On Multiple Traffic Type Integration over 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: koutsaki@grecian.net, pateraki@telecom.tuc.gr

Abstract

A new medium 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, video and data packet traffic over a wireless channel of high capacity. Depending on the number of video users admitted into the system, our protocol varies: a) the request bandwidth dedicated to resolving the voice users contention, and b) the probability with which the base station grants information slots to voice users, in order to preserve full priority for video traffic. We evaluate the voice and video packet dropping probabilities for various voice and video load conditions, and the average data message delays. As proven by the comparison with a recently introduced efficient MAC scheme (DPRMA), when integrating voice and video traffic our scheme obtains higher voice capacity and aggregate channel throughput. When integrating all three traffic types, our scheme achieves high aggregate channel throughput in all cases of traffic load.

1. Introduction

High-speed packet-switched network architectures will soon have the ability to support a wide variety of multimedia services, the traffic streams of which will have widely varying traffic characteristics (bit-rate, performance requirements). The main goal of wireless communication is to allow the user access to the capabilities of the global packet-switched network at any time without regard to location or mobility. Current and future wireless networks are and will be based on the cellular concept. In such networks, a well designed multiple access control (MAC) protocol will reduce system costs by maximizing system capacity, integrating different classes of traffic and satisfying the diverse and usually contradictory quality of service (QoS) requirements of each traffic class.

In this work, we design and evaluate a multiple access scheme which multiplexes voice traffic at the vocal activity (talkspurt) level to efficiently integrate voice (Constant Bit Rate, CBR On/Off Traffic), video (Variable Bit Rate, VBR) and bursty data traffic in high capacity picocellular environments.

Within the picocell, spatially dispersed source terminals share a radio channel that connects them to a fixed base station. The base station allocates channel resources, delivers feedback information and serves as an interface to the mobile switching center (MSC). The MSC provides access to the fixed network infrastructure. 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.

2. Voice-Video Integration

2.1 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. As shown in Figure 1 (which presents an example of the channel frame structure), each frame consists of two *types* of intervals. These are the *voice request* intervals and the *information* intervals.

Within an information interval, each slot accommodates exactly one, fixed length, packet that contains voice or video information and a header. Voice request intervals are subdivided into mini-slots and each mini-slot accommodates exactly one, fixed length, request packet. The request must include a source identifier. Since we assume that all of the voice transitions occur at the frame boundaries[1], we place all voice request intervals

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

at the beginning of the frame, in order to minimize the voice packet access delay.

Voice terminals do not exhaust their attempts for a reservation within the request intervals. *Any other free, at the time, information slot of the frame can be temporarily used as an extra request slot (ER slot)* for voice terminals [5]. The concept of reserving a minimum bandwidth for voice terminals to make reservations helps to keep the voice access delay within relatively low limits and gives clearly better performance than the PRMA [1] and quite a few PRMA-like algorithms, such as DPRMA [4], where the absence of request slots leads to a continuously decreasing probability of finding available information slots as traffic increases, and hence to greater access delays.

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

2.2 Voice and Video Traffic Models

Our primary voice traffic model assumptions are the following:

1. The speech codec rate is 32 Kbps, and voice terminals are equipped with a voice activity detector (VAD) [1]. Voice sources follow an alternating pattern of talkspurts and silence periods (on and off), and the output of the voice activity detector is modeled by a two-state discrete time Markov chain. The mean talkspurt duration is 1.0 secs and the mean silence duration is 1.35 secs.
2. All of the voice source transitions (e.g., talk to silence) occur at the frame boundaries. This assumption is reasonably accurate, taking into consideration that the duration of a frame is equal to 12 ms here, while the average duration of the talkspurt and silence periods exceeds 1 sec.
3. The number of active voice terminals, N, in the system is assumed to be constant over the period of interest. This is because the changes in the number of calls are usually on the order of tens of seconds, while the frame duration is on the order of tens of milliseconds [2].
4. The voice delay limit is equal to 40 ms.
5. The channel is error-free and without capture.
6. Reserved slots are deallocated immediately. This implies that a voice terminal holding a reservation signals the BS upon the completion of its talkspurt.

We adopt the same video traffic model as in DPRMA [4]. This model is based upon work done by Heyman, et al [10]. In this study of actual videoconferencing traffic, video frames (VFs) were found to be generated periodically and to contain a varying number of cells in each frame. The distribution of the number of cells per VF was found to be described by a gamma (or equivalently negative binomial) distribution. A Markov chain model can be constructed that demonstrates the transition from one state to the next. A "state" represents the number of video packets (cells) that a video frame contains. The transition matrix is computed as:

$$\boldsymbol{P} = \rho \boldsymbol{I} + (1-\rho) \boldsymbol{Q} \qquad (1)$$

where $\mathbf{I}$ is the identity matrix, ρ is the autocorrelation coefficient (0.98459 from [11]), and each row of the $\mathbf{Q}$ matrix is composed of the probabilities ($f0, \ldots, fK, \boldsymbol{F}K$). The quantity fK has the negative binomial distribution and represents the probability that a video frame contains k cells. The value of K in equation 1 represents the peak cell rate and $\boldsymbol{F}K = \Sigma k>K\, fK$.

The statistics for video conferencing traffic that were obtained in [10], were the result of coding a video sequence with a modified version of the H.261 standard. The results showed a peak cell generation rate of 220 cells/VF (2.112 Mbps), an average generation rate of 104.8 cells/VF (1.006 Mbps), and a standard deviation of 29.7 cells/VF (0.285 Mbps). The cell size was taken equal to 48 bytes, which is equivalent to the ATM cell size. New VFs are assumed to arrive every 40 msecs (i.e., 25 VFs per second).

2.3. Actions of Voice and Video Terminals, Base Station Scheduling and Voice Transmission Protocol

Voice terminals with packets, and no reservation, contend for channel resources using a random access protocol to transmit their request packets only during the voice request intervals. The base station broadcasts a short binary feedback packet at the end of each mini-slot indicating only the presence or absence of a collision within the mini-slot (collision (C) versus non-collision (NC)). Upon successfully transmitting a request packet the terminal waits <u>until the end of the corresponding request interval</u> to learn of its reservation slot (or slots). If unsuccessful within the request intervals of the current frame, the terminal attempts again in the request intervals of the next frame. A terminal with a reservation transmits freely within its reserved slot.

Video terminals, as already mentioned, do not have any request slots dedicated to them. This happens for two reasons:

1. Video sources "live" permanently in the system, they do not follow an ON-OFF state model like voice sources.
2. Video traffic follows a multi-state Markov model, in which however state transitions do not occur very often.

Thus, there is no need for granting request bandwidth to the video terminals, as it would be wasted in most cases. *Video terminals convey their requirements to the base station by transmitting them within the header of the first packet of their current video stream.*

To allocate channel resources, the BS maintains a dynamic table of the active terminals within the picocell. Upon successful receipt of a voice request packet, the BS provides an acknowledgment and queues the request. The BS allocates channel resources at the end of the corresponding request interval, and follows a different allocation policy for video terminals than that for voice terminals.

Video terminals have absolute priority in acquiring the slots they demand. If a full allocation is possible, the BS then proceeds to allocate any still available information slots to the requesting voice terminals. Otherwise, if a full allocation is not possible, the BS grants to the video users as many of the slots they requested as possible (i.e., the BS makes a partial allocation). The BS keeps a record of any partial allocations so that the remaining requests can be accommodated whenever the necessary channel resources become available. In either allocation type case, the BS allocates the earliest available information slots to the video terminals, which, if needed, keep these slots in the following channel frames, until the next video frame (VF) arrives.

Voice terminals which have successfully transmitted their request packets do not acquire all the available (after the servicing of video terminals) information slots in the frame. If this happened, voice terminals would keep their dedicated slots for the whole duration of their talkspurt (on average, more than 8 channel frames here), and thus video terminals, would not find enough slots to transmit in, and the particularly strict video QoS requirements would be violated. *The BS allocates a slot to each requesting voice terminal with a probability* ***p****. The requests of voice terminals which ″ fail ″ to acquire a slot, based on the above BS slot allocation policy, remain queued. The same holds for the case where 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).

Finally, in order to preserve the strict video QoS, we enforce a scheduling policy for the video terminals which prevents unnecessary dropping of video packets in channel frames within which the arrival of a new VF of a video user takes place (the details of this policy can be found in [14,15]).

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 [1], Two-Cell Stack [8], Controlled Aloha [7], Three-Cell Stack [3]). 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) [1] and PRMA-like algorithms, such as [4,6].

3. System Parameters

Each computer simulation point is the result of an average of 10 independent runs, each simulating 305,000 frames (the first 5,000 of which are used as warmup period).

The channel rate is 9.045 Mbps (from [4]). The 12 ms of frame duration accommodate 256 slots. The number of voice request slots is not fixed in the scheme. It depends on the number of video sources admitted into the system[2], and it varies accordingly between 1 and 5 slots (see Table 1). Even for the case where 5 request slots are needed, this corresponds to a 1.95% request bandwidth only. We should note that:

1. In our design, we chose the number of minislots per request interval (4), to allow for guard time and synchronization overheads, for the transmission of a generic request packet, and for the propagation delay within the picocell.
2. Because of assumption 2 of our voice traffic model, all voice request intervals are located at the beginning of each frame.
3. The maximum transmission delay for video packets is set to 40 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 maximum transmission delay for voice packets is also set to 40 msecs.
4. The allowed voice packet dropping probability is set to 0.01, whereas the allowed video packet dropping probability is set to 0.0001.

4. DPRMA

[2] The channel bandwidth consumed by each video source is large, and thus, when we examine cases with a small number of video sources, the system can accommodate a significantly larger number of voice sources. In this case, more voice request slots are needed in order to allow voice sources to enter the system without significant dropping of voice packets.

In order to justify the better performance of our scheme, we will explain the four differences of DPRMA [4] from our scheme.

The first difference exists in the scheduling mechanism for video sources. The BS does not grant the earliest available information slots. Instead, the probability that a slot is assigned is dependent upon how many slots are still needed to satisfy a user's request. Via a process described in [10], the BS *spreads the allocation of slots randomly throughout the frame*.

The second difference is the use, in DPRMA, of certain transmission rates for the video users. In DPRMA, a user continuously determines the appropriate reservation request that ensures timely delivery of the traffic awaiting transmission in its buffer. DPRMA uses 7 transmission rates.

The third difference is that DPRMA does not use either voice request slots or our idea of **p***, but adopts a PRMA-like approach for voice users, by allowing them to compete for the available information slots.

The fourth difference is that, in DPRMA, both voice and video users waste one slot when giving up their reservations. This does not happen in our scheme, 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.

5. Results and Discussion

Table 2 presents the results of both our scheme (VVI, Voice-Video-Integration) and DPRMA, for the maximum voice capacity achieved by each scheme, given the number of video users in the system. The last column of the Table shows the allocation probability **p*** for which VVI achieves this capacity.

The reasons for which VVI excels in comparison with DPRMA are (in respect to the four differences between the two schemes, presented in section 4):

1) The mechanism proposed for the video slot scheduling in DPRMA is less efficient than that of VVI. Our proposed mechanism is much more dynamic than that of DPRMA, thus achieving higher bandwidth utilization.

2) The combination of differences two and four in section 4, makes clear that our scheme again leaves less unused slots than DPRMA.

3) By using the two-cell stack random access algorithm, VVI allows the voice users to make their requests 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 VVI after they have sent their request to the BS. On the contrary, in DPRMA the ″obstacle ″ is set by using a small transmission probability when implementing the PRMA protocol for voice slot contention. The latter approach is less effective, because voice users must repeatedly enter contention in order to reserve a slot, thus leaving more slots unused. Additionally, the use of ER slots helps VVI ″exploit″ certain available slots that DPRMA leaves unused.

6. Voice-Video-Data Integration

6.1 New System Model and Protocols

In the case of the integration of all three types of traffic, *we introduce the idea* [12,13] *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. In addition, the ER slots can be used by both voice and data terminals, with priority given to voice terminals.

The aggregate data message arrivals are Poisson distributed with mean λ messages per frame. The data messages vary in length according to a geometric distribution with parameter q and mean $B=1/q$ (B is equal to 8 packets per message in our study). An upper limit on the mean data message delay, equal to 200 ms, is assumed.

The *two-cell stack* blocked access collision resolution algorithm [9] 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.

The base station scheduling is identical to the one described in section 2, with the probability **p*** taking the values shown in Table 2. Each data user is allowed to reserve just one slot per frame. This choice is explained in the next section, where we discuss our results.

6.2. Results and Discussion

Table 3 presents the results for the maximum voice capacity, obtained for various values of the number of video users and the data message arrival rate λ. It is clear, from the results presented in the Table, that the smoothest transitions (decreases) for the maximum voice capacity take place when no video users exist in the system. This is easily explained by the fact that video traffic is quite bursty. Thus, the number of voice terminals has to be drastically decreased as the data message arrival rate increases, in order to cope with the burstiness of the video traffic mainly but also with the burstiness of the data traffic, and still be able to preserve the QoS requirements for each traffic type. This explains our ″defensive″ choice of not granting, in any case, more than one information slot per frame to each data user, as this would lead to a further decrease of the maximum voice capacity in order to preserve the video QoS requirements.

The throughput achieved by our scheme is decreasing when the number of video terminals increases, as expected because of the bursty nature of the video traffic. Still, as shown in Table 3, it remains quite high for all cases, thus pointing out the good performance of the scheme[3].

Figures 7 and 8 show the mean data message delay as a function of the number of voice terminals (VTs) in the system, and the average channel load (in packets/frame). Figure 7 presents the case of the number of video terminals being equal to 3 and the data message arrival rate equal to 2 messages/frame. Figure 8 presents the case of the number of video terminals being equal to 5 and the data message arrival rate equal to 0.5 messages/frame.

We observe in both figures that the average data message delay is much smaller than the 200 msecs limit, for the maximum voice capacity in each case. This means that the system would be capable of supporting a higher data message arrival rate for the same voice capacity, in the case where the maximum allowed video dropping probability limit would be higher than 0.01%, which is again the most restraining parameter for the scheme.

7. Conclusions

In this paper we have proposed and evaluated a new multiple channel access control scheme for integrating voice, video and data packet traffic in a high capacity picocellular environment. Video traffic is offered absolute priority over voice and data traffic, due to its more stringent quality of service requirements.

Via an extensive simulation study we demonstrate that our scheme evidently excels when compared in voice-video integration to a recently introduced MAC scheme, called DPRMA. Also, the proposed scheme achieves high throughput when integrating all three traffic types, despite the very restraining video dropping probability limit.

The results achieved by our scheme are a consequence of the combination of three factors: a)our voice slots allocation policy, b)our video slots scheduling policy and c) the use of the unused information slots as extra request slots.

REFERENCES

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

[2] T. Suda and T. Bradley, ″ Packetized voice/data integrated transmission on a token passing ring local area network ″, *IEEE Transactions on Communications*, Vol. COM-37, No. 3, pp. 238-244, 1989.

[3] A. C. Cleary and M. Paterakis, ″ An Investigation of Reservation Random Access Algorithms for Voice-Data Integration in Microcellular Wireless Environments″, *Int. J. Wireless Info. Networks*, Vol. 2, No. 1, Jan. 1995, pp. 1-16.

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

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

[6] W. C. Wong and D. J. Goodman, ″ A packet reservation multiple access protocol for integrated speech and data transmission ″, *IEE Proceedings-1*, Vol. 139, Dec. 1992, pp. 607-612.

[7] D. Bertsekas and R. Gallager, ″ Data Networks ″, 2nd Ed., Prentice Hall Inc., 1992.

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

[9] M. Paterakis and P. Papantoni-Kazakos, ″ A Simple Window Random Access Algorithm with Advantageous Properties ″, *IEEE Transactions on Information Theory*, Vol. IT-35, No. 5, September 1989, pp. 1124-1130.

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

[11] H. Heeke, ″ Statistical multiplexing gain for variable bit rate video codecs in ATM networks″, *Proc. of the 4th Int. Packet Video Workshop, Tokyo, Japan, March 1991*.

[12] P. Koutsakis and M. Paterakis, ″Near-Optimal Voice-Data Integration over Third Generation Medium and High Capacity Wireless TDMA Channels″, *Proc. of the IEEE Wireless Communications and Networking Conference 1999 (WCNC '99), New Orleans, USA*.

[13] P. Koutsakis and M. Paterakis, ″Highly Efficient Voice-Data Integration over Medium and High Capacity Wireless TDMA Channels″, *accepted for publication in the ACM/Baltzer Wireless Networks Journal (to appear)*.

[14] P. Koutsakis and M. Paterakis, ″Efficient Voice-Video Integration over Wireless Channels with Adjustable Request Bandwidth″, *Proc. of the Multiaccess, Mobility and Teletraffic in Wireless Communications Workshop (MMT'99), Venice, Italy*, pp. 11-21 (supplement).

[15] P. Koutsakis and M. Paterakis, ″On Multiple Traffic Type Integration over Wireless TDMA Channels with Adjustable Request Bandwidth″, *accepted for publication in the International Journal of Wireless Information Networks (to appear)*.

3 The symbol "x", used in the last two columns of the Table, stands for the system inability to accommodate the corresponding traffic.

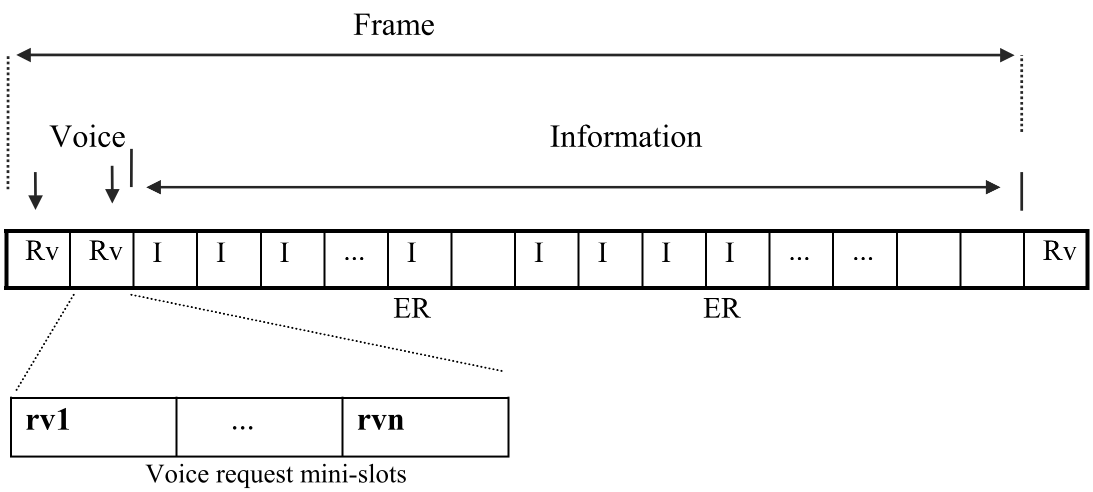

Figure1. An example of a channel frame structure showing the voice, data and information intervals within a frame.

Table1. Adjustable voice request bandwidth depending on the number of video users.

Number of Video users	Number of request slots
6	1
5	1
4	2
3	2
2	3
1	4
0	5

Table 2. Comparison of results between VVI and DPRMA.

Number of Video Users	Maximum Voice Capacity		Channel Throughput (%)		Probability p*
	VVI	DPRMA	VVI	DPRMA	
6	3	0	74.2	73.7	0.0072
5	110	99	79.6	77.7	0.03
4	205	187	82.9	80.0	0.06
3	295	291	85.5	84.8	0.085
2	386	385	88.2	88.0	0.128
1	475	475	90.6	90.6	0.18
0	587	563	96.7	92.7	1

Table 3. Maximum Voice Capacity for a set number of video users and set data message arrival rate.

λ (mes. /frame)	Second Row: Number of video users Next Rows: Maximum Voice Capacity and Throughput (%)													
	0		1		2		3		4		5		6	
0.1	566	96.3	464	91.1	377	88.6	287	85.5	201	83.2	104	79.3	0	74.3
0.5	557	96.0	433	87.1	350	85.3	264	82.9	185	82.1	90	78.2	x	x
1.0	549	96.3	410	84.9	330	83.5	244	81.1	164	80.1	70	76.5	x	x
1.5	539	96.2	400	84.8	314	82.4	229	80.2	150	79.4	57	75.9	x	x
2.0	530	96.2	385	83.8	300	81.6	215	79.4	136	78.6	44	75.3	x	x
2.5	521	96.3	367	82.4	287	81.3	201	79.0	124	78.2	34	75.2	x	x
3.0	510	96.0	362	83.1	277	80.9	192	78.7	115	78.2	27	75.6	x	x
3.5	503	96.4	345	81.8	265	80.5	181	78.5	104	78.0	16	75.3	x	x
4.0	494	96.5	338	82.2	258	80.9	173	78.7	94	77.9	6	75.2	x	X

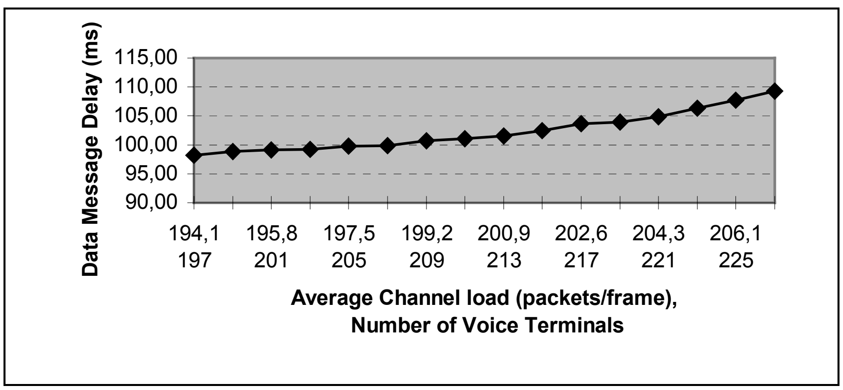

Figure 2. Data Message Delay vs. Number of Voice Terminals, for λ=2 messages/frame and 3 Video Users.

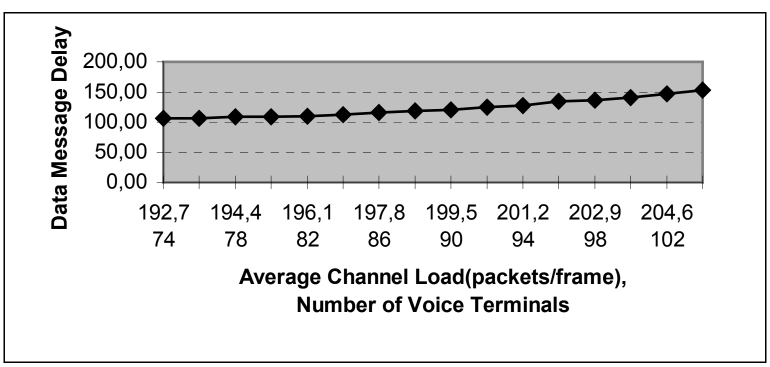

Figure 3. Data Message Delay vs. Number of Voice Terminals, for λ=0.5 messages/frame and 5 Video Users.