Performance Analysis of Packet CDMA R-ALOHA for Multi-media Integration in Cellular Systems with Adaptive Access Permission Probability

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ABSTRACT

In this paper, the Packet CDMA Reservation ALOHA protocol is proposed to support the multi-traffic services such as voice and videophone services with handoff calls, high-rate data and low-rate data services efficiently on the multi-rate transmission in uplink cellular systems. The frame structure, composed of the access slot and the transmission slot, and the proposed access permission probability based on the estimated number of contending users for each service are presented to reduce MAI. The assured priority to the voice and the videophone handoff calls is given through higher access permission probability. And through the proposed code assignment scheme, the voice service can be provided without the voice packet dropping probability in the CDMA/PRMA protocols. The code reservation is allowed to the voice and the videophone services. The low-rate data service uses the available codes during the silent periods of voice calls and the remaining codes in the codes assigned to the voice service to utilize codes efficiently. The high-rate data service uses the assigned codes to the high-rate data service and the remaining codes in the codes assigned to the videophone service. Using the Markov-chain subsystem model for each service including the handoff calls in uplink cellular systems, the steady-state performances are simulated and analyzed. After a round of tests for the examples, through the proposed code assignment scheme and the access permission probability, the Packet CDMA Reservation ALOHA protocol can guarantee the priority and the constant QoS for the handoff calls even at large number of contending users. Also, the data services are integrated efficiently on the multi-rate transmission.

I. INTRODUCTION

To realize the next generation mobile communication systems, the MAC (Medium Access Control) scheme that can use the limited radio resources efficiently and provide the required QoSs for multi-media services such as speech, image and data is needed.

To integrate voice and data packets in the packet CDMA schemes, the spreading code reservation is allowed to the voice. In [1], [2], the voice can reserve the allocated code throughout the call duration. So, there are no voice packets dropped. But, the codes cannot be utilized efficiently. In the CDMA/PRMA [3]-[5], the first voice packet of each talk-spurt for the code reservation and the data packet share the time slots to be transmitted on the single-rate transmission according to its given packet permission probability. So, the voice packets and the data packets are inclined to exceed the limited MAI (Multiple Access Interference) with the large packet error probability, And thus, the voice packets that experienced longer delay than the voice delay constraint will be dropped. But, the efficient utilization of codes during the silent periods of the voice calls can be achieved. Then, the packet CDMA scheme which can provide the voice with no packet dropping probability and use the codes during the silent periods efficiently is

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needed. In packet CDMA schemes, the assumption that all users in a cell transmit their packets using a fixedly assigned code is impossible due to the limit of the number of spreading codes and the receiver complexity in the base station. And in [3]-[5], the packet permission probabilities for each traffic need the accommodation of multi-traffic on the multi-rate transmission and the estimation of the number of contending users for each traffic. Also, the handoff calls in cellular systems must be considered and the priority and the required QoS for the handoff calls must be guaranteed through the packet permission probability scheme.

In this paper, The Packet CDMA Reservation ALOHA protocol is proposed to support the multi-traffic services such as voice and videophone services with handoff calls, high-rate data and low-rate data services efficiently on the multi-rate transmission in uplink cellular systems. The code reservation is allowed to the voice and the videophone services. The frame structure composed of the access slot and the transmission slot for the transmission request based code assignment and the proposed access permission probability adaptively given to each service users, based on the estimated number of contending users for each service, are presented to reduce MAI. The assured priority to the voice and the videophone handoff calls is given through higher access permission probability. And through the proposed code assignment scheme, the voice service can be provided without the voice packet dropping probability. The low-rate data service uses the available codes during the silent periods of voice calls and the remaining codes in the codes assigned to the voice service to utilize codes efficiently. The high-rate data service uses the assigned codes to the high-rate data service and the remaining codes in the codes assigned to the videophone service.

Considering the handoff calls in uplink cellular systems, the Markov-chain subsystem models for the voice and the videophone services are defined from the code assignment scheme and their traffic characteristics. And those for the low rate data and the high rate data services are defined. Then, the number of contending users for each service including contending users for the handoff calls is estimated using the EPA (Equilibrium Point Analysis) method [3]. The steady-state performances are simulated and analyzed from the service subsystem models. After a round of tests for the examples, through the proposed code assignment scheme and the access permission probability, the Packet CDMA Reservation ALOHA protocol can guarantee the priority and the constant QoS for the handoff calls even at large number of contending users without the voice packets dropped. Also, the data services are integrated efficiently on the multi-rate transmission.

Ⅱ. SYSTEM MODEL

The proposed protocol can support the multi-traffic services mixed of voice, videophone and data traffic. In this protocol, the time scale is organized in frames containing the access slot and the transmission slot. And the access packet and the traffic packet have the same size as the access slot and the transmission slot as Fig. 1. There are the K_a spreading codes for the access packets and a separate set of the K_t codes for the traffic packets. The $K_a - K_v$ access codes except the K_v access codes for the voice users connected are shared to access. The K_t transmission codes are composed of the K_v codes assigned to the voice service, the K_{vt} codes to the videophone service and the K_d codes to the high-rate data service. The low-rate data service can use the available codes during the silent periods of voice calls to utilize codes efficiently. In the access slot, all users having traffic packets to transmit send the access packets spread by a randomly chosen access code with given access permission probabilities to contend to acquire a transmission code.

If the access packet from a user is received successfully in the access slot, an available transmission code is allocated and the traffic packet spread by the transmission code will be transmitted immediately [1], [6]. But, if a transmission code is not allocated to the user, the traffic packet cannot be transmitted. Therefore, the MAI in the transmission slot can be maintained within the quantity permitted. To reduce the packet collisions and MAI in the access slot, the transmission code reservation using the piggyback in the traffic packet and the proposed access permission probability are presented. All the voice and the videophone service users except the data services can demand the reservation of the acquired code by sending the piggyback including the additional transmission requests to the base station. The base station reserves the allocated codes to the users for the next frame according to the information in the piggyback. Then, the users need not send the access packets at the next frame. Also, the access permission probabilities, decided every frame from the number of available transmission codes for each service and the estimated number of contending users for each service, are broadcast to all users of each service in a cell.

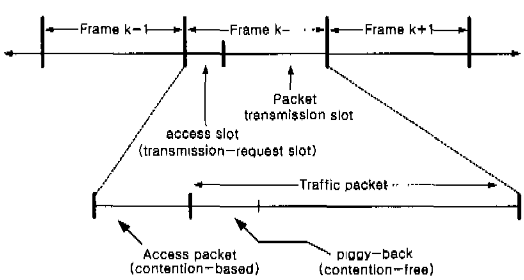
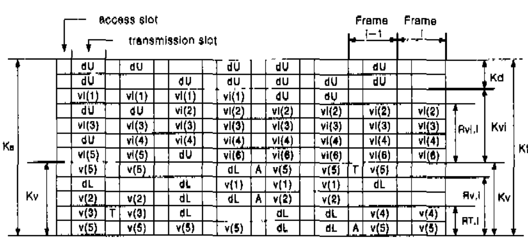


Fig. 1 Frame and slot structure



V(j), Vi(j) the j th voice service, videophone service user   dU the high−rate data service user
A : demand for the reservation of the transmission code used   dL : the low−rate data service user for talkspurts sent from the voice user in silent state
T : indication of the call terminated from the voice user

Fig. 2 Assignment scheme of transmission codes

From Fig. 2, the available codes for voice and videophone, high-rate data and low-rate data services at the ith frame are $K_v - R_{v,i}$, $K_{vi} - R_{vi,i}$, $K_d + (K_{vi} - R_{vi,i})$, $K_v - R_{t,i}$.

The $R_{t,i}$ and the $R_{vi,i}$ are the number of reserved transmission codes for the voice users in the talk-spurt state and that for the videophone users connected at the ith frame. The $R_{v,i}$ is the number of the voice users connected at the ith frame. We set the estimated number of contending users for the voice new call and the handoff call V_{N_v}, V_{H_v}, and that for the videophone new call and the handoff call $V_{N_{vi}}$, $V_{H_{vi}}$, and that for the high-rate data and the low-rate data services b_U, b_L in the uplink cellular systems. Then, the access permission probabilities for the new call and the handoff call of the voice and the videophone services P_{v_N}, P_{v_H}, P_{vi_N}, P_{vi_H} and those for the high- rate data and the low-rate data services P_{d_U} , P_{d_L} are given as (1).

$$P_{v_N} = \min\left\{1, \frac{K_v - R_{v,i}}{V_{N_v}}\right\}, \quad P_{vi_N} = \min\left\{1, \frac{K_{vi} - R_{vi,i}}{V_{N_{vi}}}\right\}$$

$$P_{v_H} = P_{vi_H} = 1 \tag{1}$$

$$P_{d_U} = \min\left\{1, \frac{K_d + (K_{vi} - R_{vi,i})}{b_U}\right\}, \quad P_{d_L} = \min\left\{1, \frac{K_v - R_{t,i}}{b_L}\right\}$$

The assured priority to the handoff calls of the voice and the videophone services is given by permitting the P_{v_H}, P_{vi_H} as 1. Also, the forcedly terminated handoff calls have the same P_{v_H}, P_{vi_H}. Because the available transmission codes are allocated firstly to the voice and the videophone users, the low-rate data service users whose access packets are received successfully can use the remaining codes in $K_v - R_{v,i}$ codes not allocated to the voice users and the $R_{v,i} - R_{t,i}$ codes in the silent periods of voice calls. The high-rate data service users accessed in the access slot can use the remaining codes in $K_{vi} - R_{vi,i}$ codes not allocated to the videophone users.

If the voice and the videophone service users acquire a transmission code in the access slot, the users can use the acquired code exclusively through the piggyback at the subsequent frames during their call duration. During the silent periods of voice calls, the piggybacks will release the allocated codes. So, the codes can be allocated to the low-rate data service users accessed in the access slot. When the voice user enters the talk-spurt state from the silent state, the user transmits the access packet, spread by the fixedly assigned access code in the K_v codes during its call duration, to demand for the reservation of the transmission code used for the talk-spurts. Then, there are no voice packets dropped. When the voice call is over, the voice user transmits the access packet including the indication of the call termination with the fixedly assigned access code. Because the silent state does not exist in the videophone traffic, the release of the allocated code through the piggyback indicates that the videophone call is over.

Ⅲ. SYSTEM ANALYSIS

From the characteristics of each service traffic and the proposed transmission code assignment scheme, the Markov-chain subsystem model for each service can be defined and the number of contending users for each service is estimated using the EPA (Equilibrium Point Analysis) method to evaluate the performance required for each service [7]. The assumption that the packets of each service are generated independently of the other services is used for the Markov -chain subsystem models. The subsystem model for each service interacts with those for other services in the case of the access packet collisions in the access slot.

A. Equilibrium Point Analysis of Voice and Videophone service subsystem

The Markov-chain subsystem models for videophone and voice services are composed of the idle ($IDLE$), the contending (CON) and the connected states as in Fig. 3 and Fig.4. The connected state for the videophone service is expressed as the $RC_{j,vi}$ that a videophone user reserves the jth transmission code in the K_{vi} codes during a its call duration. And the connected state for the voice service during a voice call is expressed as the $RC_{j,v}$. On the basis of the speech On-Off model, the connected state of the $RC_{j,v}$ is composed of the silent state and the $RT_{j,v}$ state that a voice user reserves the jth code in the K_v codes during a talk-spurt period [2], [7], [8]. If a call demand is generated, the user enters the contending state and contends for a transmission code acquisition. If the user accesses successfully in the access slot, the user enters the connected state $RC_{j,v}$ or $RC_{j,vi}$. And if the user moves to the one cell of the adjacent six cells, a handoff call will be generated and enters the contending state (CON, H) in the cell to which the user has moved.

From the subsystem models for the voice and the videophone services, the steady-state equations for the subsystems can be established. The steady-state equations for the videophone service can be made in the same procedure as in those of the voice service given in (2). And the probabilities of a transmission code acquisition for the new call and the handoff call of the voice service and the videophone service f_{v_N}, f_{v_H}, f_{vi_N}, f_{vi_H} are expressed as in (3) and (4) [3].

$$\begin{aligned}
&K_v(r_{v_N}+r_{v_H})p_{T_V}=p_{N_V}i_{v_N}=\lambda_{v_N}\tau && (IDLE)\\
&p_{N_V}i_{v_N}=K_v f_{v_N}V_{N_V} && (CON)\\
&\lambda_{v_N}\tau=K_v f_{v_N}V_{N_V}=K_v r_{v_N}(p_{T_V}+6p_{H_V}) && (RC_{j,v})\\
&\sigma_v r_{s_N}=\gamma_v r_{t_N},\quad r_{s_N}+r_{t_N}=r_{v_N} && (RT_{j,v})\\
&6K_v(r_{v_N}+r_{v_H})p_{H_V}=p_{H_V}i_{v_H}=\lambda_{v_H}\tau && (IDLE, H)\\
&p_{H_V}i_{v_H}=K_v f_{v_H}V_{H_V} && (CON, H)\\
&\lambda_{v_H}\tau=K_v f_{v_H}V_{H_V}=K_v r_{v_H}(p_{T_V}+6p_{H_V}) && (RC_{j,v})\\
&\sigma_v r_{s_H}=\gamma_v r_{t_H},\quad r_{s_H}+r_{t_H}=r_{v_H} && (RT_{j,v})
\end{aligned} \tag{2}$$

$$f_{v_N}=\frac{1}{K_v}P_{v_N}\left(1-\frac{P_{v_N}}{K_a-K_v}\right)^{V_{N_v}-1}\left(1-\frac{P_{v_H}}{K_a-K_v}\right)^{V_{H_v}} \tag{3}$$
$$\times\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)^{V_{N_{vi}}}\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)^{V_{H_{vi}}}\left(1-\frac{P_{d_U}}{K_a-K_v}\right)^{b_U}\left(1-\frac{P_{d_L}}{K_a-K_v}\right)^{b_L}$$

$$f_{v_H}=\frac{1}{K_v}P_{v_H}\left(1-\frac{P_{v_H}}{K_a-K_v}\right)^{V_{H_v}-1}\left(1-\frac{P_{v_N}}{K_a-K_v}\right)^{V_{N_v}}$$
$$\times\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)^{V_{N_{vi}}}\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)^{V_{H_{vi}}}\left(1-\frac{P_{d_U}}{K_a-K_v}\right)^{b_U}\left(1-\frac{P_{d_L}}{K_a-K_v}\right)^{b_L}$$

$$f_{vi_N}=\frac{1}{K_{vi}}P_{vi_N}\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)^{V_{N_{vi}}-1}\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)^{V_{H_{vi}}} \quad (4)$$
$$\times\left(1-\frac{P_{v_N}}{K_a-K_v}\right)^{V_{N_v}}\left(1-\frac{P_{v_H}}{K_a-K_v}\right)^{V_{H_v}}\left(1-\frac{P_{d_U}}{K_a-K_v}\right)^{b_U}\left(1-\frac{P_{d_L}}{K_a-K_v}\right)^{b_L}$$
$$f_{vi_H}=\frac{1}{K_{vi}}P_{vi_H}\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)^{V_{H_{vi}}-1}\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)^{V_{N_{vi}}}$$
$$\times\left(1-\frac{P_{v_N}}{K_a-K_v}\right)^{V_{N_v}}\left(1-\frac{P_{v_H}}{K_a-K_v}\right)^{V_{H_v}}\left(1-\frac{P_{d_U}}{K_a-K_v}\right)^{b_U}\left(1-\frac{P_{d_L}}{K_a-K_v}\right)^{b_L}$$

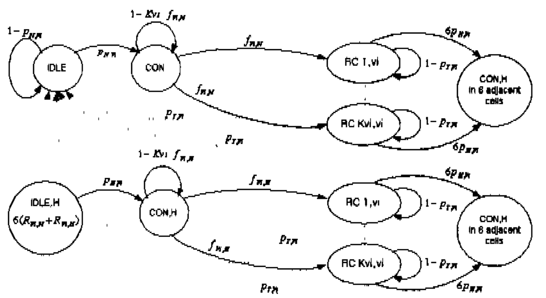


Fig. 3 The videophone service subsystem model

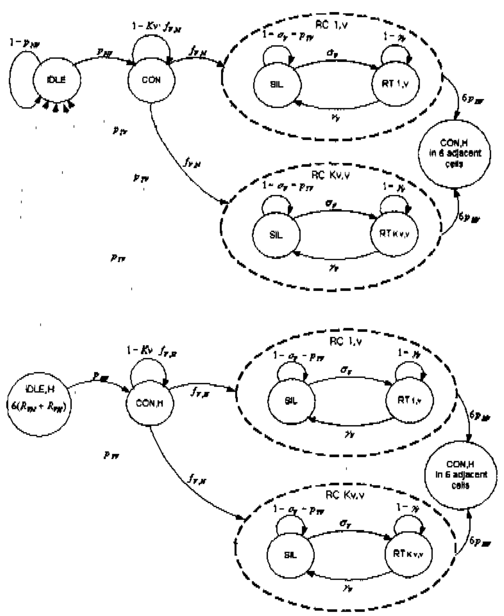


Fig. 4 The voice service subsystem model

i_{v_N} , i_{v_H} , i_{vi_N} , i_{vi_H} : the number of users in *IDLE*, *IDLE,H* of voice and videophone service subsystems

r_{t_N} , r_{t_H} , r_{vi_N} , r_{vi_H} : the reserved ratio of a code in K_v, K_{vi} codes by new calls and handoff calls from voice and videophone services

R_{t_N} , R_{t_H} , R_{vi_N} , R_{vi_H} : the number of users in $RT_{j,v}$, $RC_{j,vi}$ from voice and videophone service subsystems

$R_{t_N}=K_v r_{t_N}$, $R_{t_H}=K_v r_{t_H}$ $(0\le r_{t_N},\ r_{t_H}\le 1)$

$R_{vi_N}=K_{vi} r_{vi_N}$, $R_{vi_H}=K_{vi} r_{vi_H}$ $(0\le r_{vi_N},\ r_{vi_H}\le 1)$

R_{s_N} , R_{s_H} : the number of users in the *SIL* state of new calls and handoff calls from voice service subsystem

$R_{s_N}=K_v r_{s_N}$, $(0\le r_{s_N}\le 1)$, $R_{s_H}=K_v r_{s_H}$, $(0\le r_{s_H}\le 1)$

r_{v_N} , r_{v_H}: the connected ratio of a code in the K_v codes by new calls and handoff calls from voice service subsystem

R_{v_N} , R_{v_H}: the number of connected users in the $RC_{j,v}$ of new calls and handoff calls from voice service subsystem

$R_{v_N}=K_v r_{v_N}$, $(r_{t_N}\le r_{v_N}\le 1)$ $R_{v_H}=K_v r_{v_H}$, $(r_{t_H}\le r_{v_H}\le 1)$

σ_v: a talk-spurt generation probability of voice traffic per frame

$\sigma_v=1-\exp(-\tau/\beta_v)$ τ · length of a frame

β_v : the mean silent duration

γ_v: a talk-spurt termination probability per frame of voice traffic

$\gamma_v=1-\exp(-\tau/a_v)$, a_v: the mean talk-spurt duration

p_{T_V} , $p_{T_{vi}}$: the call termination probability per frame of voice and videophone services

$p_{T_V}=1-\exp(-\tau/t_v)$, $p_{T_{vi}}=1-\exp(-\tau/t_{vi})$

t_v , t_{vi} : mean call holding time of voice and videophone

p_{N_V} , $p_{N_{vi}}$: the new call generation probability per frame of voice and videophone services

$p_{N_V}=1-\exp(-\tau/t_{v_N})$, $p_{N_{vi}}=1-\exp(-\tau/t_{vi_N})$

t_{v_N} , t_{vi_N}: the mean new call inter-arrival time of voice and videophone

p_{H_V} , $p_{H_{vi}}$: the handoff call generation probability per frame to a cell in adjacent six cells of voice and videophone service

$p_{H_V}=(1-\exp(-\tau/t_{v_H}))/6$, $p_{H_{vi}}=(1-\exp(-\tau/t_{vi_H}))/6$

t_{v_H}, t_{vi_H} : mean call dwell time of voice and videophone in cell

λ_{v_N} , λ_{v_H} , λ_{vi_N} , λ_{vi_H} : the arrival rate of new calls and handoff calls from voice and videophone services

In (3) and (4), the probabilities of a transmission code acquisitione $f_{v_N}, f_{v_H}, f_{vi_N}, f_{vi_H}$ are the case that there are no access packet collisions. The access packet collisions do not occurs when all the contending users with the given access permission probability for each service send their access packets spread by different access codes to the access code chosen by the user. Thus, From the previously described, the number of reserved transmission codes, R_t in the $K_v$codes by the voice users in the talk-spurt state and the number of the voice and the videophone users connected R_v, R_{vi} at steady-state are given as in (5). Also, the relationship of the number of the voice new call arrivals during a frame, G_{v_N} to the number of the voice handoff call arrivals, G_{v_H} can be given as in (6) from (2).

$$R_t=R_{t_N}+R_{t_H}\,,\ R_v=R_{v_N}+R_{v_H}\,,\ R_{vi}=R_{vi_N}+R_{vi_H} \quad (5)$$

$$\frac{G_{v_H}}{G_{v_N}}=\frac{\lambda_{v_H}\tau}{\lambda_{v_N}\tau}=\frac{R_{v_H}(p_{T_V}+6p_{H_V})}{R_{v_N}(p_{T_V}+6p_{H_V})}=\frac{i_{v_H}p_{H_V}}{(R_{v_N}+R_{v_H})p_{T_V}}=\frac{6p_{H_V}}{p_{T_V}}$$

$$i_{v_H}=6(R_{v_N}+R_{v_H}),\ p_{H_V}=\frac{p_{T_V}}{6}\cdot\frac{\lambda_{v_H}}{\lambda_{v_N}},\ p_{H_{V_t}}=\frac{p_{T_{V_t}}}{6}\cdot\frac{\lambda_{vi_H}}{\lambda_{vt_N}} \quad (6)$$

B. Equilibrium Point Analysis of Data service subsystem

In Markov-chain subsystem models for the high-rate data and the low-rate data services as in Fig. 5 and Fig. 6., the Birth-Death model is used on the assumption that both more than one packet generation and more than one packet transmission are impossible during a frame. The steady-state equations at the states in the low-rate data service subsystem model can be established in the same procedure as those in the high-rate data service in (7). But the probabilities of the transmission code acquisition for the high-rate data and the low-rate data services w_{d_U}, w_{d_L} are different as in (8). And the b_u can be expressed as in (9) from (7) [3].

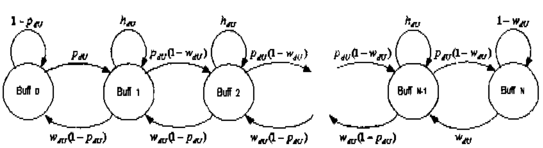


Fig. 5 The high-rate data service subsystem model

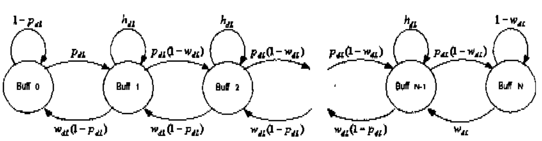


Fig. 6 The low-rate data service subsystem model

$$w_{d_U}(1-p_{d_U})b_1=b_0p_{d_U} \quad (7)$$

$$b_0p_{d_U}+w_{d_U}(1-p_{d_U})b_2=b_1(w_{d_U}(1-p_{d_U})+p_{d_U}(1-w_{d_U}))$$

$$b_{j-1}p_{d_U}(1-w_{d_U})+w_{d_U}(1-p_{d_U})b_{j+1}$$

$$=b_j(w_{d_U}(1-p_{d_U})+p_{d_U}(1-w_{d_U}))$$

$$b_{N-2}p_{d_U}(1-w_{d_U})+b_Nw_{d_U}=b_{N-1}(w_{d_U}(1-p_{d_U})+p_{d_U}(1-w_{d_U}))$$

$$p_{d_U}(1-w_{d_U})b_{N-1}=w_{d_U}b_N \qquad (2\le j\le N-2)$$

$$w_{d_U}=\binom{K_a-K_v}{1}\left(\frac{1}{K_a-K_v}\right)P_{d_U}\left(1-\frac{P_{d_U}}{K_a-K_v}\right)^{b_U-1}\left(1-\frac{P_{d_L}}{K_a-K_v}\right)^{b_L}$$

$$\times\left(1-\frac{P_{v_N}}{K_a-K_v}\right)^{V_{N_v}}\left(1-\frac{P_{v_H}}{K_a-K_v}\right)^{V_{H_v}}\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)^{V_{N_{vt}}}\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)^{V_{H_{vt}}}$$

$$w_{d_L}=\binom{K_a-K_v}{1}\left(\frac{1}{K_a-K_v}\right)P_{d_L}\left(1-\frac{P_{d_L}}{K_a-K_v}\right)^{b_L-1}\left(1-\frac{P_{d_U}}{K_a-K_v}\right)$$

$$\times\left(1-\frac{P_{v_N}}{K_a-K_v}\right)^{V_{N_v}}\left(1-\frac{P_{v_H}}{K_a-K_v}\right)^{V_{H_v}}\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)^{V_{N_{vt}}}\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)^{V_{H_{vt}}} \quad (8)$$

p_{d_U}, p_{d_L} : a packet generation probability per frame of high-rate data and low-rate data services

$p_{d_U}=1-\exp(-\tau/\beta_{d_U})$, $p_{d_L}=1-\exp(-\tau/\beta_{d_L})$

β_{d_U}, β_{d_L}: mean idle time of high-rate data and low-rate data

N : the buffer size of the data service user

b_j : the number of users in $Buff_j$ state that a user has j packets to transmit

b_U, b_L : the number of contending users for high-rate data and low-rate data services

$$b_j=\left[\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right]^{j-1}\frac{p_{d_U}}{w_{d_U}(1-p_{d_U})}b_0 \ ,\quad (1\le j\le N-1)$$

$$b_N=b_{N-1}\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}}$$

$$b_U=\sum_{j=1}^{N}b_j \quad (9)$$

$$=\frac{p_{d_U}}{w_{d_U}(1-p_{d_U})}b_0\left(\frac{1-\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)^{N-1}}{1-\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)}\right)$$

$$+\frac{p_{d_U}}{w_{d_U}(1-p_{d_U})}b_0\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)^{N-2}\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}}$$

C. Estimation of the number of contending users

From M_v, M_{vt} and M_{d_U}, M_{d_L} that are the numbers of users for the voice and the videophone services, the high-rate data and the low-rate data services in a cell, the numbers of users in each states of each service subsystems have the relationships as in (10). Also, the probabilities of a transmission code acquisition for each service have the relationships as in (11), (12).

$$M_v=M_{v_N}+M_{v_H}-i_{v_H},\quad M_{vt}=M_{vt_N}+M_{vi_H}-i_{vi_H}$$

$$M_{v_N}=V_{N_v}+i_{v_N}+R_{v_N},\quad M_{vi_N}=V_{N_{vt}}+i_{vi_N}+R_{vt_N}$$

$$M_{v_H}=V_{H_v}+i_{v_H}+R_{v_H},\quad M_{vt_H}=V_{H_{vt}}+i_{vt_H}+R_{vi_H}$$

$$M_{d_U}=b_{o,U}+b_U,\qquad M_{d_L}=b_{o,L}+b_L \quad (10)$$

$$f_{v_H}=f_{v_N}\frac{\left(\frac{P_{v_H}}{P_{v_N}}\right)\left(1-\frac{P_{v_N}}{K_a-K_v}\right)}{\left(1-\frac{P_{v_H}}{K_a-K_v}\right)}$$

$$f_{vi_H}=f_{vi_N}\frac{\left(\frac{P_{vi_H}}{P_{vi_N}}\right)\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)}{\left(1-\frac{P_{vi_H}}{K_a-K_v}\right)} \quad (11)$$

$$w_{d_U}=f_{vi_N}K_{vi}\frac{\left(\frac{P_{d_U}}{P_{vi_N}}\right)\left(1-\frac{P_{vi_N}}{K_a-K_v}\right)}{\left(1-\frac{P_{d_U}}{K_a-K_v}\right)}$$

$$w_{d_L}=f_{v_N}K_v\frac{\left(\frac{P_{d_L}}{P_{v_N}}\right)\left(1-\frac{P_{v_N}}{K_a-K_v}\right)}{\left(1-\frac{P_{d_L}}{K_a-K_v}\right)} \quad (12)$$

So, the V_{H_v} and the $V_{H_{vi}}$ can be showed in terms of the V_{N_v} and the $V_{N_{vi}}$ from the equations in (1), (2) and (11). Then, the V_{N_v} and the $V_{N_{vi}}$ can be figured out using those in (2), (6) and (10) from the observed new call arrival rates λ_{v_N} , λ_{vi_N} and the user mobility ratios $\lambda_{v_H}/\lambda_{v_N}$, $\lambda_{vi_H}/\lambda_{vi_N}$ and the ratios p_{N_v}/p_{T_v} , $p_{N_{vi}}/p_{T_{vi}}$. Also, b_U and b_L are estimated from those in (9) and (12) assuming the infinite buffer size ($N=\infty$).

D. Blocking probability for voice and videophone call

From the proposed transmission code assignment scheme in the Fig. 2, the voice and the videophone services can be provided without the packet dropping probability on the assumption that the traffic packet transmitted by the user using the allocated code is error-freely received. But the handoff calls experienced longer delay than the handoff call delay constraint will be terminated forcedly. From the equations in (2), (5) and (6) from Fig. 3 and Fig. 4, the blocking probabilities for the new call and the handoff call of the voice and the videophone services P_{B,v_N}, P_{B,v_H}, P_{B,vi_N}, P_{B,vi_H} are given as in (13). Also, the blocking probabilities for total calls of the voice and the videophone services $P_{B,v}$, $P_{B,vi}$ are given with the total offered load of voice service and videophone service G_v, G_{vi}.

$$P_{B,v_N}=\frac{V_{N_v}(1-f_{v_N}K_v)}{V_{N_v}+R_{v_N}-G_{v_N}},\quad P_{B,v_H}=\frac{V_{H_v}(1-f_{v_H}K_v)}{V_{H_v}+R_{v_H}-G_{v_H}}$$

$$P_{B,vi_N}=\frac{V_{N_{vi}}(1-f_{vi_N}K_{vi})}{V_{N_{vi}}+R_{vi_N}-G_{vi_N}},\quad P_{B,vi_H}=\frac{V_{H_{vi}}(1-f_{vi_H}K_{vi})}{V_{H_{vi}}+R_{vi_H}-G_{vi_H}}$$

$$G_v=V_{N_v}+V_{H_v}+R_v-G_{v_N}-G_{v_H},$$

$$G_{vi}=V_{N_{vi}}+V_{H_{vi}}+R_{vi}-G_{vi_N}-G_{vi_H}$$

$$P_{B,v}=\frac{V_{N_v}(1-f_{v_N}K_v)+V_{H_v}(1-f_{v_H}K_v)}{G_v}$$

$$P_{B,vi}=\frac{V_{N_{vi}}(1-f_{vi_N}K_{vi})+V_{H_{vi}}(1-f_{vi_H}K_{vi})}{G_{vi}} \quad (13)$$

To obtain the forced termination probability for the handoff call, firstly, the probabilities of voice and videophone services $P_{w_v}(j)$, $P_{w_{vi}}(j)$ are needed that the user will wait j frames to get the reservation of a transmission code. And the probability takes the geometric distribution as in (14). If we define D as the number of frames of the handoff call delay constraint, the forced termination probabilities for the voice and the videophone handoff calls P_{fT_v}, $P_{fT_{vi}}$ are equal to the probability that a handoff call will wait more than D frames to get the reservation of a code. And the probabilities can be expressed as in (15).

$$P_{w_v}(j)=(P_{B,v_H})^j(1-P_{B,v_H}),\ P_{w_{vi}}(j)=(P_{B,vi_H})^j(1-P_{B,vi_H}) \quad (14)$$

$$P_{fT_v}=\sum_{j=0}^{\infty}P_{w_v}(j)-\sum_{j=0}^{D}P_{w_v}(j)=(P_{B,v_H})^{D+1}$$

$$P_{fT_{vi}}=\sum_{j=0}^{\infty}P_{w_{vi}}(j)-\sum_{j=0}^{D}P_{w_{vi}}(j)=(P_{B,vi_H})^{D+1} \quad (15)$$

E. Data packet average delay D_{av}

If the data service user does not get a transmission code at a frame, the user can store the packet in the buffer. And the mean number of frames i that the backlogged high-rate data packet will wait to be transmitted successfully can be expressed as in (16). Therefore, the high-rate data packet generated in the user at the $Buff_j$ state will experience the mean packet transmission delay time of $(j+1)((1-w_{d_U})/w_{d_U})$ frames. As the probability of being at the $Buff_j$ state in the high- rate data service subsystem is b_j/M_{d_U} , the average transmission delay of the high-rate data packet $D_{av,U}$ and the average delay of the low-rate data packet $D_{av,L}$ can be shown as in (17).

$$Mean[i] = \sum_{i=0}^{\infty} i(1-w_{d_U})^i w_{d_U} = \frac{1-w_{d_U}}{w_{d_U}} \tag{16}$$

$$\begin{aligned} D_{av,U} &= \left(\frac{1-w_{d_U}}{w_{d_U}}\right)\sum_{j=0}^{N-1}\frac{b_j}{M_{d_U}}(j+1) \\ &= \left(\frac{1-w_{d_U}}{w_{d_U}M_{d_U}}\right)\left(\frac{P_{d_U}b_0}{w_{d_U}(1-p_{d_U})}\right) \\ &\quad\times\left(\frac{1-\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)^{N-1}}{1-\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)} - (N-1)\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)^{N-1}\right) \\ &\quad+\left(\frac{1-w_{d_U}}{w_{d_U}M_{d_U}}\right)\left\{M_{d_U}-\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}(1-p_{d_U})}\right)^{N-1} \right. \\ &\quad\left.\times\left(\frac{p_{d_U}}{w_{d_U}(1-p_{d_U})}\right)b_0\left(\frac{p_{d_U}(1-w_{d_U})}{w_{d_U}}\right)\right\} \end{aligned}$$

$$D_{av,U} = \left(\frac{1-w_{d_U}}{w_{d_U}}\right)\left(\frac{1-\rho_{d_U}p_{d_U}}{1-\rho_{d_U}}\right) \ , \ \left(\rho_{d_U} = \frac{p_{d_U}}{w_{d_U}} < 1\right) \ (N=\infty)$$

$$D_{av,L} = \left(\frac{1-w_{d_L}}{w_{d_L}}\right)\left(\frac{1-\rho_{d_L}p_{d_L}}{1-\rho_{d_L}}\right) \ . \ \left(\rho_{d_L} = \frac{p_{d_L}}{w_{d_L}} < 1\right) \tag{17}$$

Ⅳ. NUMERICAL RESULTS

Considering the packet collisions in the access slot, the steady-state performances required for each service in the uplink cellular systems are simulated and analyzed from the service subsystem models. The proposed protocol is simulated in the system that can provide the multi-rate data transmission using the random direct spreading sequence and the BPSK modulation scheme in the B bandwidth. All data rates are the multiples of the lowest bit rate R_n and they are transmitted with the same E_b/N_o. On the assumption that there is no ICI (Inter Cell Interference), the BER of the k th user in K_i users who have transmitted the $R_i$bit rate in the AWGN channel can be expressed as in (18) [9].

Therefore, the proposed protocol can provide the voice and the low-rate data services of 24kbps bit rate with the 10^{-4}BER and the videophone and the high-rate data services of 72kbps bit rate with the 10^{-5}BER by setting K_v, K_{vi}, K_d as 8, 5, 3 (E_b/N_o = 20dB, B = 4.096MHz). Also, the K_a is set 43 to guarantee the 3×10^{-4}BER to the access packet of the 24kbps bit rate from (18).

$$P_{b,i,k} = Q\left(\left[\frac{N_o}{2E_b}+\frac{1}{3N_i}\left(\sum_{j=1}^{n}\frac{R_j}{R_i}K_j-1\right)\right]^{-1/2}\right) \tag{18}$$

$$(B=1/T_c=N_iR_i)$$

$$P_b(K_a) = Q\left(\left[\frac{N_o}{2E_b}+\frac{K_a-1}{3N_{R_n}}\right]^{-1/2}\right) < 3\times10^{-4}$$

The call processing and the transmission code allocation are assumed completed simultaneously and the data service users have the infinite buffer size. The bandwidth B is set 4.096MHz as in UTRA W-CDMA and the frame rate of videophone traffic using the H.263 coding technique is set 50 frames/sec from the frame duration of 20msec [10], [11]. The forced termination probability for the handoff call is equal to the blocking probability because the D is set zero. In the voice service subsystem model, the mean talk-spurt duration α_v and the mean silent duration β_v are set as 1sec and 1.35sec. The voice activity ratio is 0.43 and the mean call holding times of the voice and the videophone services, are set 3min [4], [12]. The mean new call inter-arrival times of the voice and the videophone services are assumed 15min($p_{N_v}/p_{T_v}, p_{N_{vi}}/p_{T_{vi}}$ =0.2). Also, the activity ratio of data service traffic is 0.05 assuming that the mean active time and the mean idle time are 57sec and 1080sec. Then, respectively, the burst data of 4.104Mbits and 1.368Mbits will be generated from the high-rate data user and the low-rate data user [13].

The variation of the $\lambda_{v_H}/\lambda_{v_N}$and the $\lambda_{vi_H}/\lambda_{vi_N}$ratios show the variation of the voice and the videophone service users mobility. That is, the increase of the handoff call arrivals is proportional to the increase of the users mobility. In Fig. 7 for the blocking ratio of the voice service and Fig. 8 for that of the videophone service, the numbers of contending users for the voice service and the videophone service $V_{N_v}+V_{H_v}$, $V_{N_{vi}}+V_{H_{vi}}$

increase as M_v and M_{vi} increase respectively. And the blocking ratio for the new call increases as the handoff call arrivals with higher access permission probability increase. Also, due to the larger V_{H_v}, $V_{H_{vi}}$ when the larger λ_{vH}, λ_{viH}, the blocking ratio for the handoff call increases. But that for the entire calls do not vary and the congestion points that are M_v and M_{vi} , where the number of contending users and the blocking ratio for the services increase radically, are almost constant. From Fig. 7 and Fig. 8, the blocking ratio for the new call increases radically over the congestion point. But the blocking ratio for the handoff call is constant over the congestion point and much smaller than that for the new call. Therefore, the priority and the constant QoS guarantees to the voice and the videophone handoff calls can be provided through the proposed access permission probability but the QoS guarantee to the new calls can't.

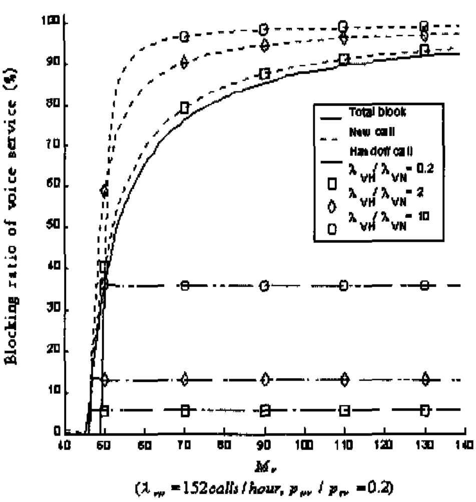


Fig. 7 Blocking ratio of voice service at each ratio of handoff call to new call arrival rate

From the proposed transmission code assignment scheme, the low-rate data users who have accessed successfully, can use the remaining codes in $K_v - R_{v,i}$ codes not allocated to the voice users and the high-rate data users can use the codes in $K_{vi} - R_{vi,i}$ codes not allocated to the videophone users at the i th frame. Therefore, respectively, the

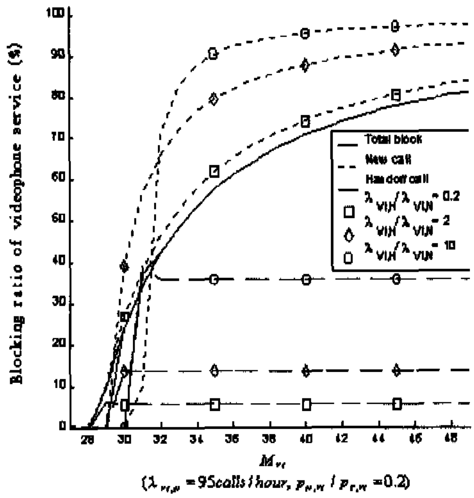


Fig. 8 Blocking ratio of videophone service at each ratio of handoff call to new call arrival rate

average data packet delay performances can be analyzed according to the numbers of contending users for the voice and the videophone service. Figures 9 and 10 show the performances according to M_{d_L} and M_{d_V}.

In Fig. 9, the average delay of the low-rate data service increases exponentially in proportion to the number of contending users for the voice service as M_{d_L} increases, and so does the average delay of the high-rate data service in proportion to that for the videophone service as M_{d_V} increases

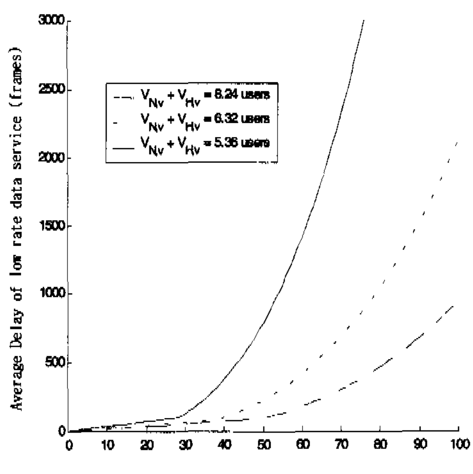
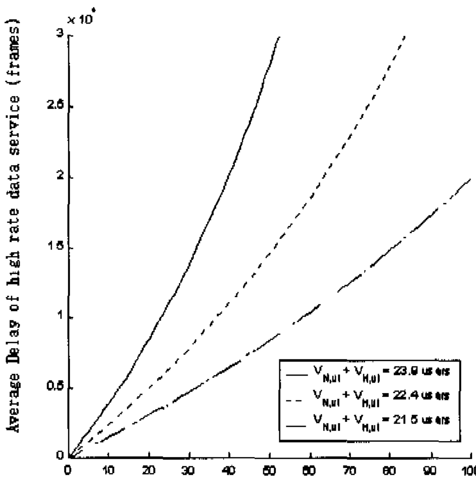


Fig. 9 Average Delay of low-rate data service at each number of contending users for voice service

in Fig. 10. But the average delay of the high-rate data service shows the much larger magnitude and gap than the low-rate data service as to the increase of the number of contending users, which is from the reason that the low-rate data service can be allocated larger codes due to the available transmission codes during the silent periods of voice calls. But, by the adjustment of the K_d codes assigned to the high-rate data service, the improved delay performance can be provided to the high-rate data service.



M_{du}

$(\lambda_{r,N} / \lambda_{r,H} = 0.5, p_{N,R} / p_{r,R} = 0.2)$

Fig. 10 Average Delay of high-rate data service at each number of contending users for videophone service

V. CONCLUSIONS

In this paper, the Distributed Request based CDMA Reservation ALOHA protocol is proposed to support the multi-traffic services such as voice and videophone services with handoff calls, high-rate data and low-rate data services efficiently on the multi-rate transmission in the uplink cellular systems. The frame structure, composed of the access slot and the transmission slot, and the proposed access permission probability based on the estimated number of contending users for each service are presented to reduce MAI. From the steady-state performances, through the proposed code assignment scheme and the access permission probability, the proposed protocol guarantees the priority and the constant QoS for the handoff calls even at large number of contending users, and the voice service can be provided without the voice packet dropping probability in the CDMA/PRMA protocols. Also, the data services are integrated with the efficient code utilization on the multi-rate transmission. And the system capacity, where the required QoS for each service is satisfied, can be obtained at each number of codes assigned to each service and the proposed protocol can support the additional multi-rate services.

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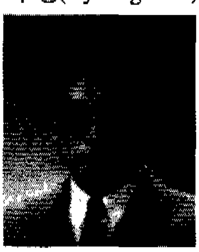

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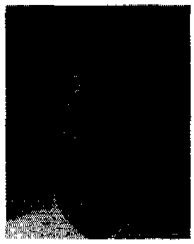

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