Near-Optimal Voice-Data Integration over Third Generation Medium and High Capacity Wireless TDMA Channels

P. Koutsakis and M. Paterakis
Dept. of Electronics and Computer Engineering
Telecommunications and Information & Computer Networks Laboratories
Technical University of Crete, 73100 Chania Greece
e-mail: {polk, pateraki}@telecom.tuc.gr

Abstract- A new multiple access control (MAC) protocol for mobile wireless communications is presented and investigated. We explore, via an extensive simulation study, the performance of the protocol when integrating voice and data traffic over two wireless channels, one of medium capacity (referring mostly to outdoor microcellular environments) and one of high capacity (referring to an indoor microcellular environment). We evaluate the voice packet dropping probability and access delay, as well as the data packet access and data message transmission delays for various voice and data load conditions. Our protocol achieves near-optimal voice sources multiplexing results along with most satisfactory voice and data performance and quality of service (QoS) requirements servicing.

I. INTRODUCTION

Future generation wireless personal communication networks (PCN) are expected to provide multimedia capable wireless extensions of fixed ATM/B-ISDN, as data and video traffic will soon gain in importance due to the continuous proliferation of small, portable and inexpensive computing devices.

In this work, we design and evaluate multiple access schemes that multiplex voice traffic at the vocal activity (talkspurt) level to efficiently integrate voice and data traffic in outdoor and indoor microcellular environments. We focus on the uplink (mobiles to base station) channel, where a MAC scheme is required in order to resolve the source terminals contention for channel access.

A well designed multiple access protocol will reduce system costs by maximizing system capacity, integrating different classes of traffic (as opposed to current wireless networks which are mostly optimized for voice communications only), and satisfying the diverse and usually contradictory quality of service (QoS) requirements of each traffic class.

II. SYSTEM MODEL

A. Channel Frame Structure

The uplink channel time is divided into time frames of equal length. The frame duration is selected such that a voice terminal in talkspurt generates exactly one packet per frame. Each frame consists of three *types* of intervals, the *voice request* intervals, the *data request* intervals and the *information* intervals. Within an information interval, each slot accommodates exactly one, fixed length, packet that contains voice or data information and a header. All request intervals (voice or data) are subdivided into an equal number of minislots and each minislot accommodates exactly one, fixed length, request packet. For both voice and data traffic, the request must include a source identifier. For data traffic, the request must also include message length in packets and perhaps QoS parameters such as priority. The data request intervals are distributed uniformly within the frame. This way, since data message arrivals occur uniformly within the frame duration (they are assumed to be Poisson), we allow the data terminals to transmit their requests soon after their messages have been generated. Since we assume that all of the voice source transitions occur at the frame boundaries, we place all voice request intervals at the beginning of the frame, in order to minimize the voice packet access delay.

The voice and data terminals do not exhaust their attempts for a reservation within the request intervals. Any other free, at the time, information slot of the frame can be temporarily used as an extra request slot (ER slots), with priority given to the voice terminals. This approach has been introduced and implemented in [2,7].

We introduce the idea that certain 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. An example of the channel frame model can be found in Fig. 1, where we present the frame structure for the high capacity channel.

The concept of reserving a minimum bandwidth for both voice and data terminals to make reservations helps to keep the access delay within relatively low limits and gives clearly better performance than the PRMA [4] and quite a few PRMA-like algorithms (e.g., [6]), where the absence of request slots leads to a continuously decreasing probability of finding available information slots as traffic increases, and hence to greater access delays. A request bandwidth of 2-3% is usually required for optimum system performance.

B. Actions of Voice and Data Terminals, and Base Station Scheduling

Voice and data terminals with packets, and no reservation, contend for channel resources using the *two-cell stack* blocked access collision resolution algorithm [9], in order to transmit their request packets only during the voice or data, respectively, request intervals. The base station broadcasts a short binary feedback packet at the end of each minislot indicating only the presence or absence of a collision within the minislot (collision (C) versus non-collision (NC)). It is assumed that the feedback information is immediately available to the terminals (i.e., before the next minislot). 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). A terminal with a reservation transmits freely within its reserved slot. Generally, a terminal that fails to transmit a request tries again in successive frames until it succeeds. However, since voice packets that age beyond the voice delay limit are dropped, a voice terminal may stop transmitting requests without ever succeeding.

The base station (BS) allocates channel resources at the end of the corresponding request interval, if available. If the resources are unavailable, the request remains queued.

We assume that the BS always allocates the earliest available information slot within the frame, and that it services every outstanding voice request before servicing any data requests. Within each priority class, the queuing discipline is assumed to be FCFS.

Finally, *we apply a low-voice-load mechanism to our scheme*. As data terminals try to transmit messages that vary in length and are, on average, much longer than one packet, it would be both unfair to them and diminishing to our system's performance to not allocate to them more than one slot per frame if resources are available. On the other hand, by allocating more than one slot per frame to data terminals, voice terminals would find a lower number of information slots available for either reservations or requests (ER slots), and our objective is to enforce voice priority. Therefore, we implement the following mechanism.

We define the *frame voice occupancy* as the ratio of {(voice reservations + voice requests) / number of information slots in the frame}. This ratio is calculated by the BS immediately after the end of the voice request slots of each frame. If the ratio is lower than a set limit, we allow data terminals with requests to acquire more than one slot in the current frame. Still, only the first allocated slot is guaranteed to data terminals with reservations in subsequent frames. The selection of the *low frame voice occupancy limit* and of the *maximum number of slots that can be allocated to data terminals within a frame* (the two parameters of our low-load mechanism) must be done carefully, so that even in the case of low voice load enough information slots will still remain available in the next frame for voice terminals who enter talkspurt to use as ER slots.

These selections should be based on the combination of the following two factors:

a) the average data message length, and

b) the channel capacity.

We introduce our numerical choices for the two low-load parameters in Section III.

C. Voice and Data Traffic Models

Voice terminals are equipped with a voice activity detector [3,4]. 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. All of the voice source transitions (e.g., talk to silence) occur at the frame boundaries, and the voice delay limit is equal to the duration of two channel frames.

The channel is error-free and without capture, and reserved slots are deallocated immediately by the BS.

Data messages are generated by a large unknown number of data terminals (theoretically infinite). The aggregate message arrivals are Poisson distributed with mean λ messages per frame. The messages vary in length according to a geometric distribution with parameter q and mean $B=1/q$.

III. System Parameters

Our simulations were conducted with the channel and low-load-mechanism parameters contained in Table I.

IV. Results

A. Voice-Data Integration in a Medium Capacity Channel (VDI-MCC)

We compare our scheme with two previously proposed efficient schemes for voice-data integration, IPRMA [8] and RRA [1,5]. The comparison proves our scheme's significantly better performance. Figures 2,3 and Table II show the comparison with RRA's respective results. We observe, from the figures, that the mean data message delay in VDI-MCC is not only much smaller than that in RRA, but is also smaller than the mean data access delay in RRA (i.e., in VDI-MCC data messages are transmitted faster than the transmission of just one data packet in RRA).

B. Voice-Data Integration in a High Capacity Channel (VDI-HCC)

In this case, we achieve a near-optimal multiplexing gain (2.17), whereas the optimal (however, impossible to achieve in practice) multiplexing gain is equal to 2.26. The constant surpassing of 96% channel throughput for all the data message arrival rates and the achievement of a throughput as high as 98.5% for very high data message arrival rates indicates the efficiency of the proposed multiplexing mechanism (see Table III). In addition, Figure 4 shows the DaD and DmD curves for VDI-HCC, for a constant number of voice terminals equal to 515 and for different data message arrival rates. We see again that VDI-HCC achieves a very high aggregate channel throughput of 96.8% while the average data message delay remains below the delay limit of 200 ms, but most importantly it is shown that the mean data message delay is slightly higher than the mean data packet access delay. This proves that the low-voice-load mechanism guarantees the quickliest possible transmission to data messages after they have succeeded in transmitting their first packet, without affecting the overall throughput, voice performance and data packet access delay.

V. FUTURE WORK

The next step of our research will be the introduction of variable bit rate compressed video sources into the high capacity channel system and the efficient multiplexing of all three diverse types of traffic.

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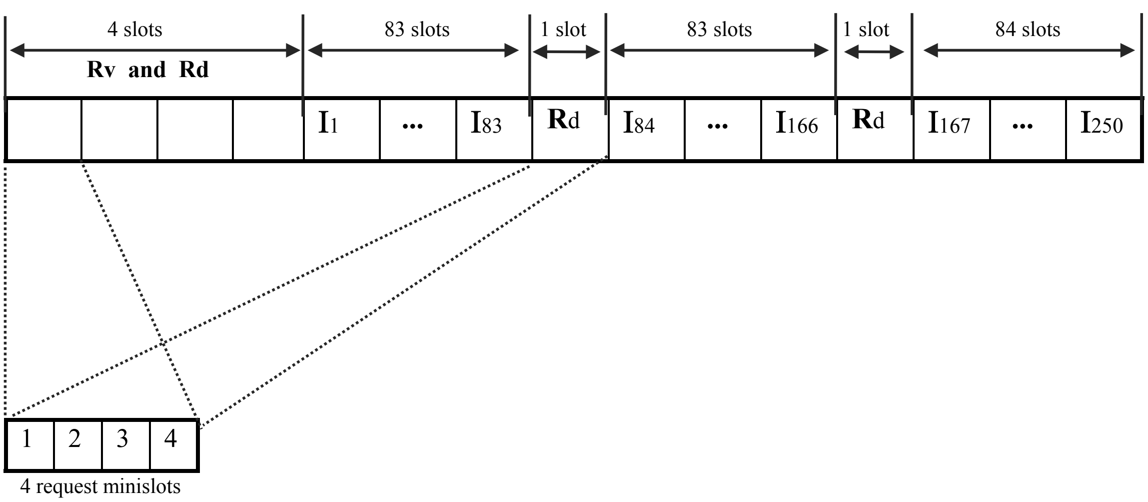

Fig. 1. Frame structure for the 9.045 Mbps channel.

TABLE I

EXPERIMENTAL SYSTEM PARAMETERS

Design Parameters	Medium Capacity Channel	High Capacity Channel
Channel Rate (Mbps)	1.8	9.045
Speech Codec Rate (Kbps)	32	32
Frame Duration (ms)	12	12
Slots per Frame	50	256
Slot duration (μs)	240	46.875
Request slots per frame	2	6
Minislots per request slot	6	6
Packet size (bytes)	53 (5 header)	53 (5 header)
Voice delay limit (ms)	24	24
Mean talkspurt duration (s)	1.41	1.41
Mean silence duration (s)	1.78	1.78
B (average data message length)	8 packets	8 packets
Low frame voice occupancy limit	95% (46 slots)	95% (238 slots)
Maximum number of slots allocated to data terminals	8	8
Max. voice dropping probability	0.01	0.01

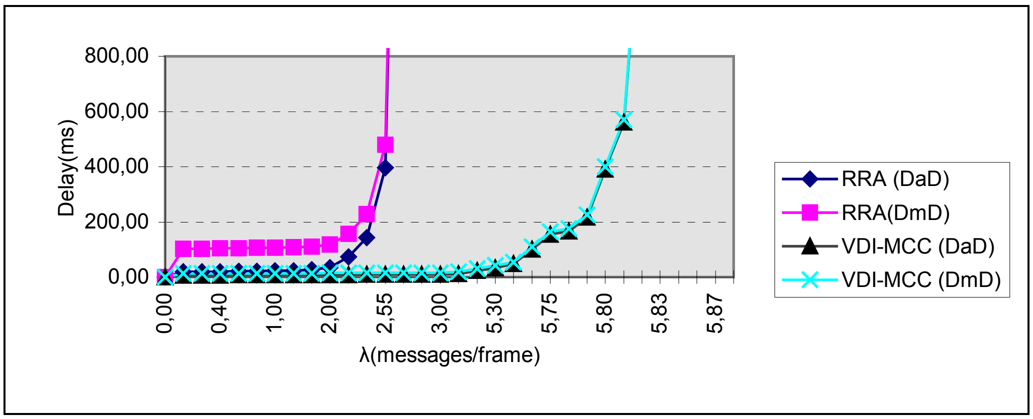

Fig. 2. Steady state mean data delays in the absence of voice traffic. (DaD = Data access delay, DmD = Data Message Delay).

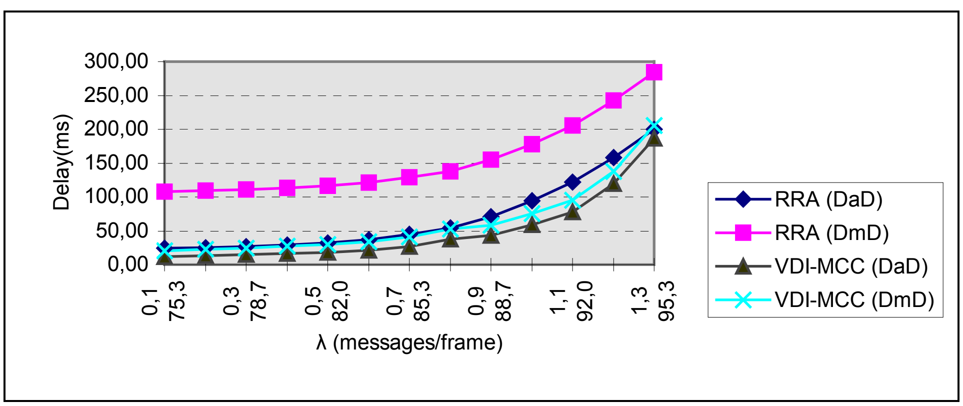

Fig. 3. Steady state mean data delays: N=80.

TABLE II

QoS COMPARISON OF VDI-MCC AND RRA.

λ (messages/frame)	Max. Voice Capacity for Pdrop<1% and DmD<200 ms							
	VDI-MCC				RRA			
	Cap.	Chan. Throu.	Pdrop (%)	DmD (ms)	Cap.	Chan. Throu.	Pdrop (%)	DmD (ms)
0.1	100	0.909	0.973	180.6	95	0.886	0.623	197.4
0.4	94	0.910	0.417	194.7	89	0.885	0.187	189.4
0.7	89	0.917	0.176	187.3	85	0.899	0.084	198.7
1.0	85	0.928	0.077	192.4	81	0.912	0.042	199.5
1.3	80	0.933	0.031	195.6	76	0.916	0.032	188.3

TABLE III

PERFORMANCE OF VDI-HCC WHEN FULFILLING BOTH THE QoS REQUIREMENTS.

λ (messages/frame)	Max. Voice Capacity for Pdrop<1% and DmD<200 ms		
	VDI-HCC		
	Capacity	Channel Throughput	DmD(ms)
0.4	538	0.960	198.6
1.0	525	0.960	187.8
2.0	511	0.966	182.3
3.0	500	0.977	188.3
4.0	486	0.982	199.6
6.0	448	0.981	189.9
8.0	409	0.979	198.2
10.0	372	0.977	187.2

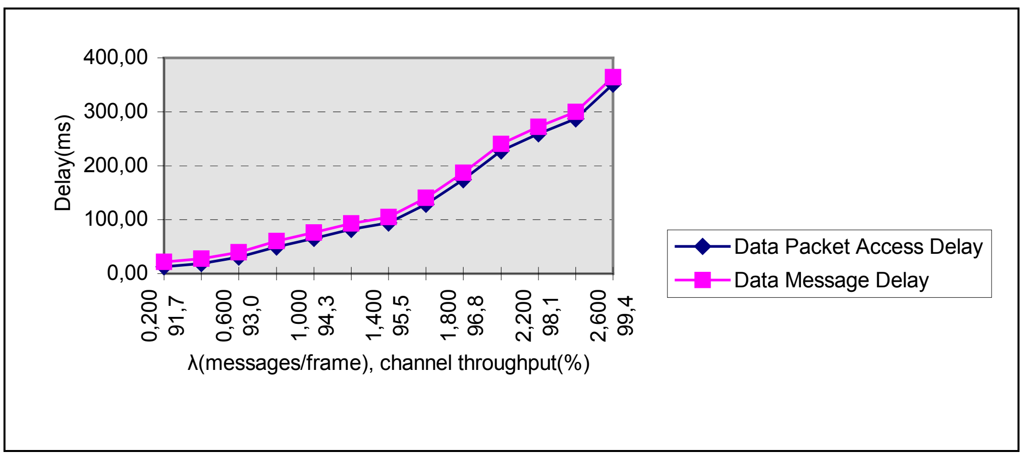

Figure 4. Steady state mean data delays with N=515 active voice terminals.