

A New Protocol for Multimedia Traffic DS-CDMA

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Abstract: In this study, a media access control protocol is proposed for multimedia traffic in slotted CDMA. The channel resource is proposed into three compartments: video, voice and data. Video packet occupies a whole frame whereas each voice and data packet only occupies a slot in the frame. The concepts of dynamic boundary is adopted in both code and time domains to meet the different requirements for video, voice and data traffic. Data traffic is allowed to use any idle slots of the voice compartment and voice traffic is permitted to use unused video slots but video traffic is not permitted to use any unused slots. An efficient channel borrowing strategy is introduced to both schemes to achieve better Quality of Service (QoS) in a CDMA-based video/voice/data integrated system. The numerical models are developed to evaluate and compare their performances. This model uses the Markov analysis to calculate the video and voice blocking probabilities. The results indicate that an improved performance can be achieved by introducing the dynamic boundary scheme and the dynamic assignment policy results in a performance close to the optimal value.

Key words: Voice compartment, voice traffic, CDMA, video packet, Markov analysis, France

INTRODUCTION

Voice, data and video are three different kinds of traffic which have different service requirements. Voice traffic requires real-time delivery but voice packets show greater tolerance to transmission errors. On the other hand, data traffic does not need real-time delivery but requires very low packet errors probability (Soroushnejad and Geraniotis, 1995). In order to support these different services in one network, using Code Division Multiple Access (CDMA), Direct Sequence (DS)-CDMA is attractive because of its flexibility in supporting multimedia traffic as well as low interference coexistence with other CDMA. The transmission quality depends on the system load: the more calls are admitted to the system, the worse it gets. Obviously, the aim is to admit calls only if the required quality levels can be met but exceeding the appropriate load level somewhat results only in a gradual degradation of quality and may occasionally be tolerated (Brand and Aghvami, 2002). In this study, researchers propose a system using dynamic channel allocation and admission control for integrated video/voice/data services. By scheduling different kinds of traffic, system is able to meet their various requirements. Dynamic boundaries are also adopted to handle traffic efficiently.

MATERIALS AND METHODS

System channel model: In wireless communication system with a centralized topology, the base station and terminals communicate with each other. The base station uses the downlink to broadcast control traffic and information traffic to terminals while the uplink channel is used by the mobile terminals for packet transmission, adopting an appropriate CDMA protocol. The CDMA network considered here is completely slotted. This research therefore, focuses on the uplink channel. In a CDMA System, a physical channel can accommodate a number of logical channels simultaneously by employing different pseudo-random codes (Lee and Miller, 1998).

Each logical channel is divided into frames. The basic unit for resource allocation is one time slot per code which is defined as a channel. In general, a video traffic is expected to have a lower call arrival rate than that of a voice and a video call requires a much higher bandwidth than a voice call. Suppose that the transmission rates of a voice and a video are R_v and R_s bit sec^{-1} , respectively. A voice call (or a data packet) occupies one channel (one time slot per code) and a video call occupies a multiple channels, l channels in a CDMA frame and we have $l = R_v/R_s$. For the voice service, each voice

source generates exactly one packet in each frame periodically, i.e., the length of one frame is equal to the duty cycle of voice packets. Researchers define R_p to be the peak rate supported by the channel; therefore video traffic is transmitted at R_p (Soong and Soh, 1998).

In a frame, the channel is able to deliver $R_p T_f$ bits where T_f is the duration of one frame. Each video packet occupies the whole frame (i.e., all the slots in the frame). Now, a frame consists of N slots and each slot carries one packet, thus according to the number of slots per frame becomes:

$$N = \frac{R_p T_f}{R_s T_f + H}$$

With H bits long header attached to each data packet. Each frame is divided into N slots in time domain (Fig. 1). In code domain, K_v , K_e and K_d is the maximum number of simultaneous transmission per slot for video, voice and data so that the expected packet error probabilities remain below their specified values, respectively where $K_v > K_e > K_d$ (Soroushnejad and Geraniotis, 1995). K_r is the number of codes used only for the reservation purpose. The collision introduced by the request packets transmitted in the same minislot using the same code is neglected here due to the small probability of its occurrence (Fleming and Xu, 1994). When there is a video traffic in order to meet the most stringent BER requirement of the video, each slot can only accommodate K_e (rather than K_v , otherwise the BER requirement of the video will be violated) simultaneous video and voice transmissions (Cheng *et al.*, 2000). In order to transmit multimedia services efficiently, two thresholds V_{max} and E_{max} are introduced. E_{max} divides the vertical channel resource into two compartments, one for video transmission and the other for voice/data transmission as shown in Fig. 1.

E_{max} is the maximum number of simultaneous video calls allowed in the system. After the video call assignment, the idle channels in the video compartment can be used by voice and data. Researchers assume $E_{max} < K_e$ so that some channels are still reserved for the voice transmission even when the video traffic is heavy. The voice/data compartment is divided into voice compartment and data compartment by V_{max} which is the maximum number of slots that can be assigned to voice terminals in each frame. Two types of packets are distinguished: request and information packets (Fig. 2). The information packets can be categorized into video, voice and data packets according to their message types. Voice and data information packets are of the same length which is equal to the duration of one normal slot while the length of a video packet is equal to a frame. Each information packet contains a synchronization preamble

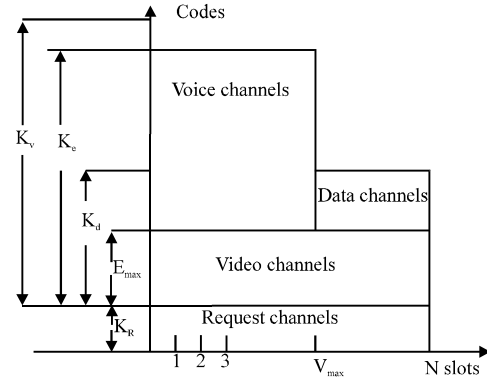


Fig. 1: Two dimensional channel resources for video/voice/data integrated traffic

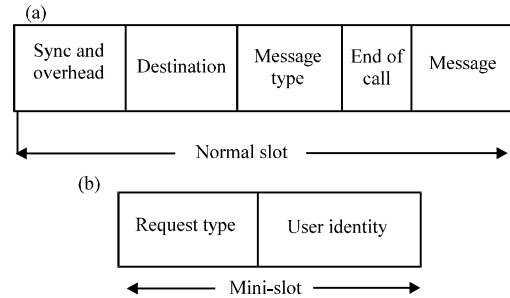


Fig. 2: Request packet; a) information packets and b) request packets

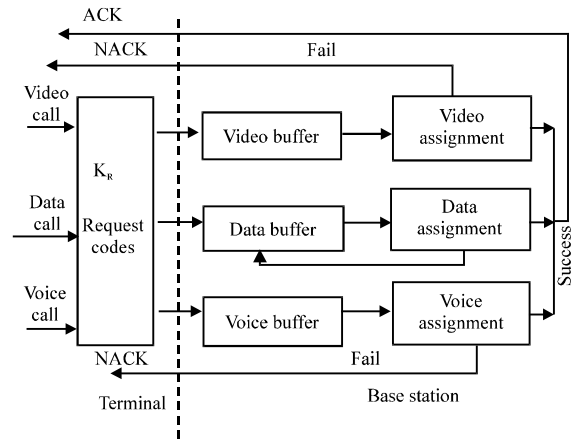


Fig. 3: Video/voice/data transmission process

with other overhead information, the destination address of the mobile receiver, the type of message and the message. In a video information packet, the message portion consists of both video and voice information. The reservation packet which is used by a user to request channel assignment, contains only the type of the required service (video, voice or data transmission) and the user identity (Soong and Soh, 1998) (Fig. 3).

Packet transmission process: A video, voice or data terminal tries to call or transmits a data message, a request packet using one of the K_R reservation codes is generated and transmitted in a randomly selected mini-slot (Wilson *et al.*, 1993):

- Each request packet arrives in frame (n - 1) at the base station which stores its packets into either the FIFO buffers of video, voice or data
- At the base station, at the end of the frame (n - 1), the mobile terminals (voice, data and video) are assigned a code and a slot
- Each code can be identified as available (ACK) or reserved (NACK) based on the feedback information from the base station at the beginning of frame n
- If the video or voice terminal receives ACK, it will use the assigned slot and code to transmit in the subsequent frames until the end of the call
- When a video or voice terminal fails to reserve a channel, it will receive NACK from the base station to inform it to discard the call due to a collision
- For a data terminal after receiving ACK, it will transmit the message using information packets in the assigned slots of frame (n+1) with the assigned codes
- All data request packets which fail to reserve the channel in the current frame will remain in the system until there are channels available for their delivery
- The duration of one frame must be larger than the round-trip propagation delay R (Wieselthier and Ephremides, 1995) to ensure that the base station can receive the information packets from the successful terminals in the assigned channels of frame (n+1) after it sends acknowledgement packets to them

Operation of multimedia subsystem: The base station will give the priority to video terminal over voice users and voice over data users in the channel assignment procedure. When all video (voice) channels are reserved then the video (voice) call request into buffer is dropped. In each frame, there are some ongoing video and voice calls. The base station has the channel resource information $N(q)$ of all slots where $N(q)$ is the total number of voice, data and video terminals transmitting in slot q. The $N(q)$ of frame (n+1) is required for the channel assignment which takes place at the end of frame (n - 1). The initial value of $N(q)$ of frame (n+1) can be easily determined by the number of total voice and video ongoing calls minus the calls with EOC = 1 in frame (n - 1). The base station also has the number of reserved video call channels (N_e) of frame (n+1). Before the video call assignment at the end of frame (n - 1), N_e includes only the number of ongoing video calls minus the video calls with EOC = 1 in frame (n - 1).

Video subsystem: In the video call assignment at the end of frame (n - 1), the base station checks the N_e if $N_e < E_{max}$, the base station will assign the channels to the new arrivals, until $N_e = E_{max}$.

Voice subsystem: Then in the voice call assignment after checking the channel resource in frame (n+1), the base station will assign the first slot to the new voice arrivals until $N(1) = K_v$ if $N_e = 0$ or until $N(1) = K_e$ when $N_e > 0$, for $K_e < K_v$. Afterwards, the base station will assign the second slot in the same way until it reaches the slot V_{max} th.

Data subsystem: The base station will assign channels to the data terminals from the first slot to N-th slot. The base station will check $N(q)$ and it will not assign slot (q+1) to a data packet until $N(q) = K_d$. Once a video call arrives or departs or a voice terminal departs, rearrangement of channels is required to make sure that all voice calls occupy the lower number of slots, in order to give more room for data transmission.

ANALYSIS OF THE MULTIMEDIA INTEGRATION PROTOCOL

System model: The users in each cell are divided into three classes of users: M_e video terminals, M_v voice terminals and M_d data terminals, a voice terminal generates new voice calls at the rate λ_v (calls/slot/terminal) and a video terminal generates new video calls with a Poisson rate λ_e (calls/slot/terminal) while a data terminal generates new message with a Poisson rate λ_d (packets/slot/terminal). There are M_v voice terminals holding conversations simultaneously in the system. The speech source is described as a pattern of talk spurts and gaps which is a slow Voice Activity Detector (VAD). The talk spurts have the average duration of $1/\lambda_v$ and the gaps have the average duration of $1/\mu_v$ where λ_v (calls/slot/terminal) is the poisson arrival rate of each terminal and μ_v (calls/slot/terminal) is the poisson leaving rate of each terminal. They are all exponentially distributed processes and statistically independent of each other. P_{gv} and P_{ev} are defined as the probabilities for an idle terminal to generate a call and for a terminal to end a voice call during a slot, respectively. That is:

$$P_{gv} = 1 - e^{-\lambda_v} \quad (2)$$

$$P_{ev} = 1 - e^{-\mu_v} \quad (3)$$

P_{gv}^f and P_{ev}^f are defined as the probabilities for an idle terminal to generate a call and for a terminal to end a voice call during a frame, respectively.

$$\begin{aligned} P_{gv}^f &= 1 - e^{-\lambda_v N} \\ P_{ev}^f &= 1 - e^{-\mu_v N} \end{aligned} \quad (4)$$

There are M_e video terminals in the system, each generates new video calls with the Poisson rate λ_e (calls/slot/terminal) and departs from the system with the Poisson rate μ_e (calls/slot/terminal). Researchers define P_{ge} and P_{ee} as the probabilities for an idle terminal to generate a call and for a terminal to end its call during a slot, respectively:

$$P_{ge} = 1 - e^{-\lambda_e} \quad (5)$$

$$P_{ee} = 1 - e^{-\mu_e} \quad (6)$$

P_{ge} and P_{ee} are defined as the probabilities for an idle terminal to generate a call and for a terminal to end a video call during a frame, respectively. That is:

$$P_{ge}^f = 1 - e^{-\lambda_e N} \quad (7)$$

$$P_{ee}^f = 1 - e^{-\mu_e N} \quad (8)$$

Video blocking probability: The video process is independent on the voice, the performance of the threshold control scheme is analyzed using a one-dimensional Markovian model (Cheng *et al.*, 2000; Kleinrock, 1976) and the video blocking probability is evaluated. Firstly, researchers define:

- X_n = The total number of the video ongoing calls in frame n
- Y_n = The total number of the video calls ending in frame n
- Z_n = The total number of the video calls arrivals in frame n

The system is depicted as a one-dimensional Markov Chain Model X_n . We define Π_x as the steady state probability of X_n and it can be determined by the following equations:

$$\begin{cases} \pi_x = \pi_x P_x \\ \sum_{x=0}^{E_{max}} \pi_x = 1 \end{cases} \quad (9)$$

where, P_x is the transition probability matrix of size $(E_{max}+1) \times (E_{max}+1)$ with elements P_{ij}^x which is defined as the one step transition probability from the current state $X_n = i$, to the next state $X_{n+1} = j$ that is:

$$P_{ij}^x = \Pr \left\{ \begin{array}{l} X_{n+1} = j \\ X_n = i \end{array} \right\} \quad (10)$$

where, $0 \leq j \leq E_{max}$ and $0 \leq i \leq E_{max}$. The number of video calls in the system during frame $(n+1)$ is (Wieselthier and Ephremides, 1995):

$$X_{n+1} = \min \{ (X_n + Z_{n+1} - Y_n), E_{max} \} \quad (11)$$

Where:

- Z_{n+1} = The number of new video call arrivals in frame $(n+1)$
- Y_n = The number of video calls that are completed in frame n

The number of video calls in the system during frame $(n+1)$ is composed of X_n ongoing calls, Z_{n+1} new arrivals which are contending for transmission in frame $(n+1)$ and less Y_n ending video calls in frame n . Researchers also note that at most (E_{max}) video calls can be accommodated simultaneously in a frame. In the n th frame, the probability of a new video call arrival is dependent on the number of the ongoing video calls that is:

$$\Pr(Z_{n+1} = z / X_n = x) = \beta(M_E - x, z, P_{ge}^f) \quad (12)$$

Where:

$$\beta(M, n, p) = \begin{cases} \binom{M}{N} p^N (1-p)^{M-N} & \text{if } 0 \leq n \leq M \\ 0 & \text{elsewhere} \end{cases} \quad (13)$$

$\beta(M, n, p)$ is the binominal distribution with parameters M, n and p where $0 \leq p \leq 1$. Also in each frame, every active video terminal finishes its call with the probability P_{ee}^f , hence researchers have:

$$\Pr(Y_n = y / X_n = x) = \beta(x, y, P_{ee}^f) \quad (14)$$

Transition probability for voice call process is determined as follows. Firstly, we consider $X_{n+1} < E_{max}$ when all video phone calls succeeded in channel reservation, that is:

$$\begin{aligned} P_{ij}^x &= \Pr(X_{n+1} = j / X_n = i) \\ &= \sum_{y=\max(0, i-j)}^i \Pr(Z_{n+1} = y + j - i / X_n = i) \\ &\quad \Pr(Y_n = y / X_n = i) \end{aligned} \quad (15)$$

Secondly, researchers consider $X_n \leq X_{n+1} = E_{max}$. In this case, all video channels were assigned to video terminals in frame $(n+1)$ and some video calls might be blocked,

since the number of new video call arrivals exceeded what the system can accommodate. Thus, we have:

$$P_{ij}^x = \Pr(X_{n+1} = j / X_n = i) \\ = \sum_{j=E_{\max}}^{M_F} \sum_{y=0}^i \Pr(Z_{n-1} = y + j - i / X_n = i) \\ \Pr(Y_n = y / X_n = i) \quad (16)$$

The performance measure used for video traffic is the video call blocking probability. Whenever, a slot is not available for a new call, the call is blocked and dropped from the system. Researchers define:

$$P_{\text{block}}^x = \frac{E(B_{x_n})}{E(X_n)}$$

Where:

$E(B_{x_n})$ = The expected number of blocked video phone calls in a frame

$E(X_n)$ = The average number of video calls arriving in a frame

Now $E(X_n)$ and $E(B_{x_n})$ can be determined as follows:

$$E(X_n) = \sum_{x=0}^{E_{\max}} \sum_{z=0}^{M_k-x} z \Pr(Z_{n-1} = z / X_n = x) \pi_x \quad (18)$$

$$E(B_{x_n}) = \sum_{x=0}^{E_{\max}} \sum_{b=0}^{M_k-E_{\max}} \sum_{y=0}^x P \quad (19)$$

$$P = \Pr(B_{x_n} = b / X_n = x, Y_n = y) \\ \Pr(Y_n = y / X_n = x) \pi$$

Voice blocking probability: Since, the voice process is dependent on the video, a two dimensional Markov process (Kleinrock, 1976) is established and the voice blocking probability is evaluated. Firstly, researchers define:

V_n = The total number of the voice ongoing calls in frame n

W_n = The total number of the voice calls ending in frame n

C_n = The total number of the voice calls arrivals in frame n

The system is depicted as a two-dimensional Markov chain model (V_n, X_n) . The steady state probability $\pi_{x,v}$ must satisfy:

$$\begin{cases} \pi_{x,v} P_{x,v} = \pi_{x,v} \\ \sum_{x=0}^{E_{\max}} \sum_{v=0}^{N_v^x} \pi_{x,v} = 1 \end{cases} \quad (20)$$

Where $P_{v,x}$ is the transition probability matrix with elements:

$$P_{b,j;a,i}^{v,x} = \Pr(V_{n+1} = b, X_{n+1} = j / V_n = a, X_n = i) \\ = \Pr(X_{n+1} = j / X_n = i) \Pr(V_{n+1} = b / V_n = a, X_n = i) \quad (21)$$

and N_v^x is defined as the number of the channels available for the voice traffic in a frame when there are x video ongoing calls in the system:

$$N_v^x = \begin{cases} K_v V_{\max} & x = 0 \\ K_v (K_E - x) & 0 < x \leq K_E \\ 0 & \text{otherwise} \end{cases} \quad (22)$$

In frame n - 1, C_{n-1} new voice arrivals request for channel assignment. In frame n, the base station will acknowledge them ACK if there are available voice channels, thus the terminals will begin to transmit voice packets in frame n+1.

Otherwise, the blocked calls will be dropped from the system. Therefore, the number of the voice calls in the system during frame n+1 is composed of V_n ongoing calls, C_{n-1} new arriving voice calls in frame n - 1, less W_n ending voice calls in frame n. Researchers have:

$$V_{n+1} = \min\{(V_n + C_{n-1} + W_n), N_v^x\} \quad (23)$$

The new voice calls, C_{n-1} are arriving dependently on the ongoing calls V_n :

$$\Pr(C_{n-1} = c / V_n = a) \beta(M_v - a, c, P_{gv}^f) \quad (24)$$

and every active voice terminal finishes its call with the probability P_{ev}^f . Hence we have:

$$\Pr(W_n = w / V_n = a) = \beta(a, w, P_{ev}^f) \quad (25)$$

The transition probability $\Pr(V_{n+1} = b / V_n = a, X_n = j)$ is evaluated by these conditions. If $V_{n+1} < N_v^x$ all voice requests succeed in channel reservation:

$$\begin{aligned}
 & \Pr(V_{n+1} = b / V_n = a, X_n = j) \\
 &= \sum_{w=\max(0, a-b)}^a \Pr(C_{n-1} = w + b - a / V_n = a) \\
 & \quad \Pr(W_n = w / V_n = a) \\
 &= \sum_{w=\max(0, a-b)}^a \beta(M_v - a, w + b - a, P_{gv}^f) \beta(a, w, P_{ev}^f)
 \end{aligned} \tag{26}$$

When $V_n < V_{n+1} = N_v^x$, all voice channels are occupied and some voice calls may be blocked, since the number of voice call arrivals exceeds what the system accommodate:

$$\begin{aligned}
 & \Pr(V_{n+1} = b / V_n = a, X_n = j) \\
 &= \sum_{b=N_v^x}^{M_v} \sum_{w=0}^a \beta(M_v - a, w + b - a, P_{gv}^f) \beta(a, w, P_{ev}^f)
 \end{aligned} \tag{27}$$

Since, the voice call performance is dependent on that of the video (Eq. 28) thus:

$$P_{\text{voice}} = \frac{E(B_{V_n})}{E(V_n)} \tag{28}$$

Where:

- $E(B_{V_n})$ = The expected number of the blocked voice calls in a frame
- $E(V_n)$ = The average number of the voice arriving in a frame

$$E(V_n) = \sum_{v=0}^{V_{\max}(K_{v-x})} \sum_{c=1}^{M_v - v} c \Pr(C_{n-1} = c / V_n = v) \pi_{v,x} \tag{30}$$

$$E(B_{V_n}) = \sum_{v=0}^{V_{\max}(K_{v-x})} \sum_{b=1}^{M_v - N_v^x} \sum_{w=0}^v PA \tag{31}$$

$$\begin{aligned}
 PA &= \Pr(B_{V_n} = b / V_n = v, W_n = w) \\
 & \quad \Pr(W_n = w / V_n = v) \pi_{v,x}
 \end{aligned}$$

$$E(B_{V_n}) = \sum_{v=0}^{V_{\max}(K_{v-x})} \sum_{b=1}^{M_v - N_v^x} PB \tag{32}$$

$$PB = \sum_{w=0}^v \beta(M_v - v, N_v^x + w + b, P_{gv}^f) \beta(v, w, P_{ev}^f) \pi_{v,x}$$

RESULTS AND DISCUSSION

The numerical evaluation results are given in terms of video call blocking probability and voice call blocking probability. In the numerical results, researchers assume a DS/CDMA system, employing a BPSK modulation with a coherent demodulation and a rate 1/2, constraint length 3, binary convolutional code with a hard decision Viterbi

decoding. As Pursley and Taipale (1987) can obtain an upper bound on the packet error probability P_E as follows:

$$P_E \leq 1 - [1 - P_u(\rho)]^L \tag{33}$$

where, L is the packet length and $P_u(\rho)$ is obtained from:

$$P_u(\rho) \leq \Gamma_{n0} \left\{ \frac{1}{2} [T(D) + T(-D)] + \frac{1}{2} D [T(D) + T(-D)] \right\}_{D=2\sqrt{\rho}} \tag{34}$$

Where:

$$\Gamma_{n0} = \binom{2n_0 - 1}{n_0} 2^{-2n_0}$$

$T(D)$ (Pursley and Taipale, 1987; Conan, 1984) is the generation function and n_0 is half of the free distance of the code (or half of the free distance plus 1 if the distance is odd). The symbol error probability ρ under a Gaussian noise channel from Pursley and Taipale (1987) is given by:

$$\rho \approx Q \left(\left[\frac{N_0}{E_b} + \frac{(K-1)}{S_f} \right]^{-1/2} \right) \tag{35}$$

Where:

- E_b = The energy per information bit in the received signal
- $N_0/2$ = The two-sided spectral density of the noise
- K = The number of packets transmitted simultaneously
- S_f = The number of chips per information bit (spreading factor)

In the voice system, each data packet is 160 bits long with 127 chips per bit and (E_b/N_0) is 12 dB. The following parameters are chosen for illustrative purposes only $K_v = 9$ and $K_v = 11$ which correspond to a packet error probability of 7.93×10^{-5} (for video) and 3.4×10^{-4} (for voice). Each video call occupies the whole frame and data rate is 86.4 Kbps (1728 bits/20 ms). The system parameters are shown in Table 1.

Figure 4 shows the video blocking probability with E_{\max} ranging from 1-5 vs., the video arriving rate λ_v . It can be seen that the video blocking probability increases when the video arriving rate increases which means the offered load of video increases. It is also noticed that video blocking probability decreases when E_{\max} increases which means more channels are available for video transmission. Figure 5 shows the blocking probability of voice versus the offered load of number of voice conversations for various V_{\max} .

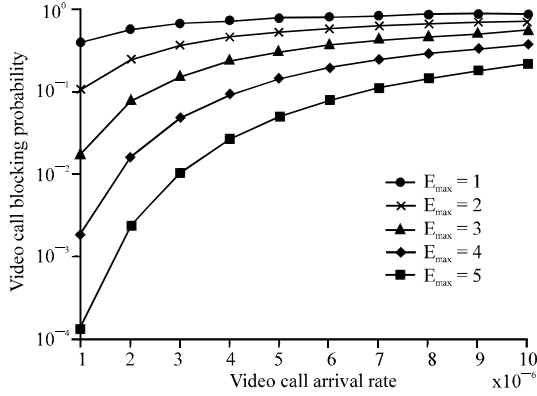


Fig. 4: Video blocking probability as function of video arrival rate λ_e for various E_{max} ($M_e = 10$)

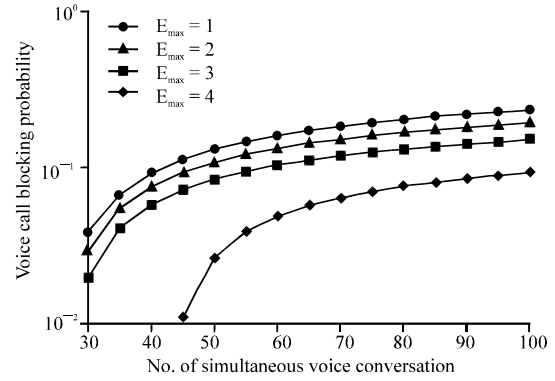


Fig. 6: Voice call blocking probability as a function of number of simultaneous voice conversations for various E_{max} ($M_e = 10$, $V_{max} = 3$, $\lambda_e = 3 \times 10^{-6}$)

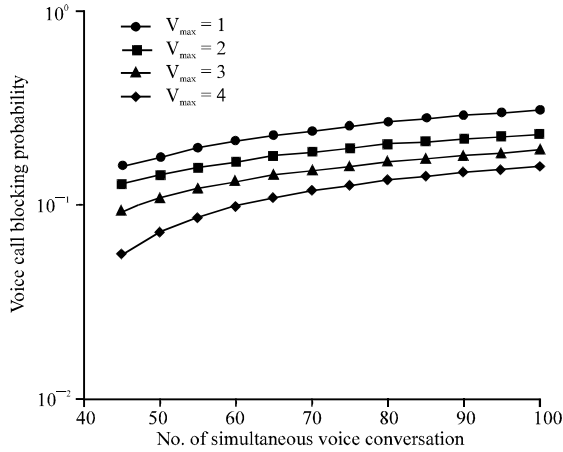


Fig. 5: Voice call blocking probability as a function of number of simultaneous voice conversations for various V_{max} ($M_e = 10$, $E_{max} = 3$, $\lambda_e = 3 \times 10^{-6}$)

It shows that the blocking probability of voice calls increases with the increase of number of simultaneous voice conversations traffic load. When V_{max} increases, more channels are available for voice terminals. Therefore, the blocking probability of voice call decreases with E_{max} and video traffic load are fixed.

From Fig. 6, it can be seen that the voice call blocking probability increases when the number of voice conversations for various E_{max} increases. This is due assigning more channels to the video transmissions when E_{max} elevated which mean less channels are available for voice calls.

CONCLUSION

In this study, researchers have proposed a slotted CDMA protocol for the multimedia (video\voice\data) traffic of the future wireless communication networks. Two dynamic boundaries V_{max} and E_{max} are introduced in both code and time domains to guarantee users with different sources rates (video\voice\data) to transmit packets effectively in the same network. These boundaries also prevent sharp performance degradation of the lower priority service when the high priority service is under heavy traffic.

In addition, the boundary V_{max} separates voice and data service into compartments, therefore they do not need to suffer the same packet error probability as occurs classical CDMA systems. Furthermore, the system can achieve an optimal performance, by adjusting V_{max} and E_{max} . The analytical model, Markov analysis is derived to serve as a building block for the mathematical evaluation of its performance. The Markov analysis is used to evaluate the video and the voice call blocking probabilities.

Table 1: System parameters of slotted CDMA

Definition	Symbols	Values
Energy per bit to noise power spectral density (dB)	E_b/N_0	12
CDMA channel rate (kb sec ⁻¹)	R_c	22937.6
Spreading factor	S_f	128
Channel rate after coding (kb sec ⁻¹)	R_{ac}	179.2
Channel rate before coding (kb sec ⁻¹)	R_p	89.6
Header per packet (bit)	H	64
Voice source rate (kb sec ⁻¹)	R_s	8
Video source rate (kb sec ⁻¹)	R_v	86.4
Information bit per packet (bit)	$R_p T_f$	160
Frame duration (sec)	T_f	0.02
Number of the slots per frame	N	8
Total number of video terminals	M_e	10
Total number of request channels	K_R	1
Average duration of talk spurt (sec)	$T_f/\lambda_v N$	1
Average duration of silent gaps (sec)	$T_f/\mu_v N$	1.35
Average duration of video call (sec)	$T_f/\mu_e N$	180
Video packet error probability requirement	P_E^v	7.93×10^{-1}
Voice packet error probability requirement	P_E^v	3.4×10^{-4}
Packet round-trip delay (sec)	R	0.02

RECOMMENDATIONS

Finally, researchers suggest the study of the channel utilization performance and due to the complexity and computer accuracy problems caused by the Markov analysis method so it must to find a method for evaluate the average data delay.

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