A new framework for multiple access and call admission control in wireless cellular networks

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A B S T R A C T

In recent work, we have introduced MI-MAC (Multimedia Integration Multiple Access Control), a new access control protocol for wireless cellular networks. MI-MAC was shown to be a good candidate for next generation wireless cellular networks, due to its superior performance in comparison to other (TDMA and WCDMA-based) protocols in the literature when integrating various types of multimedia traffic. In this paper we propose the combination of MI-MAC with a new efficient call admission control (CAC) mechanism, which will prevent bursty H.264 users from entering the network if system stability is not guaranteed. We proceed to evaluate the ability of our framework to efficiently integrate streams from latest technology video encoders with other types of packet traffic over noisy wireless networks, especially in the case of significant handoff loads. To the best of our knowledge, this is one of the first works in the literature to study the integration of H.264 streams with other types of multimedia traffic over wireless cellular networks.

1. Introduction

In recent work [1], we introduced and evaluated a new multiple access scheme which was shown to efficiently integrate voice (Constant Bit Rate, CBR, On/Off Traffic), email and web traffic with MPEG-4 and H.263 video streams (Variable Bit Rate, VBR) in high capacity picocellular wireless systems with burst-error characteristics. In this work we continue the performance evaluation of the scheme. The new elements of this work, in comparison to [1], are:

(a) the integration of streams from the latest technology video encoding (H.264) with voice and WAP (Wireless Application Protocol) traffic;
(b) the use of a new CAC mechanism which prevents bursty H.264 video users from entering the network if network stability is not guaranteed;
(c) the use of a different channel error model than the one used in [1];
(d) in [1], in order to facilitate the comparison with other protocols of the literature and given that the protocols were evaluated over one cell of the network, no traffic was considered to be arriving from other cells (handoff traffic). This assumption is waived in the present work, where a significant portion of the traffic in our simulations is considered to be handoff.

We focus on the uplink (wireless terminals to base station) channel, where a MAC scheme is required in order to resolve the source terminals contention for channel access. We compare the results of our framework with those of DPRMA, a well-known MAC protocol for wireless networks.

The paper is organized as follows. In Section 2 we analyze our proposed scheduling and CAC schemes, and present the various traffic types used in our study. Section 3 includes the details of the adopted channel error model. The system parameters are presented in Section 4. In Section 5 we discuss our simulation results, and in Section 6 we present our conclusions.

2. Multiple traffic type integration

2.1. 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 the ATM cell size for reasons of comparison with DPRMA [2]; however, the nature of our results remains the same, regardless of the packet size, therefore the scheme could be used in any GSM-type network). 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 WAP 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 2.2.1), we place the voice and data request interval at the beginning of the frame, in order to minimize the voice packet access delay. Request slots can be shared by voice and data terminals, in this priority order. 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 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 (this assumption is again made in order to facilitate our scheme's comparison with the DPRMA protocol [2], which makes the same assumption). Regarding handoff video terminals, it is assumed that their current bandwidth requirements are known to the new Base Station (BS) through interaction with the last BS that serviced the video call.

The frame structure parameters have been chosen as follows:

(a) For all the examined scenarios of system load (a vast number of scenarios has been studied), we tried to find a maximum request bandwidth which would suffice for voice and data terminals. This was found, via simulations (both in [1] and in the present work), to be equal to three request slots.

(b) We design the protocol so that we can enforce a fully dynamic mechanism for the use of the request bandwidth: the number of request slots is variable per channel frame (between 1 and 3, which is the maximum number, as explained above), and depends on the total voice and data channel load in each frame. In the cases when less than three request slots are needed for the end of the voice and data terminals’ contention, the Base Station signals all user terminals for the existence of additional information slots in the current frame. Also, any free information slot of the current channel frame can be temporarily used as an extra request slot (ER slot) [1] (the use of a slot as an ER slot is conveyed to the terminals by the BS after the end of the request interval in each channel frame).

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. Reserved slots are deallocated immediately.

The allowed voice packet dropping probability is set to 0.01, and the maximum transmission delay for voice packets is set to 40 ms [2].

2.2.2. WAP traffic model

We adopted the WAP traffic model presented in [3] (corresponding to the WAP release 1.2.1) in our work. WAP sessions consist of requests for a number of decks, performed by the user. The number of decks is modeled by a geometric distribution with mean equal to 20 decks and the packet size by a log2-normal distribution. To cover the influence of different applications, four different types of user profiles are introduced: email, news, m-commerce and common (referring to mixed traffic traced from a WAP server in real operation). The size of a wap request message in [3] ranges on average between 82 and 112 bytes, depending on the specific user profile, i.e., it ranges between 2 and 3 ATM packets in size. The standard deviation of the size of a wap request message ranges between 16.5 and 84.7 bytes, i.e., between 1 and 2 ATM packets.

The arrival process of WAP sessions is chosen to be Poisson with rate λ/WAP sessions per second, with an upper limit on the average WAP request transmission delay equal to 2 s. Given that the average size of a WAP request is quite small in terms of number of packets, it is clear that we adopt the widely accepted assumption that data traffic is delay-tolerant. Still, if we take into consideration that estimations of GSM networks’ SMS transmission delays refer to delays of 2–30 s [4] (SMS messages have a payload of 140 bytes [5], i.e., similar to a WAP request), the upper bound set in this work for WAP request transmission is quite strict.

2.2.3. H.264 video streams

H.264 is the latest international video coding standard. It was jointly developed by the Video Coding Experts Group (VCEG) of the ITU-T and the Moving Picture Experts Group (MPEG) of ISO/IEC. It uses state-of-the-art coding tools and provides enhanced coding efficiency for a wide range of applications, including video telephony, video conferencing, TV, storage (DVD and/or hard disk based, especially high-definition DVD), streaming video, digital video authoring, digital cinema, and many others [6].

The 3rd Generation Partnership Project (3GPP) standardizing the Universal Mobile Telecommunications System (UMTS) has approved the inclusion of H.264/AVC (Advanced Video Coding) as an optional feature in release 6 of its packet oriented mobile multimedia telephony [7] and streaming service [8] specifications.

In our study, we use the trace statistics of actual H.264 streams from the High Definition (HD) Video Trace Library of [9]. The video streams correspond to videoconference traffic; they have been ex-

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**Fig. 1.** Dynamic frame structure for the 9.045 Mbps channel, frame duration 12 ms.
extracted and analyzed from a camera showing the Sony Digital High Definition Video Camera Demo and they have an I-P-B frames quantization of 28-28-30.

The streams have a mean bit rate of 455 Kbps, a peak rate of 6.63 Mbps, and a standard deviation of 2 Mbps (this type of video traffic is much burstier than the MPEG-4 traffic used in [1]). New video frames (VFs) arrive every 33.3 ms. We have set the maximum transmission delay for video packets to 33.3 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 [2].

In our work in [10] we focused on the accurate fitting of the marginal (stationary) distribution of video frame sizes of single video traces. We have studied three different long sequences of H.264 VBR encoded videos from [9]. After investigating the possibility of modeling the traces with a number of well-known distributions, our results have shown that the best fit among these distributions for modeling a single movie is achieved for all traces examined with the use of the Weibull distribution.

The three traces are, respectively:

1. The demo from the Sony Digital HD Video Camera, which we use in this work.
2. A talk-show program (“KAET’s Horizon”).
3. A documentary (“KAET’s From Mars to China”).

Based on the good fit of the Weibull distribution for modeling a single movie, the behavior of both single and multiplexed HD H.264 video traces from VBR coders can be accurately captured and used for efficient resource allocation in wireless cellular systems. The case of modeling multiplexed video streams is especially significant for our study, since numerous sources are multiplexed in the uplink channel. A Discrete Autoregressive Model of order 1 (DAR(1) model) was used to model the behavior of video users, as DAR(1) provides an easy and practical method to compute the transition matrix in the Markov chain that it creates (the states of which represent the number of packets transmitted by the user). DAR(1) is based only on four physically meaningful parameters, i.e., the mean, peak, variance and the lag-1 autocorrelation coefficient \( \rho \) of the offered traffic. Our results have shown that our model captures with high accuracy the behavior of multiplexed H.264 video users.

Since the channel rate considered in this work is 9.045 Mbps, the trace parameters (mean, peak) of traces 2, 3 above were such that they could not be accommodated by the network. For this reason, we only use in our simulation study the Sony Demo trace, as explained earlier in this Section; an additional reason for this choice is that we are considering a cellular network, therefore we are considering mobile devices with limited screen size.

2.3. Actions of voice, video and data terminals, base station scheduling and transmission protocols

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 by the base station. Upon successfully transmitting a request packet the terminal waits until the end of the 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 base station by transmitting them within the header of the first packet of their current video frame.

It is a common assumption in the literature that the dissatisfaction of a wireless cellular subscriber who experiences forced call termination while moving between picocells is higher than that of a subscriber who attempts to access the network for the first time and experiences call blocking; for this reason we offer full priority to handoff traffic. This means that voice terminals who have been hand-offed to the cell are the first to attempt to transmit their requests in the request minislots at the beginning of the frame request interval; when their contention is finished, they are followed by hand-offed data terminals, then by voice terminals originating from within the cell and finally by data terminals originating from within the cell. The above prioritization by “isolating” each type of traffic and letting it contend only with traffic of the same type is feasible due to the use of the two-cell stack reservation random access algorithm, as it will be explained at the end of this Section.

Video terminals have the highest priority in acquiring the slots they demand (again, with priority given to handoff video terminals). If a full allocation is not possible, the BS makes a partial allocation and keeps a record of all 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. This allocation takes place at the end of the first extra request interval after the arrival of a new VF. Video terminals keep these slots in the following channel frames, if needed, until the next video frame (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 (more details on this “resuffling” policy can be found in [11]).

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 80 channel frames here), and thus video terminals would not find enough slots to transmit in; hence, the particularly strict video QoS requirements would be violated. Consequently, 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. 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).

The BS also “preempts” WAP reservations (both handoff and those originating from within the cell) in order to service voice requests. When WAP reservations are canceled, the BS notifies the affected data terminal and places an appropriate request at the front of the WAP request queue. No data preemption is executed by the BS to favor video users. This design choice was made to avoid very significant increases in data delay (due to the “greediness” of video users in terms of bandwidth and QoS requirements) and to allow voice traffic (which is restricted by the \( p \) policy) the small advantage of solely “exploiting” the preemption mechanism.

In our study, we adopt the two-cell stack reservation random access algorithm [11], due to its operational simplicity, stability and relatively high throughput when compared to PRMA-like algorithms, such as [2]. Another important reason for the choice of this algorithm, as mentioned earlier in this Section, is that it offers a clear indication of when voice contention has ended, and therefore it supports the prioritization mechanism used for voice and data access to the request minislots. The two-cell stack blocked access collision resolution algorithm [11] is adopted for use by the data terminals in order to transmit their data request packets.
2.4. Call admission control

As we discussed in [1], 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. However, almost all of these mechanisms suffer, as noted by their authors, 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. For this reason, we combined in [1] the use of our MAC scheme with the use of a traffic policing scheme which we proposed in that work. The efficiency of the traffic policing mechanism, which helped define malicious users after they had entered the network, did not, however, cancel the need for an efficient CAC scheme at the entrance of the network. Therefore, we proposed in [12] a new CAC mechanism for the transmission of H.263 videoconference traffic over wireless cellular networks. The basic idea of our mechanism is that, based on a model we have built for H.263 videoconference traffic, a wireless provider should precompute the estimated traffic from various traffic scenarios which can take place in the network. This can be done based on the provider’s client database and specifically on the traffic profile declared by each client in his contract with the provider. Therefore, many traffic scenarios which will be encountered in the network will be a priori known to the provider in terms of the actual network resources (bandwidth) which will be needed in order to satisfy the QoS requirements of video users. Certainly, not all traffic scenarios can be precomputed, due to the very large number of all possible traffic loads; however, as explained in [12], with the use of an adequate number of precomputed scenarios and our accurate video model, an online simulation can be quickly conducted when a non-precomputed traffic load occurs in the system, in order to compute the “deviation” between the bandwidth needed currently and the “closest” precomputed traffic scenario.

In our new framework, in this paper, we have implemented a CAC scheme similar to that of [12], with the significant difference that this CAC scheme is based on our HD H.264 traffic modeling (which is different from the H.263 traffic modeling used in [12]), due to the differences in the video encoding schemes and, consequently, in the traffic characteristics; this translates in the use of the Weibull distribution for modeling HD H.264 videoconference traffic, instead of the Pearson V distribution which was used in [12]). An efficient CAC scheme for H.264 videoconference traffic is proposed here for the first time in the literature; the use of a CAC scheme is especially important, as it does not allow into the network traffic which, added to the already existing network traffic in the cell under study, would cause system instability and the violation of the QoS requirements of users already transmitting in that cell.

Additionally to the CAC scheme for video traffic, we implement a CAC scheme for voice and data traffic, which checks whether voice and data users are complying to their declared mean parameters. The check is made over time windows of one minute. The CAC scheme for voice and data traffic can be very simple, due to the lack of significant burstiness of data traffic in the uplink channel and the fact that voice traffic is actually Constant Bit Rate when a user is in “talkspurt” mode, and is not transmitting at all when the user is in “silence” mode.

3. Channel error model

We use a simplified Fritchman Markov model (from [13]) to emulate the process of packet transmission errors. This model is less bursty than the channel error model used in [1], and is chosen in order to make a fairer comparison with DPRMA, which in [2] was evaluated on an error-free wireless channel. The Markov model used (presented in Fig. 2) comprises of six states. State $S_0$ represents the “good state” and all other states represent the “bad states”. When the channel is in state $S_0$, it can either remain in this state or make the transition to state $S_1$ (with probability $p_0$). When the channel is in a bad state, the transition is either to the next higher state or back to state $S_0$, based on the status of the currently received packet. This means that the channel does not remain in any of the “bad states” for more than 1 slot. With this model, it is only possible to generate burst errors of length equal to five slots at most. The transition probabilities ($p_0, p_1, \ldots, p_5$) of the error model are (0.06, 0.100324, 0.164083, 0.149606, 0.526316, 0), respectively. The probability that the channel is in a good state is $p_{\text{good}} = 0.999995$, and the total probability for a transition from a bad state to the good state is $p_{\text{bad-good}} = 0.8924$.

4. System parameters

The channel rate is assumed equal to 9.045 Mbps. The 12 ms of frame duration accommodate 256 slots. These parameters are taken from [2], where the DPRMA scheme (with which we compare the performance of our new framework) was introduced.

The value of the probability $p^*$ is chosen equal to 9%, as in [1]. Many other values of $p^*$ have also been tried out through simulation (both in [1] and in the present work), and it has been found that the chosen value gives very satisfactory results for all the examined cases of video load (generally, all values of $p^*$ between 7% and 10% were found to provide similar results).

5. Results and discussion

We use computer simulations executed on Pentium-IV workstations to study the performance of the new framework, combining the MI-MAC scheme with our call admission control mechanism. Each simulation point is the result of an average of 10 independent runs (Monte-Carlo simulation), each simulating
305,000 frames (the first 5000 of which are used as warm-up period).

5.1. DPRMA

The basic differences of the DPRMA protocol [2] and the proposed new framework are the following.

1. The BS in DPRMA does not use a reshuffling policy as our framework does, and does not grant the earliest available information slots to video users. Instead, it uses a process which tends to spread the allocation of slots randomly throughout the frame.

2. The authors in [2] use a video traffic model based on H.261 videoconferencing traffic (i.e., a model for video traffic from past technology encoders); also, the authors in [2] consider an abstract simplified model for data traffic (not referring to a specific type of data traffic), with which data packets (i.e., not messages) are generated according to a Poisson process.

3. DPRMA uses certain transmission rates for all types of users; 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; one pair of up- and down-threshold levels is implemented for data users, and one pair for voice users.

4. DPRMA uses neither request slots nor our idea of $p^*$, but adopts a PRMA-like approach for voice and data users, allowing them to compete for the available information slots by transmitting their packets according to a probability ($P_{v} = 0.05$ is the voice transmission probability and $P_{d} = 0.007$ is the data transmission probability).

5. In DPRMA, both voice and video users waste one slot when giving up their reservations.

6. DPRMA employs data preemption in favour of both video and voice users (not just for voice users, as our framework does).

7. DPRMA does not employ a CAC mechanism, as our framework does.

5.2. Results and discussion

Table 1 presents the simulation results when integrating all three traffic types: voice, HD H.264 video streams and WAP sessions, for both DPRMA and our framework. For various video loads and for different, fixed arrival rates of WAP sessions ($\lambda_{WAP}$ sessions per second), we present the voice capacity of each scheme, as well as the corresponding channel throughput. In all the results presented in the Table, we consider that 15% of the total traffic originates from handoff calls. We examine the cases of 1–4 video users (no more than 4 video users can be accommodated while receiving the strict QoS they demand), $\lambda_{WAP}$ being equal to 20, 40, and 60 sessions/second, (i.e., data traffic ranging from about 300 Kbps to 1 Mbps), and we observe from our results that in our framework, for a given number of video terminals, as $\lambda_{WAP}$ increases, the channel throughput increases as well. This shows 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 less 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 $\lambda_{WAP}$ increases, as WAP request delays quickly exceed the set upper bound and the system becomes unable to accommodate these traffic loads for a larger number of voice users.

The data preemption policy is not the only reason that our framework achieves better throughput results than DPRMA (their difference in throughput ranges from 4.37% to 11.27%, with an average of 6.62%). The other reasons are:

1. The use of our reshuffling policy ensures a timely slot allocation to video users.
2. The use of an efficient CAC mechanism ensures that new video users will be admitted into the system only if the network is capable of accommodating the new load, after the addition of the new video user.
3. The use of a number of transmission rates in DPRMA does not ensure that the terminal will be allocated the maximum possible number of slots in each frame, based on its needs.
4. By using the two-cell stack reservation random access algorithm, our framework 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 ($p^*$) is set in our scheme after they have sent their request to the BS, therefore they will wait in the queue at the BS for a possible slot allocation without having to further con-

### Table 1

<table>
<thead>
<tr>
<th>Number of video users</th>
<th>$\lambda_{WAP}$ (sessions/second)</th>
<th>Voice capacity (Maximum number of voice terminals)</th>
<th>Channel throughput (%)</th>
<th>Voice packet dropping for handoff users (%)</th>
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<tbody>
<tr>
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<td>MI-MAC+CAC framework</td>
<td>DPRMA</td>
<td>MI-MAC+CAC framework</td>
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<td>32</td>
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serve that the WAP data message delay increases quickly as to correspond to approximately 50% channel utilization. We observ
handoff calls; the combined voice and video load has been chosen present in the system, and 5% of the total traffic originates from
session arrival rate, when 195 voice users and two video users are included in this calculation).

dramatically over 62%, reaching up to 71%) for low and medium video throughput decreases, due to the very bursty nature of video traff
voice capacity with the use of our scheme ranges from 10.79% to DPRMA in all the examined cases of traffic loads; the increase in
satisfactory channel throughput results (steadily and often signifi-
c communal voice and video traffic over wireless cellular networks. Our framework's performance is evaluated by integrating voice, bursty H.264 video and WAP packet traffic over a noisy wireless channel of high capacity and under significant handoff loads. With the combination of the efficiency of the MAC scheme and the accuracy of the CAC mechanism (which does not overestimate users' needs as many CAC schemes in the literature do for bursty traffic), our framework is shown to outperform a well-known protocol and to achieve high aggregate channel throughput and relatively low data transmission delays in all cases of traffic loads examined, while preserving the Quality of Service (QoS) requirements of each traffic type. This is one of the first works in the literature, to the best of our knowledge, to study the integration of latest video technology encoding streams with other types of multimedia traffic over wireless cellular networks.

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References
[8] 3GPP, TSG Services and System Aspects, Transparent End-to-End Packet- Switched Streaming Service (PSS) (Ref. 6), version 6.8.0.