分碼多工無線區域網路的一個奇妙整合協定分析
ANALYSIS OF A NOVEL INTEGRATED PROTOCOL FOR CDMA WIRELESS LANS

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摘要

本篇論文提出一個分碼多工無線區域網路之整合協定。在此以時框爲基礎的協定下，語音及資料可有效率地在分碼多工無線區域網路上交換。此協定對於語音首先用一握手程序以預留一純語音傳送時段，然後剩下時框則分配給純資料傳送者。資料傳送基本上以 ALOHA 協定爲基礎。資料傳輸時間長度可根據語音數量、長度及系統容量臨界値而定。在此協定下，通話品質及資料錯誤率得以控制保持。在此我們提出連續時間馬可夫鏈模型以對此協定之特性表現作數學分析。數據顯示在此協定下，語音忙線率及損失率、資料損失率、及網路流通率都有優異表現。

關鍵詞：無線區域網路，分碼多工，整合 協定，馬可夫鏈。

ABSTRACT

This paper proposes an integrated protocol for CDMA wireless LANs. Voice and data can be exchanged over the wireless LAN based on this frame-based protocol. Modified from our previously proposed RBHS protocol [2], the protocol can accommodate integrated services on CDMA LANs with quality of service(QoS). This protocol uses a handshake procedure for voice calls first, followed by a reserved period for pure voice transmission. Then the remaining time of the frame is allocated to the data users for transmissions based on ALOHA protocol in nature. The length of data transmission period can be varied according to the number of voice calls initiated, voice packet length and the system threshold. Performance analysis using continuous-time Markov chain models are proposed and numerical results in voice call blocking probability, packet loss rate, and network throughput are presented.

Keywords: WLAN, CDMA, Integrated Protocol, Markov Chain.

1. INTRODUCTION

A wireless local area network (WLAN) is usually designed with the following characteristics: transportable, support of variable and high-speed data, multimedia support, reliable and secure, short range coverage, and rapidly deployable which may be co-exitent with current LANs or may operate independently[1,11].

CDMA LANs have many advantages over other wireless LANs [1,12]. Spread spectrum communications enjoy many unique characteristics such as low probability of detection and interception, allowing simultaneous transmissions, multipath rejection, and larger soft capacity [1,5]. However, we must note that there exists a threshold effect[1,2,3,4,7,8,9]. Note that our scheme can be applied to broadband CDMA networks as well.

2. SYSTEM MODELS AND ASSUMPTIONS

A. CDMA SYSTEM MODEL

We consider a DS/CDMA with coherent BPSK system. We assume each radio unit can select one of the finite number of spreading code, PN sequences $C=(C_1,C_2,\ldots C_U)$ which can be used to transmit packets to the base station. At the base station, there are U receivers, where $U=U_v+U_d$. Packet collision can only occur when two or more units choose the same spreading code. For simplicity, we can assume each user is assigned a unique code.

We assumed N chips per symbol(bit) duration. The bit error probability can be calculated [6] as follows.

$$P_b(K)\cong\left[\left(\frac{K-1}{3N}+\frac{N_0}{2E_b}\right)^{-0.5}\right]+$$
$$\frac{1}{6}Q\left[\left(\frac{(K-1)(N/3)+\sqrt{3}\sigma}{N^2}+\frac{N_0}{2E_b}\right)^{-0.5}\right]+$$
$$\frac{1}{6}Q\left[\left(\frac{(K-1)(N/3)-\sqrt{3}\sigma}{N^2}+\frac{N_0}{2E_b}\right)^{-0.5}\right] \quad (1)$$

where

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

We denote the threshold for voice packets as K_v and the threshold for data packets as K_d. For a packet with L bits, if we set the acceptable voice packet error rate is $P_{voice}=10^{-3}$, then since

$$P_{voice}\le\left[1-P_b(K)\right]^L \quad (2)$$

we have

$$P_b(K_v)\le 1-\left(P_{voice}\right)^{1/L} \quad (3)$$

Substituting into previous equation, we can find the approximate threshold of K_v. Similar result can be obtained for K_d as follows.

$$P_b(K_d)\le 1-\left(P_{data}\right)^{1/L} \quad (4)$$

A more accurate model which accounts for the channel activities has been proposed in [13] and further developed in [1]. In which, the error occurs according to a Poisson process. simulation results by [13] have proven the validity of this model. The error rate can be calculated as follows. Assume the bit duration is B_d seconds, then

$$P_r\{no\ \ error\ \ occurs\ \ during\ B_d\}=e^{-B_d\varepsilon(K)}$$
$$=1-P_b(K) \quad (5)$$

From EQ(5), we have

$$\varepsilon(K)=-\ln\left[1-P_b(K)\right]/B_d \quad (6)$$

Specifically, we have the voice error rate as

$$\varepsilon_v(K)=-\ln\left[1-P_b(K)\right]/B_d \quad (7)$$

and the data bit error rate is then as

$$\varepsilon_v(K)=-\ln\left[1-P_b(K)\right]/B_d \quad (8)$$

EQ(7) is the upper bound of voice error rate. On the other hand, for a data packet, reliable transfer is required. Furthermore, we need usually some kind of forward error control for data packets. In that case, EQ(8) is the symbol error rate rather than is the bit error rate.

B. TRAFFIC ASSUMPTIONS

Assume the voice traffic generated by each voice unit is Poisson with rate λ_v, and the data traffic with rate λ_d. The arrival of traffic will not be scheduled once the previous packet is in service. Assume one frame is total τ seconds, then the probability that an idle voice unit will generate packet for transmission at next frame is

$$P_v=1-e^{-\lambda_v\tau} \quad (9)$$

Similarly, the probability that an idle data user will generate packet for transmission at next frame is

$$P_d=1-e^{-\lambda_d\tau} \quad (10)$$

The packet length of voice packet is exponentially distributed with parameter μ_v and data packet is μ_d. In our protocol, we assume that a packet transmission will be forced to finish in a frame time.

3. THE PROPOSED HP PROTOCOL

Our proposed protocol is illustrated in Figure 1. The frame structure consists of a control period and a transmission period. The control period is the handshaking time τ_1 seconds. τ_1 is equal to the request time and reply time, preamble synchronization time, processing time, and round-trip delay. The transmission period is the remaining time in a frame for transmission which is $\tau-\tau_1$. Generalized from our previous work [2,4,13], this protocol utilizes the receiver-based handshake (RBHS) procedure before a real-time service is admitted, after that formal transmission of voice packets starts. Data transfers are not allowed until all voice packets are through. The remaining time in a frame period is then allocated to data users. They will contend for transmissions based on Aloha protocol in nature. We call this a Handshake-Aloha (HA) protocol. Since formal transmission of voice packets will start immediately after the control period, transmissions are batch processing in nature.

The handshake procedure is based on the same rationale of CSMA/CD protocol. By the handshake procedure, the intended transmitter is able to test the channel quality using the very short handshake packet. Boundary between voice transmission and data transmission can be either fixed or movable based on voice traffic types. It can be changed on frame-by-frame basis if necessary. This protocol can be furnished with the IEEE 802.11 standard and TCP/UDP/IP protocols.

4. PERFORMANCE EVALUATION

Performance measures will focus on the voice call blocking rate and packet loss rate of real-time packets, in addition to the system throughput. To ensure fairness among users and to guarantee service, the base station plays as a decision maker which is important in allocating the channel resource.

A. PROBABILITY OF BLOCKING VOICE CALLS

An idle voice unit will initiate handshaking according to a Poisson process. The handshake will be successful if no error occurs. To compute the probability of successful handshaking, we will use a primary Markov chain and an auxiliary chain. The first chain, $X_1(t)$, is used to model the behavior of network voice traffic. The states are the number of users who initiate handshaking. We denote the state space as Ω_1. For U_v voice users, there will $|\Omega_1| = U_v + 1$. For convenience, we can index the state 0 as 1, the state 1 as 2, ..., the state U_v as $U_v + 1$. The transition rates diagram is shown in Figure 2. We denote the generator matrix as A_1. Then by solving $\Pi_1 A_1 = 0$, $\sum_i \pi_i = 1$, we can find the steady state probabilities π_i. To calculate the probability of successful handshaking, we focus on a tagged voice user. and form the second Markov chain, $X_2(t)$. The states are the number of handshaking users, plus two absorbing states: *success* and *failure*. The state space is denoted as Ω_2. For convenience, we can index the state *failure* as 1, the state *success* as 2, the state 1 as 3,..., and the state $U_v = U_v + 2$. The transition rates diagram is shown in Figure 3. The generator matrix A_2 can thus be found.

By A_2, we can compute the probability transition matrix P_{t2}, and obtain the probability of entering a specific absorbing state, starting from any transient states. In fact, P_{t2} has the form

$$P_{t2} = \begin{pmatrix} I & 0 \\ R_2 & Q_2 \end{pmatrix}$$

where I is a 2×2 Identity matrix, R_2 is a $(|\Omega_2| - 2) \times 2$ matrix, and Q_2 is a $(|\Omega_2| - 2) \times (|\Omega_2| - 2)$. Let

$$sum(i) = \sum_{\substack{k=1 \\ k \neq i}}^{|\Omega_2|} A_2(i,k)$$

Then, for i=3,4,....., $|\Omega_2|$,

$$R_2(i-2, j) = A_2(i,j) / sum(i), \qquad j = 1,2;$$

$$Q_2(i-2, j) = A_2(i, j+2) / sum(i), j = 1, \ldots, |\Omega_2| - 2.$$

The fundamental matrix is given as

$$N_2 = (I - Q_2)^{-1} \tag{11}$$

Let C_2 be the matrix whose entries are the probabilities from transient states to the absorbing states. In other words,

$C_2(i,1) = \Pr\{$*entering the failure state, starting from transient state i*$\}$

$C_2(i,2) = \Pr\{$entering the success state, starting from transient state $i\}$

Then, we have

$$C_2 = N_2 R_2 \tag{12}$$

Note that C_2 is a $(|\Omega_2| - 2) \times 2$ matrix. Thus, the average probability of successful handshaking is given as

$$P_{success} = \sum_{i=2}^{|\Omega_1|} \pi_1(i) C_2(i+1,2) \tag{13}$$

and the average probability that a voice call is blocked is given as

$$P_{blocked} = \sum_{i=2}^{|\Omega_1|} \pi_1(i) C_2(i+1,1) \tag{14}$$

Obviously, it can be verified that

$$P_{success} = 1 - P_{blocked}$$

We note that with successful handshaking, the voice user will begin transmission at the beginning of the transmission period. Suppose there are A voice users initiate calls during previous frame period. Note that A is a random variable which ranges from 0 to U_v. Since A is an important parameter for the size of the

Markov chain in next section, we have to determine the value of A first. We note the following fact:

$$P_r\{A = a | U_i \text{ active calls among } U_v \text{ users}\}$$

$$= \begin{cases} \binom{U_i}{a} P_{success}^{a} P_{blocked}^{U_i-a}, & if \quad U_v \geq U_i \geq a; \\ 0, & if \quad U_i < a \end{cases}$$

and

$$P_r\{U_i \text{ active calls among } U_v \text{ users}\}$$

$$= \binom{U_v}{U_i} P_v^{U_i}(1-P_v)^{(U_v-U_i)}$$

Thus

$$\Pr\{A = a\} =$$

$$\begin{cases} \sum_{U_i=1}^{U_v} \binom{U_v}{U_i} P_v^{U_i}\left(1-P_v\right)^{(U_v-U_i)} \binom{U_i}{a} P_{success}^{a} P_{blocked}^{U_i-a}, if \; U_i \geq a, \\ 0, & if \; U_i < a. \end{cases} \quad (15)$$

The expectation of A is given as

$$\bar{A} = \sum_{a=1}^{U_v} a \Pr\{A = a\}$$

B. PROBABILITY OF VOICE PACKET LOSS

If the number of successful handshaking is A, then we can form a Markov chain to observe the behavior of a tagged packet. Since the formal transmission of voice packets are a batch arrivals process, the states of this chain are A, A-1, ..., 1, plus three aborbing states: *end*, *failure*, and *success*. We can denote the state space as Ω_3, with size $|\Omega_3| = A+3$. Again, for convenience, we can index the state *end* as 1, the state *failure* as 2, the state *success* as 3, the state 1 as 4, the state 2 as 5, ..., the state A as A+3. The tagged voice packet will finally enter either absorbing state. The transition rates diagram is shown in Figure 4, the expected length of period is $1/\mu_t$. Thus, we can find the generator matrix A_3. The probability transition matrix P_{t3} has the form:

$$P_{t3} = \begin{pmatrix} I & 0 \\ R_3 & Q_3 \end{pmatrix}$$

where I is a 3×3 Identity matrix, R_3 is a ($|\Omega_3|$ -3) $\times$ 3 matrix, and Q_3 is a ($|\Omega_3|$ -3) $\times$ ($|\Omega_3|$ -3). Let

$$sum(i) = \sum_{\substack{k=1 \\ k \neq i}}^{|\Omega_3|} A_3(i,k)$$

Then, for i=4,5,....., $|\Omega_3|$,

$$R_2(i-3, j) = A_3(i,j) / sum(i), \qquad j = 1,2,3;$$

$$Q_2(i-3, j) = A_3(i, j+3) / sum(i), j = 1, \ldots\ldots, |\Omega_3| - 3.$$

The fundamental matrix is given as

$$N_3 = \left(I - Q_3\right)^{-1} \quad (16)$$

Let C_3 be the matrix whose entries are the probabilities from transient states to the absorbing states. In other words,

$C_3(i,1) = \Pr\{$transmission ended before finishing, starting from transient state i $\}$

$C_3(i,2) = \Pr\{$entering the failure state, starting from transient state i $\}$

$C_2(i,3) = \Pr\{$entering the success state, starting from transient state i $\}$

Then, we have

$$C_3 = N_3 R_3 \quad (17)$$

Note that C_3 is a ($|\Omega_3|$ -3) $\times$ 3 matrix. Thus, the average probability that the tagged voice transmission is successful and finished is given as

$$P_{v(success)} = \sum_{a=1}^{U_v} \Pr\{A = a\} C_3(a,3) \quad (18)$$

where $\Pr\{A = a\}$ is given in EQ(15). The average probability that the tagged voice call is lost is given as

$$P_{v(loss)} = \sum_{a=1}^{U_v} \Pr\{A = a\} C_3(a,2) \quad (19)$$

and the average probability that the tagged voice transmission is successful but forced to finish at the end of period is given as

$$P_{v(end)} = \sum_{a=1}^{U_v} \Pr\{A = a\} C_3(a,1) \quad (20)$$

C. PROBABILITY OF DATA PACKET LOSS

Data users are allowed to contend for transmissions during the remaining time of a frame period based on the ALOHA nature. For a frame period of τ, the control period $\tau_1 = 1/\mu_h$, and voice transmission period $\tau_2 = 1/\mu_t$, the remaining period is $\tau_3 = \tau - \tau_1 - \tau_2 = 1/\mu_f$, where we assume the remaining period is also an exponentially distributed with parameter $1/\mu_f$.

We use two Markov chains to model the behavior of the channel activity and the behavior of a tagged data packet respectively. The first chain $X_4(t)$ is one-dimensional, the states are the number of transmitted data packets. The state space is denoted as Ω_4. Obviously, for U_d data users, $|\Omega_4| = U_d + 1$. For convenience, we can index the state 0 as 1, the state 1 as 2, ..., the state U_d as U_d+1. The transition rates diagram is shown in Figure 5. We denote the generator matrix as A_4. Then by solving $\Pi_4 A_4 = 0$, $\sum_i \pi_4(i) = 1$, we can find the steady state probabilities Π_4.

To calculate the probability of loss of a data packet as well as successful transmission, we focus on a tagged data user and form the auxiliary Markov chain, $x_5(t)$. The states of this chain are U_d, U_d -1, ..., 1, plus three aborbing states: *end*, *failure*, and *success*. We can denote the state space as Ω_5, where $|\Omega_5| = U_d + 3$. Again, for convenience, we can index the state *end* as 1, the state *failure* as 2, the state *success* as 3, the state 1 as 4, the state 2 as 5, ..., the state U_d as U_d+3. The tagged data packet will finally enter either absorbing state. The transition rates diagram is shown in Figure 6, where we have assumed that the length of transmission period is exponentially distributed with parameter μ_f. The expected length of period is $1/\mu_f$. Thus, we can find the generator matrix A_5. The probability transition matrix P_{t5} has the form:

$$P_{t5} = \begin{pmatrix} I & 0 \\ R_5 & Q_5 \end{pmatrix}$$

where I is a 3×3 Identity matrix, R_5 is a $(|\Omega_5|-3) \times 3$ matrix, and Q_5 is a $(|\Omega_5|-3) \times (|\Omega_5|-3)$. Let

$$sum(i) = \sum_{\substack{k=1 \\ k \neq i}}^{|\Omega_5|} A_5(i,k)$$

Then, for $i = 4,5, \ldots, |\Omega_5|$,

$R_5(i-3,j) = A_5(i,j)/sum(i), j = 1,2,3;$

$Q_5(i-3,j) = A_5(i,j+3)/sum(i), j = 1,\ldots,|\Omega_5|-3.$

The fundamental matrix is given as

$$N_5 = (I - Q_5)^{-1} \quad (21)$$

Let C_5 be the matrix whose entries are the probabilities from transient states to the absorbing states. In other words,

$C_5(i,1) = \Pr\{$*transmission ended before finishing, starting from transient state* i $\}$

$C_5(i,2) = \Pr\{$*entering the failure state, starting from transient state* i $\}$

$C_5(i,3) = \Pr\{$*entering the success state, starting from transient state* i $\}$

Then, we have

$$C_5 = N_5 R_5 \quad (22)$$

Note that C_5 is a $(|\Omega_5|-3) \times 3$ matrix. Thus, the average probability that a data transmission is successful and finished is given as

$$P_{d(success)} = \sum_{i \in \Omega_4} \pi_4(i) C_5(i,3) \quad (23)$$

and the average probability that a data packet is lost is given as

$$P_{d(lost)} = \sum_{i \in \Omega_4} \pi_4(i) C_5(i,2) \quad (24)$$

and the average probability that a data transmission is successful but forced to finish at the end of period is given as

$$P_{d(end)} = \sum_{i \in \Omega_4} \pi_4(i) C_5(i,1)$$
$$= 1 - P_{d(success)} - P_{d(lost)} \quad (25)$$

D. THROUGHPUT PERFORMANCE

The throughput contributed by a specific user can be defined as

$S_u = \frac{1}{\tau}$(*Time in* τ *spent by a user for successful transmission*)

where τ is the frame period. Let S_{uv} be the contribution from a voice user, and S_{ud} be the contribution from a data user. Then

$S_{uv} = \sum_{i \in \Omega_1} \lambda_v \pi_1(i) \Pr\{$*handshake is successful starting from* i $\} \bullet [\Pr\{$*successful transmission and finished*$\} \bullet \{$*modified average voice packet length*$\} + \Pr\{$*successful transmission but ended before finishing*$\} \bullet \{$*voice transmission period*$\}]$

where $\{$*modified average voice packet length*$\}$ means the average length of those voice packets whose lengths are smaller then the voice transmission period, and

$\{$*voice transmission period*$\} = \frac{1}{\mu_l}$

To find $\{$*modified average voice packet length*$\}$, we note the fact for two exponentially distributed random variables, X_1 and X_2, with parameters μ_v and μ_l respectively, we have

$$\Pr\{X_2 > X_1\} = \frac{\mu_v}{\mu_v + \mu_l}$$

Therefore,

$$\{\text{modified average voice packet length}\} = \frac{\mu_v}{\mu_v + \mu_l} \frac{1}{\mu_v} = \frac{1}{\mu_v + \mu_l}$$

Thus,

$$S_{uv} = \sum_{i \in \Omega_1} \lambda_v \pi_1(i) C_2(i+1,2) [P_{v(success)} \frac{1}{\mu_v + \mu_l} + P_{v(end)} \frac{1}{\mu_l}] \quad (26)$$

Similarly, we have

$$S_{ud} = \sum_{i \in \Omega_4} \lambda_d \pi_4(i) C_5(i+1,3) \frac{1}{\mu_d + \mu_f} + C_5(i+1,1) \frac{1}{\mu_f}] \quad (27)$$

The total throughput contributed by all data users normalized to the frame period is then as

$$S = U_d S_{ud} \tag{28}$$

5. NUMERICAL RESULTS

In the following numerical computations, we assume that there are $U_v = 10$ voice users and $U_d = 10$ data users. The bit error probabilities when different number of signals are transmitted are given as: $P_b(k) = 10^{-12}$, 10^{-11}, 10^{-10}, 10^{-8}, 10^{-7}, 5×10^{-7}, 10^{-6}, 5×10^{-6}, 10^{-5}, for K=1, 2, ..., 10 respectively. These data are then used to calculate $\varepsilon_v(k)$ in EQ(7) and $\varepsilon_d(k)$ in EQ(8), assuming $B_d = 1\mu$ seconds. The average blocking rate of voice calls can be obtained by EQ(14). The result is shown in Figure 7, for blocking rate vs. λ_v.

Figure 8 shows the average voice packets loss rate vs. μ_v by using EQ(19). Figure 9 shows the average data packet loss rate vs. λ_d. Figure 10 shows the average throughput vs. μ_v. Figure 11 shows the average throughput vs. μ_d.

6. CONCLUSION

We have demonstrated an integrated protocol for CDMA wireless LANs. This protocol has fully utilized the characteristic of CDMA techniques, and provides QoS for different service types. Analysis using continuous-time markov chains is demonstrated and numerical results are presented.This protocol can also be adapted to multimedia communications on wireless LANs.

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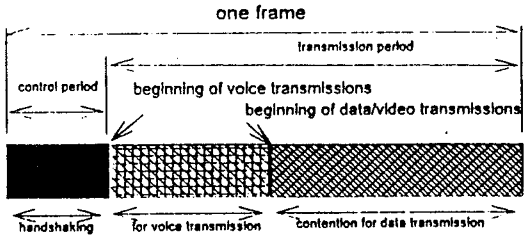


Figure 1. The Handshake-ALOHA(HA) protocol.

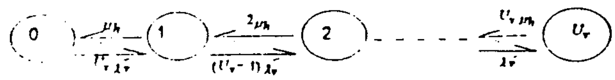


Figure 2. The transition rates diagram of the markov chain for handshaking activities..

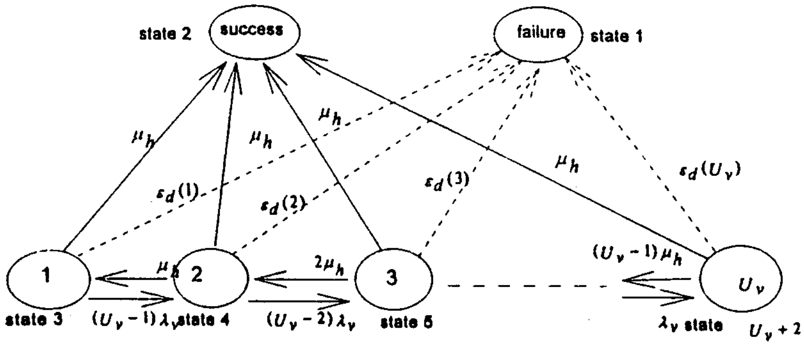


Figure 3. The transition rates diagram of the auxiliary markov chain for handshaking activities.

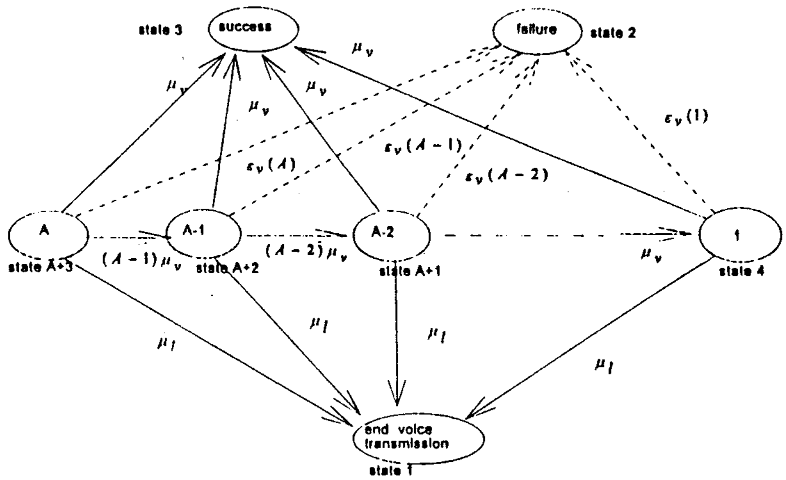


Figure 4. The transition rates diagram of the markov chain for voice transmission.

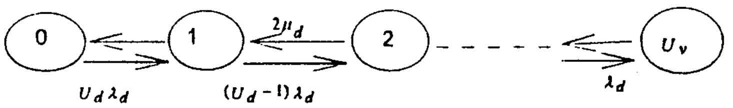


Figure 5. The transition rates diagram of the markov chain for data traffic.

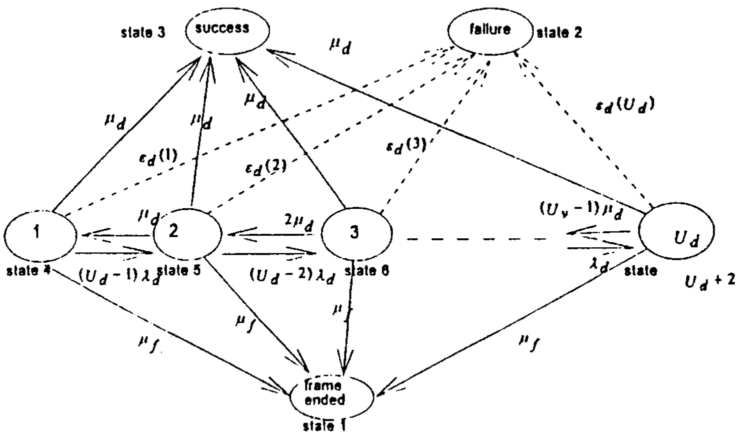


Figure 6. The transition rates diagram of the auxiliary markov chain for data transmission.

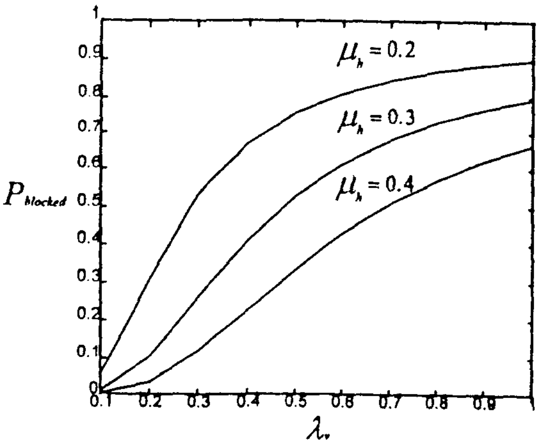


Figure 7. The average voice blocking rate vs. λ_v.

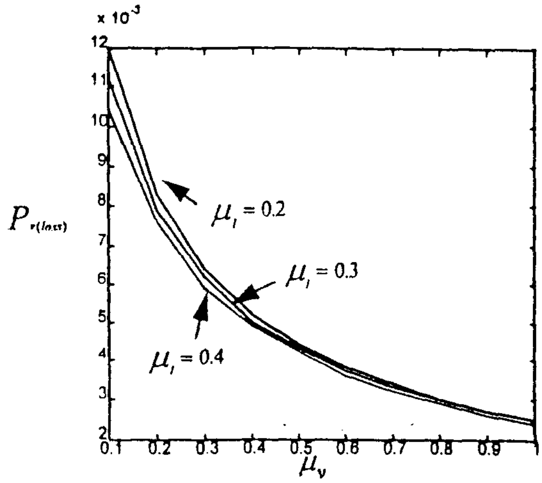


Figure 8. The average voice dropping rate vs. μ_v.

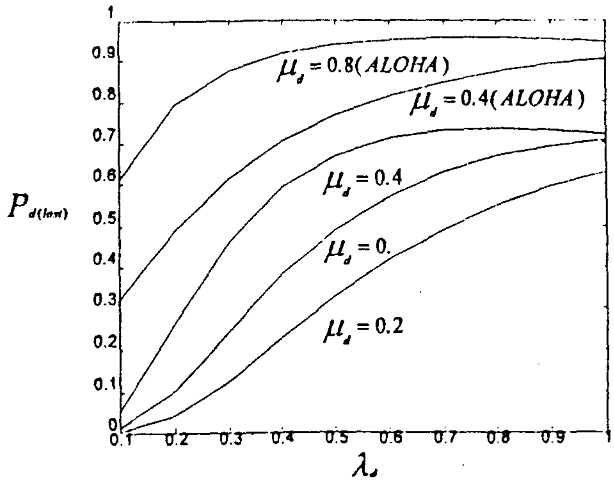
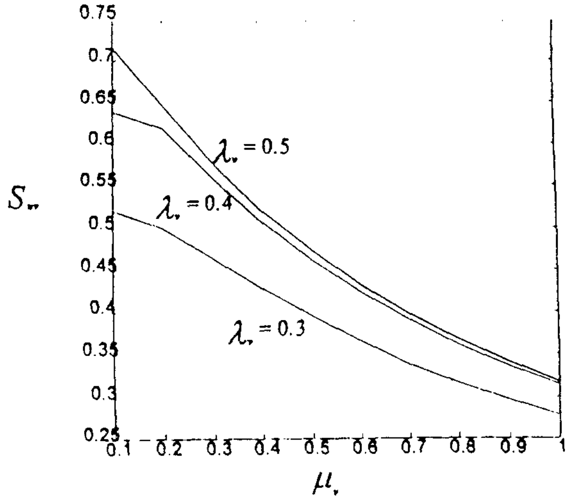


Figure 9. The average data loss rate vs. λ_d where μ_l is fixed at 0.3.



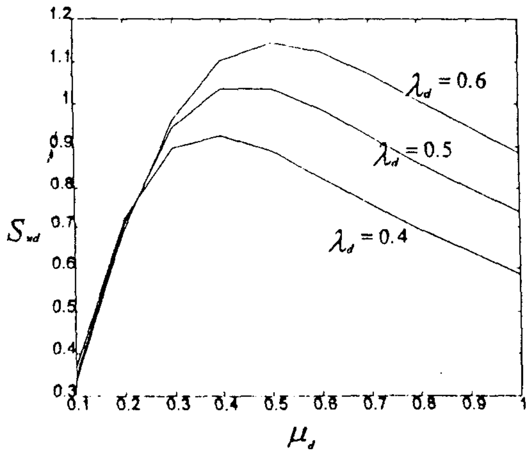


Figure 11. The throughput contributed by data users vs. μ_d with μ_l is fiexd at 0.3..