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Integrating Voice, Video, and E-mail Data Packet Traffic over Wireless TDMA Channels with Errors

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In this paper, we explore, via an extensive simulation study, the performance of a new medium access-control (MAC) protocol when integrating voice, video, and e-mail data packet traffic over a wireless channel of high capacity, with errors. Depending on the number of video users admitted into the system, our protocol varies (a) the request bandwidth dedicated to resolving the voice users contention and (b) the probability with which the base station grants information slots to voice users, in order to preserve full priority for video traffic. We evaluate the voice and video packet-dropping probabilities for various voice and video load conditions and the average e-mail data message delays. Our scheme achieves high aggregate channel throughput in all cases of traffic load despite the introduction of errors in the system.

KEY WORDS: Multiple traffic type integration; channel errors; QoS.

1. INTRODUCTION

High-speed packet-switched network architectures will soon have the ability to support a wide variety of multimedia services, the traffic streams of which will have widely varying traffic characteristics (bit rate, performance requirements). The main goal of wireless communication is to allow the user access to the capabilities of the global packet-switched network at any time without regard to location or mobility. Current and future wireless networks are and will be based on the cellular concept. In such networks, a well-designed multiple-access control (MAC) protocol will reduce system costs by maximizing system capacity, integrating different classes of traffic, and satisfying the diverse and usually contradictory quality of service (QoS) requirements of each traffic class.

In this work, we design and evaluate a multiple access scheme that multiplexes voice traffic at the vocal activity (talkspurt) level to efficiently integrate voice (constant bit rate, CBR on/off traffic), video (variable bit rate, VBR), and bursty data traffic in high-capacity picocellular environments. All transmissions are subject to error due to noise over the wireless channel.

Within the picocell, spatially dispersed source terminals share a radio channel that connects them to a fixed base station. The base station allocates channel resources, delivers feedback information, and serves as an interface to the mobile switching center (MSC). The MSC provides access to the fixed network infrastructure. We focus on the uplink (mobiles to base station) channel, where a MAC scheme is required in order to resolve the source terminals' contention for channel access.

2. VOICE-VIDEO-DATA INTEGRATION

2.1. Channel Frame Structure

The uplink channel time is divided into time frames of equal length. The frame duration is selected such that a voice terminal in talkspurt generates exactly one packet

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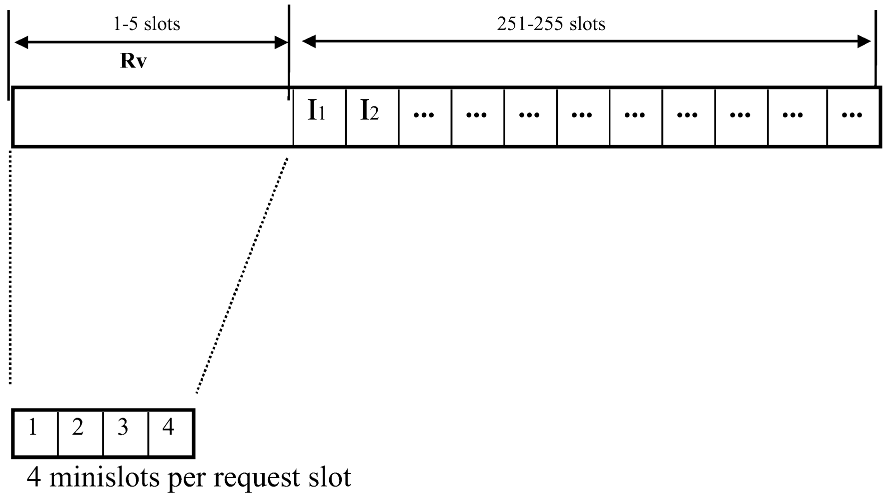

Fig. 1. Frame structure for the 9.045 Mbps channel.

per frame. As shown in Fig. 1 (which presents the channel frame structure), each frame consists of two *types* of intervals. These are the *voice and data request* intervals and the *information* intervals.

Within an information interval, each slot accommodates exactly one fixed-length packet that contains voice or video information and a header. Voice and data request intervals are subdivided into minislots and each minislot accommodates exactly one fixed-length request packet. The request must include a source identifier. Because we assume that all of the voice transitions occur at the frame boundaries,[2] we place all request intervals at the beginning of the frame, in order to minimize the voice packet access delay. *We introduce the idea* [12] *that the request slots can be shared by voice and data terminals (first by voice terminals and, after the end of voice contention, by data terminals),* in order to optimize the use of the request bandwidth.

Voice and data terminals do not exhaust their attempts for a reservation within the request intervals. *Any other free, at the time, information slot of the frame can be temporarily used as an extra request slot (ER slot)* for voice and data terminals [5]. The ER slots can be used by both voice and data terminals, with priority given to voice terminals. The concept of reserving a minimum bandwidth for voice and data terminals to make reservations helps to keep the voice access delay within relatively low limits and gives clearly better performance than the PRMA [1] and quite a few PRMA-like algorithms, such as DPRMA [4], where the absence of request slots leads to a continuously decreasing probability of finding available information slots as traffic increases, and hence to greater access delays.

[2] The explanation for this assumption will be given in Section 2.2.

No request slots are used for the video terminals for two reasons that will be be analyzed in Section 2.3.

2.2. Voice, Video and Data Traffic Models

Our primary voice traffic model assumptions are the following:

1. The speech codec rate is 32 Kbps, and voice terminals are equipped with a voice activity detector (VAD) [1]. Voice sources follow an alternating pattern of talkspurts and silence periods (on and off), and the output of the voice activity detector is modeled by a two-state discrete-time Markov chain. The mean talkspurt duration is 1.0 s and the mean silence duration is 1.35 s.
2. All of the voice source transitions (e.g., talk to silence) occur at the frame boundaries. This assumption is reasonably accurate, taking into consideration that the duration of a frame is equal to 12 ms here, while the average duration of the talkspurt and silence periods exceeds 1 s.
3. The number of active voice terminals, N, in the system is assumed to be constant over the period of interest. This is because the changes in the number of calls are usually on the order of tens

of seconds, while the frame duration is on the order of tens of milliseconds [2].

4. The voice delay limit is equal to 40 ms.
5. The channel is without capture.
6. Reserved slots are deallocated immediately. This implies that a voice terminal holding a reservation signals the BS upon the completion of its talkspurt.

Our video traffic model is based upon work done by Heyman *et al.* [10]. In this study of actual videoconferencing traffic, video frames (VFs) were found to be generated periodically and to contain a varying number of cells in each frame. The distribution of the number of cells per VF was found to be described by a gamma (or equivalently negative binomial) distribution. A Markov chain model can be constructed that demonstrates the transition from one state to the next. A "state" represents the number of video packets (cells) that a video frame contains. The transition matrix is computed as

$$\mathbf{P} = \rho\mathbf{I} + (1-\rho)\mathbf{Q} \tag{1}$$

where $\mathbf{I}$ is the identity matrix, ρ is the autocorrelation coefficient (0.98459 from [11]), and each row of the $\mathbf{Q}$ matrix is composed of the probabilities $(f0, \ldots, fK, \boldsymbol{F}K)$. The quantity fK has the negative binomial distribution and represents the probability that a video frame contains k cells. The value of K in Eq. (1) represents the peak cell rate, and $\boldsymbol{F}K = \Sigma k > K\, fK$.

The statistics for video conferencing traffic that were obtained in [10] were the result of coding a video sequence with a modified version of the H.261 standard. The results showed a peak cell-generation rate of 220 cells/VF (2.112 Mbps), an average generation rate of 104.8 cells/VF (1.006 Mbps), and a standard deviation of 29.7 cells/VF (0.285 Mbps). The cell size was taken equal to 48 bytes, which is equivalent to the ATM cell size. New VFs are assumed to arrive every 40 msecs (i.e., 25 VFs per second).

We adopt the data traffic model based on statistics collected on e-mail usage from the Finnish University and Research Network (FUNET) [14]. The probability distribution function $f(x)$ for the length of the data messages of this model was found to be well approximated by the Cauchy (0.8, 1) distribution. The packet interarrival time distribution for the FUNET model is exponential.

The maximum transmission delay for video packets is set to 40 ms, with packets being dropped when this deadline is reached. That is, all video packets of a VF must be delivered before the next VF arrives. The maximum transmission delay for voice packets is also set to 40 ms.

The allowed voice packet dropping probability is set to .01, whereas the allowed video packet dropping probability is set to .0001.

2.3. Actions of Voice, Video, and Data Terminals, Base Station Scheduling, and Voice-Data Transmission Protocol

Voice and data terminals with packets, and no reservation, contend for channel resources using a random access protocol to transmit their request packets only during the voice-data request intervals, with absolute priority given to voice terminals. The base station broadcasts a short binary feedback packet at the end of each minislot, indicating only the presence or absence of a collision within the minislot (collision, C, versus noncollision, NC). Upon successfully transmitting a request packet the terminal waits *until the end of the corresponding request interval* to learn of its reservation slot (or slots). If unsuccessful within the request intervals of the current frame, the terminal attempts again in the request intervals of the next frame. A terminal with a reservation transmits freely within its reserved slot.

Video terminals, as already mentioned, do not have any request slots dedicated to them. This happens for two reasons:

1. Video sources "live" permanently in the system; they do not follow an ON-OFF state model like voice sources.
2. Video traffic follows a multistate Markov model, in which, however, state transitions do not occur very often.

Thus, there is no need for granting request bandwidth to the video terminals, as it would be wasted in most cases. *Video terminals convey their requirements to the base station by transmitting them within the header of the first packet of their current video frame.*

To allocate channel resources, the BS maintains a dynamic table of the active terminals within the picocell. Upon successful receipt of a voice or data request packet, the BS provides an acknowledgment and queues the request. The BS allocates channel resources *at the end of the corresponding request interval* and follows a different allocation policy for video terminals than for voice terminals.

Video terminals have absolute priority in acquiring the slots they demand. If a full allocation is possible, the BS then proceeds to allocate any still-available information slots to the requesting voice terminals. Otherwise, if a full allocation is not possible, the BS grants to the video users as many of the slots they requested as possible (i.e., the BS makes a partial allocation). The BS keeps a record of any partial allocations so that the remaining requests can be accommodated whenever the necessary channel resources become available. In either allocation-type case, the BS allocates *the earliest available* information slots to the video terminals, which, if needed, keep these slots in the following channel frames until the next video frame (VF) arrives.

Voice terminals that have successfully transmitted their request packets *do not* acquire all the available (after the servicing of video terminals) information slots in the frame. If this happened, voice terminals would keep their dedicated slots for the whole duration of their talkspurt (on average, more than 8 channel frames here), and thus video terminals would not find enough slots to transmit in and the particularly strict video QoS requirements (the maximum allowed video packet-dropping probability is only .0001) would be violated. *The BS allocates a slot to each requesting voice terminal with a probability* $\boldsymbol{p}^*$. The requests of voice terminals that "fail" to acquire a slot, based on the above BS slot allocation policy, remain queued. The same holds for the case where the resources needed to satisfy a voice request are unavailable. Within each priority class, the queuing discipline is assumed to be first-come, first-served (FCFS).

We study two cases of incorporating e-mail data users into the system. In the first case, each data user is allowed to reserve *just one slot per frame,* which is guaranteed until the completion of the e-mail message transmission. The "defensive" choice of granting only one slot per frame to data users is easily explained by the fact that video traffic is quite bursty. Thus, the number of voice terminals has to be drastically decreased as the data message arrival rate increases, in order to cope with the burstiness of the video traffic mainly but also with the burstiness of the data traffic, and still be able to preserve the QoS requirements for each traffic type. This explains our "defensive" choice, as this would lead to a further decrease of the maximum voice capacity in order to preserve the video QoS requirements.

The second case is the one in which the BS "preempts" data reservations in order to service voice requests. Thus, whenever new voice requests are received and every slot within the frame is reserved, the BS attempts to service the voice requests by canceling the appropriate number of reservations belonging to data terminals (if any). When data reservations are canceled, the BS notifies the affected data terminal and places an appropriate request at the front of the data request queue.

Finally, in order to preserve the strict video QoS, we enforce a scheduling policy for the video terminals that prevents unnecessary dropping of video packets in channel frames within which the arrival of a new VF of a video user takes place (the details of this "reshuffling" policy can be found in [13] where we study a different data model and we do not consider transmission errors and preemption of data users).

Quite a few reservation random-access algorithms have been proposed in the literature for use by contending voice terminals to access a wireless TDMA channel (e.g., PRMA [1], Two-Cell Stack [8], Controlled Aloha [7], Three-Cell Stack [3]). In our study, we adopt the *two-cell stack* reservation random-access algorithm, due to its operational simplicity, stability, and relatively high throughput when compared to the PRMA (Aloha-based) [1] and PRMA-like algorithms, such as [4,6]. The *two-cell stack* blocked-access collision resolution algorithm [9] is adopted for use by the data terminals in order to transmit their data request packets. This algorithm is of the window type, with FCFS-like service.

3. CHANNEL ERROR MODELS

We use a two-state Markov model and an N-state Markov model to emulate the process of packet transmission errors (from [15]). In the two-state Markov model, the channel switches between a "good state" and a "bad state", s0 and s1, respectively. Packets are transmitted correctly when the channel is in state s0, and errors occur when the channel is in state s1. The N-state Markov model (presented in Fig. 2) comprises 6 states in the uplink channel, which is here under study. State s0 represents the "good state" and all other states represent the "bad states." When the channel is in state s0, it can either remain in this state, with probability $1 - \mathbf{p}0$, or make the transition to state s1, with probability $\mathbf{p}0$. When the channel is in state sn, $n \in [1,4]$, the transition of the channel state is either to the next higher state (with probability $\mathbf{p}n$) or back to state s0 (with probability $1 - \mathbf{p}n$), based on the status of the currently received data packet. This means that the channel does not remain in one of the "bad states" for more than 1 slot. If the channel is in the last state (s5), it will always return to state s0. With this model, it is only possible to generate burst errors of length $N - 1$ at most. The transition probabilities of the two channel error models are presented in Table I. The only difference between our models and the ones in [15] is that we have changed the value of the probability Pgood, i.e.,

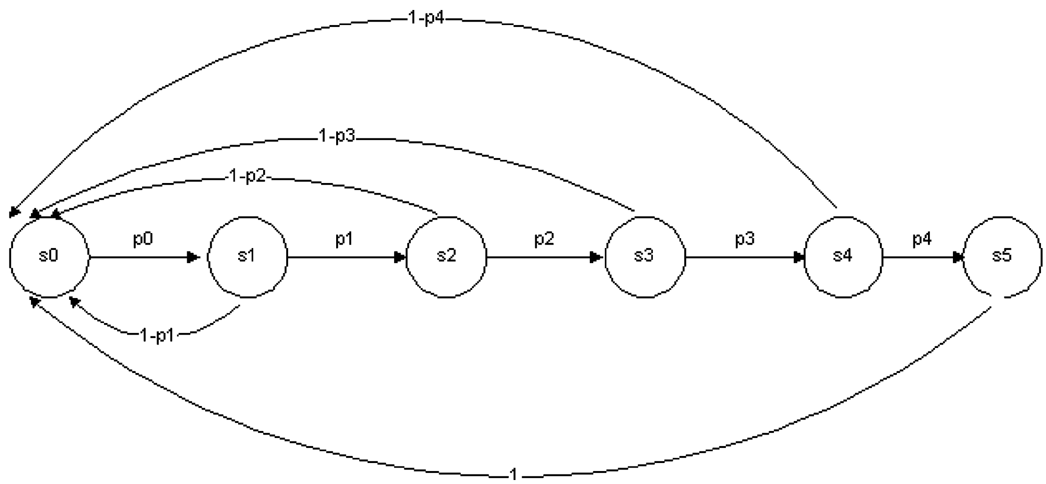

Fig. 2. *N*-state Markov model.

the steady-state probability that the channel is in the good state. In [15], this probability is equal to .9328, whereas in our models it is larger and equal to .99995. This is necessary in order to be able to accommodate the type of video traffic we study, the QoS requirements of which are very strict.

4. SYSTEM PARAMETERS

The channel rate is 9.045 Mbps (from [4]). The 12 ms of frame duration accommodate 256 slots. The number of request slots shared by voice and data users is not fixed in the scheme. It depends on the number of video sources admitted into the system,[3] and it varies accordingly between 1 and 5 slots (see Table II). Even for the case where 5 request slots are needed, this corresponds to a 1.95% request bandwidth only. We should note that

1. In our design, we chose the number of minislots per request interval (4) to allow for guard time and synchronization overheads, for the transmission of a generic request packet, and for the propagation delay within the picocell.
2. Because of assumption 2 of our voice traffic model, all voice request intervals are located at the beginning of each frame.
3. The average e-mail data message length has been found (by simulation) to be 80 packets. We do not impose an upper limit on the average e-mail

Table I. Channel Error Model Parameters

	Two-State	*N*-State
P0		0.0000446
P1		0.100324
P2		0.164083
P3		0.149606
P4		0.526316
P5		0.000000
Pr (Good)	0.99995	0.99995
Pr (Good-Bad)	0.0000235	0.0000446
Pr (Bad-Good)	0.46945	0.8924

Table II. Adjustable Voice Request Bandwidth Depending on the Number of Video Users, and Allocation Probability for Voice Users

Number of Video Users	Number of Request Slots	Probability **p***
6	1	.0072
5	1	.03
4	2	.06
3	2	.085
2	3	.128
1	4	.18
0	5	1

[3] The channel bandwidth consumed by each video source is large, and thus, when we examine cases with a small number of video sources, the system can accommodate a significantly larger number of voice sources. In this case, more voice request slots are needed in order to allow voice sources to enter the system without significant dropping of voice packets.

Table III. Data Message Delay and Maximum Voice Capacity for 0 Video Users and Set Data Message Arrival Rate

	2-State Error Model				N-State Error Model			
	No Preemption		Preemption		No Preemption		Preemption	
λ	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)
0.01	560	1800.11	577	2789.77	560	1834.25	578	2869.28
0.05	552	1385.20	568	2247.47	553	1496.08	569	2326.41
0.1	543	1362.49	558	2013.51	545	1328.40	562	1912.56
0.15	520	1276.43	540	2009.45	522	1316.62	542	1999.47
0.2	498	1288.58	512	1986.35	500	1247.36	515	2017.59
0.25	481	1360.92	498	2156.70	483	1401.23	500	2096.91
0.3	452	1294.40	474	2221.39	453	1276.42	477	2203.17
0.35	439	1270.84	462	2089.96	443	1309.57	465	2109.73
0.4	428	1313.36	442	2127.52	430	1365.86	446	2193.26

Table IV. Data Message Delay and Maximum Voice Capacity for 1 Video User and Set Data Message Arrival Rate

	2-State Error Model				N-State Error Model			
	No Preemption		Preemption		No Preemption		Preemption	
λ	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)
0.01	460	1998.02	468	2307.69	461	1897.09	469	2411.13
0.05	430	1356.45	436	1943.35	432	1425.12	437	2000.67
0.1	406	1293.18	418	2007.11	409	1386.55	420	1987.36
0.15	352	1304.26	361	1966.25	356	1247.53	366	1943.25
0.2	331	1226.53	341	1860.96	334	1268.46	345	1966.64
0.25	318	1300.12	329	1932.29	321	1295.89	334	1997.39
0.3	300	1267.68	313	2091.16	304	1326.74	319	1962.81
0.35	286	1271.06	294	2065.38	293	1311.37	298	2102.52
0.4	275	1326.88	285	2001.55	281	1308.19	289	2096.31

data message delay, as this is a type of traffic that can withstand a delay of a number of seconds or even more. Thus, we simply evaluate the average e-mail data message delay in our study.

5. RESULTS AND DISCUSSION

Each computer simulation point is the result of an average of 10 independent runs, each simulating 305,000 frames (the first 5,000 of which are used as a warm-up period). Tables III, IV, V, and VI present the results of our scheme (VVEDI, i.e., voice, video, and e-mail data integration) for different numbers of video users within the system. We present the maximum voice capacity and the average e-mail data message delay for different e-mail message arrival rates (λ messages/frame) and for both channel error models examined.

Our results show that, for both error models, the data preemption mechanism helps significantly to increase the voice capacity. Also, for all data message arrival rates, we observe that in the presence of the *N*-state error model, our scheme achieves slightly better results. This can be explained based on the observation that although the two error models have the same probability of good-bad and bad-good state transitions, the *N*-state model is less bursty, due to the fact that this model can only generate burst errors of at most length $N - 1$, i.e., 5 slots in this case.

Figure 3 presents the channel throughput (%)

Table V. Data Message Delay and Maximum Voice Capacity for 3 Video Users and Set Data Message Arrival Rate

	2-State Error Model				*N*-State Error Model			
	No Preemption		Preemption		No Preemption		Preemption	
λ	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)
0.01	235	2002.47	247	2436.95	235	1946.36	248	2371.22
0.05	214	1370.18	227	1864.26	215	1488.61	229	2009.64
0.1	194	1402.36	206	1923.90	195	1438.04	208	1978.92
0.15	178	1439.45	189	1912.68	180	1377.93	191	1964.37
0.2	165	1415.36	176	1877.19	167	1364.82	178	1953.56
0.25	151	1329.94	161	1950.82	153	1325.53	164	2049.72
0.3	138	1356.27	148	1994.30	142	1417.71	152	1946.35
0.35	130	1296.54	139	2012.80	134	1386.80	143	1976.39
0.4	121	1423.83	131	1970.45	124	1401.68	135	2075.29

Table VI. Data Message Delay and Maximum Voice Capacity for 5 Video Users and Set Data Message Arrival Rate

	2-State Error Model				*N*-State Error Model			
	No Preemption		Preemption		No Preemption		Preemption	
λ	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)	NV	EDmD(ms)
0.01	78	1909.64	88	2425.31	78	1863.75	89	2384.62
0.05	68	1503.21	76	2107.91	69	1587.29	78	2117.83
0.1	52	1388.48	60	2008.30	52	1493.37	62	2186.53
0.15	36	1401.29	44	2124.52	37	1426.01	47	2099.90
0.2	29	1432.30	38	2056.48	30	1508.42	40	2126.85
0.25	22	1464.42	30	1961.07	24	1453.47	32	2091.46
0.3	14	1381.36	21	2023.34	16	1398.29	22	2104.71
0.35	5	1376.28	13	2178.56	7	1441.72	15	2132.96
0.4	x		2	2155.87	x		4	2188.13

achieved by our scheme (number of slots used/frame, divided by the number of information slots in the frame) for the 2-state error model for different numbers of video users and without preemption. Figure 4 presents the throughput achieved by our scheme for the *N*-state error model for different numbers of video users with preemption. Thus, these two figures present the "minimum" and "maximum" set of throughputs achieved by

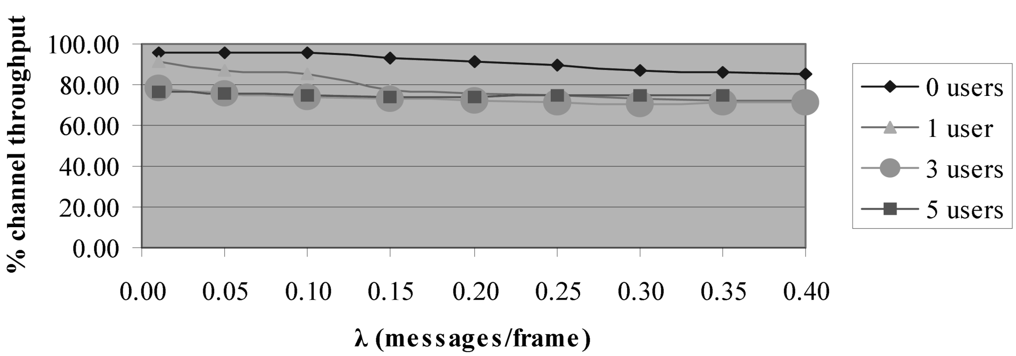

Fig. 3. 2-state error model, no preemption.

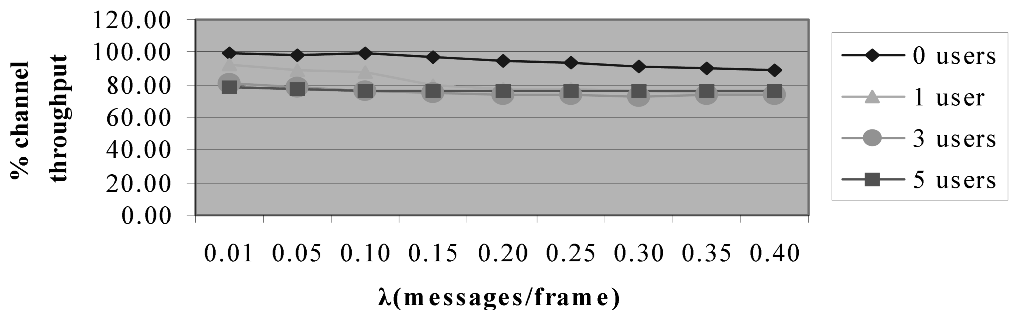

Fig. 4. *N*-state error model, with preemption.

our scheme for all the data message arrival rates under study. As shown in both figures, even for the case of five video users, in which the system is heavily loaded with bursty and demanding sources and has to cope with the transmission errors, the throughput achieved is quite high.

The reasons VVEDI achieves such good results (steadily above 70% throughput) are

1. Our proposed video slot allocation mechanism is very dynamic, thus achieving higher bandwidth utilization.
2. The use of the probability **p*** for the allocation of slots to voice terminals ensures the absolute priority of the very demanding video traffic in the system.
3. With the above proposed mechanism and the use of ER slots, our scheme "exploits" the maximum amount of slots within the frame.
4. The data preemption policy proves to be very effective, as it imposes just a small extra delay on e-mail data messages (of the order of 600–1000 ms, which is totally acceptable for e-mail traffic), while at the same time it helps our system to increase its voice capacity significantly (i.e., around 3% in the case of one video user, around 6% in the case of three video users).

Table VII includes results that have been extensively presented in [13]. In that work (the scheme called VVDI), we have integrated voice, video (the same models as in this work), and a different data model (from [3]): The aggregate message arrivals are Poisson distributed with mean λ messages per frame. Additionally, we assumed that the messages vary in length according to a geometric distribution with parameter q and mean $B = 1/q$. B is expressed in packets per message, and the steady-state data rate packet arrival is equal to λB packets/frame (B is equal to 8 in our study, i.e., an e-mail message in our present study has 10 times more packets than a data message in VVDI). Also, an upper limit on the mean data message delay, equal to 200 ms, is assumed, whereas in this work no such limit has been set. Other significant differences between VVDI and VVEDI are, as already mentioned, the absence of channel errors and the preemption policy in VVDI. We present the results in Table VII in order to compare them with the results presented in Tables VIII and IX (this is the reason we have chosen the e-mail data message arrival rate to be 10 times smaller than the data message arrival rate in VVDI, so that the average data packet load will be the same in all cases examined). It is shown that in the case of no preemption for e-mail data users, this scheme achieves a throughput about 7% less than VVDI for two video users and about 4% less than VVDI for four video users. In the case of preemption of e-mail data users, this difference is even smaller (about 5% for two video users and 3% for four video users). Taking into account the facts of the

Table VII. Maximum Voice Capacity and Channel Throughput for a Set Number of Video Users and Set Poisson Data Message Arrival Rate

λ (mes./frame)	Maximum Voice Capacity and Throughput (%)			
	2 Video Users		4 Video Users	
0.1	377	88.6	201	83.2
0.5	350	85.3	185	82.1
1.0	330	83.5	164	80.1
1.5	314	82.4	150	79.4
2.0	300	81.6	136	78.6
2.5	287	81.3	124	78.2
3.0	277	80.9	115	78.2
3.5	265	80.5	104	78.0
4.0	258	80.9	94	77.9

Table VIII. Maximum Voice Capacity and Channel Throughput for a Set Number of Video Users and Set E-Mail Data Message Arrival Rate, under the 2-State Error Model

λ (mes./frame)	Maximum Voice Capacity and Throughput (%)							
	2 Video Users				4 Video Users			
	No Preemption		Preemption		No Preemption		Preemption	
0.01	322	80.3	336	82.7	165	78.7	176	80.6
0.05	299	77.6	314	80.2	140	75.7	150	77.4
0.1	278	75.7	294	78.4	128	75.3	137	76.8
0.15	264	74.9	281	77.8	113	74.3	121	75.7
0.2	254	74.8	270	77.5	104	74.4	112	75.8
0.25	239	73.8	253	76.2	98	75.0	106	76.3
0.3	227	73.4	241	75.8	88	74.9	97	76.4
0.35	218	73.5	228	75.2	75	74.3	83	75.6
0.4	208	73.4	217	74.9	63	73.8	69	74.8

presence of channel errors in VVEDI, and the much larger data message size, which is a "burden" for the channel and can cause the "loss" of information slots (in the sense that they could have been allocated to newly arrived video packets, which need them much more urgently), it is once again shown that the performance of our scheme is highly satisfactory.

6. CONCLUSIONS

In this paper, we have proposed and evaluated a new multiple-channel access-control scheme for integrating voice, video, and e-mail data-packet traffic in a high-capacity picocellular environment. Video traffic is offered absolute priority over voice and data traffic, due to its more stringent quality of service requirements. Voice traffic can "preempt" data traffic in order to preserve its priority over data traffic.

Via an extensive simulation study, we demonstrate that the proposed scheme achieves high throughput when integrating all three traffic types, despite the very restraining video dropping probability limit and the presence of errors in the packet transmission.

The results achieved by our scheme are a consequence of the combination of four factors: (a) our voice slots allocation policy, (2) our video slots scheduling policy, (3) the preemption of data packets, and (4) the use of the unused information slots as extra request slots.

Table IX. Maximum Voice Capacity for a Set Number of Video Users and Set E-Mail Data Message Arrival Rate, under the *N*-State Error Model

λ (mes./frame)	Maximum Voice Capacity and Throughput (%)							
	2				4			
	No Preemption		Preemption		No Preemption		Preemption	
0.01	323	80.4	338	83.0	166	78.9	177	80.7
0.05	302	78.2	315	80.4	142	76.1	152	77.7
0.1	280	76.0	297	78.9	130	75.6	139	77.2
0.15	268	75.6	284	78.3	115	74.7	125	76.4
0.2	258	75.5	272	77.8	107	74.9	115	76.3
0.25	244	74.7	257	76.9	100	75.3	108	76.7
0.3	230	73.9	245	76.5	92	75.6	99	76.7
0.35	221	74.0	233	76.0	78	74.8	86	76.1
0.4	212	74.0	222	75.7	66	74.3	72	75.4

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