Integrated Wireless Access for Videoconference From MPEG-4 and H.263 Video Coders With Voice, E-mail, and Web Traffic

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***Abstract*—In this paper, a new medium access control protocol for wireless communications, named Multimedia Integration Multiple Access Control (MI-MAC), is presented and investigated. We explore, via an extensive simulation study, the performance of MI-MAC when integrating voice, e-mail data, and web packet traffic with either MPEG-4 or H.263 videoconference streams over a noisy wireless channel of high capacity. Our scheme, one of the first in the literature that considers the integration of MPEG-4 or H.263 streams with other types of packet traffic over wireless networks, achieves high aggregate channel throughput in all cases of traffic load, while preserving the quality of service (QoS) requirements of each traffic type.**

***Index Terms*—MAC protocol, MPEG-4 and H.263 video, voice-video-data integrated access, wireless cellular communications, wireless channel errors.**

I. Introduction

THE success of emerging high-speed mobile integrated-services network architectures will depend on their ability to arbitrate among various data sources with different quality of service (QoS) requirements over shared wireless links. As the traffic streams of the sources will have widely varying traffic characteristics (bit-rate, performance requirements), the new air interfaces of beyond 3G wireless networks should have in common a high degree of flexibility and capacity, in order to support the very different traffic types.

A well-designed multiple access control (MAC) protocol should provide an efficient mechanism to share the limited bandwidth resources and satisfy the diverse and usually contradictory QoS requirements of each traffic class (voice, video, data).

In this paper, we describe the design and evaluation of a multiple access scheme which efficiently integrates voice (Constant Bit Rate, CBR, On/Off Traffic), bursty e-mail, and web requests traffic with either MPEG-4 or H.263 video streams (Variable Bit Rate, VBR) in high capacity picocellular systems with burst-error characteristics. To the best of our knowledge, this is one of the first papers in the relevant literature that considers the integration of actual MPEG-4 or H.263 streams with other types of packet traffic over wireless networks.

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II. Multiple Traffic Type Integration

Within the picocell, spatially dispersed source terminals share a radio channel that connects them to a fixed base station (BS). The BS 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. Since the BS is the sole transmitter on the downlink channel, it is in complete control of the downstream traffic, using time division multiple access (TDMA) to relay information to the users. Thus, we focus on the uplink (wireless terminals to BS) channel, where a MAC scheme is required in order to resolve the source terminals' contention for channel access.

A. Contribution of This Work

We use the MAC schemes that we designed in [1] and [2] as a basis for the design of the multimedia integration MAC (MI-MAC). However, in contrast to [1] and [2], the work presented in this paper considers actual video traces from the latest technology encoders, introduces a much more dynamic design of the channel frame structure, incorporates web traffic, sets upper delay bounds for web and e-mail traffic, and studies system performance under a channel error model recently proposed in the literature.

B. Channel Frame Structure

The uplink channel time is divided into time frames of fixed length. The frame duration is selected such that a voice terminal in talkspurt generates exactly one packet per frame. (Packet size is considered to be equal to 53 bytes, 48 of which contain information; i.e., the packet size is equivalent to the ATM cell size. This choice is made for reasons of results comparison with another efficient protocol of the literature.) 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* interval (by data, we refer to both e-mail and web traffic) and the *information* interval.

Since we assume that all of the voice sources state transitions occur at the channel frame boundaries (this assumption will be explained in Section II.C), we place the voice and data request interval at the beginning of the frame, in order to minimize the voice packet access delay.

We introduce the idea [1] 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.

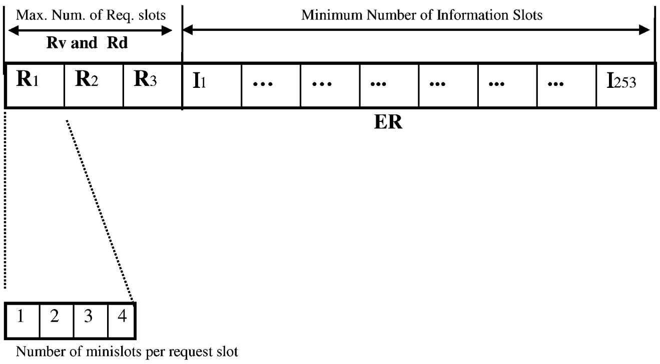

Fig. 1. Dynamic Frame structure for the 9.045-Mbps channel, frame duration 12 ms.

No request slots are used for the video terminals. Since video sources are assumed to "live" permanently in the system (they do not follow an ON-OFF state model like voice sources) and the duration of our simulation study is long (we simulate one hour of system performance), we assume without loss of generality that they have already entered the system at the beginning of our simulation runs; thus, there is no need for granting request bandwidth to the video terminals.

In [1] and [2], the request interval at the beginning of each frame consisted of a number of request slots, shared by voice and data users. The number of request slots was variable, depending on the channel load in video terminals, and each request slot comprised four minislots, each one accommodating exactly one fixed-length request packet. (The number of minislots in a request slot cannot, of course, be increased at will, since a large number of minislots would mean that their duration would be very short and would not suffice for the request packet to be transmitted to the BS and for the BS to send an acknowledgement to the requesting terminals, or for guard time and synchronization overheads.)

In this study, we chose the frame structure parameters as follows.

1) For all the examined scenarios of system load, we tried to find a maximum request bandwidth which would suffice for voice and data terminals. This was found, via simulations, to be equal to 3 request slots (i.e., 12 minislots).
2) In [1] and [2], the request bandwidth varied depending solely on the video load. Therefore, the variable request bandwidth mechanism was static for a given video load, which resulted in a waste of request bandwidth (minislots remained unused) when just a small number of voice and data requests awaited transmission. In this work, we enforce *a fully dynamic mechanism* for the use of the request bandwidth: The number of request slots is variable *per frame* (between 1 and 3, which is the maximum number, as explained above) and depends on the total channel load in each frame (i.e., the total voice and data loads, which are, naturally, restricted by the corresponding video load). In the cases when less than 3 request slots are needed for the end of contention, the BS signals all user terminals for the existence of the additional information slots in the current frame. This dynamic design of the frame structure leads to a possible exploitation of one or two more slots of the frame as information slots, when the number of requesting voice and data terminals in a frame is low and the contention among them ends quickly. Fig. 1 shows the case of maximum request bandwidth and minimum information bandwidth, respectively.

Within an information interval, each slot accommodates exactly one fixed-length packet that contains voice, video, or data information and a header.

Any free information slot of the current channel frame can be temporarily used as an extra request (ER) slot [3]. ER slots can be used by both voice and data terminals (again, as in the standard request slots, first by voice terminals and, after the end of voice contention, by data terminals). ER slots are subdivided into a fixed number (again, equal to four) of minislots.

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 packet reservation multiple access (PRMA) [4] and quite a few PRMA-like algorithms, such as dynamic PRMA (DPRMA) [5], where the absence of request slots leads to a continuously decreasing probability of finding available information slots as traffic level increases, and, hence, to greater access delays [1].

C. Traffic Models

1) Voice Traffic Model: Our primary voice traffic model assumptions are the following.

1) The speech codec rate is 32 kb/s [3], 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 s and the mean silence duration is 1.35 s.
2) The number of active voice terminals N in the system is assumed to be constant over the period of interest.
3) 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.
4) Reserved slots are deallocated immediately. This implies that a voice terminal holding a reservation signals the BS upon the completion of its talkspurt (the same assumption is made for slots reserved by data terminals).

The allowed voice packet dropping probability is set to 0.01, and the maximum transmission delay for voice packets is set to 40 ms [5].

2) E-mail Data Traffic Model: We adopt the data traffic model based on statistics collected on e-mail usage from the Finnish University and Research Network (FUNET) [6]. The probability distribution function f(x) for the length of the e-mail 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, and the average e-mail data message length has been found (by simulation) to be 80 packets. A quite strict (considering the nature of this type of traffic) upper bound is set on the average e-mail transmission delay, equal to 5 s. The reason for this strict bound is that mobile users sending e-mails will probably be quite demanding in their QoS requirements, as they will expect service times similar to those of short message service traffic.

3) Web Traffic Model: We adopt the http traffic model from [7], which is largely but not solely based on the model presented in [8]. This latter model has recently often been used by researchers in the field (e.g., [9], [10]). According to [7], the distributions of the random variables concerning the composition of web requests are the following:

1) size of web request: lognormal $(5.84, 0.29)$, with mean $= 360$ bytes and standard deviation $= 106.5$ bytes.
2) number of web requests per www session: lognormal (1.8, 1.68), with mean $= 25$ pages and standard deviation $=$ 100 pages.
3) web request viewing time: weibull (α, β), truncated at a maximum of 15 min (if the viewing time is longer than 15 min, a new session will follow), with mean $= 39.5$ s and standard deviation $= 92.6$ s.

Other relevant random variables include the size of the main object of the requested http session, the number of inline objects, and the size of the inline objects. However, as we study the uplink channel in this work (i.e., only the web requests sent from the mobile terminals to the BS), only the distributions of the random variables presented above are needed for our model.

The arrival process of www sessions is chosen to be Poisson with rate λ_{web} sessions per second, with an upper limit on the average web request transmission delay equal to 3 s. Given that the average size of a web request is 360 bytes, i.e., 7.5 packets (less than a tenth of the average e-mail message size), it is clear that we consider web traffic to be the most delay-tolerant. Still, this upper bound is again strict, as in the case of e-mail traffic, and the reason for this choice is, as in the case of e-mail traffic, to test system performance when incorporating users with very demanding QoS.

4) MPEG-4 and H.263 Video Streams: The MPEG initiated the new MPEG-4 standards in 1993 with the goal of developing algorithms and tools for high efficiency coding and representation of audio and video data to meet the challenges of video conferencing applications. The standards were initially restricted to low bit rate applications but were subsequently expanded to include a wider range of multimedia applications and bit rates. The most important addition to the standards was the ability to represent a scene as a set of audiovisual objects. The MPEG-4 standards differ from the MPEG-1 and MPEG-2 standards in that they are not optimized for a particular application but integrate the encoding, multiplexing, and presentation tools required to support a wide range of multimedia information and applications. In addition to providing efficient audio and video encoding, the MPEG-4 standards include such features as the ability to represent audio, video, images, graphics, text, etc. as separate objects, and the ability to multiplex and synchronize these objects to form scenes. Support is also included for error resilience over wireless links, the coding of arbitrarily shaped video objects, and content-based interactivity such as the ability to randomly access and manipulate objects in a video scene [11].

In our study, we use the trace statistics of actual MPEG-4 streams from the publicly available library of frame size traces of long MPEG-4 and H.263 encoded videos provided by the Telecommunication Networks Group at the Technical University of Berlin [12]. The video streams have been extracted and analyzed from a camera showing the events happening within an office. We have used the high quality version of the video, which has a mean bit rate of 400 kb/s, a peak rate of 2 Mb/s, and a standard deviation of 434 kb/s. New video frames (VFs) arrive every 40 ms. We have set the maximum transmission delay for video packets 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 allowed video packet dropping probability is set to 0.0001 [5].

H.263 is a video standard that can be used for compressing the moving picture component of audiovisual services at low bit rates. It adopts the idea of PB frame, i.e., two pictures being coded as a unit. Thus a PB-frame consists of one P-picture which is predicted from the previous decoded P-picture and one B-picture which is predicted from both the previously decoded P-picture and the P-picture currently being decoded. The name "B-picture" was chosen because parts of B-pictures may be bidirectionally predicted from the past and future pictures. With this coding option, the picture rate can be increased considerably without increasing the bit rate much [13].

In our study, we use again the trace statistics of actual H.263 streams from [12], and in particular the streams of the same movies that we study with MPEG-4 encoding. We have used the unspecified target bit rate (VBR) coding version of the movie, which has a mean bit rate of 91 kb/s, a peak rate of 500 kb/s, and a standard deviation of 32.7 kb/s. In this case, new VF arrives every 80 ms. We have set the maximum transmission delay for video packets to 80 ms, with packets being dropped when this deadline is reached. That is, again, all video packets of a VF must be delivered before the next VF arrives. The allowed video packet dropping probability is again set to 0.0001 [5].

D. Actions of Voice, Video, and Data Terminals, BS Scheduling, and Voice-Data Transmission Protocols

Voice and data (e-mail and web) 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 by the BS. The BS 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. They convey their requirements to the BS by transmitting them within the header of the first packet of their current VF, in the first idle channel information slot.

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 (fixed or extra) request interval*, and follows a different allocation policy for video terminals than that for voice terminals.

Video terminals have the highest 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 as possible of the slots they requested (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 type of allocation case, the BS allocates *the earliest available* information slots to the video terminals. This allocation takes place at the end of the first extra request interval after the arrival of a new VF (in the rare case of a full frame, i.e., a frame with no ER slots, the allocation takes place at the end of the fixed request interval of the next channel frame). Video terminals keep these slots in the following channel frames, if needed, until the next VF arrives.

Also, 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 [1], where it was first introduced). It should be pointed out that, if the arrivals of VFs were not strictly regular, the implementation of our reshuffling policy in the real world would practically be inapplicable, as the BS would not know when a new VF would arrive. Still, our reshuffling policy can be implemented, as the strict regularity in the arrival of VFs is an inherent characteristic of both the MPEG-4 and H.263 encoded movies.

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 80 channel frames here), and thus video terminals would not find enough slots to transmit in; hence, the particularly strict video QoS requirements (the maximum allowed video packet dropping probability is only 0.0001) would be violated. Consequently, *the BS allocates a slot to each requesting voice terminal with a probability* $\boldsymbol{p}^*$ [1], [2]. In [1] and [2], the probability $\boldsymbol{p}^*$ for the allocation of slots to voice users varied according to the video load. In this study, a near-optimal value of $\boldsymbol{p}^*$ has been found through extensive simulations, which works well for all video loads examined. The requests of voice terminals which "-fail-" to acquire a slot, based on the above BS slot allocation policy, remain queued. The same holds for the case when 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).

In addition, the BS "preempts" e-mail and web reservations (starting with web reservations, as web traffic is more delay-tolerant) 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 e-mail and web request queue. No data preemption is executed by the BS to favor video users. This design choice was made for two reasons.

1) The first reason is that we have studied a scenario with data preemption in favor of video, and our simulation results have shown a very significant increase in data delay. The reason for this increase is the "greediness" of video users in terms of bandwidth and QoS requirements (video packet dropping was about 25% smaller in this case). Still, the significant increase in data message transmission delays led to smaller channel throughputs achieved by the system, as, for the same traffic loads, the upper bound set on web and e-mail data delay was exceeded much more quickly than in the case when no data preemption in favor of video was implemented.
2) The second reason is that voice traffic, although less demanding than video traffic in its maximum packet dropping requirements, has equally strict delay requirements with video traffic. Given that voice traffic is much restricted by the $\boldsymbol{p}^*$ policy, in order to facilitate video traffic transmission, we allow voice users the small advantage of solely "exploiting" the preemption mechanism.

Among data terminals, e-mail users have a higher priority than web users and are serviced first by the BS.

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 [4], two-cell stack [14], controlled Aloha [15], three-cell stack [16]). 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) [4] and PRMA-like algorithms, such as those in [5] and [17]. Another important reason for the choice of this algorithm is that it offers a clear indication of when voice contention has ended (this happens when the two consecutive "noncollision" signals are transmitted by the BS in the downlink); therefore, this algorithm supports the prioritization mechanism used for voice and data access to the requested minislots.

The two-cell stack blocked access collision resolution algorithm [14] is adopted for use by the data terminals in order to transmit their data request packets. This algorithm is of window type, with FCFS-like service.

E. Call Admission Control Versus Traffic Policing

Quite a few efficient call Admission control (CAC) algorithms have been proposed in the literature for the transmission of voice, data, and multimedia traffic over wireless networks (e.g., [18]–[21]). However, all these mechanisms suffer from a necessary conservatism in their estimation of the channel bandwidth consumed by the multiplexed sources, in order to preserve system stability and the users' QoS requirements. In [22], we have reached the conclusion that the adoption of a CAC mechanism for the transmission of videoconference traffic from MPEG-4 and H.263 video coders in a wireless network leads to significant throughput deterioration in comparison to the adoption of a strict traffic policing mechanism; i.e., traffic control is implemented much more effectively inside the system than at its entrance.

Therefore, in this work we have modified the traffic policing mechanism of [22] in order to be implemented for multimedia traffic, and not solely for video, as was the case in [22], where video users of various coding qualities where considered and in the case of high packet dropping users were asked to decrease their demands by moving to a lower quality coding. According to our modified mechanism, the system checks the average video and voice packet dropping in regular time periods and, if either exceeds its set upper bound in two consecutive checks, the call of the last user (video, voice, or data) that entered the system is terminated, as this user is considered to be responsible for the deterioration in QoS requirements. This procedure continues until both the video and voice packet dropping are below their set upper bounds. In [22], an optimal check-time period was chosen to be equal to 1 min. In this work, where the problem is more complicated due to the existence of various types of traffic in the system, we have found via extensive simulations that an optimal check-time period is equal to 0.4 min, i.e., 2000 channel frames (therefore, the check-time "window" of two consecutive checks is equal to 0.8 min). As in [22], significantly smaller windows have the disadvantages of increased system complexity and the occasional problem of the system exceeding the upper bound for only a short time period and activating the call termination mechanism although there may be no need for such action; significantly larger windows, on the other hand, have the problem that excessive video or voice packet dropping is often identified with delay and therefore QoS guaranteed to the users is violated for an unacceptable amount of time.

III. Channel Error Model

The most widely adopted wireless channel error model in the literature is the Gilbert-Elliot model [23], [24]. The Gilbert-Elliot model is a two-state Markov model where the channel switches between a "good state" (always error-free) and a "bad state" (error-prone). However, many recent studies have shown that the Gilbert-Elliot model fails to predict performance measures depending on longer-term correlation of errors [25], minimizes channel capacity [26], and leads to a highly conservative allocation strategy [27].

A better choice for a more robust error model for wireless channels is the model presented in [28], which we adopt in our study. This model, with the use of the short and long error bursts, makes more accurate predictions of the long-term correlation of wireless channel errors than the Gilbert-Elliot model. The error model consists of a three-state discrete-time Markov chain, where one state is the "good state" (error-free) and the other two states are the "bad states," the *long bad* and the *short bad* state (the Markov chain is shown in Fig. 2). A transmission is successful only if the channel is in the "good state" (G); otherwise, it fails. The difference between the long bad (LB) and short bad (SB) states is the time correlation of errors: LB corresponds to long bursts of errors, SB to short ones.

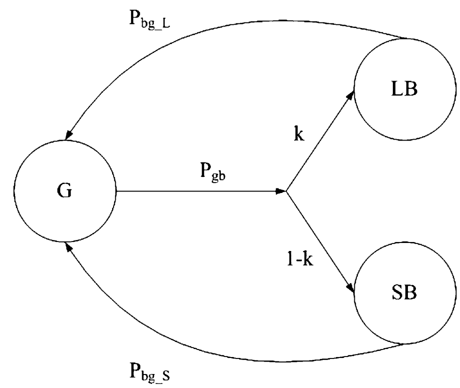

Fig. 2. Channel Error Model.

TABLE I
Error Model Parameters

P_{good}=0.99992
B_G=1/p_{gb}=65160 slots
B_{SB}=1/p_{bg_S}=2.38 slots
B_{LB}=1/p_{bg_L}=59.53 slots
k=0.05

The parameters of the error model are presented in Table I. The average number of error bursts, in slots, experienced when the states LB and SB are entered, are, respectively, given by $\mathrm{B_{LB}} = 1/\mathrm{p_{bg\_L}}$ and $\mathrm{B_{SB}} = 1/\mathrm{p_{bg\_S}}$, where $\mathrm{p_{bg\_S}}$ is the transition probability from state SB to G, and $\mathrm{p_{bg\_L}}$ is the transition probability from state LB to G. Similarly, the average number of consecutive error-free slots is given by $\mathrm{B_G} = 1/\mathrm{p_{gb}}$, where $\mathrm{p_{gb}}$ is the probability to leave state G. The parameter k is the probability that the Markov chain moves to state LB, given that it leaves state G; k also represents the probability that an error burst is long (i.e., the fraction of long bursts over the total number of error bursts).

We have chosen in our study the value of the probability $\mathrm{P_{bad}}$, i.e., the steady-state probability that the channel is in a bad state, to be equal to 8×10^{-5}; this value has been chosen in order to test an "almost worst" case scenario for our system, as the video packet dropping probability is set to 10^{-4} and, by choosing a value of bad state probability larger than the upper bound on video packet dropping, the strict QoS requirement of video users will certainly be violated. The values for $\mathrm{p_{gb}}$ and for the parameter k have been taken from [28], as well as the ratio between $\mathrm{p_{bg\_S}}$ and $\mathrm{p_{bg\_L}}$. The value for $\mathrm{p_{bg\_L}}$ is derived from the steady-state behavior of the Markov chain, for the bad state probability chosen.

IV. System Parameters

The channel rate is 9.045 Mb/s. The 12 ms of frame duration accommodate 256 slots. These parameters are taken from [5], where the DPRMA scheme (with which we compare the performance of MI-MAC) was introduced.

The value of the *probability* $\boldsymbol{p}^*$ is chosen equal to 0.09 (9%). Other values of $\boldsymbol{p}^*$ have also been tried out through simulation, and it has been found that the chosen value gives very satisfactory results for all the examined cases of video load. Due to the complexity of the scheme, the optimal $\boldsymbol{p}^*$ value is extremely difficult to obtain analytically.

V. Results and Discussion

We use computer simulations executed on Pentium-IV workstations to study the performance of MI-MAC. Each simulation point is the result of an average of 10 independent runs (Monte Carlo simulation), each simulating 305 000 frames (the first 5 000 of which are used as warm-up period).

A. DPRMA

The DPRMA protocol [5] was inspired by PRMA [4] and proposed for accommodating multimedia traffic, as PRMA was optimized for systems for voice traffic only.

The basic differences of DPRMA and MI-MAC are the following.

The first difference exists in the scheduling mechanism for video sources. The BS in DPRMA does not use a reshuffling policy like MI-MAC and does not grant the earliest available information slots to requesting users. The BS first identifies which slots are currently unallocated and determines how many such slots exist. Next, it examines each of these slots in sequential order to determine if the slot will be assigned. Throughout the process, the BS maintains a record of how many slots Sn the user n (the user currently serviced by the BS) still needs. Every time a slot is successfully assigned, Sn is decremented. In addition, the BS keeps track of the number of available slots Sc that have not yet been considered for assignment. Each time a new slot is considered, Sc is decremented. As the BS sequentially considers each available slot, it assigns each one with probability Pa, where $Pa = Sn/Sc$. Thus, the probability that a slot is assigned is dependent upon how many slots are still needed to satisfy a user's request. This process tends to *spread the allocation of slots randomly throughout the frame*.

The second difference concerns the video and data traffic considered in each scheme. DPRMA uses a video traffic model from [29], which is based on H.261 videoconference traffic (i.e., a model for video traffic from past technology encoders). Also, DPRMA considers an abstract model for data traffic (not referring to a specific type of data traffic), with which data packets (i.e., not messages) are generated independently from each other according to a Poisson process.

The third difference is the use of certain transmission rates in DPRMA for all types of users. In DPRMA, a user continuously determines the appropriate reservation request that ensures timely delivery of its traffic. Newly generated packets are queued in a buffer as they await transmission. As the size of the queue grows, the user increases its reservation request to avoid excessive transmission delay. If the queue length subsequently decreases, the user then requests a lower reservation rate to avoid running out of packets. The buffer size that corresponds to an increase or decrease in the reservation request is defined as a threshold. DPRMA uses seven threshold levels, and, respectively, seven transmission rates for video users (the lowest rate is 70.667 kb/s and the highest 4.523 Mb/s). One pair of up- and down-threshold levels is implemented for data users, and one pair for voice users.

The fourth difference is that DPRMA uses neither request slots nor our idea of $\boldsymbol{p}^*$, but adopts a PRMA-like approach for voice and data users, by allowing them to compete for the available information slots by transmitting their packets according to a probability ($\mathrm{P_{tv}} = 0.05$ for voice, $\mathrm{P_{td}} = 0.007$ for data traffic). No transmission probability is needed for video users because in DPRMA, as in MI-MAC, video users are considered to have obtained reservations prior to the beginning of the simulation.

The fifth difference is that, in DPRMA, both voice and video users waste one slot when giving up their reservations. This does not happen in MI-MAC because of the VAD used for voice terminals and because the BS knows exactly when a video user has transmitted all the packets of its VF (since video users convey this information to the BS, whereas in DPRMA they convey only at times a reservation request rate in order to keep the content of their video packet buffers below certain thresholds).

The sixth difference is that DPRMA employs data preemption in favour of both video and voice users.

B. DPRMA*

Since DPRMA was evaluated for different types of multimedia traffic than the ones considered in MI-MAC, we have modified DPRMA very slightly, in order to be able to make comparisons with MI-MAC. We will refer to this modified protocol as DPRMA*. DPRMA* has four differences in comparison to DPRMA:

First, it is implemented on the same types of multimedia traffic as MI-MAC, and second, its performance is evaluated under the same channel error model as MI-MAC.

Third, in DPRMA, it was found that in a data-only system (for the data model used in [5]) nearly identical performance was achieved when $0.006 < \mathrm{P_{td}} < 0.1$. Since data users are given lowest priority, the lowest possible value of $\mathrm{P_{td}} = 0.007$ was chosen in the scheme. In DPRMA*, where two types of data traffic (e-mail and web) are considered and web traffic is much more delay-tolerant, the web data transmission probability is kept to the lowest possible value of $\mathrm{P_{tdw}} = 0.007$, whereas for e-mail data traffic the transmission probability is chosen equal to $\mathrm{P_{tde}} = 0.0105$, (i.e., 50% higher than $\mathrm{P_{tdw}}$). Our results have shown that this choice for $\mathrm{P_{tde}}$ provides much smaller average access delays for e-mail traffic than the delays provided by lower values of $\mathrm{P_{tde}}$ and does not severely influence the QoS of voice traffic, as was the case with much higher values of $\mathrm{P_{tde}}$.

TABLE II
VOICE CAPACITY AND CHANNEL THROUGHPUT FOR VARIOUS MPEG-4 VIDEO LOADS

Number of Video Users	**Voice Capacity (Maximum Number of Voice Terminals)**		**Channel Throughput (%)**	
	MI-MAC	**DPRMA***	**MI-MAC**	**DPRMA***
9	0	x	43.9	x
8	77	70	51.9	50.5
7	124	109	54.8	52.0
6	190	171	60.9	57.6
5	238	212	64.0	59.4
4	279	265	65.9	63.3
3	336	328	70.5	68.9
2	378	373	72.6	71.5
1	402	399	71.7	70.9

TABLE III
VOICE CAPACITY AND CHANNEL THROUGHPUT FOR VARIOUS MPEG-4 VIDEO AND E-MAIL DATA LOADS, FOR $\lambda_{\mathrm{web}} = 1$ SESSION/SECOND

Number of Video Users	**λ_{email} (messages/frame)**	**Voice Capacity (Maximum Number of Voice Terminals)**		**Channel Throughput (%)**	
		MI-MAC	**DPRMA***	**MI-MAC**	**DPRMA***
1	0.1	314	308	61.79	60.80
	0.2	302	288	62.94	60.59
	0.4	268	244	63.54	58.65
3	0.1	273	265	64.78	63.46
	0.2	260	243	65.76	63.21
	0.4	227	201	66.53	62.35
5	0.1	180	171	59.02	57.48
	0.2	167	151	60.00	57.34
	0.4	146	112	62.79	56.29
7	0.1	77	65	51.58	50.60
	0.2	63	43	52.39	49.11
	0.4	26	x	52.49	x

Fourth, for "fairness" reasons, i.e., for being able to compare the "best" possible version of DPRMA* with MI-MAC, we have slightly changed the number of threshold levels' transmission rates for video users. In the case of H.263 traffic, which is less demanding in bandwidth, six transmission rates were used (and, respectively, six threshold levels, calculated in the same way as in DPRMA). The lowest rate was equal to 35.333 kb/s and the highest rate was equal to 1.13 Mb/s. In the case of MPEG-4 traffic, which has a much lower mean rate than the H.261 traffic used in DPRMA, but a higher standard deviation and comparable peak rate, we have used eight transmission rates (and, respectively, 8 threshold levels). The lowest rate was equal to 35.333 kb/s and the highest rate was equal to 4.523 Mb/s.

No other differences exist between DPRMA and DPRMA*.

C. Results for MPEG-4 Video Traffic

Table II presents the simulation results of MI-MAC and DPRMA*, when integrating only voice and MPEG-4 video streams and satisfying the QoS requirements of both traffic types. As expected, the very bursty nature of the video streams leads to a significant decrease of the maximum voice capacity (and, consequently, of the channel throughput) as the number of video users in the system increases. It should be pointed out that the reason for which the channel throughput is smaller in the case of one video user than in the case of two video users in the system (for both MI-MAC and DPRMA*) is explained by the fact that one video stream is burstier than the superposition of multiple video streams. As the number of video terminals increases, their traffic superposition is smoother; however, the load increase is great and leads to the decrease of the channel throughput, due to the burstiness and strict QoS requirements of video users. It is evident from all the results presented in Table II that MI-MAC outperforms DPRMA* for all the combined loads, and their difference in throughput ranges from a minimum of 0.8% to a maximum of 4.6%. It should also be noted that DPRMA* fails to accommodate nine video users (this inability is symbolized by "x" in the table), in contrast to MI-MAC which is capable of accommodating this video traffic load.

The above explanation for the decrease/increase in channel throughput/number of video users and the burstiness of a single stream stands also for the results of MI-MAC shown in Table III. Table III presents the simulation results when integrating all four traffic types: voice, MPEG-4 video streams, e-mail data, and web sessions, for both MI-MAC and DPRMA*. For various video loads and for a fixed arrival rate of web sessions (λ_{web} sessions per second), we present the voice capacity of each scheme for different e-mail message arrival rates (λ_{email} messages/frame), as well as the corresponding channel throughput. We examine the cases of λ_{email} being equal to 0.1, 0.2, and 0.4 messages/frame, (i.e., 256 kb/s, 512 kb/s, and 1.024 Mb/s, respectively), and we observe from our results that in MI-MAC, for a given number of video terminals, as λ_{email} increases, the channel throughput increases as well. This proves the efficiency of our data preemption mechanism, which allows the incorporation of larger data message arrival rates into the system without significant reduction of the voice capacity or violating the strict QoS requirements of video and voice traffic. The reason for the reduction of the voice capacity, despite the data preemption mechanism in favor of voice, is the fact that data users are not preempted in favor of video users as well, and thus fewer voice users can enter the system in order to preserve the strict QoS requirements of the video traffic.

The results of DPRMA* show that the choice of preemption of data users in favor of both video and voice users leads to throughput deterioration when λ_{email} increases, as e-mail message and web request delays quickly exceed the set upper bounds and the system becomes unable to accommodate these traffic loads for a larger number of voice users. Of course, the data preemption policy is not the only reason that MI-MAC achieves better throughput results than DPRMA* (their difference in throughput ranges from 1% to 6.5%). The other reasons which cause MI-MAC to excel are as follows.

1) Our reshuffling policy ensures a much more timely slot allocation to video users than the mechanism making

random spreading of allocated slots that is used in DPRMA and DPRMA*.

2) The use of a number of transmission rates in DPRMA* increases system complexity without ensuring that the terminal will be allocated the maximum possible number of slots in each frame, based on its needs. The use of the transmission rates does guarantee that, in the long run, the terminal will be well serviced, but this policy is proven inadequate from our results, as the very strict video packet dropping requirement asks for the best possible *short-term* (i.e., for every VF) allocation.
3) By using the two-cell stack random access algorithm, MI-MAC allows voice users to make their requests to the BS more effectively than DPRMA, which uses the PRMA algorithm for that purpose. The "obstacle" put to the voice users in acquiring a slot ($\boldsymbol{p}^*$) is set in MI-MAC thus: After they have sent their request to the BS, they will wait in the queue at the BS for a possible slot allocation *without having to further contend*. On the contrary, in DPRMA the "obstacle" to voice and data users is set by using a small transmission probability for contention. The latter approach is less effective because voice and data users must *repeatedly enter contention* in order to reserve a slot, thus leaving more information slots unused. Additionally, the use of ER slots helps MI-MAC "exploit" certain available slots that DPRMA leaves unused.
4) Unlike MI-MAC, in DPRMA*, a slot is wasted each time a user gives up its reservation.

The results in both Tables II and III show that MI-MAC achieves very satisfactory channel throughputs (over 60% and occasionally over 70%) when the number of video users is between 1 and 5, i.e., for low and medium capacity video traffic. When the number of video users becomes higher, the throughput decreases below 60%, due, again, to the very bursty nature of video traffic and its very stringent QoS requirements. If, however, the QoS requirements for video traffic were more "tolerant" [e.g., if we considered the acceptable upper bound of the video packet dropping probability to be 0.001 instead of 0.0001, which would be reasonable, given the nature of the movie (videoconferencing)], then the throughput of our scheme would increase considerably (by about 8–10%, as observed from our simulations, but not shown here because the focus of the paper is on results under very strict video QoS requirements).

Fig. 3 presents the e-mail data message delay versus the e-mail message arrival rate, when 200 voice users and 3 MPEG-4 video users are present in the system (no web traffic is considered in this scenario). We observe that the average e-mail message delay in MI-MAC is steadily much lower than in DPRMA*, for the reasons explained above. Also, as expected, the data message delay increases quickly in both schemes as λ increases. This quick increase is a result of the fact that newly arriving e-mail data messages are preempted by voice and video users in DPRMA*, and preempted by voice users and queued in favor of newly arriving video streams in MI-MAC. As shown in Fig. 3, for an arrival rate λ_{email} larger than 0.32 messages/frame (which, added to the load of the other traffic types, corresponds to a channel throughput of 58.57%) the system is unable to sustain the channel load, as the average e-mail data message delay exceeds the assumed delay limit of 5 seconds. The same upper bound is exceeded by DPRMA* for an arrival rate larger than 0.24 messages/frames.

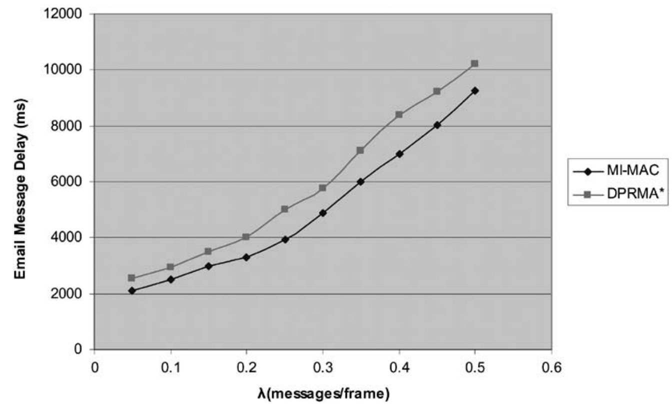

Fig. 3. Average e-mail message delay versus e-mail message arrival rate, for $\text{Nvoice} = 200$ and MPEG-4 video users $= 3$.

One more significant point that needs to be made is that all the comparisons between MI-MAC and DPRMA* are made based on the maximum traffic load which can be accommodated by each scheme. DPRMA*, like DPRMA, lacks a traffic policing mechanism such as the one used in MI-MAC, which can guarantee a balanced performance of the scheme when new users enter the system. However, for "fairness" reasons again, we limit our comparisons to only the scheduling efficiency of the two schemes.

D. *Results for H.263 Video Traffic*

The comments made in Section V.C on the results of MI-MAC for MPEG-4 traffic stand for the respective results for H.263 traffic as well.

Table IV presents the simulation results of MI-MAC and DPRMA*, when integrating voice and H.263 video streams and satisfying the QoS requirements of both traffic types. Again, as in the case of the MPEG-4 video streams, the bursty nature of the video streams leads to a significant decrease of the maximum voice capacity (and, consequently, of the channel throughput) as the number of video users in the system increases. Still, in this case, as we can see from the results in Table IV, and as expected from the different traffic parameters of the H.263 video streams, a significantly larger number of video users can be supported (up to 59, when no voice users are present in the system, whereas DPRMA* cannot accommodate this traffic load). Of course, this huge "gain," when comparing the number of H.263 movies that the system is able to accommodate with the respective number of MPEG-4 movies, is obtained at the cost of video quality (which is quite better when using MPEG-4 encoding). From the throughput results of Table IV, we observe that the channel throughput ranges from 65% to 70% and is clearly larger than the throughput achieved by DPRMA*, for all the examined loads. The reason that MI-MAC's throughput does not "fall" as low as 50%, which happens when studying the MPEG-4 encoding of the same movies (see the results in Table II), is that

TABLE IV
VOICE CAPACITY AND CHANNEL THROUGHPUT FOR VARIOUS H.263 VIDEO LOADS

Number of Video Users	Voice Capacity (Maximum Number of Voice Terminals)		Channel Throughput (%)	
	MI-MAC	DPRMA*	MI-MAC	DPRMA*
59	0	x	65.5	x
50	60	48	65.6	63.6
45	96	81	66.0	63.5
40	130	109	66.2	62.6
35	164	140	66.3	62.4
30	198	179	66.5	63.3
25	234	218	67.0	63.8
20	269	256	67.3	65.2
15	307	294	68.1	65.9
10	347	338	69.2	67.7
5	386	383	70.2	69.7

TABLE V
VOICE CAPACITY AND CHANNEL THROUGHPUT FOR VARIOUS H.263 VIDEO AND E-MAIL DATA LOADS, FOR $\lambda_{\rm web} = 1$ SESSION/SECOND

Number of Video Users	λ (messages/frame)	Voice Capacity (Maximum Number of Voice Terminals)		Channel Throughput (%)	
		MI-MAC	DPRMA*	MI-MAC	DPRMA*
10	0.1	307	299	66.91	65.58
	0.2	289	272	67.05	64.24
	0.4	254	229	67.49	63.35
20	0.1	226	215	64.53	62.71
	0.2	208	187	64.67	61.19
	0.4	175	145	65.44	60.47
30	0.1	151	135	63.16	60.51
	0.2	134	106	63.46	58.83
	0.4	102	67	64.41	58.62
40	0.1	80	58	62.46	58.81
	0.2	64	28	62.93	56.97
	0.4	31	x	63.71	x

the H.263 encoding of these movies has a considerably smaller standard deviation/mean ratio.

Table V presents the simulation results of MI-MAC and DPRMA*, when integrating voice, H.263 video streams, e-mail data, and web sessions. We present again the voice capacity and the corresponding channel throughput for a fixed $\lambda_{\rm web}$ rate and for different values of $\lambda_{\rm email.}$ When examining the cases of $\lambda_{\rm email}$ being equal to 0.1, 0.2, and 0.4 messages/frame, we observe once more that, for a given number of video terminals, as $\lambda_{\rm email}$ increases, the channel throughput in MI-MAC increases as well. It should also be noted that

1) the reason for the considerably larger channel throughput achieved in this case (than in the case of the MPEG-4 movies) by MI-MAC is again the smaller standard deviation/mean ratio of the H.263 movies;

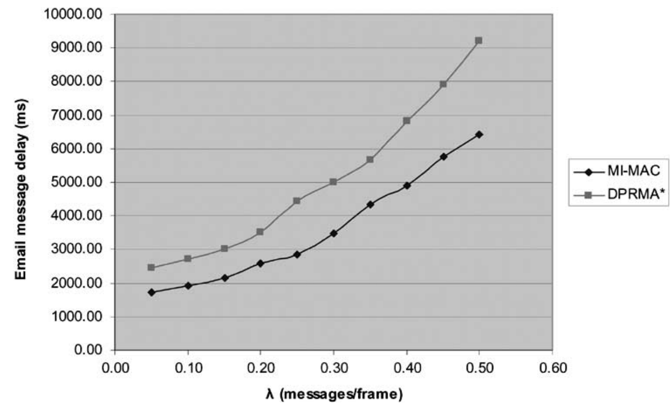

Fig. 4. Average e-mail data message delay versus e-mail message arrival rate, for $\rm Nvoice = 155$ and H.263 video users $= 20$.

2) the reasons for choosing a relatively low web session arrival rate (1 session/s) are the limited priority of web traffic compared to the other three types of traffic and the long duration of a web session (due to the large viewing time between web requests) compared to the duration of our simulation study. Based on the traffic parameters of our web model, it can easily be computed that a web session needs, on average, around 16–17 min to be completed. (This amount of time does not include the download time for the corresponding web content of each request, as in this work we do not study the downlink channel) Therefore, although our web traffic metric for system performance is the average web request transmission delay, we wanted to estimate the web session delay as well, which would be difficult if a vast number of web sessions were still unfinished. Still, in the results presented in Fig. 5 we have studied system behavior with larger web session arrival rates.

Fig. 4 presents the e-mail data message delay versus the e-mail message arrival rate for MI-MAC and DPRMA*, when 155 voice users and 20 H.263 video users are present in the system (again, as in Fig. 3, no web traffic is considered in this scenario). The total channel load of voice and video users has been selected to be equal to the corresponding load used in Fig. 3. We observe again that MI-MAC achieves much smaller e-mail data message delays and that the e-mail data message delay increases quickly as $\lambda_{\rm email}$ increases, but in this case the data message delays are not as high as those experienced by data users when MPEG-4 video users are present in the system. This again happens due to the fact that MPEG-4 video streams are much burstier and bandwidth-consuming than the corresponding H.263 ones. As shown in Fig. 4, for an arrival rate $\lambda_{\rm email}$ larger than 0.41 messages/frame, the system is unable to sustain the channel load, as the average e-mail data message delay exceeds the assumed delay limit of 5 s. The same upper bound is exceeded by DPRMA* for an arrival rate larger than 0.3 messages/frames.

Fig. 5 presents the average web request delay versus the web session arrival rate, when 83 voice users, 25 H.263 video users, and an e-mail arrival rate of $\lambda_{\rm email} = 0.2$ messages/frame are

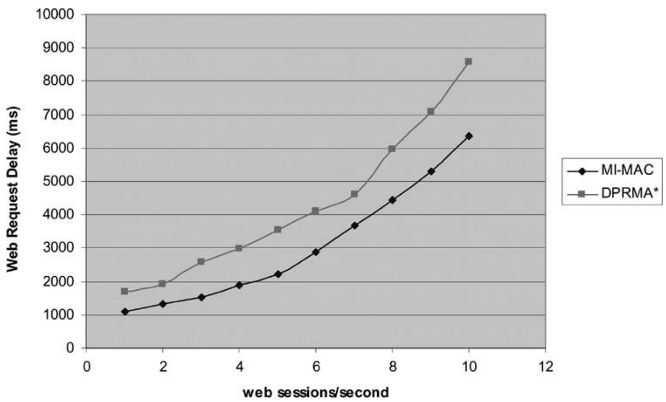

Fig. 5. Average web request delay versus web session arrival rate, for $\text{Nvoice} = 83$, H.263 video users $= 25$, and $\lambda_{\text{email}} = 0.2$ messages/frame.

present in the system. The combined load of the three traffic types has been chosen to correspond to a 50% channel utilization. As in Figs. 3 and 4, where we focused on the behavior of the e-mail message delay, we observe that the web data message delay increases quickly as λ_{web} increases, for both MI-MAC and DPRMA*. This quick increase is once more a result of the fact that new web requests (arriving in sessions) are preempted by voice and video users in DPRMA*, and preempted by voice users and queued in favor of newly arriving video streams in MI-MAC. As shown in Fig. 5, for an arrival rate λ_{web} higher than 6 sessions/s (which corresponds to a channel throughput of 55.3%) the system is unable to sustain the channel load in MI-MAC, as the average web request delay exceeds the assumed upper delay bound of 3 s. The same upper bound is exceeded by DPRMA* for an arrival rate higher than 4 sessions/s.

E. General Comments and Discussion

Our results for MI-MAC, for both cases of MPEG-4 or H.263 video traffic multiplexed with voice and data traffic, can be used for the development of a nonconservative CAC mechanism (this work is currently in progress from our group), which, based on a large set of precomputed traffic scenarios, could allow or deny the entrance to the system to a newly arriving voice, video, or data terminal. For example, with the use of our results, where the system is working at its "limits" (i.e., with the maximum combined traffic load for which QoS for all traffic types is not violated), the behavior of the system for different combined traffic loads can be predicted. Such a mechanism could work in cooperation with a scheme for estimating the equivalent bandwidth of multiplexed sources, as the total number of possible traffic scenarios is infinite and therefore only a limited number can be computed. This mechanism would also help restrict the conservatism that schemes like [18]–[22] necessarily had, in order to provide the required QoS. The equivalent bandwidth estimation will still be a factor of conservatism,[1] but this conservatism will be very limited, since the original traffic load will be the maximum "bearable" by the system.

[1]Equivalent bandwidth algorithms have been shown in the relevant literature to provide quite conservative estimations in comparison to the actual sources' behavior.

One more comment that needs to be made is that, as the results from our tables and figures make clear, the maximum throughput achieved by MI-MAC is close to 73%. Hence, it could be argued that, since many information slots remain unused in each frame, these slots can be used as ER slots and therefore the existence and the adjustability of dedicated request slots is not critical to system performance. However, this is correct only in the case on which we focus in this study (i.e., in the case that video traffic is present in the system), and the system cannot be designed on the hypothesis that there will necessarily be active video users in a cell. It is often the case that only voice and data users exist in the cell, and in this case many MAC protocols in the literature have shown that channel throughput (in the absence of the very bursty and demanding in QoS video traffic) exceeds 80% and is often higher than 90%, even close to 98% [30]–[32]. Therefore, the existence of a dedicated adjustable number of request slots is critical to the overall system performance as, in their absence, and with the limited availability of ER slots, voice and data terminals would find it difficult to access the channel.

Finally, it should be pointed out that an efficient MAC scheme requires a dynamic adaptive error handling capability. Video packet loss, due to violation of the maximum video packet transmission delay limit, can result in significant damage of the transmitted image, especially if the dropped packets belong to a VF containing significant information. This corresponds, for both MPEG-4 and H.263 traffic, to the loss of an I-frame, since P- and B-frames are used for prediction, by exploiting their similarity with previously encoded frames. The loss of an I-frame leads to a decoding error for the whole group of pictures related to the specific I-frame, for both MPEG-4 and H.263 streams. Therefore, it could happen that the average video packet dropping probability in the system remains below 10^{-4}, but the packets lost belonged to I-frames, and therefore significant information is lost at the receiver. For this reason, considerable work has been conducted and presented in the literature on the development of forward error correction (FEC) schemes for wireless networks [33]–[35]. Still, in order to minimize the impact of information loss and at the same time decrease the channel load, FEC schemes need to be combined with efficient buffer management techniques. Towards this direction, the technique presented in [36] can be modified for the wireless environment. More specifically, [35] proposes a combination of a FEC scheme for ATM networks with the partial packet discard (PPD) scheme proposed in [37] and [38]. The PPD scheme proposes dropping of subsequent arriving cells from a block which is already known to be lost. This policy can be used in our scheme, considering that an I-frame is generated from the video coder in each change of "state" in the video Markov model. The idea is that a terminal which has failed to transmit its I-frame with the desired QoS (i.e., it has experienced large packet dropping) does not transmit the subsequent P- and B-frames, since the I-frame contained the critical information on which the next frames' prediction is based. This policy can help relieve the "burden" of the network from the transmission of less significant information and therefore improve the QoS for all other video, voice, and data users, while at the same time it can significantly restrict video loss at the receiver. Still, all FEC schemes increase the number of cells being transmitted, and a tradeoff exists between the redundant information transmitted for FEC purposes and the improvements in QoS provided by the FEC mechanism. A work on this subject from our group is also currently in progress.

F. Comparison With WCDMA MAC Protocols

As MI-MAC is based on TDMA, we proceed on a *conceptual performance comparison* with some efficient wideband CDMA (WCDMA) MAC schemes from the literature. The lack of a direct comparison is also due to the fact that, to the best of our knowledge, no WCDMA MAC schemes proposed in the literature have addressed the problem of integrating actual video traces from the latest technology encoders with voice and data traffic.

WCDMA is an asynchronous scheme, using a wide (5 MHz) carrier to achieve high data rates. It can be categorized into pure WCDMA and wideband time-division CDMA (TD-CDMA). Pure WCDMA uses frequency division duplex (FDD) to organize the uplink and downlink transmissions, while wideband TD-CDMA uses time-division duplex.

In [39], the authors show that the common assumption that the random access channel in a WCDMA MAC has comparable throughput to that of slotted Aloha is not correct, and that resource utilization is significantly higher than e^{-1}. Nevertheless, in [39] system throughput reaches a maximum of 74% (i.e., merely larger than the maximum throughput achieved by MI-MAC) *although no video traffic is present*, which leads to a logical expectation of a severe deterioration in system throughput if video traffic were to be introduced in the system.

Another efficient WCDMA MAC scheme was proposed in [40]. This scheme studies the integration of multimedia (voice, data, videoconference) traffic and is shown to achieve very high throughput (close to 90% of the theoretical maximum) when the load from all traffic types is increased. However, this result is obtained due to the authors' choice of considering CBR videoconference traffic of low rate and very "loose" maximum packet delay requirements, voice traffic of a much lower rate than the one used in our study and data traffic of a very low rate with no delay requirements. Hence, in a scenario with bursty VBR videoconference, and much more demanding voice and data traffic system throughput, performance is again expected to significantly deteriorate.

One more efficient FDD WCDMA MAC scheme is proposed in [41]. Six types of traffic are integrated in the scheme (voice, audio, CBR video, VBR video, computer data messages, and e-mail messages). The voice traffic model used is the same as the one in our scheme; the average e-mail message size is slightly smaller than the one in our study, and the VBR video traffic mean and peak rates are similar to the ones used in our study for H.263 videos. However, once again, VBR video traffic is based on a video model. Throughput results are presented in [41] only for a scenario with voice, audio, CBR video, and VBR video (i.e., without computer data and e-mail). In these results, the scheme's throughput, in packets/frame, for the QoS requirements considered in our work for voice and VBR video traffic is lower than the throughput achieved by MI-MAC in all the examined cases of voice-video and voice-video-data integration.

VI. Conclusion

In this paper, we have proposed and investigated the performance of a new medium access control protocol for wireless communications. Our protocol, MI-MAC, integrates voice, MPEG-4 or H.263 video, e-mail, and web packet traffic over a noisy wireless channel of high capacity. With the use of a dynamic frame structure and an efficient scheduling policy which we design and propose, our scheme, which is one of the first in the literature to study the integration of MPEG-4 or H.263 video streams with other traffic types, is shown to achieve high aggregate channel throughput in all cases of traffic load examined, while preserving the QoS requirements of each traffic type.

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