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Simulation Based Analysis of Random Access CDMA Networks

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Abstract

As large mobile networks have become commonplace, the allocation of resources has become critical. The most important and restricted resource is the system bandwidth for this kind of networks. In order to use the system bandwidth efficiently, multiple access systems are used. For multiple access systems, MAC (medium access control) protocols have primary effect in the throughput and delay performance of the system. Among MAC protocols, the ALOHA scheme has enjoyed the advantage of simplicity, however, the throughput of ALOHA decreases under heavy traffic conditions. Alternative random access systems have been proposed based on code division multiple access (CDMA) systems to improve the throughput [1,2]. Random access CDMA systems have been shown to efficiently allocate scarce radio communication channels (bandwidth) in such a way that users with bursty traffic can share the same frequency without significant degradation to the overall throughput. However, large system simulation of random access CDMA networks with unslotted implementation has been difficult. Therefore, in this paper, we develop a simulation model in OPNET to analyze random access CDMA system for different channel models. Such simulation can provide means to observe and report the system specifications and performance of the large CDMA networks under different circumstances.

Keywords: CDMA, fading channels, MAC protocols, OPNET, random access networks

1. Introduction

In recent years, there has been an increased demand for mobile data communications. To handle the bursty nature of data traffic and to efficiently allocate the resources among the users, packet-based multiple access protocols must be used. But most random access networks such as ALOHA typically suffer from collisions. However, if CDMA based MAC protocols are used some of the collided packets can be extracted correctly. So CDMA random access systems have drawn much attention and much literature has been devoted to improve system performance.

Most of the analyses of random access CDMA systems are based on slotted systems or circuit switched systems. In a slotted system transmission time is divided into slots, which consist of a packet interval time and a guard time. All users must synchronize their transmission to the beginning of the slot. The performance analysis of the slotted system is easy and the system performance only depends on the number of interfering packets (or users) within a slot.

Unslotted systems are easy to implement because they do not require synchronization. However, their performance analysis is very difficult since the number of interfering users fluctuates during the packet interval. Most of the performance analysis of unslotted ALOHA depends on the perfect capture while the number of interference is assumed to be constant. [2].

The primary objective of this paper is to develop a simulation model in OPNET to analyze random access CDMA system for different channel models. OPNET is the industry's leading environment for network modeling and simulation, allowing us to design and study communication networks, devices, protocols, and applications with flexibility. OPNET's object-oriented modeling approach and graphical editors mirror the structure of actual networks and network components. Moreover, OPNET supports all network types and technologies.

The rest of this paper is organized as follows. Next section describes different channel models. Section III provides an overview of forward error correction. In section IV, we describe CDMA system model and CDMA probability of error expressions. In section V, we provide on overview of random access network models. In section VI, we explain our simulation model and attributes. We present and evaluate simulation results in section VII. Concluding remarks are offered in section VIII.

2. Channel Models

We start by considering transmission of binary phase shift keying (BPSK) signals over additive white Gaussian noise (AWGN) channels. The BPSK signal is one dimensional; therefore, their geometric representation is simply the one dimensional vector $s_1 = \sqrt{E_b}, s_2 = -\sqrt{E_b}$ where Eb is the energy of the transmitted signal. Assume that s1 is transmitted then the received signal is

$$r = s_1 + n$$

where n is the additive white Gaussian noise (AWGN) component with zero mean and variance σ2=N0/2. Assuming coherent reception, the conditional probability distribution function of r is given by

$$p(r \mid s_1) = \frac{1}{\sqrt{\pi N_0}} \exp\left(-\frac{(r-\sqrt{E_b})^2}{N_0} \right).$$

Given s1 is transmitted the probability of error is simply the probability that r<0, i.e.

$$\begin{aligned} P(e \mid s_1) &= \int_{-\infty}^{0} p(r \mid s_1) dr \\ &= \frac{1}{\sqrt{\pi N_0}} \int_{-\infty}^{0} \exp\left(-\frac{(r-\sqrt{E_b})^2}{N_0} \right) dr \\ &= Q\left(\sqrt{\frac{2E_b}{N_0}} \right) \\ &= Q\left(\sqrt{2SNR} \right) \end{aligned}$$

where SNR=Eb/N0 is the signal to noise ratio. Assuming that the transmitted symbols are equally likely, the probability of bit error for binary PSK is given by

$$P_b = Q\left(\sqrt{2SNR} \right) \tag{1}$$

The wireless communications channels can be characterized by multipath fading and shadowing as depicted in multipath fading results from signal scattering, which may occur for many reasons, such as reflections off buildings, trees, hills, and other objects and from satellite motion and atmospheric effects. Shadowing is the result of blockage from buildings, trees, and other factors. Typically two models are used to characterize the multipath fading. Rayleigh fading model is used if there is no strong line-of-sight (LOS) component. In the presence of a strong LOS component, the Ricean fading model is used.

For fading channels, we assume that there is also a multiplicative noise in addition to AWGN. Assuming BPSK transmission, the received signal is

$$r = \alpha s_1 + n$$

The instantaneous SNR is given by

$$\text{SNR} = \gamma = \frac{E_b}{N_0}\alpha^2 .$$

The average bit error rate can found by averaging over the distribution of the channel as

$$\overline{P_b} = \int_0^\infty P_b(\gamma)\, f_\gamma(\gamma) d\gamma \tag{2}$$

On the other hand, shadowing can be modeled as a log-normal fading. The effects of shadowing on the transmitted signal are typically longer-term than those of multipath fading. Intuitively, trees or buildings can obstruct communications over several seconds. Thus, slow variations in the signal level are assumed to be Gaussian distributed in dB's. Shadowing is typically modeled by log-normal distribution. A random variable is log-normally distributed if its logarithm is normally distributed. Typically, the log-normal distribution is defined based on natural logarithm. However, the log-normal model for mobile communications systems is based on dB scale. Thus, we need to make a base change in order to derive the mean (in linear scale) of the log-normal distribution defined in dB scale.

Let X1 be a random variable such that ln(X1) with N(μ1, σ12) is a normally distributed random variable with mean μ1 and variance σ12.

Then the probability distribution function (PDF) of X1 is given by

$$f_{X_1}(x) = \begin{cases} \dfrac{\exp\left(-\dfrac{1}{2\sigma_1^2}\left(\ln(x)-\mu_1\right)^2\right)}{x\sigma_1\sqrt{2\pi}} & x>0 \\ 0 & otherwise \end{cases}$$

The expected value of X1 can be obtained as follows:

$$E[X_1] = \int_0^\infty x f_{X_1}(x)dx$$

$$= \int_0^\infty x \frac{\exp\left(-\dfrac{1}{2\sigma_1^2}\left(\ln(x)-\mu_1\right)^2\right)}{x\sigma\sqrt{2\pi}} dx$$

$$= \int_0^\infty \frac{\exp\left(-\dfrac{1}{2\sigma_1^2}\left(\ln(x)\right)^2\right)\exp\left(\dfrac{\ln(x)}{\sigma_1^2}\mu_1\right)\exp\left(-\dfrac{\mu_1^2}{2\sigma_1^2}\right)}{\sigma_1\sqrt{2\pi}} dx$$

$$= \exp\left(\frac{\sigma_1^2}{2}+\mu_1\right)$$

On the other hand let X2 be a random variable such that log10(X2) with N(μ2, σ22) is a normally distributed random variable with mean μ2 and variance σ22. Then the probability distribution function (PDF) of X2 is given by

$$f_{X_2}(x) = \begin{cases} \dfrac{\exp\left(-\dfrac{1}{2\sigma_2^2}\left(\log_{10}(x)-\mu_2\right)^2\right)}{x\ln(10)\sigma_2\sqrt{2\pi}} & x>0 \\ 0 & otherwise \end{cases} \quad (3)$$

The expected value of X2 can be written as

$$E[X_2] = \int_0^\infty x f_{X_2}(x)dx$$

$$= \int_0^\infty x \frac{\exp\left(-\frac{1}{2\sigma_2^2}(\log_{10}(x)-\mu_2)^2\right)}{x\ln(10)\sigma_2\sqrt{2\pi}}dx$$

$$= \int_0^\infty \frac{\exp\left(-\frac{(\ln(x))^2}{2\sigma_2^2(\ln(10))^2}\right)\exp\left(\frac{\ln(x)\mu_2}{\sigma_2^2\ln(10)}\right)\exp\left(-\frac{\mu_2^2}{2\sigma_2^2}\right)}{\sigma_2\ln(10)\sqrt{2\pi}}dx$$

In writing the third equation we use the following property

$$\log_{10}(x) = \frac{\ln(x)}{\ln(10)}$$

Expressing σ2ln(10)= σ1 and μ1=μ2ln(10), we obtain the expectation

$$E[X_2] = \exp\left(\frac{(\sigma_2\ln(10))^2}{2} + \mu_2\ln(10)\right).$$

Now assume that X_f is a random variable such that $X_{f,\mathrm{dB}} = 10\log_{10}X_f$ is a normally distributed random variable with mean μ dB and variance σ2 dB. The expected value of X_f can be obtained as

$$E[X] = \exp\left(\frac{\left(\frac{\sigma\ln(10)}{10}\right)^2}{2} + \frac{\mu\ln(10)}{10}\right)$$

The log-normal fading model uses the random variable X_f to define the fading channel factor of the received signal in linear scale: $X_{f,\mathrm{dB}} = 10\log_{10}X_f$. In cellular mobile communications, it is typically assumed that the mean of $X_{f,\mathrm{dB}}$ zero. This means that the fading channel factor in dB will be evenly distributed around 0 dB which corresponds to 50% cumulatively. In Figure 1, BER rate for different channel models are simulated for BPSK signaling.

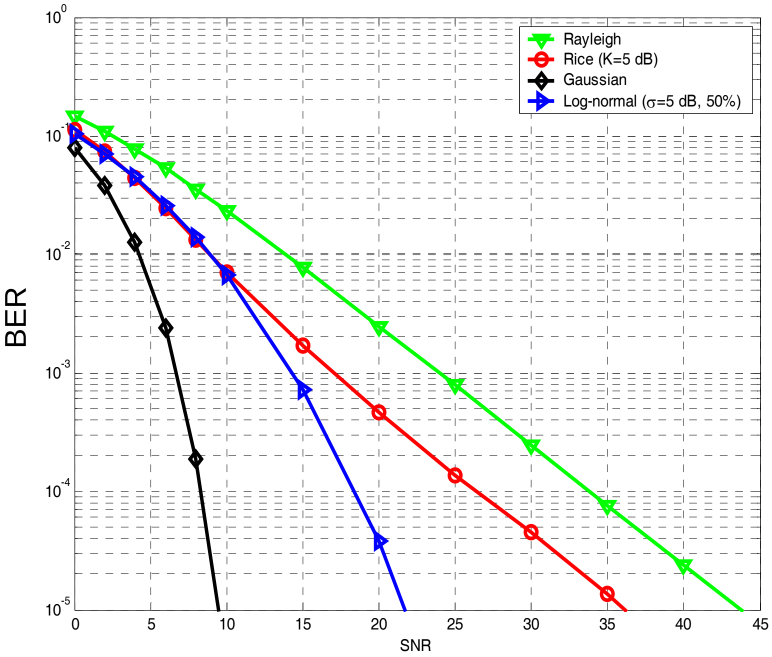

Figure 1. Bit error rates of different fading channels

3. Forward Error Correction (FEC)

The transmission of information over an air link always results in some degradation in the quality of the information. In digital links, we measure the degradation of the information content of a signal in terms of the BER. If desired, we can improve the quality of a digital signal by the use of error correction techniques. The systems that can detect and correct errors use forward error correction (FEC).

The FEC is a technique for adding redundant bits to a data stream in such a way that one or more errors in the data stream can be corrected. Our simulation model assumes convolutional FEC coding. Convolutional codes are generated by a tapped shift register and two or more modulo-2 adders wired in a feedback network.

In this section, the BER performance of binary transmission with convolutional coding over different channels is presented. To measure the BER performance, a MATLAB simulation script is written to encode the transmitted bits with a rate ½ and a constraint length K=9 convolutional encoder. The generator polynomials are $G_1 = 1 + x^2 + x^3 + x^4 + x^8$ and $G_2 = 1 + x + x^2 + x^3 + x^5 + x^7 + x^8$. Since the data rate (and transmission bandwidth) is doubled due to coding, the each output bit carriers half of the energy of the uncoded bits. The BER of uncoded transmission has an analytic solution as we have derived in section 2. However, we can generally obtain a union bound for BER of coded waveforms. Due to this difficulty, the BER simulation curves obtained here are used as modulation table in OPNET. Figure 2 shows the coded and uncoded BERs over various log-normal fading channels.

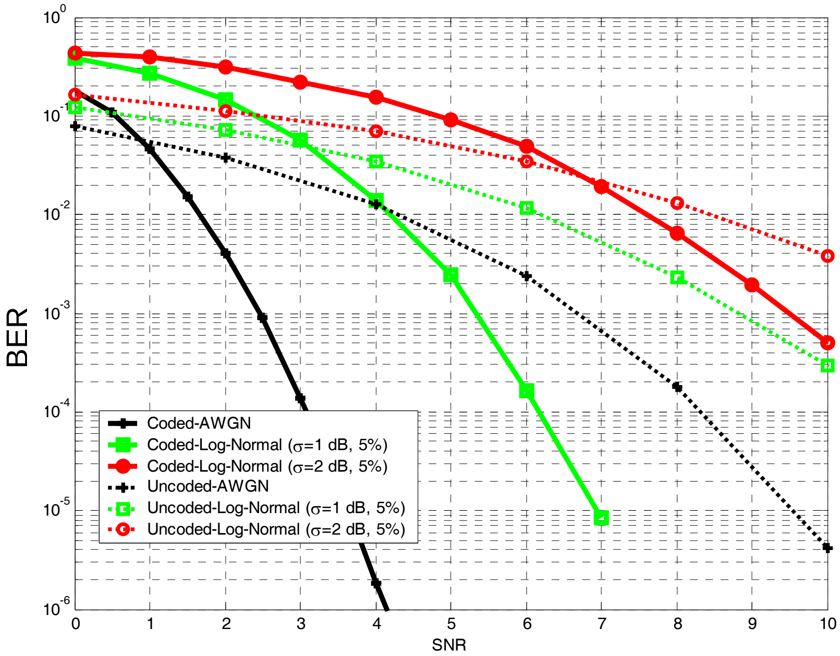

Figure 2. Bit error rate of coded log-normal fading channel for different log-normal fading conditions as compared to uncoded case

4. CDMA System Model

It was shown that multiple-access interference can be modeled as Gaussian noise [5]. We will use the probability of error formulas with the Gaussian interference over CDMA systems. First, we give a brief overview of the random access CDMA system model. In a CDMA system the received signal at the base station from the kth user is given by [3]

$$s_k(t-\tau_k) = \sqrt{2P_k}\, a_k(t-\tau_k) b_k(t-\tau_k) Cos(\omega_c t + \varphi_k)$$

where bk(t) is the data sequence for user k, ak(t) is the pseudo-noise (PN) spreading sequence for user k, τk is the delay of user k, Pk is the received power of user k, wc is the carrier frequency and φk is the carrier phase offset of user k. The data signal bk(t) is a sequence of unit amplitude, positive and negative, rectangular pulses and the bit period is Tb. Superimposed on the data signal is a much faster sequence of chips ak(t), composed also of unit amplitude, positive and negative, rectangular pulses and chip period is Tc. It is assumed that the bit period is an integer multiple of chip period such that Tb=NTc where N is the spreading gain.

The received signal containing the desired user, K-1 undesired users, and the noise is given by

$$r(t) = \sum_{k=0}^{K-1} s_k(t - \tau_k) + n(t)$$

where n(t) is the white Gaussian noise with two sided power spectral density N0/2. This signal along with the receiver is illustrated in Figure 3.

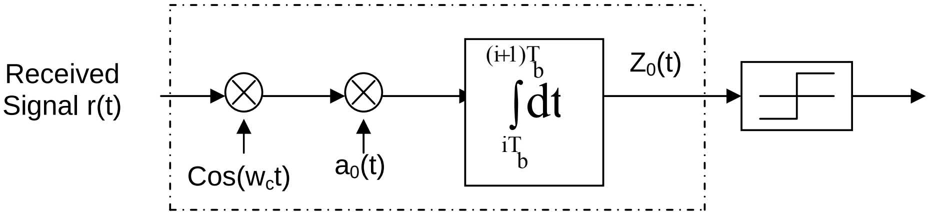

Figure 3. CDMA receiver model

The received signal is mixed down to baseband, multiplied by the PN sequence of the desired user and integrated over 1 bit period by the receiver. Assuming that the receiver is delay and phase synchronized with user 0, the decision statistics for jth bit of user 0 is given by

$$Z_o = \int_{jT_b}^{(j+1)T_b} r(t) a_0(t) \cos(w_c t) dt.$$

Using the Gaussian approximation (GA) described in [4], the average bit error probability is found to be

$$P_e = Q\left(\sqrt{\frac{1}{\frac{1}{3N}\sum_{k=1}^{K-1}\frac{P_k}{P_0}+\frac{N_0}{2T_bP_0}}}\right).$$

where Q(.) function can be defined in an integral form as

$$Q(v) = \frac{1}{\sqrt{2\pi}}\int_{v}^{\infty}\exp(-\frac{x^2}{2}).$$

Assuming all the users have equal power

$$P_e = Q\left(\sqrt{\frac{1}{\frac{K-1}{3N}+\frac{N_0}{2T_bP_0}}}\right). \qquad (4)$$

Assume that we express the delays and phases of the interfering signals as random vectors S = (S1, S2,…,SK) and Φ = (Φ1, Φ2, …, ΦK) where Sk is a uniform random variable taking values in the range [0 1] and Φk is a uniform random variable taking values in the range [0 2π]. Given these two random vectors the variance of the multiple access interference from the k'th user has been found to be [5]

$$E[Z_k \mid S_k, \Phi_k] = N\left(S_k^2 - S_k + 0.5\right)\left[1+\cos\left(2\Phi_k\right)\right]$$

where Zk is the variance of the multiple access interference from the k'th user. The total multiple access interference (MAI) variance is thus

$$\Psi = \sum_{k=2}^{K} Z_k .$$

If the interference from users is identical then the variance Ψ of the MAI is given by

$$\Psi = (K-1)Z_k$$

for some k.

The probability of data bit error is then approximated by

$$P_e = Q\left(\frac{N}{\sqrt{\Psi + \frac{N_0}{2E_b}N^2}}\right)$$

If the interfering signals are not chip and phase synchronous with the desired signal, then the average interference from kth user is found by substituting E[Sk2 – Sk]=-1/6 and E[cos(2Φk)] which produces

$$E[Z_k] = \frac{N}{3} \qquad and \qquad E[\Psi] = \frac{(K-1)N}{3}$$

The average bit error probability is then

$$P_e = Q\left(\frac{N}{\sqrt{\frac{(K-1)N}{3} + \frac{N_0}{2E_b}N^2}}\right) = Q\left(\sqrt{\frac{1}{\frac{(K-1)}{3N} + \frac{N_0}{2E_b}}}\right)$$

which is the standard Gaussian approximation (GA) result as stated in equation 3. If interfering signals are chip and phase aligned with the desired signal, then E[Zk | Sk=0, Φk=0]=N resulting in

$$P_e = Q\left(\frac{N}{\sqrt{(K-1)N + \frac{N_0}{2E_b}N^2}}\right) = Q\left(\sqrt{\frac{1}{\frac{(K-1)}{N} + \frac{N_0}{2E_b}}}\right)$$

which represents worst case scenario. Using similar reasoning, interfering signals that are chip aligned with random phases have

$$P_e = Q\left(\sqrt{\frac{1}{\frac{(K-1)}{2N} + \frac{N_0}{2E_b}}}\right)$$

and phase aligned interfering signals with random chip delays produce

$$P_e = Q\left(\sqrt{\frac{1}{\frac{2(K-1)}{3N}+\frac{N_0}{2E_b}}}\right).$$

Finally, in a general case, in which the received power from each user is unequal, the probability of bit error for the jth user is given by

$$P_e(j) = E\left[Q\left(\sqrt{\frac{P_j}{\frac{\sum_{k\neq j} P_k}{3N}+\frac{N_0}{2E_b}}}\right)\right] \tag{5}$$

Using the simplified improved Gaussian approximation (SIGA) [4], the average bit error probability is given by

$$P_e \approx \frac{2}{3}Q\left(\sqrt{\frac{N^2}{2(\mu_\psi+\frac{No}{2Eb}N^2)}}\right)+\frac{1}{6}Q\left(\sqrt{\frac{N^2}{2(\mu_\psi+\sqrt{3}\sigma_\psi+\frac{No}{2Eb}N^2)}}\right) \tag{6}$$

$$+\frac{1}{6}Q\left(\sqrt{\frac{N^2}{2(\mu_\psi-\sqrt{3}\sigma_\psi+\frac{No}{2Eb}N^2)}}\right)$$

When all users are assumed to have unit power, their mean and variance are given by

$$\mu_\varphi = \frac{N}{6}(K-1),$$

$$\sigma_{\psi}^{2} = \frac{K-1}{4}\left[\frac{23N^{2}}{360} + N\left(\frac{1}{20}+\frac{K-2}{36}\right) - \frac{1}{20} - \frac{K-2}{36}\right].$$

Figure 4 shows BER results obtained by the MATLAB simulations, GA and SIGA approximations (Equation 5 & Equation 6) by varying number of users, with spreading gain of N=31 in the absence of background noise. We see that the simulation results match with SIGA even with small number of users present in the system. As the number of users increases in the system GA approaches SIGA. Therefore, in the OPNET implementation uses the GA with unequal powers to compute SINR.

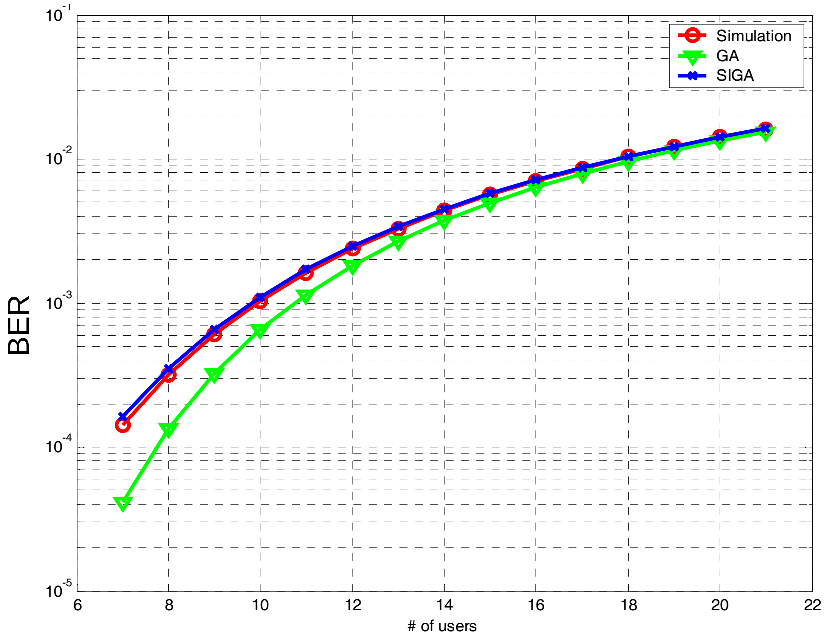

Figure 4. Verification of the proposed approximations in the absence of thermal noise

5. Random Access Network Models

Random access networks such as ALOHA typically suffer from collisions. Figure 5 shows such scenario with a narrowband ALOHA system. Random access packet networks can be implemented as unslotted or slotted system. In a slotted system transmission time is divided into slots, which consist of a packet interval time and a guard time. All users must synchronize their transmission to the beginning of the slot. The performance analysis of the slotted system is easy and the system performance only depends on the number of interfering packets (or users) within a slot.

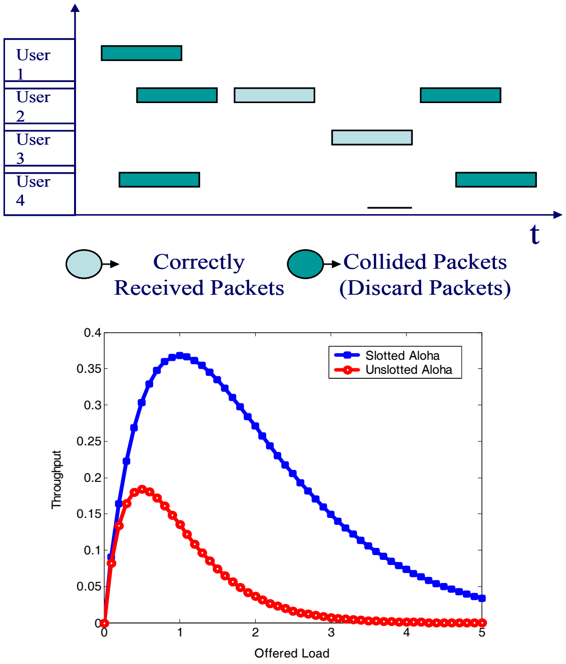

Figure 5. Collisions in a narrowband unslotted ALOHA system and throughput of slotted and unslotted ALOHA systems

Unslotted systems are easy to implement because it does not require synchronization. However, its performance analysis is very difficult since the number of interfering users fluctuates during the packet interval. Most of the performance analysis of unslotted ALOHA depends on the perfect capture while the number of interference is assumed to be constant.

Previously we have stated the variance of the multiple access interference (MAI) as a function of random delays and phases as

$$\Psi(s_2,\ldots s_K,\phi_2,\ldots,\phi_K)=\sum_{k=2}^{K}N\left(1-2s_k+2s_k^2\right)\cos^2\phi_k$$

where (1+cos(2Φk)) is replaced with 2cos2(Φk). To compute the average packet error rate, we need an estimate of the MAI. However, the MAI is random. Thus, we need to have its distribution. The distribution of the MAI has been derived in [5]. Using this distribution the average probability of bit error can be written as

$$P_e=E\left[Q\left(\frac{N}{\sqrt{\Psi}}\right)\right]=\int_0^{\infty}Q\left(\frac{N}{\sqrt{x}}\right)f_{\Psi}(x)dx \tag{7}$$

where $Q\left(\frac{N}{\sqrt{x}}\right)$ is the conditional probability of bit error given the random delays and random phases and $f_{\Psi}(x)$ is the distribution of the MAI (in power). If there are L bits in a packet the packet success means that all the bits in a packet were transmitted successfully. Note that by conditioning on the differential delays and phases between the desired and all interfering symbols there is conditional independence of data bit success from data bit to data bit within the desired packet. Therefore packet success rate, PS can be expressed as

$$P_S=\int_0^{\infty}\left(1-Q\left(\frac{N}{\sqrt{x}}\right)\right)^L f_{\Psi}(x)dx$$

This method accounts for the bit-to-bit error dependencies within the packet and called IGA-D technique in [5]. MATLAB simulations account for the bit-to-bit error dependence within the desired packet. However, it is nearly impossible to efficiently model the distribution $f_{\Psi}(x)$ in OPNET. Therefore, [5] suggests using the approximate technique to derive the probability of the packet success. If the effect of bit-to-bit error dependence is ignored as suggested in [5], an approximation of the packet success rate can be written as a function of bit error rate P_e, i.e.,

$$\hat{P}_S = (1 - P_e)^L .$$

According to [6], this approximation will provide a lower bound to throughput of the system.

6. OPNET Implementation

Here we describe the simulation model for the random access CDMA networks in OPNET. Our model consists of two parts: designing specific OPNET transceiver pipeline stages and the specific receiver and transmitter nodes. Our transmitter nodes generate packets with random (i.e., exponential) inter-arrival times. As the receiver receives the packets, the throughout statistics is collected in the process module to be reported in the end of simulation. We have modified a number of pipeline stages in order to calculate BER rate and the total power of interfering users during a packet interval. A simple network is configured in OPNET to test the modified pipeline stages and to compare the OPNET results with the theoretical as well as simulation results obtained by MATLAB. In this configuration the user terminals are located circularly around the base-station in middle. In order to implement different power levels we add another circle with different radius. All the users have same transmit power so the distance between the terminals and base station determines the received power.

We will give a brief description of the modified pipeline stages to establish OPNET simulation environment. The modified pipeline stages are shown in Figure 6.

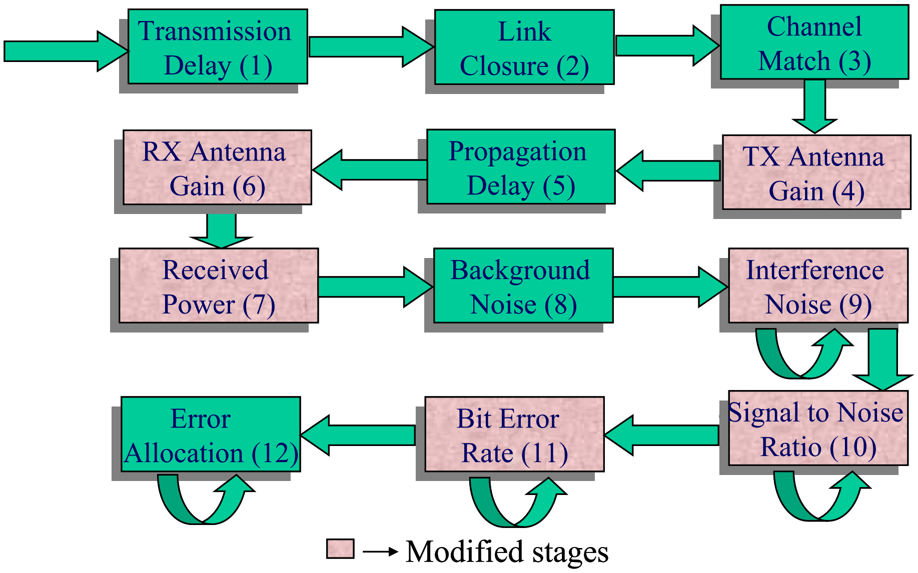

Figure 6. Modified radio transceiver pipeline stages in Opnet

Stage 0: Receiver Group

This stage is executed once in the start of the simulation for each pair of transmitter and receiver channels to determine the feasibility of communication. It is not executed on a per-transmission basis. It assigns receiver groups for all transmitter channels. We use the default receiver group stage where all receivers are in the receiver group of all transmitters.

Stage 1: Transmission Delay

This stage computes the time required for the transmission of a packet. The transmitter stops transmitting new packets during this time interval and continues transmitting packets in queue after transmission of the current packet.

Transmission delay result is used in conjunction with the result of the propagation delay stage to compute the time at which the packet completes reception at the links destination.

We use the default transmission delay stage to compute the transmission time

Stage 2: Link Closure

This stage runs right after the transmission delay stage. The goal of the closure stage is to determine if the transmitted signal can physically attain the candidate receiver channel. This is determined by checking whether the line connecting the transmitter and receiver intercepts with the earth. If the transmitted signal can physically attain the candidate recover channel then the packet continues transmission through remaining stages. If the closure can not be maintained then the Simulation Kernel will discontinue the pipeline execution between the transmitter and receiver channels for this particular transmission (future transmissions between the channels are not prevented). In our simulations the default link closure stage was sufficient.

Stage 3: Channel Match

The channel match stage occurs immediately after the link closure stage. The purpose of this stage is to classify the transmission with respect to the receiver channel. One of the three possible categories must be assigned to the packet, namely, valid, noise, or ignored. A packet is considered a valid packet if the channel attributes of the transmitter and receiver nodes match. In our simulation we use default link closure stage.

Stage 4: Transmitter Antenna Gain

The purpose of the transmitter antenna gain stage is to compute the gain provided by the transmitter's associated antenna, based on the direction of the vector leading from the transmitter to the receiver. The Simulation Kernel does not itself use this result, but it is typically factored into the received power computation of stage 7. We use omni directional antenna at the transmitter.

Stage 5: Propagation Delay

The purpose of this stage is to calculate the amount of time required for the packet's signal to travel from the radio transmitter to the radio receiver. This result is generally dependent on the distance between the source and the destination. The Kernel uses this result to schedule a beginning-of-reception event for the receiver channel that the packet is destined for. In addition, the propagation delay result is used in conjunction with the result of the transmission delay stage to compute the time at which the packet completes reception (i.e., the time at which the last bit finishes arriving is the time at which the packet begins transmission added to the sum of the transmission delay and the propagation delay). We use the default propagation delay stage. In the default stage the propagation delay is computed assuming that the packets travel with the light of speed.

Stage 6: RX Antenna Gain

The receiver antenna gain stage is the earliest stage associated with the radio receiver rather than the transmitter. The purpose of the receiver antenna gain stage is to compute the gain provided by the receiver's associated antenna, based on the direction of the vector leading from the receiver to the transmitter. The Simulation Kernel does not itself use this result, but it is typically factored into the received power computation of the stage 7. The concept of receiver antenna gain is identical to that of transmitter antenna gain, except that it is due to the physical configuration and implementation of the antenna associated with the receiver platform. We use omni directional receiver antenna.

Stage 7: Received Power

The purpose of this stage is to compute the received power of the arriving packet's signal (in watts). In general, the calculation of received power is based on factors such as the power of the transmitter, the distance separating the transmitter and the receiver, the transmission frequency, and transmitter and receiver antenna gains. We have made two major modifications to the default received power stage.

In the default received power stage "signal lock" attribute of the radio receiver object is used to prevent simultaneous correct reception of multiple packets. Therefore the first arriving packet in a receiver node is assigned as a valid packet. The packets that arrive after this packet collides wit this packet are regarded as noise packet. Since our system is a CDMA based system, the simultaneous reception of multiple packets needs to be enabled. In our simulations, we remove the signal lock property so that the system can receive multiple packets.

The most important modification at the stage is in the calculation of the received power. In the default pipeline stage free space propagation is assumed without any type of fading. However, we want to model different channel models. The modified pipeline stage computes the path loss based on the free space propagation and the log-normal fading as described in Channel Models section (Equation 3). Therefore, the received power in Watts is then estimated by multiplying transmitted signal power, transmitter antenna gain, path loss, receiver antenna gain, and the random log-normal variable. The user of the module can determine the variance of the log normal variable as well as the user can determine the mean shift of the log normal distribution. Moreover if the user wants to simulate the scenario without log normal gain factor, the user can set the log-normal variance to 0 in OPNET.

Stage 8: Background Noise

The purpose of this stage is to represent the effect of all noise sources, except for other concurrently arriving transmissions, since these are accounted for by the interference noise stage. The expected result is the sum of the power of other noise sources, measured at the receiver's location and in the receiver channel's band. Typical background noise sources include thermal or galactic noise, emissions from neighboring electronics, and otherwise unmodeled radio transmissions (e.g., commercial radio, amateur radio, and television, depending on frequency).

We use the default background noise stage and we adjust the noise figure (NF) to meet our specifications. The necessary modification is made so that the user can specify any other NF when setting the receiver attributes in OPNET.

Stage 9: Interference Noise

Interference noise stage may be executed for a packet under two circumstances: the packet arrives at its destination channel while another packet is already being received; or the packet is already being received when another packet arrives where we assume that all packets are valid packet in our system. Clearly, the first circumstance can occur at most once for each packet, and the second may occur any number of times depending upon the transmission activities of other transmitters in the model. Note that a single invocation of this stage can be shared by the pipelines of the two packets.

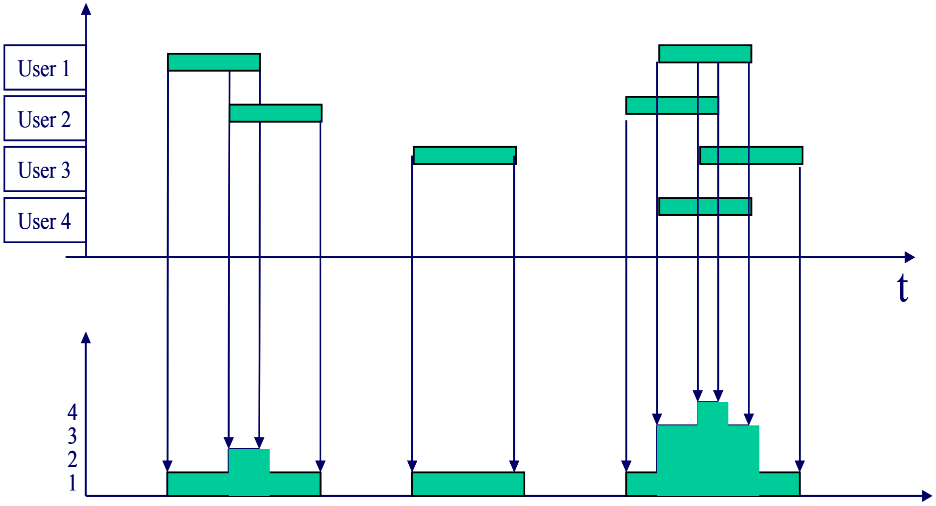

Figure 7. Concept of accumulation of interference in OPNET radio transceiver

The purpose of this stage is to account for the interactions between transmissions that arrive concurrently at the same receiver channel. The interference noise stage is expected to augment the value of an accumulator in each valid packet by the received power of the interfering packet. When a packet (valid or invalid) completes reception, the Kernel automatically subtracts its received power from the noise accumulator of all valid packets that are still arriving at the channel. In this manner, the accumulator reflects only the current noise level.

Stage 10: Signal to Noise Ratio

Signal to noise ratio stage may be executed for a packet under three circumstances: (1) the packet arrives at its destination channel; or (2) the packet is already being received and another packet arrives; or (3) the packet is already being received and another packet completes reception. Clearly, the first circumstance occurs exactly once for each packet, and the second and third may occur any number of times depending upon the transmission activities of other transmitters in the model. The three types of invocations define intervals over which a packet's average power SNR is taken to be constant (of course, this is an approximation when mobility is involved, since SNR would be continuously varying).

The purpose of SNR stage is to compute the current average power SNR result for the arriving packet. This calculation is usually based on values obtained during earlier stages, including received power, background noise, and interference noise. The SNR of the packet is an important performance measure that supports determination of the receiver's ability to correctly receive the packet's content.

Stage 11: Bit error Rate

It may be executed for a packet under three circumstances: (1) the packet completes reception at its destination channel, or (2) the packet is already being received and another packet arrives, or (3) the packet is already being received and another packet completes reception. These circumstances correspond to the ends of periods during which the packet's SNR is taken to be constant. The purpose of the BER stage is to derive the probability of bit errors during the past interval of constant SNR. This is not the empirical rate of bit errors, but the expected rate, usually based on the SNR. In general, the bit error rate provided by this stage is also a function of the type of modulation used for the transmitted signal.

In this stage the machine takes the modulation table from the receiver and transmitter attributes. OPNET is designed to simulate packet level modulation attributes; therefore, it is difficult and unnecessary to design a Viterbi decoder which needs bitwise operation. Instead of using complex receivers we form a modulation table (or BER table) for convolutional coded one user system. The user can change the modulation table in order to simulate different coding rate and constraint lengths for convolutional codes. Note that this method greatly increases the flexibility of the programming in OPNET.

Stage 12: Error Allocation

Error allocation stage is always executed immediately upon return from the bit error rate stage. The purpose of the error allocation stage is to estimate the number of bit errors in a packet segment where the bit error probability has been calculated and is constant. This segment may be the entire packet, if no changes in bit error probability occur over the course of the packet's reception. Bit error count estimation is based on the bit error probability (obtained from stage 11) and the length of the affected segment.

Stage 13: Error Correction

Error correction stage is invoked when a packet completes reception, immediately after the final return of the error allocation stage, with no simulation time elapsing in between. Exactly one invocation of this stage occurs for each packet.

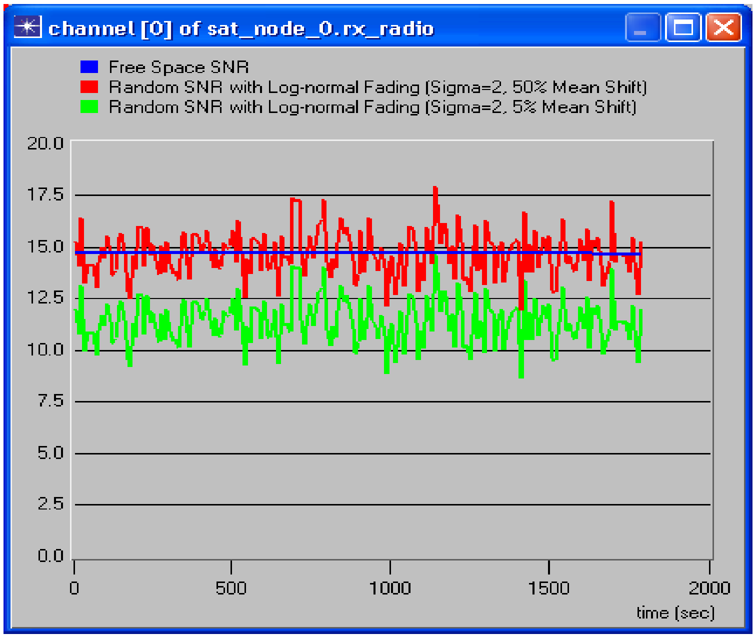

Figure 8. Single user SNR values for different fading statistics

7. Simulation Results

Our simulation model in OPNET uses two different power level users. Moreover, we use one circle to simulate only one power level. In Figure 9, we compare the capacity of these systems. We use N=31 spreading code in order to generate BER rate curves for uncoded and coded case. We use Equation 1 and MATLAB coded BPSK results in order to form these curves. We insert these BER curves into the OPNET simulations. The dark (blue) curve in Figure 9 represents uncoded one power level simulations. The light (red) curve shows the uncoded two power levels simulations. The lighter (green) curve shows coded two power levels simulations. All the curves has the ALOHA throughput structure. They had a peak point where the capacity is maximized. The maximum channel capacity for the coded case is higher then the uncoded cases. With using coding, we decrease the BER so we can have more simultaneous users that we can serve correctly.

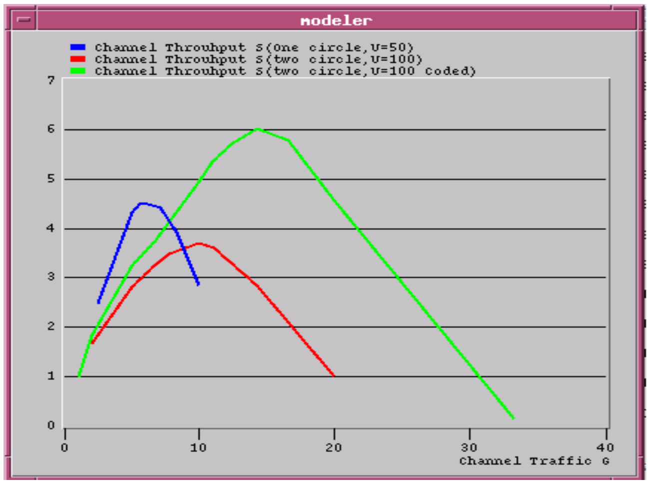

Figure 9. Results with different power levels

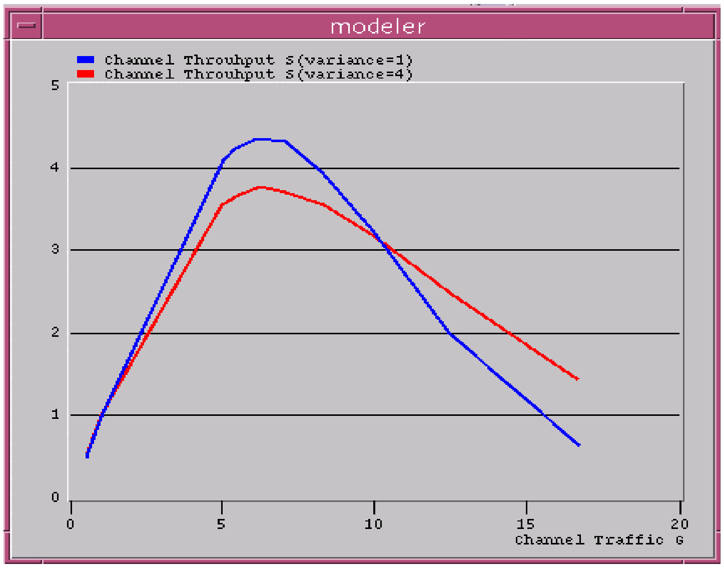

Figure 10. OPNET results for log-normal distribution

In Figure 10, we compare the capacity for one power level. But we change the log normal distribution variance (Equation 3). In these simulations, we assume that the signal to noise ratio equals to 20 dB. The result in Figure 10 shows that with lower variance we can achieve higher maximum capacity. Moreover, we verify this with MATLAB simulations. The results of the MATLAB simulations are shown in Figure 11.

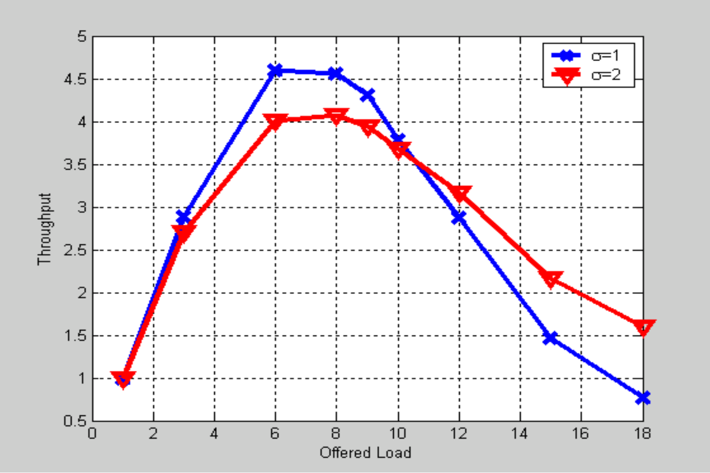

Figure 11. MATLAB results for log-normal distribution

8. Conclusions

OPNET simulation methods for CDMA system for coded and uncoded systems have been discussed. Instead of using many BER tables (modulation curves) for different number of users, we proposed to use GA in the pipeline stage of OPNET for both coded and uncoded cases. Thus, only the necessary bit error rate is calculated for every segment of the packet according to number of interfering users and E_b/N_o. We choose the SIGA because of its simplicity and accuracy. Moreover, we investigate our OPNET results with MATLAB results in order to verify them. The results show that using coding in random access networks improves the capacity. We investigate this result for different power level users and different channel conditions.

We present an OPNET simulation of random access CDMA system for different channel models. Such simulation can help us to observe and report the system specifications and performance of the large CDMA networks.

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