

Dynamic Bandwidth Reservation Scheduling for the Integrated Wireless Access of H.263 Videoconference Traffic with Voice and E-mail Packet Traffic

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ABSTRACT

A medium access control (MAC) protocol for wireless multimedia communications is presented and investigated. We explore, via an extensive simulation study, the performance of the protocol when integrating voice, H.263 variable bit rate (VBR) video and bursty e-mail data packet traffic over a wireless picocellular system of high capacity. Our scheme achieves high aggregate channel throughput in all examined cases of traffic loads, while satisfying the Quality of Service (QoS) requirements of each traffic type.

I. SYSTEM MODEL-INTEGRATED ACCESS

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. Since the base station is the sole transmitter on the downlink channel, it is in complete control of the downstream traffic, using Time Division Multiple Access (TDMA) to relay information to the users. Thus, we focus on the uplink (mobiles to base station) channel, where a MAC scheme is required in order to resolve the source terminals contention for channel access.

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 (ATM size, 53 bytes) per frame. Each frame consists of two *types* of intervals. These are the *voice and data request* intervals and the *information* intervals.

Within an information interval, each slot accommodates exactly one, fixed length, packet that contains voice, video or data information and a header. Voice and data 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. By using more than one minislot per request slot, a more efficient usage of the available request bandwidth is possible. Since we assume that all of the voice source transitions occur at the frame boundaries, we place all request intervals at the beginning of the frame, in order to minimize the voice packet access delay. *We use the idea introduced in [9] 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.

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 slot) for voice and data terminals [4]. The ER slots can be used by both voice and data terminals, with priority given to voice terminals. The concept of reserving a minimum bandwidth for voice and data terminals to make reservations helps to keep the voice access delay within relatively low limits and gives clearly better performance than the PRMA [1] and quite a few PRMA-like algorithms, such as DPRMA [3], 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 video sources are assumed to “live” permanently in the system (since video session durations are in general much longer than voice talkspurts), they do not follow an ON-OFF state model like voice sources. Thus, there is no need for granting request bandwidth to the video terminals, as it would be wasted in most cases.

We consider the channel to be error-free and without capture.

A. Voice and Data 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. 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 [1].
3. The voice delay limit is equal to 40 ms [3], and the allowed voice packet dropping probability is set to 0.01.
4. Reserved slots are deallocated immediately.

We adopt the data traffic model based on statistics collected on email usage from the Finish University and Research Network (FUNET) [11]. The probability distribution function $f(x)$ for the length of the data messages of this model was found to be well approximated by the *Cauchy* (0.8,1) distribution. The packet inter-arrival time distribution for the FUNET model is exponential.

B. H.263 Video Streams

H.263 is a video standard that can be used for compressing the moving picture component of audio-visual services at

low bit rates. As such, it is appropriate for low-bit rate videoconferencing applications on the go [14].

In our study, we use the trace statistics of actual H.263 streams from [13]. The video streams have been extracted and analyzed from a camera showing the events happening within an office. We have used three coding versions of the movie:

The first has a mean bit rate of 16 Kbps and a peak rate of 84 Kbps, the second has a mean bit rate of 64 Kbps and a peak rate of 320 Kbps, and the third has a mean bit rate of 256 Kbps and a peak rate of 1.4 Mbps. The maximum transmission delay for the video packets of a video frame is equal to the time before the arrival of the next video frame, with packets being dropped when the deadline is reached. 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. 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 until the end of the corresponding request interval to learn of its reservation slot (or slots). If unsuccessful within the request intervals of the current frame, the terminal attempts again in the request intervals of the next frame. A terminal with a reservation transmits freely within its reserved slot.

Video terminals, as already mentioned, do not have any dedicated request slots. *They convey their requirements to the base station by transmitting them within the header of the first packet of their current video frame.*

Upon successful receipt of a voice or data request packet, the BS provides an acknowledgment and queues the request. The BS allocates channel resources at the end of the corresponding request interval, and follows a different allocation policy for video terminals than that for voice terminals.

Video terminals have the highest priority in acquiring the slots they demand. If a full allocation is possible, the BS then proceeds to allocate any still available information slots to the requesting voice terminals. Otherwise, if a full allocation is not possible, the BS grants to the video users as many 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 80 channel frames here), and thus video terminals, would not find enough slots to transmit in, and the particularly strict video QoS requirements (the maximum allowed video packet dropping probability is only 0.0001) would be violated. Consequently, *the BS allocates a slot to each requesting voice terminal with a probability 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).

In order to efficiently incorporate email data users into the system, the BS "preempts" data reservations in order to service voice requests. When data reservations are "preempted" (canceled), the BS notifies the affected data terminal and places an appropriate request at the front of the data request queue.

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 "reshuffling" policy can be found in [10], where we study different video and data models, and we do not consider preemption of data users).

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 [7], Controlled Aloha [6], Three-Cell Stack [2]). 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 these in [3,5].

The *two-cell stack* blocked access collision resolution algorithm [8] 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.

II. SYSTEM PARAMETERS

The channel rate is 9.045 Mbps (from [3]). The 12 ms of frame duration accommodate 256 slots. The number of request slots shared by voice and data users is fixed, and equal to 2. It has been found, through simulation, that this number of request slots suffices for voice and data users in all the examined cases of video load. We should also note that:

1. In our design, we chose the number of minislots per request interval to be equal to 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. The average email data message length is 80 packets. We do not impose an upper limit on the average email data message delay, as this is a type of traffic that can withstand a delay of a number of seconds or even more.
3. The value of the *probability p^** is chosen equal to 0.1 (10%). Other values of p^* have also been tried out through simulation, and it has been found that the chosen value gives very satisfactory results for all the examined cases of video load.

III. RESULTS AND DISCUSSION

Tables 1, 2 and 3 present the simulation results of our scheme, when integrating voice and H.263 video streams coded with different target mean bit rates, and satisfying the QoS requirements of both traffic types. We observe that, in all cases, the bursty nature of the video streams leads to a significant decrease of the maximum voice capacity (and, consequently, of the aggregate channel throughput) as the number of video users in the system increases. Still, there is a very large difference between the channel throughputs achieved for 16 and 64-Kbps target mean bit rate video coding, in comparison to the throughputs achieved for 256-Kbps target mean bit rate video coding. In the latter case, the throughput is very low, especially in the case where more than three 256-mean bit rate video users access the system. This result can be explained by both the facts: a) that the mean target bit rate of the users in this case is much higher, and b) that the peak-to-mean bit rate ratio is again (as in the cases of 16 and 64-Kbps coding) quite high, causing the peak to be 1.4 Mbps. Therefore, it is the burstiness of the video coding that hinders the system from reaching higher channel throughputs. However, the results presented in Tables 1 and 2, as well as the results for 256-Kbps video coding with less than 3 video users present in the system show that our scheme achieves quite high (and in certain cases, very high) throughputs while satisfying the very strict Quality of Service (QoS) requirements of each traffic type.

Tables 4, 5 and 6 present the simulation results of the scheme, when integrating all three traffic types: voice, H.263 video streams, and email data.

We present the voice capacity for different email message arrival rates (λ messages/frame) and the corresponding channel throughput. We examine the cases of λ being equal to 0.01, 0.05, 0.1 and 0.3 messages/frame, (i.e., 25.6 Kbps, 128 Kbps, 256 Kbps and 768 Kbps, respectively), and we observe from our results that, for a given number of video terminals, as λ increases, the channel throughput remains more or less stable. This proves the efficiency of our data preemption mechanism, which allows the incorporation of larger data message arrival rates into the system without significant reduction of the voice capacity or violating the strict QoS requirements of video and voice traffic. The "x" symbol in the last row of the table means that this load can not be supported by the system. 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 video traffic.

The results presented in all the Tables show that the examined scheme achieves good channel throughput (steadily over 60 and occasionally close to 80 percent) for all cases except, again, for the case of servicing more than 2 video users of 256-Kbps target mean bit rate coding. If, however, the QoS requirements for video traffic were more "tolerant" (e.g., if we considered the acceptable upper limit of the video dropping probability to be 0.001 instead of 0.0001, which would be reasonable, given the nature of the movie (videoconferencing)), then the throughput of our

scheme would be considerably larger (by about 8-10%, as observed from our simulations).

The average email data message delay (in msec) does is, in all the examined cases, lower than 2 seconds, which is a completely acceptable delay for email traffic. It does not increase linearly with the increase of the data message arrival rate, as would be expected for a given number of video terminals. This happens mainly because of the burstiness and the exceeding variance of the Cauchy distribution, which produces the length of the data messages of our model. Another reason for this is the fact the voice capacity decreases with the increase of the data message arrival rate.

A final comment that needs to be made is that we do not further "penalize" the data users, by using a second preemption mechanism which would also cancel data reservations in favor of video reservations, because: a) video users are already given the highest priority, and b) even in the case of abnormally high data message arrival rates (e.g., for $\lambda=2$, which corresponds to 160 data packets/frame, i.e. 6.25 Mbps of data), data users with newly generated messages and without slot reservations would acquire on average only 2 slots per frame (i.e., 1 slot/arriving message).

IV. CONCLUSIONS

The reasons for which our scheme achieves good results are:

- 1) Our proposed video slot allocation mechanism is very dynamic.
- 2) The use of the probability p^* for the allocation of slots to voice terminals ensures the absolute priority of the very demanding video traffic in the system.
- 3) With the proposed above mechanism and the use of ER slots, our scheme "exploits" the maximum amount of slots within the frame.
- 4) The data preemption policy proves itself to be very effective, as it imposes just a small extra delay on email data messages, while at the same time it helps to increase the voice capacity significantly.

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Video Users	329	320	300	250	200	150	100	50	30	10
Voice Capacity	1	11	36	102	173	235	303	395	434	496
Through-Put (%)	67.95	67.65	67.70	68.47	70.03	70.03	70.96	76.64	78.75	83.41

Table 1. Voice Capacity and Channel Throughput for 16-Kbps target mean bit rate.

Video Users	76	70	60	50	40	30	20	10	5	3
Voice Capacity	4	31	82	143	215	280	349	422	472	498
Through-Put (%)	61.16	60.98	61.62	63.55	67.44	70.21	74.02	78.48	82.36	85.30

Table 2. Voice Capacity and Channel Throughput for 64-Kbps target mean bit rate.

Video Users	10	9	8	7	6	5	4	3	2	1
Voice Capacity	9	39	69	104	143	196	233	288	360	444
Through-Put (%)	33.06	35.05	37.01	39.89	42.91	48.75	51.87	57.64	66.67	76.32

Table 3. Voice Capacity and Channel Throughput for 256-Kbps target mean bit rate.

Messages/ No. video users		0.01	0.05	0.1	0.3
80	Cap	333	327	323	292
	EMmD (ms)	1472.81	1869.37	1013.01	963.86
	Throug. (%)	72.79	72.65	72.58	72.18
160	Cap	215	209	206	172
	EMmD (ms)	879.18	967.73	940.56	1031.18
	Throug. (%)	69.49	69.40	69.20	69.19
240	Cap	113	108	104	71
	EMmD (ms)	980.34	1027.42	969.56	1012.04
	Throug. (%)	68.67	68.41	68.35	68.24
329	Cap	×	×	×	×
	EMmD (ms)	-	-	-	-
	Throug. (%)	-	-	-	-

Table 4. Voice-Video-Data Integration Results for 16-Kbps target mean bit rate video.

Messages/ No. video users		0.01	0.05	0.1	0.3
20	Cap	349	315	307	276
	EMmD	927.20	1373.47	2101.68	1204.69
	Throug. (%)	74.14	74.03	73.93	73.84
40	Cap	212	177	165	138
	EMmD	1150.50	1621.38	1278.88	1255.33
	Throug. (%)	67.76	66.96	66.57	66.33
60	Cap	81	77	73	41
	EMmD	1063.11	1277.93	1585.66	1267.73
	Throug. (%)	61.66	61.59	61.56	61.57
76	Cap	3	×	×	×
	EMmD	980.25	-	-	-
	Throug. (%)	61.18	-	-	-

Table 5. Voice-Video-Data Integration Results for 64-Kbps target mean bit rate video.

Messages/ No. video users		0.01	0.05	0.1	0.3
1	Cap	434	412	402	370
	EMmD	876.41	879.17	1017.85	1134.65
	Throug. (%)	76.35	75.84	75.62	75.28
4	Cap	226	220	217	185
	EMmD	982.87	1034.69	936.05	1031.26
	Throug. (%)	51.32	50.29	50.60	50.73
7	Cap	103	99	96	66
	EMmD	1478.11	664.75	848.12	974.53
	Throug. (%)	39.77	39.46	39.41	39.32
10	Cap	8	4	×	×
	EMmD	1091.08	862,20	-	-
	Throug. (%)	33.08	33,10	-	-

Table 6. Voice-Video-Data Integration Results for 256-Kbps target mean bit rate video.