System Capacity of an Integrated Voice and Data CDMA Network in Channel Load Sensing Protocol

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In this paper, we analyze an integrated voice and data system over a CDMA Unslotted ALOHA with Channel Load Sensing Protocol, considering the effect of each traffic. We can find that the threshold for the number of data transmissions is very effective to improve both performances of the throughput for data and the Erlang capacity for voice users. Furthermore, to optimize this system, we define the system capacity. As a result, we obtain an optimum threshold for data to make the best use of the channel capacity.

1 Introduction

Recently, integrated service technique has imposed itself as an essential demand in the universe of the modern communications. A lot of multiple access techniques have been used to fulfill these requirements. However, it is important that the one which adapts the best to dynamic traffic in wireless multimedia communications is chosen. Among these techniques that has attracted a great deal of attention is the Code Division Multiple Access (CDMA)[1].

In this paper, we focus on voice signal and data packet integration over an asynchronous CDMA system[2]-[5]. Since voice traffic is required for real time delivery, the voice users are required to get a reservation for available channel before they send their voice signals. When they try to get the reservation, they send reservation packets to the central station.

Once the reservation is accomplished, voice signal can be sent until voice call end. While data users can transmit their packets randomly. Data packets are allowed to retransmit their packets until the packets are received successfully.

We apply CDMA Unslotted ALOHA, which is the combination of Pure Aloha and CDMA[6]-[7], to the transmission protocol for the reservation packet and data packet. Furthermore, we use Channel Load Sensing Protocol(CLSP) to limit the number of ongoing transmission[6]-[8]. In CLSP, we can control the number of simultaneous transmissions under a certain threshold so that we can expect the decrease in the effect of multiple access interference(MAI) and we can expect to maintain the signal quality more than a required level, e.g. the bit error rate$< 10^{-3}$ for voice signal. Therefore, the signal quality is dependent on the threshold of CLSP.

In a system accommodating two different media, the quality required for each medium and the characteristics of each medium should be considered. For the quality, voice signals show greater tolerance to transmission errors than data packets. Accordingly, we consider the system in which each medium is controlled by different threshold to correspond to the quality required for each medium[5]. The main purpose of this paper is to investigate the effect of the threshold.

For voice users, we analyze the blocking probability and the Erlang capacity which depends on the data packet traffic, as well as voice traffic. We also analyze the throughput for data packets with consideration of not only data packet traffic, but also voice traffic. Effects of the thresholds on these performance are investigated.

Furthermore, we investigate the effect of thresholds on this system from the view point of total capacity of the system. We define the system capacity which is the sum of the throughput for data packets and the Erlang capacity for voice users. By using this capacity, we obtain an optimum threshold for data in which we can make the best use of channel capacity.

This paper is organized as follows. Section 2presents the system model. In Section 3 analysis for integrated voice and data system. In Section 4, system performances are evaluated and concluding remarks are presented in Section 5.

2 SYSTEM MODEL

In Fig.1, system model is depicted. A CDMA Unslotted ALOHA with channel load sensing protocol(CLSP) is considered. The network may consist of a large number of independent voice and data users and a central station. All users have their own CDMA codes(random signature). Only up-link is considered. For the simplification of this analysis, we make some assumptions as follows. Perfect power control is assumed, ensuring that all transmissions are received with equal power at the central station. Moreover, the same data rate and the same processing gain are used by both voice and data users. In CLSP, the central station monitors the channel load which is the number of ongoing voice and data transmissions. If the channel load is bellow a certain threshold, then the central station broadcasts all users information of the allowance of transmission. Otherwise, when the channel load is more than threshold, it broadcasts them information of the rejection of transmission.

We consider the system in which the generation of data packets and reservation packets for voice call are controlled by different threshold designed for each medium with correspond to the signal quality or priority. Main purpose of our analysis is to investigate the effect of the thresholds on performances of both media.

Once a reservation is accomplished, the voice user can transmit their continuous voice signal by using their own CDMA code until the voice call ends. In our analysis, perfect reservation is assumed, ensuring that all of the voice users can get the reservation from the first trial.

Data users transmit their packets to the central station randomly in according to the control signal from the central station. We assume that the data packet is transmitted successfully if, and only if, all its bits succeeded, and no error correcting code is applied. If the data packet is not transmitted successfully, the packet would be retransmitted.

BPSK modulation of the transmitted bit stream for both of voice and data is assumed.

2.1 Model of Voice Traffic

With an infinite number of voice users, Poisson process is used to model voice reservation packet arrivals with arrival rate λ_v. Moreover, under our assumption of perfect reservation, the length of voice call from the first trial for reservation

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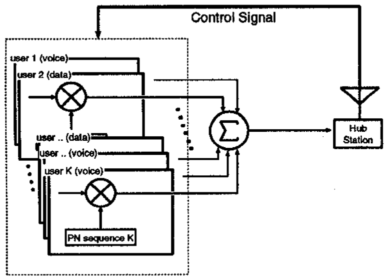

Figure 1: System Model

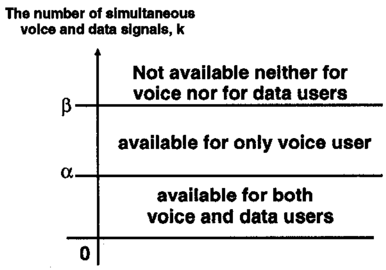

Figure 2: definition of thresholds

until voice call ends is modeled as exponentially distributed and the voice call is terminated with death rate μ_v. The voice offered load,G_v is defined as the mean number of voice calls generating in the average length of voice calls, therefore, $G_v = \lambda_v \cdot \mu_v$.

2.2 Model of Data Traffic

A Poisson process is used as a model for the generation of data packets. An infinite number of data users is assumed. Packet generation rate is λ_d, which includes the generation rate of retransmission packets. The death rate $\mu_d(k_{d1})$ at which a packet service is terminated is given for fixed packet length as [7],

$$\mu_d(k_{d1}) = \frac{k_{d1}}{T_d}$$

where T_d is the length of data packets and k_{d1} is the number of interfering data packets at 1st bit in a tagged data packet. The offered load of data, G_d, is defined as the mean number of data packets generating in a data packet duration. Therefore, $G_d = \lambda_d \cdot T_d$.

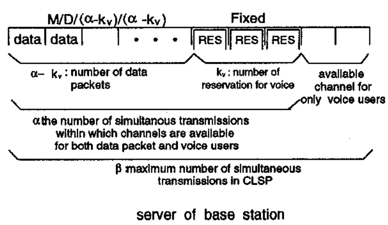

Figure 3: The Arrival of Data Packets

3 Performance analysis for Integrated System

3.1 Throughput Analysis for Data

In CLSP, the number of established voice users and data packets simultaneously transmitted is limited up to a certain threshold, which is the number of maximum available channels. We denote the thresholds for data and voice as α and β, respectively, and always satisfied with $\beta > \alpha$ because voice signals show greater tolerance to transmission errors than data packets. As shown in Fig. 2, if the number of voice and data simultaneous transmissions k is less than α, both voice and data users are allowed to send their packets(reservation packets for the case of voice users). If $\alpha \leq k < \beta$, data users are prohibited to send their packets while voice users are allowed to send their reservation packets. If $\beta \leq k$, both of users are prohibited to send them. Suppose that there is no other medium in the system, a data packet traffic can be modeled by $M/D/\alpha/\alpha$ [1] queueing system while $M/M/\beta/\beta$ queueing system can model a voice signal traffic [10]. But since voice and data exist simultaneously in this system, these models cannot express each of traffic exactly. Therefore, both of the two traffic is dependent on each other and the number of simultaneous voice and data transmissions fluctuates too complicatedly to express the fluctuation of that by Malkov state transition diagram as in the case of data only system[7]. This fact makes this analysis difficult.

To get over this difficulty, we focus on a fact that the number of established voice users can be considered as a constant during a data packet transmission. This fact is explained as follows. Since for voice calls, the length is very large and the generation rate is very few in compared with each of data packets, there are few changes in the fluctuation of the number of established voice users in one data packet duration. Therefore, we can consider the number of established voice user as a constant during a one data packet transmission. Furthermore, with the assumption that bit errors are caused by the effect of multiple access interference and additive white Gaussian noise (AWGN), the bit error probability of an asynchronous one media CDMA system for BPSK modulation is given in [9]. We expand it and obtain the bit error probability for an integrated voice and data system as follows,

$$BER(k_d, k_v) = Q[(\frac{k_d + k_v}{3N} + \frac{N_0}{2E_b})^{-0.5}], \quad (1)$$

where k_d,k_v is the number of simultaneous data and voice transmissions respectively, and the same processing gain N is designed for data packets and voice signals, N_0 is two-sided spectral density of Gaussian noise and $Q(x)$ is given by

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty \exp(-u^2/2)du \ . \quad (2)$$

Since voice and data users are controlled by the central station with the number of transmissions of both media, the generation of voice and data is not independent of each other. But with the assumption that the number of established voice users is constant value k_v in a one data packet duration, the data packet traffic can be modeled as $M/D/(\alpha - k_v)/(\alpha - k_v)$. This concept is figured in 3. This model expresses well the situation that when the number of

[1] $M/D/\alpha/\alpha$: service arrival time/ service duration time/ the maximum number of simultaneous transmissions/the number of buffers (M : the exponential distribution, D : fixed)

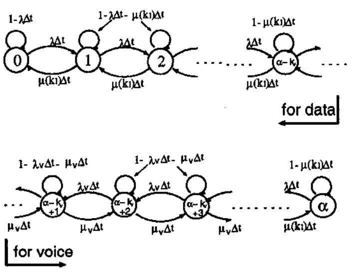

Figure 4: State Transision Diagram

established voice users k_v, is more than data threshold α, data packets are hardly generated. So that data packets can exist only in the case of $k_v < \alpha$. We obtain k_v from the steady state probability for $M/M/\beta/\beta$, as

$$P_\beta(k_v) = \frac{(G_v)^{k_v}/k_v!}{\sum_{m=0}^{\beta}(G_v)^m/m!}, \tag{3}$$

where m is the number of established voice users.

Therefore, we can obtain k_d as conditional probability with k_v.

$$P_\alpha(k_d|k_v) = \frac{(G_d)^{k_d}/k_d!}{\sum_{m=0}^{\alpha-k_v}(G_d)^m/m!} \tag{4}$$

The mean number of data packets at the server is

$$E_d = \sum_{k_v=0}^{\beta} G_d \cdot (1 - P_\alpha(k_d|k_v)) \cdot P_\beta(k_v), \tag{5}$$

Figure 4 shows the state transition of the number of interfering data packets for data packet analysis in CLSP. We approximate that the interference level is constant over a bit period and we assume that Δt equals to a bit interval. Since this data packet traffic can be thought as M/D/$(\alpha - k_v)/(\alpha - k_v)$ queueing system, the channel can be thought as $\alpha - k_v$-server station for data packets. All arriving data packets that find all $\alpha - k_v$ servers of the channel busy will be rejected. Therefore, there is no state transition beyond $\alpha - k_v - 1$. We define the probability $P(k_d, i, k_{d1})$. The number of interfering data packets is k_{d1} on first bit in a tagged data packet. The packet is transmitted successfully from first bit to $i-1$-th bit, and the number of interfering packets becomes k_d on i-th bit. $P(k_d, i, k_{d1})$ is as follows.

Case $i = 1$;

(a) $k_{d1} \leq \alpha - k_v - 1$; (the number of interfering data packets is less than or equals to $\alpha - k_v - 1$ on first bit.)

Using the steady state probability for the M/D/$(\alpha-k_v)/(\alpha-k_v)$ queueing system, we obtain,

$$P(k_d = k_{d1}, i = 1, k_{d1} \leq \alpha - k_v - 1) = \frac{G_d^{k_{d1}}/k_{d1}!}{\sum_{k_{d1}=0}^{\alpha-k_v-1} G)_d^{k_{d1}}/k_{d1}!}. \tag{6}$$

(b) $k_{d1} > \alpha - k_v - 1$; (the number of interfering data packets is more than $\alpha - k_v - 1$ on first bit.)

$$P(k_d = \alpha - k_v - 1, i = 1, k_{d1} > \alpha - k_v - 1) = 0 \tag{7}$$

Case $i > 1$;

(a)$k_d < \alpha - k_v - 1$; (the number of interfering data packets is below $\alpha - k_v - 1$.)

With an assumption that the transition of the number of interfering packets may occur every Δt seconds, the $P(k_d, i, k_{d1})$ is conditioned on the bit error probability, we obtain from [7],

$$\begin{aligned} P(k_d < \alpha - k_v - 1, i, k_{d1}) = \\ & P(k_d, i-1, k_{d1}) \cdot (1 - \mu_d(k_{d1})\Delta t - \lambda_d \Delta t) \\ & \cdot(1 - BER(k_d, k_v)) \\ + \quad & P(k_d + 1, i-1, k_{d1}) \cdot \mu_d(k_{d1})\Delta t \\ & \cdot(1 - BER(k_d + 1, k_v)) \\ + \quad & P(k_d, i-1, k_{d1}) \cdot \lambda_d \Delta t \cdot (1 - BER(k_d - 1, k_v)) \end{aligned} \tag{8}$$

(b)$k_d = \alpha - k_v - 1$;(the number of all packets on the server equals to a threshold $\alpha - k_v$.)

In this case, we can also obtain from Fig. 4.

$$\begin{aligned} P(k_d = \alpha - k_v - 1, i, k_{d1}) = \\ & P(k_d, i-1, k_{d1}) \cdot (1 - \mu_d(k_{d1})\Delta t) \cdot (1 - BER(k_d, k_v)) \\ + \quad & P(k_d - 1, i-1, k_{d1}) \cdot \lambda_d \Delta t \cdot (1 - BER(k_d - 1, k_v)) \end{aligned} \tag{9}$$

(c)$k_d > \alpha - k_v - 1$;(the number of all packets on the server is more than $\alpha - k_v - 1$)

$$P(k_d > \alpha - k_v - 1, i, k_{d1}) = 0 \tag{10}$$

Thus, averaging all over the possible values for k_d, k_{d1}, k_v, we obtain the probability of packet success Q and throughput S as follows,

$$Q = \sum_{k_d=0}^{\infty} \sum_{k_{d1}=0}^{\infty} P(k_d, L, k_{d1}) \cdot (1 - BER(k_d, k_v)) \ , \tag{11}$$

$$S = E_d \cdot \sum_{k_v=0}^{\alpha} Q \cdot \frac{(G_v \cdot \rho)^{k_v}/k_v!}{\sum_{m=0}^{\beta}(G_v \cdot \rho)^m/m!} \ , \tag{12}$$

where E_d is obtained by (5), ρ is voice activity factor.

3.2 Blocking Probability for Voice Users

Performance analyses for voice users is presented. Once voice users have got a reservation, they can send their signals until the voice call end. Therefore, performance for voice users is dependent on whether they can get a reservation or not. We evaluate this system for voice users by the blocking probability, which is the probability that the users cannot send their reservation packet due to the control signal from the central station. After that we obtain the Erlang capacity.

The blocking probability is dependent on the number of established voice users and ongoing data packets at the time when voice users want to transmit their reservation packet. Since we consider the stationary process, we can obtain the number of data packet transmissions as a steady state probability. We assume that the number of ongoing data packets k_d is obtained as a steady state probability for $M/D/\alpha/\alpha$.

Thus the voice traffic can be modeled as $M/M/(\beta - k_d)/(\beta - k_d)$. Therefore, we obtain the blocking probability for voice users from the Erlang B formula as a conditional probability with the number of ongoing data packets k_d.

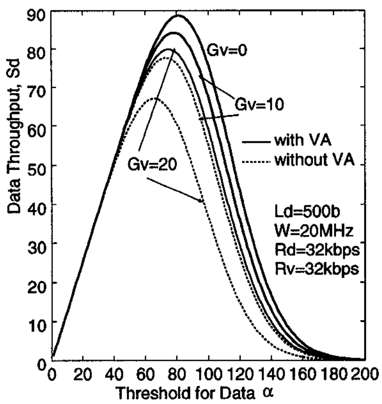

Figure 5: Effects of Voice Offered load on Data Throughput

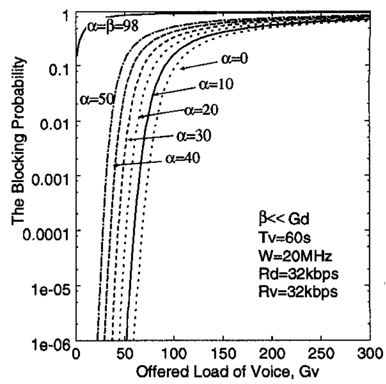

Figure 7: The Blocking Probability for the case of $\beta << G_d$

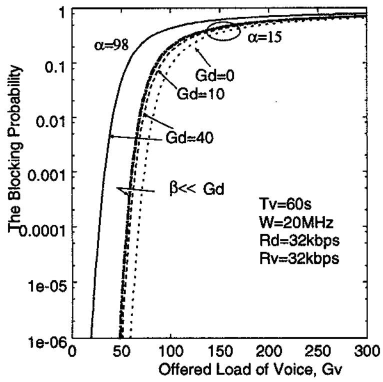

Figure 6: The Blocking Probability

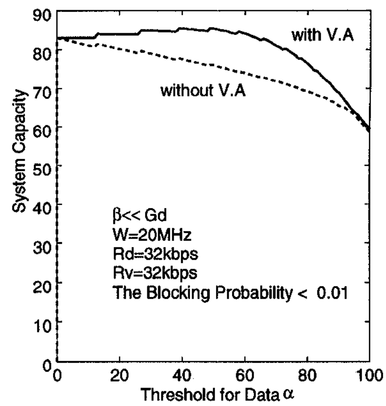

Figure 8: System Capacity

$$P_{blocking}(k_v|k_d) = \frac{G_v^{k_v}/(k_v)!}{\sum_{m=0}^{\beta-k_d} G_v^m/m!} \tag{13}$$

Averaging all over the possible value for the number of data packets k_d, the blocking probability is obtained as a function of G_v.

$$P_{blocking}(G_v) = \sum_{k_d=0}^{\alpha} \sum_{k_v=0}^{\beta-k_d} P_{blocking}(k_v|k_d) \cdot P_\alpha(k_d) \tag{14}$$

In this equation, the effect of voice activation is not included. Because the central station counts the number of established voice users, but not the number of ongoing voice signals.

Erlang capacity is defined as the maximum offered load satisfied with the condition that the blocking probability is less than 1 %. Thus, we obtain as the following,

$$E_{rlang} = G_v \tag{15}$$

where, G_v is chosen as the maximum value satisfied with the condition that $P_{blocking}(G_v) < 0.01$.

4 Numerical Results

With parameters shown below, some numerical results are performed. From (1), we obtain the value of $\beta = 98$ which satisfies the following equation, $BER(0,\beta) \leq 10^{-3}$. The value of 10^{-3} is a probability that is considered necessary for acceptable voice performance at the decoder.

In CLSP, the number of transmitted data packet is controlled below the threshold for data α. So if we consider a large G_d, the number of transmitted packets is close to the threshold α. In Fig.5, the throughput performance for data is plotted versus data thresholds. The condition of $\beta << G_d$ may be enough to investigate the effect of data threshold on the data throughput performances. We can find easily that the throughput degrades as the voice offered load G_v increases. While we can obtain the significant improvement by the effect of voice activation.

Figure 6 shows the blocking probability for voice users as a parameter of data offered load, G_d. Threshold for data is fixed at $\alpha = 15$. Moreover, to compare them with that for the case without data threshold, performance for $\alpha = 98 (= \beta)$ is also shown. First, we discuss about the performance for the case of $\alpha = 15$. We can find the performance for $G_d = 0$ is an optimum. Performance for the asymptotic case, $\beta << G_d$, is also depicted. We can obtain the Erlang capacity $E_{rlang} = 82$ for the case of $G_d = 0$. The Erlang capacity is obtained from (15). We can obtain $E_{rlang} = 69$ for the case of $G_d = 40$. It is significantly improved in compared with that of no data threshold case, $\alpha = 98$, in which we can obtain the Erlang capacity only $E_{rlang} = 42$, if $G_d = 40$. Moreover, it is interesting that even if $\beta << G_d$, we can obtain $E_{rlang} = 68$ in the case of $\alpha = 15$.

Furthermore, to investigate the effect of data threshold on the blocking probability, the asymptotic case is shown in Fig. 7 as a parameter of α. The performance degrades as α becomes larger.

From these results, we can reach to a conclusion that the data throughput can be increased at the expense of the Erlang capacity for voice users, vice versa. This conclusion expects us to obtain a point at which we can make both performances for both media as high as possible at the same time. To investigate the point, we define the system capacity which is the sum of throughput for data users and the Erlang capacity for voice users. We derive it from throughput for data and Erlang capacity for voice users, as follows.

$$C_{system} = S + E_{rlang} \quad (16)$$

This merit of figure indicates how effectively the capacity of channels is used. The reason for what we use throughput and the Erlang capacity for deriving the system capacity is as follows. Suppose that we obtain the Erlang capacity E_{rlang} in a certain data offered load G_d, and then the average number of data packets on the server is E_d from (5). Since the signal quality required for voice signal is maintained due to the limitation of the maximum number of simultaneous transmissions by CLSP, the system can support the voice offered load of E_{rlang} at the maximum. While all of data packets of E_d cannot be supported in the system due to the signal quality required for data packet. At the time, we need to consider the throughput which is the product of E_d and the packet success probability Q. Therefore, with consideration of the relation of the signal quality required for both media and the number of simultaneous transmissions, we can obtain the system capacity as the sum of the throughput for data and the Erlang capacity for voice users.

In Fig.8, we plot the system capacity for the case of considering voice activation(V.A) and the case without V.A. Both performances are significantly improved due to the data threshold in compared with the case of no threshold for data $\alpha = 98$. In the case of with V.A, performance is increasing slightly until $\alpha = 60$. Therefore, for the case of with V.A, we can find an optimum data threshold between $\alpha = 40$ and $\alpha = 60$. In which, we have a good balance between the actual transmissions of data and voice so that we can make the best use of the channel capacity.

Table1: Parameters

Parameter	values
The average length of voice call	$T_r = 60s$
Voice Activity rate	$\rho = 0.4$
Data packet length	$L_d = 500bits$
Bit rate for Voice	$R_v = 32kbps$
Bit rate for Data	$R_d = 32kbps$
Total band width	$W = 20MHz$
Processing gain for Voice and Data	$N = W/R_v = W/R_d \approx 312$
Threshold for voice	$\beta = 98$
The ratio of a signal and white Gaussian noise	$E_b/N_0 \to \infty$

5 Conclusion

In this paper, we have analyzed an integrated voice and data system over a CDMA Unslotted ALOHA with CLSP. We have found the threshold for data packet is very effective to improve the performance for both media.

We have obtained the throughput and for data packet users, and the blocking probability and the Erlang capacity for voice users. All performances are improved by the effect of threshold for data.

From the measurement of system capacity, we have obtained an optimum threshold for data to make the best use of the channel capacity in this system.

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