Performance and Capacity of a Voice/Data CDMA System with Variable Bit Rate Sources

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Abstract

This paper presents an approach to study the outage probability of voice calls in an integrated voice/generalized data wireless DS-CDMA network, where both voice and data can be considered as originated from variable bit rate sources. Emphasis is on voice outage probability in anticipation of control strategies for data. Taking into account the imperfections in power control for such CDMA systems, the effect of data transmission variability on the Erlang capacity of voice is analyzed (on the uplink case). All the analysis is reduced to the evaluation of means and variances of the relevant random variables of the model (number of users of a certain service, bit rates, received powers). The results presented here on the interplay between the data and the voice calls could be useful in determining control and admission policies for both the services. Illustrative results are given for a CELP speech coder and fax (both real time and store and forward). Future directions in capacity studies under optimum power control and data retransmission policies are discussed.

1 Introduction

As is well known, the direct sequence CDMA system capacity is interference limited [1]. In addition, a major problem in CDMA systems is the near-far effect due to which a nearby interferer can disrupt the reception (at the base station) of a highly attenuated desired signal. As a result, power control of the users in the system is required. Therefore the actual quality of service (QoS) of a CDMA system is clearly a function of the number of active users as well as the tightness of the power control in the system.

In [2], an approach was presented to study the Erlang capacity of a voice only CDMA system. For applications where multi-media services need to be provided, there is a need for better understanding of the nature of the relationships between the different QoS requirements of the services in the system and the capacity of each type of service. Voice traffic is well understood through many years of telephone experience. It is well characterized by a Poisson arrival process and a usual holding time of a few minutes. Also, the statistical characterization of the output of the variable bit rate codecs, developed recently, are starting to be available [3, 4]. A particular concern is the uncertainties with the data (or actually, any non-voice) traffic. These uncertainties affect not only the capacity, but also have implications upon control, i.e., both admission control and the control of set up calls. Note that uncertainties arise here not only because of variances of certain random variables, but also because of uncertainties of the means and variances. For example, in fax service, there is a high level of uncertainty about the information content of a page after source coding (from a few tens of kbytes for a text page only and a few hundred of kbytes for a highly complex picture, depending on the compression ratio) [5]. A modeling methodology is needed which facilitates sensitivity studies of uncertain parameters. The work presented here provides a step in studying the interplay between different types of services in a DS-CDMA system. We develop a generalized Erlang capacity formulation for voice and generalized data. The calculations are reduced to using means and variances. Since data, but not voice, can be delayed, emphasis is on voice outage probability in anticipation of control strategies for data. One such data control strategy, using related analysis methodology, is studied in [6].

The analytic simplicity is achieved using an approximation from [7]. This approximation was also used in [8], in a somewhat different way, for an Erlang capacity type problem for packet CDMA. We illustrate the formulation with a CELP speech coder and fax. Section 2 provides the general formulation and Section 3 gives the source models for voice and fax. Illustrative numerical results are in Section 4 and concluding comments in Section 5.

2 System Model

We consider a DS-CDMA network consisting of numerous mobile subscribers communicating with one base station (corresponding to a single cell). The subscribers can request voice calls as well as data/fax calls. If the channel bandwidth is W, the approximated measure of the interference rejection capability is given by the processing gain $G_{v,i} = W/R_{v,i}$ for voice calls that are using, in that instant, the bit rate $R_{v,i}$. Similarly, the corresponding processing gain for data/fax is given as $G_{d,i} = W/R_{d,i}$ where $R_{d,i}$ is the i^{th} bit rate allowed for data/fax. Furthermore, we will denote by K_v and K_d , the number of ongoing voice calls and data calls respectively. Since we are considering an integrated voice-data/fax wireless system, typically each type of service may require a different QoS (BER or FER) and, for this reason, we will denote the received bit energy due to each bit in a packet by $E_{b,v}$ and $E_{b,d}$ for voice and data/fax respectively. This corresponds to different power control requirements for the voice and data/fax services.

Remark: Data BER requirements can be very stringent, and in practice require additional coding and ARQ. While this is not explicitly included in the numerical examples, the methodology can easily incorporate that. The effect of other-cell interference can be included in the analysis by means of an f-factor as in [2].

2.1 Effect of data on voice

We will consider a desired voice user and treat all other voice users as well as data users as interferers. The selection of a desired voice user is arbitrary since, without loss of generality, we assume all voice users to belong to the same class of service. Therefore the bit energy over total interference density ratio of any voice user reflects (on the average) that of all other voice users as well. Then, based on the QoS requirement for voice calls, we will formulate a condition for outage of voice calls where the effect of data/fax calls is also considered. With direct sequence BPSK, we obtain the ratio of the despread bit energy to noise density at the desired user as

$$(\frac{E_b}{\eta})_v = \frac{\frac{E_{b,vo}}{N_0}}{1 + \frac{1}{G_v^{'}}\sum_{i=1}^{K_v-1}\alpha_{vi}\frac{E_{b,vi}}{N_0} + \frac{1}{G_d^{'}}\sum_{i=1}^{K_d}\alpha_{di}\frac{E_{b,di}}{N_0}}, \tag{1}$$

where N_0 is the background noise power spectral density and η is the total interference spectral density ($\eta = N_0 + I_0$). The parameters $G_v^{'}$ and $G_d^{'}$ are defined by the ratio between the spreading bandwidth and the average bit rate: $G_v^{'} = W/E[R_v]$ and $G_d^{'} = W/E[R_d]$. The bit rate variability is in the α's. The variables α_{vi} for voice user i and α_{dj} for data/fax user j are defined as $\alpha_{vi} = R_{vi}^{(c)}/E[R_v]$ and $\alpha_{dj} = R_{dj}^{(c)}/E[R_d]$, where $R_{vi}^{(c)}$ and $R_{dj}^{(c)}$ are the current voice and data/fax gross bit rates of user i and j, respectively. In other words, α is the current gross bit rate normalized with respect to the average gross bit rate. In section 3, we will give specific examples of those random variables.

The QoS requirement for voice can be specified as $(\frac{E_b}{\eta})_v \geq \gamma_v$ where γ_v is the power protection ratio for the voice calls. This translates to a specific requirement on the bit error rate for voice calls. Typically $K_v - 1$ and K_d are random variables. For the case of voice calls, K_v can be modeled as a Poisson random variable with parameter λ_v/μ_v where the voice calls arrive into the system with rate λ_v and have an exponential service time with mean $\frac{1}{\mu_v}$. It can be easily shown that $K_v - 1$ is a random variable with probability mass function given by

$$Pr\{K_v - 1 = n \mid K_v > 0\} = \frac{e^{\frac{-\lambda_v}{\mu_v}}(\frac{\lambda_v}{\mu_v})^{n+1}}{(n+1)!}\frac{1}{1-e^{\frac{-\lambda_v}{\mu_v}}}, \tag{2}$$

with the mean and the variance given by

$$E[K_v - 1 \mid K_v > 0] = \frac{\lambda_v/\mu_v}{1-\exp(-\lambda_v/\mu_v)} - 1, \tag{3}$$

$$Var[K_v - 1 \mid K_v > 0] = \frac{\frac{\lambda_v}{\mu_v}}{(1-e^{\frac{-\lambda_v}{\mu_v}})^2}\{1 - e^{(\frac{-\lambda_v}{\mu_v})} - \frac{\lambda_v}{\mu_v}e^{(\frac{-\lambda_v}{\mu_v})}\}. \tag{4}$$

For a perfectly power controlled system, the $E_{b,v}/N_0$ and $E_{b,d}/N_0$ are constants, while for the case of imperfections in power control, these are modeled as lognormal random variables with mean m and standard deviation σ, that reflect the inaccuracy in the power control loops [2]. Based on the QoS requirement specified above, the condition for outage of voice calls can be rewritten as

$$Z < \gamma_v, \tag{5}$$

where

$$Z = \frac{E_{b,vo}}{N_0} - \beta_v \sum_{i=1}^{K_v-1} \alpha_{vi} \frac{E_{b,vi}}{N_0} - \beta_d \sum_{i=1}^{K_d} \alpha_{di} \frac{E_{b,di}}{N_0}, \tag{6}$$

with $\beta_v = \gamma_v/G_v^{'}$, and $\beta_d = \gamma_v/G_d^{'}$. In general, we let $K_v - 1$ and K_d to be random variables and we assume that $E_{b,vi}/N_0$ and $E_{b,di}/N_0$ to be independent lognormal random variables with means m_v, m_d, and variances σ_v^2, σ_d^2 , respectively. We assume that the random variables $E_{b,vi}/N_0$ are identically distributed with mean m_v and variance σ_v^2, because the power is varied such that the power to current bit rate ratio $P_{vi}/R_{vi}^{(c)}$ is kept constant, and similarly for data calls of each service class as well. If we define $X = \frac{E_{b,vo}}{N_0}$, $S_v = \beta_v \sum_{i=1}^{K_v-1} \alpha_{vi} \frac{E_{b,vi}}{N_0}$, and $S_d = \beta_d \sum_{i=1}^{K_d} \alpha_{di} \frac{E_{b,di}}{N_0}$, then the outage probability for voice calls can be written as

$$Pr\{Z < \gamma_v\} = Pr\{X < \gamma_v + S\}, \tag{7}$$

where $S = S_v + S_d$. The outage probability in (7) can be rewritten as

$$Pr\{Z < \gamma_v\} = \int_0^\infty Pr\{X < \gamma_v + s \mid S = s\} f_S(s)\, ds, \tag{8}$$

where we have conditioned on, and then averaged out the random variable S. Furthermore, since X is a lognormal random variable, we have

$$Pr\{Z < \gamma_v\} = 1 - \int_0^\infty Q\left(\frac{\ln(\gamma_v + s) - E(Y)}{\sqrt{Var(Y)}}\right) f_S(s)\, ds, \tag{9}$$

where $Y = \ln X$ is a Gaussian random variable with mean $E[Y]$ and variance $Var(Y)$. The relationships for evaluating $E[Y]$ and $Var(Y)$ are given as [9]

$$E[X] = \exp(E[Y] + \frac{1}{2}Var(Y)), \tag{10}$$

$$Var(Y) = \ln((CV(X))^2 + 1), \tag{11}$$

where the Coefficient of Variation of X is defined as $CV(X) = \sqrt{Var(X)}/E[X]$. To calculate the outage probability in (9), we use the approximation used in [7], and it is given as

$$Pr\{Z < \gamma_v\} = 1 - \frac{2}{3}Q\left(\frac{\ln(\gamma_v + \mu_S) - E(Y)}{\sqrt{Var(Y)}}\right) \tag{12}$$

$$-\frac{1}{6}Q\left(\frac{\ln(\gamma_v + \mu_S + \sqrt{3}\sigma_S) - E(Y)}{\sqrt{Var(Y)}}\right) - \frac{1}{6}Q\left(\frac{\ln(\gamma_v + \mu_S - \sqrt{3}\sigma_S) - E(Y)}{\sqrt{Var(Y)}}\right),$$

where μ_S and σ_S^2 are the mean and variance of the random variable S, respectively, and can be computed as [10]

$$\mu_S = \beta_v E[K_v - 1] m_v m_{\alpha_v} + \beta_d E[K_d] m_d m_{\alpha_d}, \tag{13}$$

$$\begin{aligned}\sigma_S^2 = \beta_v^2 \{ & E[K_v - 1](Var(\alpha_v)\sigma_v^2 + Var(\alpha_v)m_v^2 + \sigma_v^2 m_{\alpha_v}^2) \\ & + Var(K_v - 1)\,(m_v m_{\alpha_v})^2\} + \beta_d^2\,\{E[K_d](Var(\alpha_d)\sigma_d^2 + \\ & Var(\alpha_d)m_d^2 + \sigma_d^2 m_{\alpha_d}^2)\; + Var(K_d)\,(m_d m_{\alpha_d})^2\}.\end{aligned} \tag{14}$$

where m_{α_v}, $Var(\alpha_v)$, and m_{α_d}, $Var(\alpha_d)$, are the means and the variances of the RVs α_v and α_d, respectively. We will use (12) for evaluating the outage probability for voice calls for the different cases of data traffic considered. The approximation from [7] is exact when the function is a fifth degree polynomial and the random variable S is normal and it is fairly robust to deviations from those conditions. When the active number of voice and data calls get large and are Poisson, the Central Limit Theorem for Filtered Poisson processes [11] can be invoked to show that S tends to a normal random variable.
Remark: Note again that we are concentrating on the effect of data upon voice QoS (in anticipation of control strategies for data).

3 Source Models and Services

To use the methodology of section 2, only simple static source models are needed, that is, the fractions of time that the source is at a given rate and power are needed.

3.1 Voice Service

The flexibility of CDMA-based communication systems can support variable bit rate transmission [12, 13]. The removal of the fixed bit rate constraint allowed the development of a class of speech codecs able to provide different output bit rates depending on the time varying local characteristics of speech. The variable bit rate codec considered here is the CELP (Codebook Excited Linear Predictive) codec designed for the CODIT project [4]. This codec has seven different modes of operations which correspond to seven different output rates. Silence periods are encoded with the lowest number of bits (so as to provide comfort noise) while the active voice periods are encoded with an increasing number of bits as the information content of the "sound" increases. The source coding process, as well as the information transmission on the air interface, is organized in fixed periods of time called "frames". The frame length is 10 ms. The instantaneous bit rate is identified by the number of bits (i.e. the physical packet length) carried by each frame. In this case the radiated power (and then, the created interference) is directly related to the instantaneous bit rate and, therefore, no power exceeding what is strictly needed in a certain instant will be transmitted. The occurrence probability of each bit rate has been estimated by means of measurements. The measurement results for the case of voice only (no background noise)

Mode	*Net Bit Rate*	$r/E[r]$	*Occurrence Probability*
1	$r_1 = 400$	0.0647	$p_1 = 0.40$
2	$r_2 = 3200$	0.517	$p_2 = 0.05$
3	$r_3 = 8500$	1.37	$p_3 = 0.14$
4	$r_4 = 12500$	2.02	$p_4 = 0.15$
5	$r_5 = 7200$	1.16	$p_5 = 0.10$
6	$r_6 = 12000$	1.94	$p_6 = 0.12$
7	$r_7 = 16000$	2.59	$p_7 = 0.04$

Table 1: Bit Rate Statistics for Voice: $E[r] = 6185$ bps, $\sigma_r = 5346$ bps

are shown in Table 1. It should be noted that there is another advantage allowed by the use of a variable rate codec. Due to the different sensitivity to the errors of the seven possible codec outputs, it is possible to design an unequal error protection coding scheme in order to cope with the impairments introduced by the channel. Here, for simplicity of illustration we assume to have equal error protection provided by a channel code with rate ρ (as a consequence, the gross bit rate will be $R_i = r_i/\rho$). The formulation, however, permits the inclusion of unequal error protection.

3.2 Fax Service

In this section we will briefly describe the problems related to the fax service provision in a mobile communication network [14, 15] and then we will make some reasonable assumptions about the requested bit rates and their occurrence probabilities. We will consider the G3 facsimile terminals that are the most commonly adopted. A "mobile fax" service can be provided in two different ways:
a) as a transparent (or real time) service
b) as a non-transparent (or store and forward) service.

In case a) the facsimile information is delivered to the destination before the call between the two fax terminals is released. In case b) the facsimile message is stored temporarily in a buffer before its transmission over the radio interface; in this case the call established between the originating terminal and "the mobile network border" is usually released before the information is delivered to the final destination. In both cases appropriate interworking units (Fax Adaptors - FA) must be inserted in the network in order to allow the fax terminals to communicate over the air interface. The FA are located in the mobile station (between the fax terminal and the radio interface) and in the Mobile Control Node (MCN), between the mobile radio environment and the PSTN (Public Switched Telephone Network), as shown in fig. 1.

3.2.1 Transparent Service

In this case the FA is simpler because its main task is to convert the G3 facsimile signals into a binary information stream suitable for the transmission over the radio interface. No protocol conversion is performed. The mobile network between the two FAs appears like an additive noise channel with a delay τ (see fig. 2), The only error protection available is provided by the error control coding designed for the mobile radio channel (in addition to the optional Error Control Mode of G3 fax transmission). According to the T.30 Recommendation [16], the control messages are (usually) transmitted at 300 bit/s while the image information is transmitted at a maximum bit rate of 9600 bit/s (14400 bit/s in recent machines and in case of very good connections). Here we assume that the transmission of the control messages and of the image information are performed at 300 bit/s and 9600 bit/s, respectively. We assume also that the mobile station is the sender and that it has to transmit a 3 page document whose information content is about 60 kbytes each. On the basis of the signal sequence for the transmission of a 3 page document [16, 5, 17] and taking into account the control message lengths, we obtain the following approximated occurrence probabilities of the two bit rates (see Table 2). In this probability estimation, we considered

Net Bit Rate	$r/E[r]$	*Occurrence Probability*
$r_1 = 300$	0.0346	0.1
$r_2 = 9600$	1.11	0.9

Table 2: Bit Rate Statistics for FAX: $E[r] = 8670$ bps, $\sigma_r = 2790$ bps

that the MS, excluding the image information transfer phase, is always transmitting at 300 bit/s. In fact, in a CDMA system, in order to maintain synchronization, enable power control and to facilitate the channel estimation, the bit rate cannot be reduced exactly to zero. At least a few hundred bit/s are always needed. As a consequence, the MS, having completed the call set up procedure, is supposed to transmit always at 300 bit/s, excluding the periods of time related to the training sequences and image information transfer. For the same reason, when the MS is receiving a fax, on the uplink there is always a net bit rate of 300 bit/s. It should also be remarked that the probabilities of Table 2 have been estimated assuming that the message transmissions are always successful and neglecting the processing time at the two ends. But, taking into account the enormous variability of the information content of a page, the relative impact of the neglected parameters should be minor. The transparent approach to the fax transmission has the advantage of the simplicity and provides a "sense of immediacy" (the sender has always an immediate indication about the transmission result). The main disadvantage consists, of course, in a non optimal efficiency in the use of the radio channel.

3.2.2 Non Transparent service

In this case the fax adapter is more complicated because it has to be able to store the image information and to handle call management procedures. On the other hand, the higher complexity and cost of the FA allows a more efficient use of the radio resource. In fact, once the call between the sender and the FA has been released, the fax information stored in the FA can be delivered like a constant bit rate file transfer, optimizing the use of the radio channel. It is, for example, likely to assume a constant bit rate transmission, higher or lower depending on the system load. Of course, in this case the information transmission over the radio channel is more reliable because, appropriate radio link protocols can be designed and implemented between the two FAs. The disadvantage consists in losing the "sense of immediacy" (the sender usually has no confirmation of the fax delivery to the destination before the call is cleared off). Taking account of the advantages and disadvantages of the two possible approaches to fax transmission over a mobile radio channel, a possible solution is to have available both options. In a mixed voice/fax service environment, an intelligent control entity can choose real time transmission when the system is not heavily loaded by voice and store and forward otherwise. Therefore, by studying the variations (mean and variance) of the data/fax traffic, and its impact on voice capacity, we could develop protocols for admission, and after call set up.

4 Numerical Results

In this section we use the models previously described so as to provide few numerical examples of evaluation of outage probability for voice service in case of an integrated voice/fax CDMA wireless system. Two main cases are considered:
1) K_d is a random variable i.e., the number of fax call arrivals is modeled as a Poisson process with parameter λ_d/μ_d (transparent case);
2) K_d is deterministic i.e., there is a fixed number of active calls on the uplink (non transparent or store and forward case).

For both situations the outage probability is evaluated first taking into account the source variability and then on the basis of the average source bit rates only (i.e., considering a continuous transmission performed at the average bit rate). This is made in order to compare the impact of the bit rate variance compared to the calculations based on the mean value only.

Remarks :. For the illustrative results to be displayed we choose the voice power control error, σ_v, to be 2 dB. For data power control error, σ_d, we used both 2 dB and 4 dB to show the effect of possibly larger errors in controlling the data. Note that while we display results with fixed means for E_b/N_0's for varying parameters, the E_b/N_0's actually change with these parameters. This is because it is the E_b/η's that are power controlled. The selection of the E_b/N_0's for multiple services is considered as a power optimization problem in [18, 19]. The combination of the two approaches will appear in a future paper.

1) **Real time fax :** If K_d is a Poisson random variable with parameter λ_d/μ_d the mean and the variance of the random variable S are given are given by the equations (13) and (14), where $E[K_d] = Var(K_d) = \lambda_d/\mu_d$. This could be a likely model when the fax is provided as transparent service. Both voice and fax call arrivals are Poisson and, depending on the average offered fax traffic, by means of (12) we can get different plots of the outage probability for voice versus λ_v/μ_v , as shown in fig. 3. Simulation results are also shown for reference. The approximations work extremely well when the number of voice users is large (for probability of outage 1% and larger) owing to the fact that the random variable S is approaching a normal random variable. The same kind of evaluation has been made

on the basis of the average bit rates only. In this case the transmission is supposed to be continuous (i.e.,$\alpha_v = \alpha_d = 1$, $m_{\alpha_v} = m_{\alpha_d}$, and $Var(\alpha_v) = Var(\alpha_d) = 0$) and the results are depicted in fig. 4. Comparing the two diagrams it turns out that, the outage probability based on the mean bit rates only is a little more optimistic than that one evaluated taking into account the bit rate variations. An interesting two-dimensional capacity diagram is illustrated in fig. 5. In case of an outage probability for voice of 1%, the diagram shows the relationship between the average voice traffic and the average fax traffic.
2) **Store and forward fax :** In this case, we assume that the allowed number of active fax users is controlled by the number of voice users. Hence, K_v and K_d are deterministic. Therefore, $E[K_v] = K_v$, $E[K_d] = K_d$, and $Var(K_v) = Var(K_d) = 0$. This could be a reasonable model of the situation where the fax is provided as a non transparent (or store and forward) service: on the basis of the current number of voice calls, if we do not want to exceed a certain outage probability value, we are not allowed to provide more than K_d contemporaneous fax calls. Also in this case, the results based on the average bit rate only are more optimistic than those obtained considering the variable bit rate sources (see fig. 6 and 7).

5 Conclusions and Future Directions

In this paper we presented an approach for the evaluation of the outage probability of voice calls in an integrated voice/data DS-CDMA wireless network with VBR sources. The variability of the sources has been described by the occurrence probabilities of the various possible bit rates. Numerical examples were presented for a CELP variable rate speech coder and fax (both transparent and non-transparent). Imperfections in the power control loop were considered in the analysis. Although the results are presented for equal error protection and same bit energy to background noise density ratio for all the bit rates, the methodology can be readily extended to the unequal error protection case. Two different cases have been studied: the case of Poisson arrivals of fax calls and the case of fixed number of fax calls (determined on the basis of the current number of voice calls). For both cases the results obtained with the variable bit rate sources have been compared with those based on the average bit rates only. As expected, evaluations based on the average bit rates provide more optimistic results. Finally, a two-dimensional capacity diagram provides the average voice traffic versus the average fax traffic for fixed voice outage probability. Two dimensional capacity diagrams like this can be used for algorithms responsible for the admission control of new calls in mixed service environments. Another example of such capacity diagrams is in [20]. Extensions of the analysis include other cell interference, unequal error protection and quality requirements for the different service classes.

Future directions related to this work include considering situations of optimum power control as well as as retransmission schemes for data messages. While this work has considered the signal-to-interference ratio as a QoS measure to define Erlang capacity, an alternate definition is possible based on total interference power (as in [2]). It is shown in [21] that the above outage condition is more stringent and hence ensures feasibility of the power control problem. An approximate analysis that takes into account imperfect power control, activity factors and different quality requirements is also presented.

Another aspect of further investigation is the guarantee of individual QoS requirements for voice and data. For delay-tolerant (but high fidelity) data services, one possible approach is to design retransmission strategies that take into account voice QoS. In [6], a protocol for retransmission of data messages in an integrated voice/data CDMA system is analyzed. Voice calls are given preemptive priority over data calls and the protocol for data calls is controlled by the QoS (Quality of Service) requirement for the voice calls. The QoS criterion for voice calls is specified by a target SIR (signal-to-interference ratio) that each voice call has to meet in order to avoid outage. The admission of data calls into the system is achieved by means of a "persistence state" (as in [23]) that controls the probability of each data user gaining access to the system. Imperfections in power control for both the voice and data messages are included while studying the effect of the protocol on average performance measures like throughput and blocking probability for data service. Stability conditions are derived in [22], delineating parameter regions for stable operation.

In real cellular environments, it is not the instantaneous SIR dropping below the threshold that determines outage. It is, in fact, a certain duration of time of the SIR below a threshold that determines outage. The notion of minimum duration outages has been introduced in [24] where an outage is defined as the signal-to-interference ratio (SIR) going below a threshold and staying below longer than a certain "minimum duration". Such a notion of outage includes the time correlation in the signals which is important in real systems owing to mobility, fading and power control.

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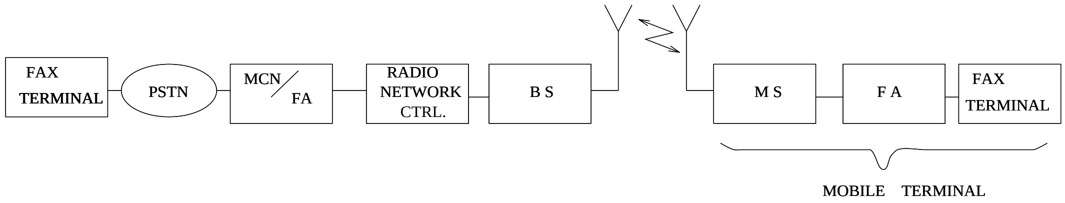

Figure 1: Mobile Fax Service

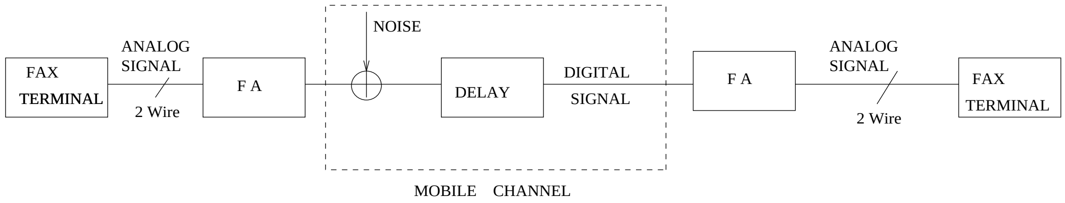

Figure 2: Mobile Fax Service (Real-Time)

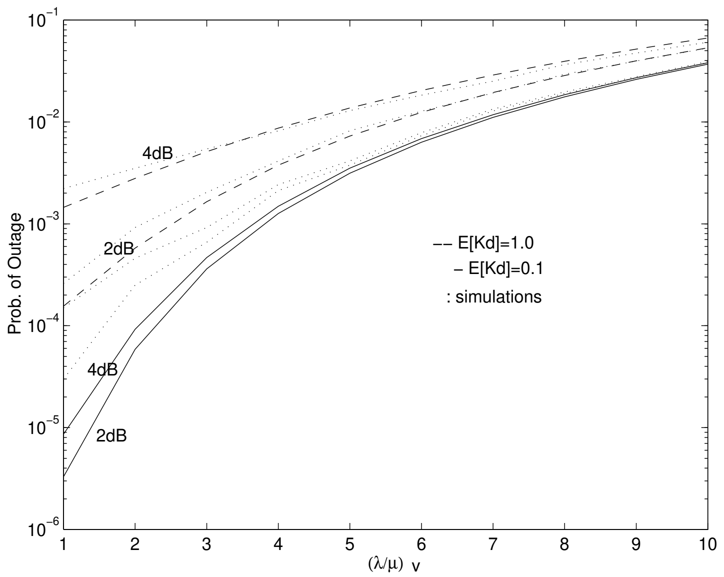

Figure 3: Outage Probability for Voice with Transparent Fax (VBR)
The plots are shown for $\sigma_d = 2dB$, and $\sigma_d = 4dB$. $m_v = m_d = 20dB$, $\sigma_v = 2dB$, $W = 1MHz$, and $\gamma_v = 5.0$

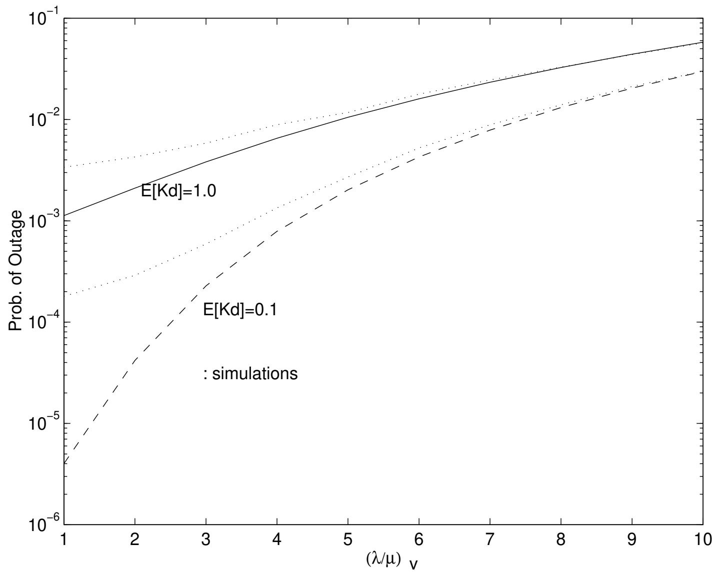

Figure 4: Outage Probability for Voice with Transparent Fax (Average BR)
The average bit rate is chosen for the voice calls and for the data for this evaluation. $m_v = m_d = 20dB$, $\sigma_v = 2dB$, $\sigma_d = 4dB$, $W = 1MHz$, and $\gamma_v = 5.0$.

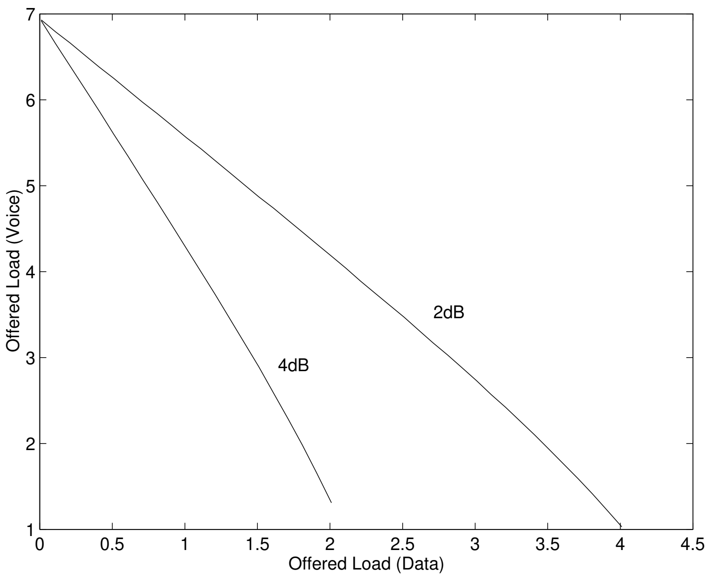

Figure 5: Two-Dimensional Capacity
The outage probability for voice calls is fixed at 0.01. The offered load of voice calls λ_v/μ_v is shown versus that for Fax calls λ_d/μ_d (transparent case). The plots are shown for $\sigma_d = 2dB$, and $\sigma_d = 4dB$. $m_v = m_d = 20dB$, $\sigma_v = 2dB$, $W = 1MHz$, and $\gamma_v = 5.0$

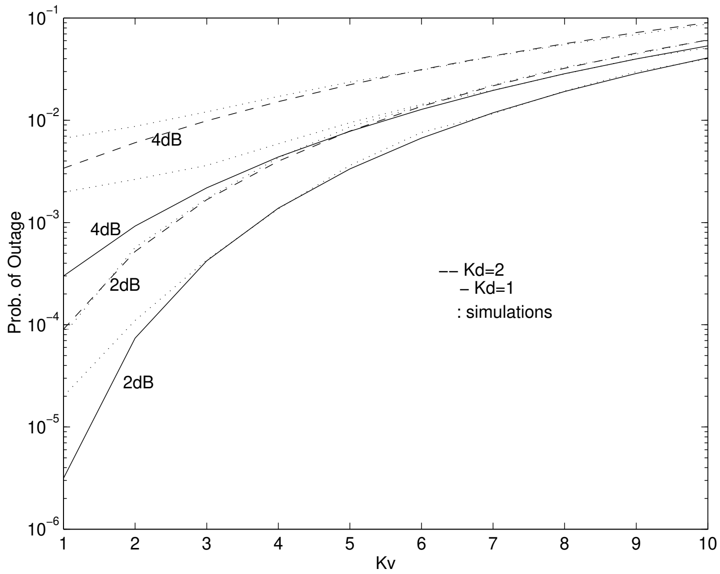

Figure 6: Outage Probability for Voice with Non Transparent Fax (VBR)
The plots are shown for $\sigma_d = 2dB$, and $\sigma_d = 4dB$. $m_v = m_d = 20dB$, $\sigma_v = 2dB$, $W = 1MHz$, and $\gamma_v = 5.0$.

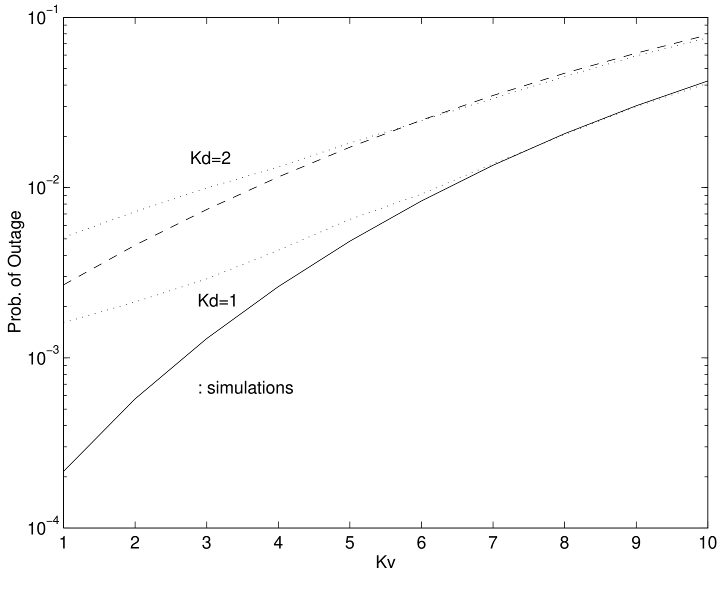

Figure 7: Outage Probability for Voice with Non Transparent Fax (Average BR)
The average bit rate is chosen for the voice calls and for the data for this evaluation. $m_v = m_d = 20dB$, $\sigma_v = 2dB$, $\sigma_d = 4dB$, $W = 1MHz$, and $\gamma_v = 5.0$.