Packetized Voice, Video and Email Integrated Transmission over Wireless TDMA Networks with Bursty Channel Errors

P. Koutsakis and M. Paterakis

Dept. of Electronics and Computer Engineering
Telecommunications and Information & Computer Networks Laboratories
Technical University of Crete, 73100 Chania Greece
tel: +30 821 37343, e-mail: polk@telecom.tuc.gr, pateraki@telecom.tuc.gr

ABSTRACT

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 high capacity picocellular system, 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 email data message delays. Our scheme achieves high aggregate channel throughput in all cases of traffic load, despite the introduction of errors in the system.[√]

I. VOICE-VIDEO-DATA INTEGRATION

A. Channel Frame Structure

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.

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 per frame. As shown in Figure 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 mini-slots and each mini-slot accommodates exactly one, fixed length, request packet. The request must include a source identifier. Since we assume that all of the voice transitions occur at the frame boundaries[1], we place all request intervals at the beginning of the frame, in order to minimize the voice packet access delay. *We introduce the idea [6] 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 [3]. 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 [2], 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.

No request slots are used for the video terminals, because of two reasons, which will be be analyzed in section 2.3.

B. 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 secs and the mean silence duration is 1.35 secs.
2. All of the voice source transitions (e.g., talk to silence) occur at the frame boundaries. This assumption is

√ *This research has been funded by the Greek General Secretariat of Research and Technology and the European Social Fund, through PENED '99 grant no 99ED 192.*

[1] The explanation for this assumption will be given in section 2.2.

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 sec.
3. The voice delay limit is equal to 40 ms.
4. The channel is without capture.
5. 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 [5]. 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.

The statistics for video conferencing traffic that were obtained in [5], 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 email usage from the Finish University and Research Network (FUNET) [7]. 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 inter-arrival time distribution for the FUNET model is exponential.

The maximum transmission delay for voice and video packets is set to 40 msecs. That is, all video packets of a VF must be delivered before the next VF arrives. The allowed voice packet dropping probability is set to 0.01, whereas the allowed video packet dropping probability is set to 0.0001.

C. 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 mini-slot indicating only the presence or absence of a collision within the mini-slot (collision (C) versus non-collision (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 multi-state 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.

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 which 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 would be violated. *The BS allocates a slot to each requesting voice terminal with a probability **p***.* The requests of voice terminals which ″ fail ″ to acquire a slot, based on the above BS slot allocation policy, remain queued. Within each priority class, the queuing discipline is assumed to be First Come First Served (FCFS).

We study two cases of incorporating email 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 email message transmission. The "defensive" choice 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 new voice requests. 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 which 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 [6]).

In our study, we adopt the *two-cell stack* [4] reservation random access algorithm for use by contending voice terminals, due to its operational simplicity, stability and relatively high throughput when compared to the PRMA (Aloha-based) [1] and PRMA-like algorithms, such as [2]. A slightly different blocked access collision resolution algorithm is adopted for use by the data terminals in order to transmit their data request packets.

II.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 [8]). 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 [8]) 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-**p**0, or make the transition to state s1, with probability **p**0. When the channel is in state sn, n € [1,4], the transition of the channel state is either to the next higher state (with probability **p**n) or back to state s0 (with probability 1-**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 at most length N-1. The transition probabilities of the two channel error models are presented in Table 1. The only difference between our models and the ones in [8] is that we have changed the value of the probability Pgood, i.e., the steady state probability that the channel is in the good state. In [8], this probability is equal to 0.9328, whereas in our models is larger, and equal to 0.99995. This is necessary, in order to be able to accommodate the very strict QoS requirements of the video traffic.

III. SYSTEM PARAMETERS

The channel rate is 9.045 Mbps (from [2]). 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[2], and it varies accordingly between 1 and 5 slots (see Table 2). 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 email data message length has been found (by simulation) to be 80 packets. We do not impose on upper limit on the average email 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 email data message delay in our study.

IV. 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 warmup period). Table 3 presents the results of our scheme (VVEDI, i.e., Voice,Video, and Email Data Integration) for three video users within the system. We present the maximum voice capacity and the average email data message delay for different email message arrival rates (λ messages/frame) and for both channel error models examined.

Our results show that, for the cases of both the error models, the data preemption mechanism helps significantly in the increase of 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

[2] 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.

fact that this model can only generate burst errors of at most length N-1, i.e., 5 slots in this case.

Figure 2 presents the throughput (%) 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 3 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 our scheme for all the data message arrival rates under study. As shown in both figures, even for the case of 5 video users, in which the system is heavily loaded with bursty and demanding sources and it has to cope with the transmission errors, the throughput achieved is quite high. The reasons for which 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 proposed above mechanism and the use of ER slots, our scheme "exploits" the maximum amount of slots within the frame.
4) The data preemption policy proves itself to be very effective, as it imposes just a small extra delay on email data messages (of the order of 600-1000 ms, which is totally acceptable for email traffic), while at the same time it helps our system to increase its voice capacity significantly (i.e., around 3% in the case of 1 video user, around 6% in the case of 3 video users).

REFERENCES

[1] S. Nanda, D. J. Goodman and U. Timor, ″Performance of PRMA: A packet voice protocol for cellular systems″, *IEEE Transactions on Vehicular Technology*, Vol. 40, pp. 584-598, 1991.

[2] D. A. Dyson and Z. J. Haas, ″A Dynamic Packet Reservation Multiple Access Scheme for Wireless ATM″, *Proceedings of the IEEE MILCOM'97.*

[3] N. M. Mitrou, G. L. Lyberopoulos and A. D. Panagopoulou, ″Voice and Data Integration in the Air-Interface of a Microcellular Mobile Communication System″, *IEEE Transactions on Vehicular Technology*, Vol. 42, No. 1, Feb. 1993, pp. 1-13.

[4] A. C. Cleary and M. Paterakis, ″On the Voice-Data Integration in Third Generation Wireless Access Communication Networks″, *European Transactions on Telecommunications and Related Technologies*, Vol. 5, No. 1, Jan.-Feb. 1994, pp. 11-18.

[5] D. Heyman, T. V. Lakshman, A. Tabatabai and H. Heeke, ″Modeling teleconference traffic from VBR video coders″, *International Conference on Communications '94,* vol. 3, pp. 1744-1748, 1994.

[6] P. Koutsakis and M. Paterakis, ″On Multiple Traffic Type Integration over Wireless TDMA Channels with Adjustable Request Bandwidth″, *International Journal of Wireless Information Networks,* Vol. 7, No.2, pp.55-68, 2000.

[7] Q. Pang, A. Bigloo, et al., ″Service Scheduling for General Packet Radio Service Classes″, *Proc. of the IEEE Wireless Communications and Networking Conference 1999 (WCNC '99), New Orleans, USA.*

[8] Chi-Yuan Hsu, Antonio Ortega, ″Rate Control for Robust Video Transmission over Burst-Error Wireless Channels″, *IEEE Journal on Selected Areas in Communications, Vol. 17, No.5,May1999.*

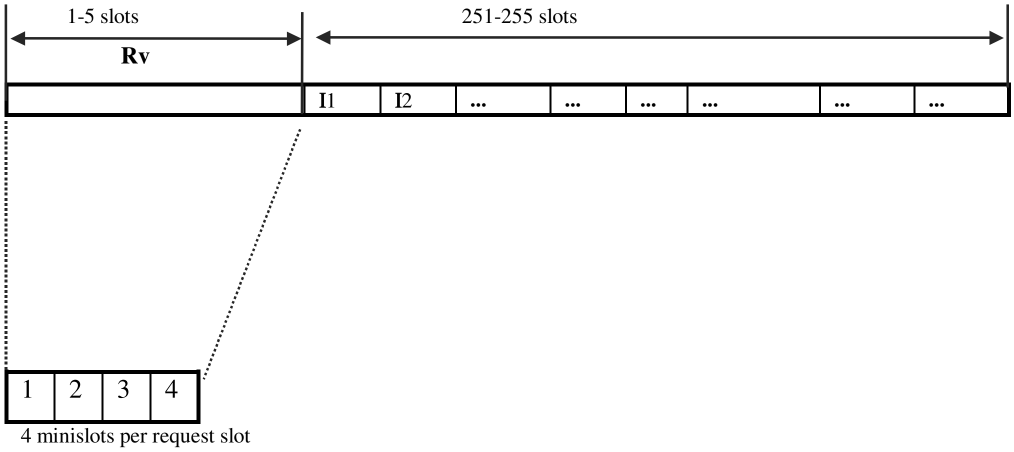

Figure1. Frame structure for the 9.045 Mbps channel.

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

Table1. Channel error models' parameters.

Number of Video users	**Number of request slots**	**Probability p***
6	1	0.0072
5	1	0.03
4	2	0.06
3	2	0.085
2	3	0.128
1	4	0.18
0	5	1

Table2. Adjustable voice request bandwidth depending on the number of video users, and allocation probability for voice users.

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 3. Data Message Delay and Maximum Voice Capacity for 3 video users and set data message arrival rate.

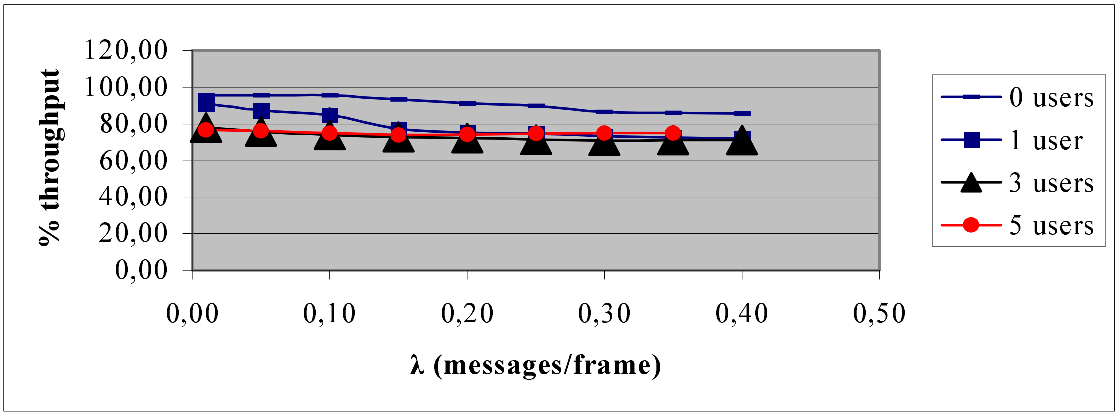

Figure2. 2-state error model, no preemption.

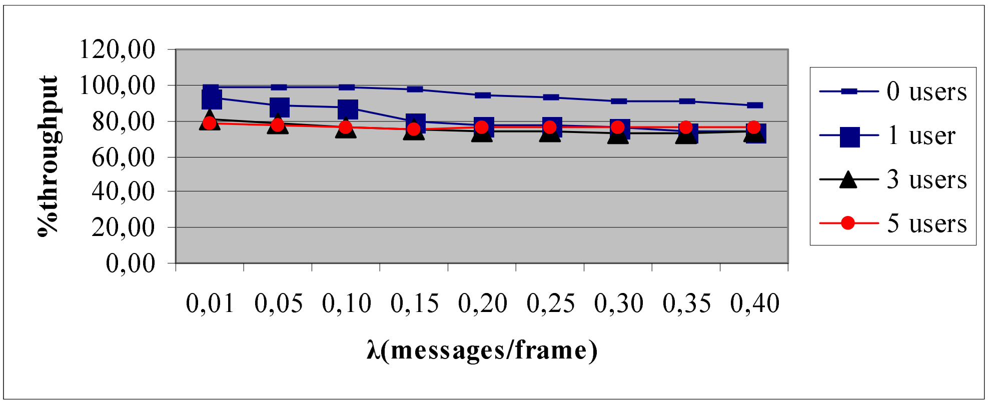

Figure3. N-state error model, with preemption.