Integrating Multimedia Traffic with Strict QoS in Wireless Cellular Networks

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ABSTRACT

We have recently introduced in the literature MI-MAC (Multimedia Integration Multiple Access Control), a new access control protocol for next generation wireless cellular networks which showed superior performance in comparison to other (TDMA and WCDMA-based) protocols when integrating various types of multimedia traffic. In this work, which to the best of our knowledge is one of the first in the literature to study the integration of H.264 streams with other types of multimedia traffic, we continue our work on MI-MAC. We evaluate the protocol's ability 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. We also discuss the differences between accommodating video traffic from previous technology encoders and H.264.

I. 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 paper we continue the performance evaluation of the scheme by discussing its performance when integrating streams from the latest technology video encoding (H.264) with voice and WAP (Wireless Application Protocol) traffic, under a different channel error model than the one used in [1]. Most importantly, 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 portion of the traffic in our simulations is considered to be handoff and therefore has very strict Quality of Service (QoS) requirements (hand-offed users expect to continue to receive the same QoS as in the previous cell of the network).

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, and we compare MI-MAC with DPRMA [3], a well-known MAC protocol for wireless networks.

II. MULTIPLE TRAFFIC TYPE INTEGRATION

A. 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, but the nature of our results is independent of the packet size and implementable in GSM-type networks). As shown in Figure 1 (which presents the channel frame structure), each frame consists of two types of intervals. These are the *voice and data request* 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 II.B), 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 DPRMA). 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 3 request slots are needed for the end of the voice and data terminals' contention, the Base Station signals all user

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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)* [2] (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).

B. Traffic Models

B1. Voice Traffic Model

Our primary voice traffic model assumptions are the following:

1. The speech codec rate is 32 Kbps [2]. The output of the voice activity detector (VAD, [1]) is modeled by a two-state discrete time Markov chain. The mean talkspurt duration is 1.0 seconds and the mean silence duration is 1.35 seconds.
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 second.
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 [3].

B2. WAP Traffic Model

We adopted the WAP traffic model presented in [4] (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 [4] 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 λWAP sessions per second, with an upper limit on the average WAP request transmission delay equal to 2 seconds. 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 seconds [10] (SMS messages have a payload of 140 bytes, i.e., similar to a WAP request), the upper bound set in this work for WAP request transmission is quite strict.

B3. 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 [5].

In our study, we use the trace statistics of actual H.264 streams from the High Definition Video Trace Library of [9]. The video streams correspond to videoconference traffic; they have been 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 msecs. We have set the maximum transmission delay for video packets to 33.3 msecs, 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 [3].

C. Actions of Voice, Video and Data Terminals, Base Station Scheduling, and Voice-Data 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 "reshuffling" policy can be found in [1]).

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 [6], due to its operational simplicity, stability and relatively high throughput when compared to the PRMA [7] and PRMA-like algorithms, such as [3]. 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 [6] is adopted for use by the data terminals in order to transmit their data request packets.

III. CHANNEL ERROR MODEL

We use a simplified Fritchman Markov model (from [8]) 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 the present work in order to make a fair comparison with DPRMA, which in [3] was evaluated on an error-free wireless channel. The Markov model used (presented in Figure 2) comprises of 6 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 5 slots at most. The transition probabilities (p_0, p_1,…p_5) of the error model are (0.0000446, 0.100324, 0.164083, 0.149606, 0.526316, 0), respectively. The probability that the channel is in a good state is p_{good}=0.99995, and the total probability for a transition from a bad state to the good state is $p_{bad\text{-}good}$ =0.8924.

IV. 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 [3], where the DPRMA scheme (with which we compare the performance of MI-MAC) 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).

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 basic differences of the DPRMA protocol [3] and MI-MAC are the following.

1. The BS in DPRMA does not use a reshuffling policy like MI-MAC, 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 [3] use a video traffic model based on H.261 videoconference traffic (i.e., a model for video traffic from past technology encoders); also, the authors in [3] 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 7 threshold levels, and, respectively, 7 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 $\boldsymbol{p}^*$, 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_{tv}=0.05 and P_{td}=0.007, respectively).

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 MI-MAC does).

B. Results and Discussion

Table 1 presents the simulation results when integrating all three traffic types: voice, H.264 video streams and WAP sessions, for both MI-MAC and DPRMA. For various video loads and for different, fixed arrival rates of WAP sessions (λ_{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 λ_{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 MI-MAC, for a given number of video terminals, as λ_{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 λ_{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 MI-MAC achieves better throughput results than DPRMA (their difference in throughput ranges from 4.05% to 11.27%, with an average of 6.51%). The other reasons are:

1. The use of our reshuffling policy ensures a timely slot allocation to video users.

2. 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.

3. By using the two-cell stack reservation 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 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 contend* (as in DPRMA). Additionally, the use of ER slots helps MI-MAC "exploit" certain available information slots that DPRMA leaves unused.

4. The fact that, in DPRMA, a slot is wasted each time a user gives up its reservation.

Also, it should be emphasized that MI-MAC achieves much lower handoff video and handoff voice packet dropping results than DPRMA, due to the provision, in our scheme, of full priority to handoff calls both in terms of transmitting their requests and in terms of slot allocation. This is clearly shown in the last column of Table 1, where even for the voice capacities that DPRMA is able to accommodate (which are significantly smaller than MI-MAC), in almost all the cases the voice packet dropping for handoff users is higher than that of MI-MAC.

The results in Table 1 show that MI-MAC achieves quite satisfactory channel throughput results (steadily and often significantly over 60%, reaching up to 71%) for low and medium video traffic loads, despite the high burstiness of video traffic. When the number of video users becomes higher, the throughput decreases, due to the very bursty nature of video traffic and its very stringent QoS requirements. Finally, it should be noted that in terms of voice capacity, MI-MAC clearly outperforms DPRMA in all the examined cases of traffic loads; the increase in voice capacity with the use of MI-MAC ranges from 10.79% to 675%, with an average of 25.15% (the 675% increase case was not included in this calculation).

In comparison to the results in [1], where we considered the integration of H.263 video traffic with other types of traffic over cellular networks, the results in Table 1 are similar in nature and the maximum throughput achieved is close to that achieved in [1]. However, on average the throughput achieved in the present work is lower (by about 2%) than the throughput achieved in [1]. The reason is the much higher burstiness of H.264 video traffic, considered here, in comparison to H.263 traffic. The same conclusion is derived by all our results when integrating H.264 video traffic with other types of multimedia traffic.

Figure 3 presents the average WAP request delay versus the WAP session arrival rate, when 195 voice users and 2 video users are present in the system, and 5% of the total traffic originates from handoff calls; the combined voice and video load has been chosen to correspond to approximately 50% channel utilization. We observe that the WAP data message delay increases quickly as λ_{WAP} increases, for both MI-MAC and DPRMA; this quick increase is once more a result of the fact that new WAP 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 the Figure, for an arrival rate λ_{WAP} higher than 20 sessions/second the system is unable to sustain the channel load in MI-MAC, as the average WAP request delay exceeds the assumed upper delay bound of 2 seconds. The same upper bound is exceeded by DPRMA for an arrival rate higher than 9 sessions/second.

VI. CONCLUSIONS

In this work, we have further investigated the performance of a reservation medium access control protocol for wireless multimedia communications which we have recently introduced in the literature. Our protocol, MI-MAC, is evaluated when integrating voice, H.264 video and WAP packet traffic over a noisy wireless channel of high capacity. With the use of a dynamic TDMA frame structure and an efficient scheduling policy, our scheme 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 load examined and for various handoff traffic loads, while preserving the 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

[1] P. Koutsakis, S. Psychis and M. Paterakis, ″Integrated Wireless Access for Videoconference from MPEG-4 and H.263 Video Coders with Voice, E-mail and Web Traffic″, *IEEE Transactions on Vehicular Technology,* Vol. 54, No.5, pp.1863-1874, 2005.

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

[3] D. A. Dyson and Z. J. Haas, "A Dynamic Packet Reservation Multiple Access Scheme for Wireless ATM", *Mobile Networks and Applications (MONET) Journal,* Vol. 4, No. 2, 1999, pp. 87-99.

[4] P. Stuckmann and C. Hoymann, "Dimensioning GPRS Networks for Coexisting Applications based on WAP and Conventional Internet Protocols", *in Proceedings of the European Wireless Conference 2002,* Florence, Italy.

[5] G. J. Sullivan, P. Topiwala and A. Luthra, "The H.264/AVC Advanced Video Coding Standard: Overview and Introduction to the Fidelity Range Extensions", *in Proc. of the SPIE Conference on Applications of Digital Image Processing XXVII, Special Session on Advances in the New Emerging Standard:H.264/AVC*, Denver, USA, August 2004.

[6] M. Paterakis, L. Georgiadis and P. Papantoni-Kazakos, ″A Full-Sensing Window RA Algorithm for Networks with Strict Delay Constraints″, *Algorithmica*, Vol. 4, 1989, pp. 313-328.

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

[8] C. Y. Hsu, A. Ortega and M. Khansari, "Rate Control for Robust Video Transmission over Burst-Error Wireless Channels", *IEEE Journal on Selected Areas in Communications*, Vol. 17, No.5, 1999.

[9] [Online]http://trace.eas.asu.edu/hd/index.html

[10] ETSI. Digital Cellular Telecommunications System (Phase 2+); Technical Realization of the Short Message Service (SMS); Point-to-Point (PP). (GSM 03.40)

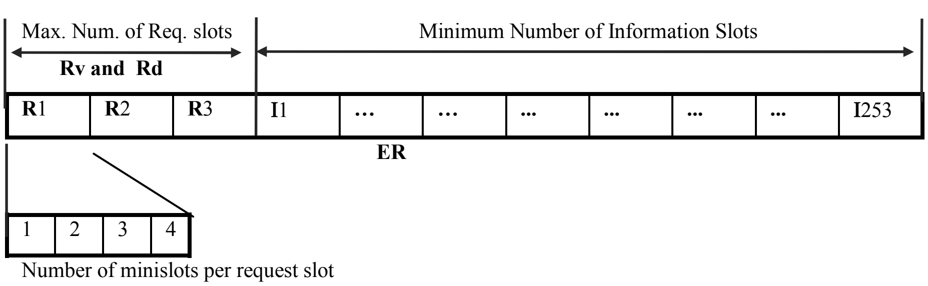

Figure1. Dynamic Frame structure for the 9.045 Mbps channel, frame duration 12 ms.

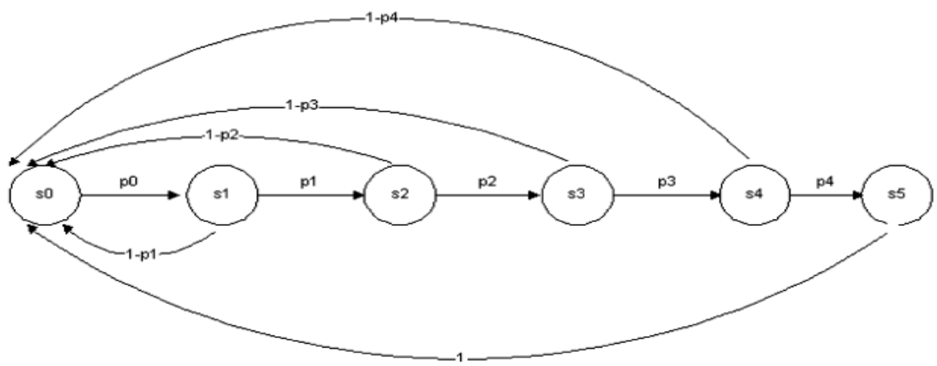

Figure 2. Channel Error Model.

Number of video users	λ_{WAP} (sessions/ second)	Voice Capacity (Maximum Number of Voice Terminals)		Channel Throughput (%)		Voice Packet Dropping for Handoff Users (%)	
		MI-MAC	DPRMA	MI-MAC	DPRMA	MI-MAC	DPRMA
1	20	349	315	67.18	62.26	0.41	0.64
	40	334	294	67.65	60.98	0.52	0.71
	60	291	248	68.02	58.42	0.57	0.75
2	20	297	267	68.98	63.95	0.48	0.60
	40	284	241	69.70	63.01	0.59	0.65
	60	252	185	70.87	59.60	0.64	0.73
3	20	194	173	62.02	57.97	0.45	0.47
	40	180	151	63.25	57.17	0.53	0.56
	60	161	114	65.08	56.54	0.71	0.68
4	20	85	62	53.34	49.65	0.51	0.70
	40	64	40	53.77	48.73	0.62	0.79
	60	31	4	54.01	48.29	0.72	0.64

Table 1. Voice Capacity and Channel Throughput for MI-MAC and DPRMA under various H.264 video and WAP data loads (handoff traffic=15%).

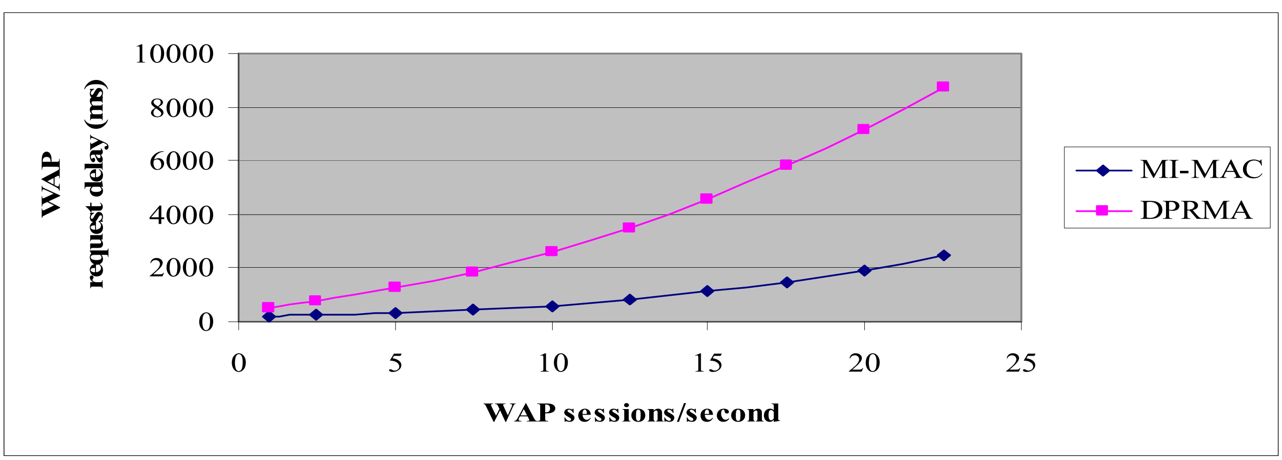

Figure 3. Average WAP request delay versus WAP session arrival rate, for Nvoice=195, H.264 video users=2, handoff traffic=5%.