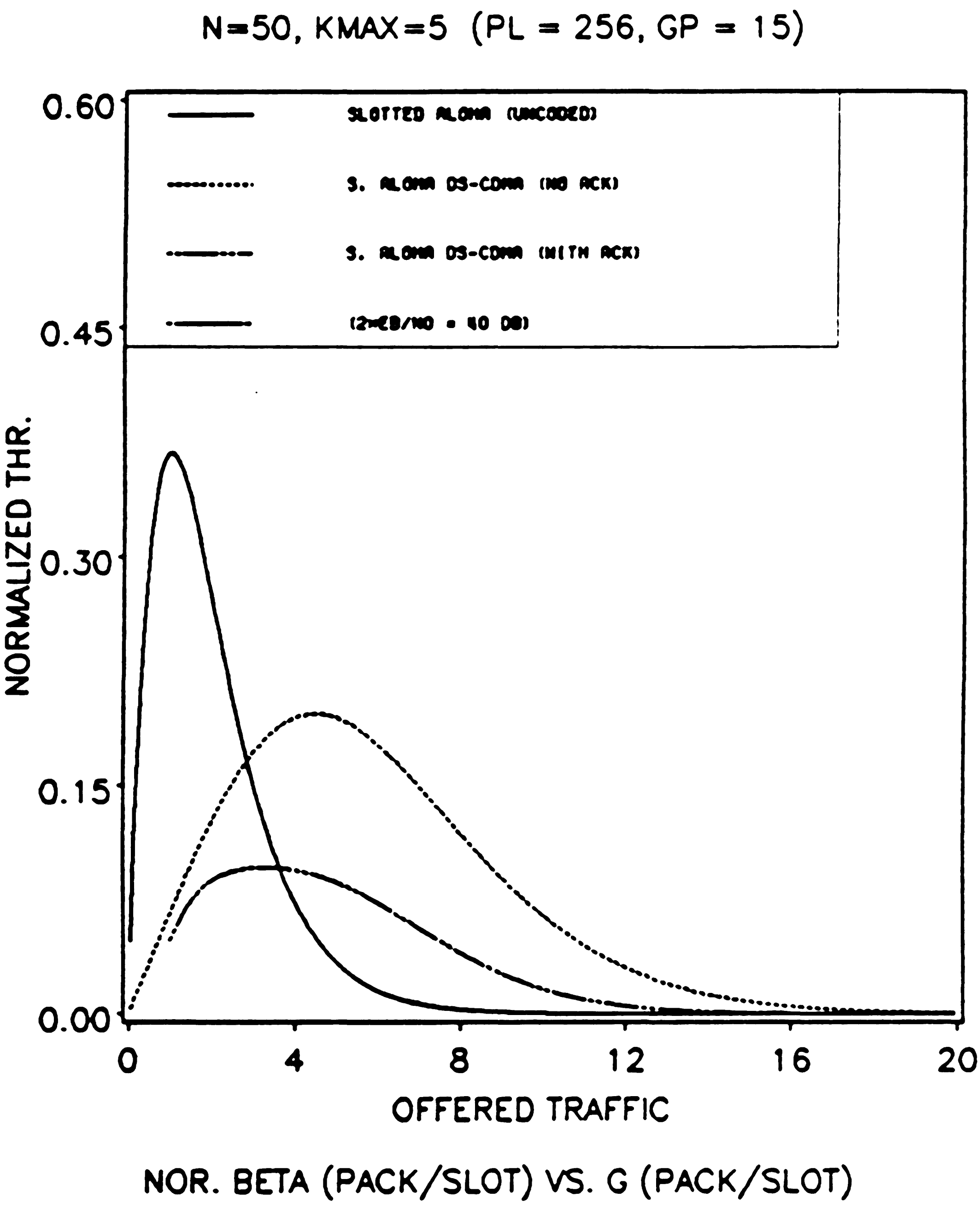

Figure 16. Normalized Throughput versus Offered Traffic (kmax = 5)

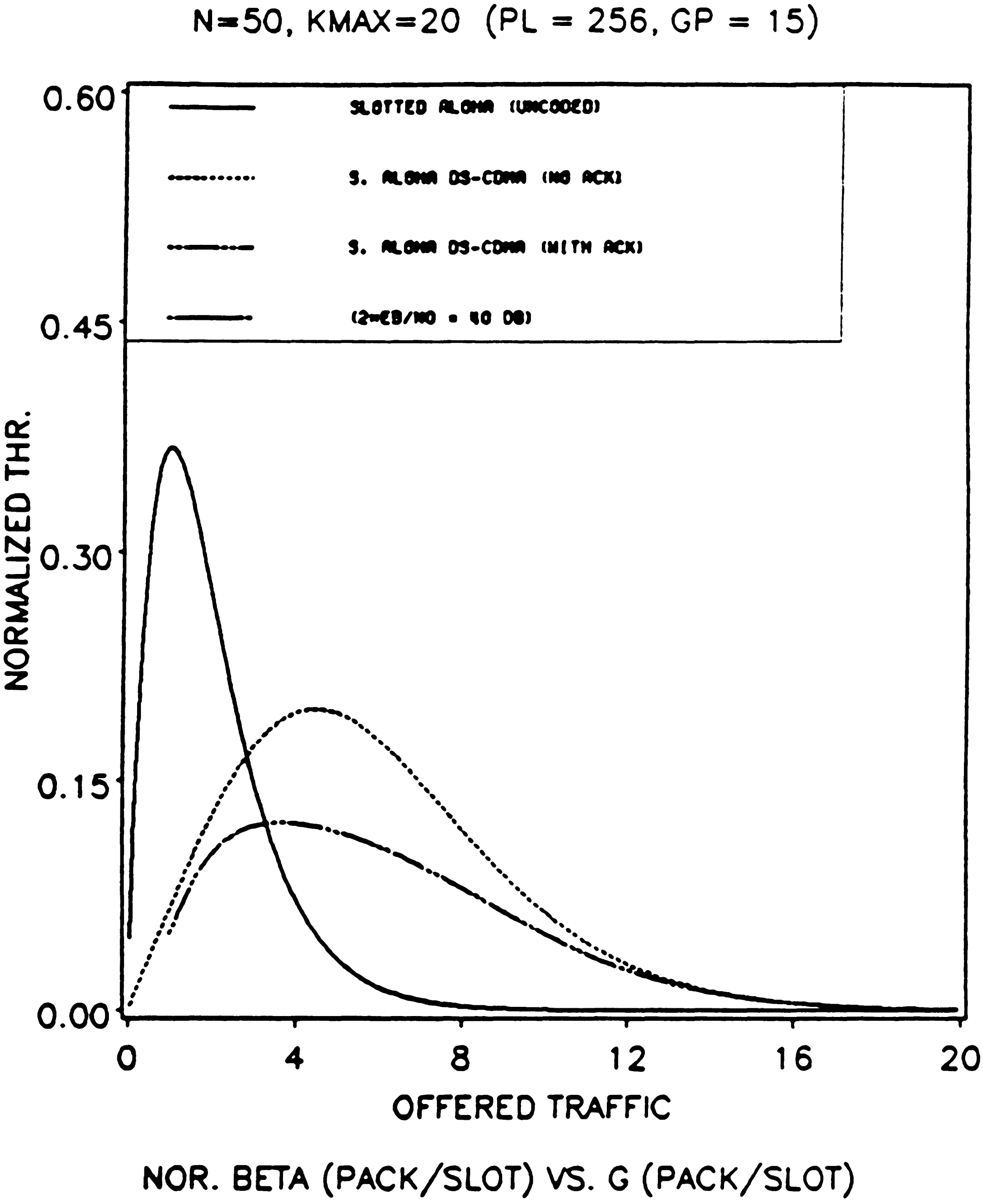

Figure 17. Normalized Throughput versus Offered Traffic (kmax = 20)

4.0 CONCLUSION

The primary contribution of this work is that it provides a mathematical formulation for directly computing the throughput of an unslotted asynchronous CDMA communication system where we have specified the link bit error probability function. This idea provides a basis for assessing the performance improvement which will be gained by using code division in place of the conventional time division as the multiple access technique for the network. Figures 15,16,and 17 show the performance of such a multiple access technique.

It is true that Time Division Multiple Access (TDMA) satellite communication offers efficient transmission of a wide variety of services, but it, unfortunately, suffers from network timing and message security. One attractive feature of CDMA is that it does not require the network synchronization and that it can be (and usually is) operated in an asynchronous manner. Code division of baseband signals using PN sequences is the method adopted to spread the spectrum for the communication of VSAT's with the hub station. It is used to achieve reliable random access transmission system. However, the dominant reason for considering CDMA is the need, in addition, for some type of external interference rejection capability, and because of its advantage that it is relatively easy to add additional users to the network by just defining a code for the user to be added. Furthermore, in this satellite communication system, any number of VSAT's may transmit signals to the hub through the satellite (the multiaccess inbound channel) and also any signal received by the satellite transponder from the HUB is beamed back to the earth stations, VSAT's, (the broadcast outbound channel). This broadcast signal is received by all the VSAT's covered by the transponder beam using time division multiplexing. Queueing theory is applied to compute the buffer size and average packet delay. A model is developed using the stop-and-wait error detection and retransmission scheme to control possible errors experienced by the transmission of a packet. This model is applied to the case where errors are independent and can occur during data and/or acknowledgement transmission with probability p and p'. The method of computation rested on analytical study. However, the results could be used as a design guide if the mean packet loss rate due to buffer overflow is specified. Finally, we looked at the interaction of both the TDM and ALOHA DS-CDMA channels as one multiple access procedure star network as a whole, and an explicit formula was proposed to obtain the average service time of a packet when transmitted along any one of the links.

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