PRMA with Reservation Subframe Protocol for Multimedia Services in Mobile Communication Networks

Yaser Abdalla
Department of Electrical Engineering
University of Washington
Seattle, Washington 98195-2500
yasera@danube.ee.washington.edu

Didem Kivanc
Department of Electrical Engineering
University of Washington
Seattle, Washington 98195-2500
didemk@u.washington.edu

Hui Liu *
Department of Electrical Engineering
University of Washington
Seattle, Washington 98195-2500
hliu@ee.washington.edu

Abstract- These paper introduces a reservation subframe protocol for multimedia services in a 3rd generation mobile telecommunication system. The Packet Reservation Multiple Access (PRMA) protocol is modified to achieve more efficient bandwidth management. Simulation results show that the resulting multiple access multimedia system can serve a large number of users with low probability of packet loss, enabling the use of high compression rates for both voice and video services.

I INTRODUCTION

The support for the forth-coming multimedia services in the next generation wireless cellular networks requires efficient use of the spectrum [1]. In the third generation mobile systems, a variety of media such as voice, data, and video are coexistent. A key issue in such a system is the design of suitable multiple access control protocol to efficiently arbitrate multiple terminals on a common shared channel. Time Division Multiple Access (TDMA) has emerged as one of the main candidates for 3rd generation multimedia systems due to its high throughput. Contention based TDMA is an inefficient solution for multimedia mobile networks. The Packet Reservation Multiple Access (PRMA) protocol that was introduced by Goodman [5,6], takes the advantage of the bursty nature of speech and data traffic. Typical voice calls last only a few minutes, and conversations are interspersed with pauses, between sentences, words and even syllables, which add up to over 50% of the total conversation time [7]. A speech activity detector at the voice terminal can distinguish between active and silent periods. Thus several voice conversations can be statistically multiplexed to improve the utilization of network resources. A disadvantage of this version of PRMA is that an entire slot is wasted in the event of a collision [8]. This makes PRMA infeasible under high load conditions [9]. This paper introduces a new modified PRMA protocol with reservation subframes to serve digital voice, video and data streams. The performance of the protocol is studied via computer simulation. In the next section, we will introduce the system model.

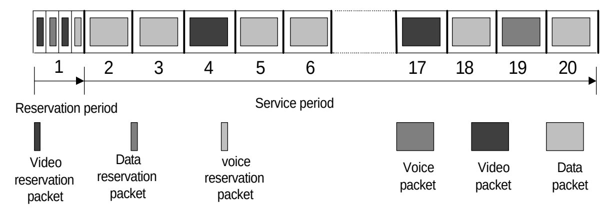

Figure 1, Slots and fram structure for PRMA with reservation subframe protocole

II SYSTEM MODEL

The mobile communication system under consideration supports a multitude of applications, such as voice, video, data, e-mail, fax, etc. with different service requirements. In this study, we provide three classes of service, two delay sensitive real-time service classes (voice, video) and one delay insensitive data class. Each of these traffic classes has unique characteristics such as activity factor and time-outs, as well as specific QoS requirements. Voice and video traffic differ in bandwidth requirements as well as resilience to packet loss.

Both voice and video are assumed to have the same delay requirements. Uplink and downlink transmission are multiplexed through Time Division Duplexing (TDD). The simulation study considers only uplink transmission. The transmission on both bands occurs in frames of length T_f. The frame length is chosen to coincide with the packet generation rate of the most abundant traffic class, voice.

From simulation parameters in Table 1, the amount of source information per packet is $R_{ss} T_f$ and the total packet length is $(R_{ss} T_f + H)$ bits. The channel carries $R_C T_f$ bits. And the number of slots N is given by

$$N = \left\lfloor \frac{R_c T_f}{R_{ss} T_f + H} \right\rfloor \quad (1)$$

Where $\lfloor c \rfloor$ is the largest integer X.

So, we find N = 20 slots/frame, the slot duration is Ts = 1 ms, the size of a packet is C = 704 bits, of which 640 bits are information, and 64 are header.

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A. Traffic Model

As PRMA is employed to support voice, video and data services on the same network. Each type of these services differs in its bandwidth requirements, sensitivity to delay and packet loss. Permission probability for each class is varied to allow access to the maximum number of users while still satisfying these requirements. The traffic models for the three classes are outlined below.

A.1. Voice Traffic

Voice is the primary class of traffic in this mobile communication network. For voice traffic, we use the model by Brady [10], in which speech sources alternate between an *ON* state ("talkspurt") during which they generate a constant stream of packets, and an *OFF* state ("principle gap") during which they are silent. The duration of a talkspurt (principle gap) is assumed to be exponentially distributed with a mean of $T_{voice\ ON}$ = 1 sec ($t_{voice\ off}$ = 1.35 sec) [11]. Voice terminals generate a steady flow of 32 kbps during the *ON* state. Due to the robustness of voice conversation, speech can be reconstructed at the distension user with acceptable quality provided that the voice packet loss is within the limit of 1%. Voice packets that suffer from a delay of 40 msec are dropped. The Percentage of dropped packets is the measure QoS.

A.2 Data Traffic

Data terminal generates data messages with a variable length (number of packets per message), which is buffered in first in first out (*FIFO*) buffer according to the order of their arrival. Data users cannot afford to drop packets at all, but tolerate some delay and can be allocated to slots, which are not reserved in the present frame. The average arrival rate of data messages is set to 50 message/sec (m_d = 50). The value of data permission probability is studied to reach the maximum number of user within the limit of delay. It is assumed that the data is originated from an independent Poisson input stream with an average arrival rate m_d, so that the mean inter-arrival time of data massage stream is $t_d = 1/\ m_d = 0.02$ sec.

A.3Video Traffic

The traffic model for video sources is assumed to resemble that for voice users. During the *ON* phase, video sources generate constant bit stream. The transmission time is exponentially distributed with mean $T_{video\ ON}$ = 180 sec. The *OFF* phase is also exponentially distributed, with mean $T_{video\ Off}$ = 20 ms. We assume that a video coder, such as MPEG [12] is employed at the video terminals, which requires a very low packet loss rate in order to reconstruct the video at the receiver. The instantaneous bit rate of a video codec varies greatly depending on the video sequence complexity. According to studies [13], a video of a moving object in a moving background can require anywhere from 20 kbps to 80 kbps for acceptable quality. Buffering at the transmitter and at the base station, within the bounds of the latency requirements, will reduce the variability of video output. Video users are allocated 3 slots per frame, leading to a throughput of 96 kbps, which leads to acceptable latency.

III PROTOCOL DESCRIPTIONS

The simulation procedure is outlined below:

1. When new data arrives, the terminal of class *q* goes from silent to active state.
2. The active terminal is allowed to transmit a reservation packet in the next reservation slot with probability $\boldsymbol{P_{pq}}$
3. If the terminal is allowed to transmit, it randomly picks a reservation channel and sends the reservation packet in it.
4. If some other terminal transmits a reservation packet on the same reservation channel, there is a collision. The terminal tries again in the next reservation slot.
5. If no other terminals transmit in the same slot, the terminal will enter the FIFO queue.
6. If there are no terminals of class 1, 2..., q waiting for an information slot, and *Nq* information slots are available, then the terminal is allocated *Nq* channels. Otherwise, the user will drop any packets delayed more than $\boldsymbol{D_{q\ max}}$.
7. If the terminal has any more packets left, it will continue to wait (go to step 6).

Once the terminal is allocated a channel, if it is not delay sensitive (data terminal), it can only transmit one packet, and then must release the channel. If it is delay sensitive (voice or video terminal), it will continue to transmit packets until its buffer is empty.

IV SIMULATION ASSUMPTIONS

The simulation is carried on a single cell and the BS is located in the middle of the cell.

All the mobile terminals (voice, data, and video) generate packets with a fixed length (640 bit) and the overhead is added to each packet before the transmission process. The gross packet (640 bit information + 64 bit header) is fit in one time slot of the PRMA frame. Each mini slot can carry 176 bits. Each slot includes a guard band at both ends to allow for unpredictable mobile to base packet arrival times which are very small. For simplicity, this guard bands are ignored and assumed that the wireless channel is error free. Any terminal wants to transmit will first need to synchronize with the frame timing derived from the BS on the down link. The simulation parameter for the intended system is shown in table (1), and the result was carried out by changing the number of simultaneous users and the permission probability for each class (one variable per simulation run), the results are studied and discussed.

V SIMULATION RESULTS

In this section, we study the effect of permission probability for the different service classes and the number of users in the system. For every set of parameters, the system is run for 5 x 10^6 time slots. The average delay for data users, and the probability of packet loss for voice and video users is plotted as a function of these probabilities. In simulations, we have allocated 1 of the N = 20 time slots to reservation, and 19 slots for information. The reservation slot is further divided into 4 minislots (reservation channels) as shown in Fig. 1.
The number of users and simulation parameters are set equal to the values shown in Table 1, and the quality of service requirements is outlined in Table 2.

The network is tested in different number of users in each class until the stability units approached, then we choose M_s = 16, M_v = 3, and M_d =17 users, while the effect of permission probability is shown in the following figures.

From Fig. 2, by increasing voice permission probability, there is degradation in video drop probability, while voice traffic gets better performance in the drop probability.

Fig. 3 shows the effect of introducing new voice users into the system. From the results, we see that with three active video users, up to 16 voice users can be added to the systems without significantly degrading the access for data users. Most interesting is the effect of video users. From Fig. 5, the maximum number of video users that can be accommodated by this system is 6, since there are only 19 traffic slots. Above 4 video users, the drop rate for video users increases dramatically, while that for voice users improves. This is because video users start waiting in the queue for at least one slot, and once a video user stops transmitting, immediately fill in the three vacant slots that he left behind. The queue for video users increases, so that by the time they get to the front of the queue, a significant portion of their active period is over. They do not keep the slot they reserve for very long, and when they are finished, voice users get precedence for the slots they leave. The voice packet loss rate drops close to that of a system with no video users. Data users get almost no access, so data throughput also drops dramatically.

From Fig. 4,5, all of the voice users combined can use a maximum of 16 slots per frame. This does not mean, however, that voice users always get a reservation directly after successfully transmitting a reservation packet. Since we are dealing with a non-preemptive system, if there all 3 video users are transmitting when the 11th voice user enters a talkspurt, there will not be any channels available (3 x 3 + 10 -19). Thus video traffic can cause packet loss in voice services even though voice has higher priority, particularly as the number of voice and video users becomes very large (Fig. 5). Data traffic as shown in Fig 6,7, on the other hand only

TABLE 1
Simulation Variables

Definition	Notation	Units	Parameter value
Channel rate	R_c	Kb/s	720
Voice source rate	R_{ss}	Kb/s	32
Video source rate	R_{sv}	Kb/s	96
Frame duration	T_f	ms	20
Slot duration	T_S	ms	1
Over head	H	b	64
Speech activity detector			Slow
Maximum delay	D_{max}	ms	40
Permission probability speech	P_{Ps}		0.1-.9
Permission probability data	P_{Pd}		.01-.9
Permission probability video	P_{Pv}		0.1-.9
Number of voice users	M_s		16
Number of data users	M_d		17
Number of video users	M_v		3

TABLE 2
Simulation Variables

Class	Unit	Value
Voice Drop Probability	Non	
Video Drop Probability	Non	
Data average delay	msec	300

contributes to congestion by increasing the probability of packet collision while in the reservation stage. Since data users cannot reserve slots in consecutive frames, voice or video users automatically replace them. In fact, data users can only use slots, which would otherwise remain empty, or be used by other data users, thus filling in the gaps in the bursty voice and video transmissions, improving system efficiency
From Fig 6, a larger value for permission probability will cause more collisions during reservation, causing packet delay and loss for all users. A smaller value will prevent the data user from transmitting its packet as quickly as possible. These parameters lead to a channel utilization of close to 0.9 without degrading the packet loss rates of the real time users.

VI CONCLUSIONS

In this paper we have examined the performance of PRMA with reservation subframe protocol for multimedia services over a wireless network. This protocol can be implemented in conjunction with GSM or CDMA protocols, to yield good capacity for all services. The network is shown to be capable of carrying a large number

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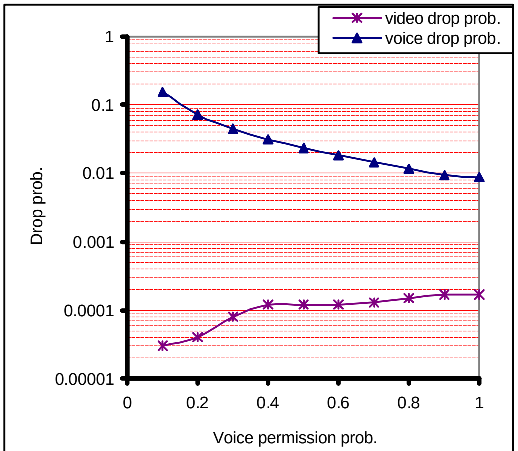

Figure 2. Variation of voice and video drop prob. with video permission prob.

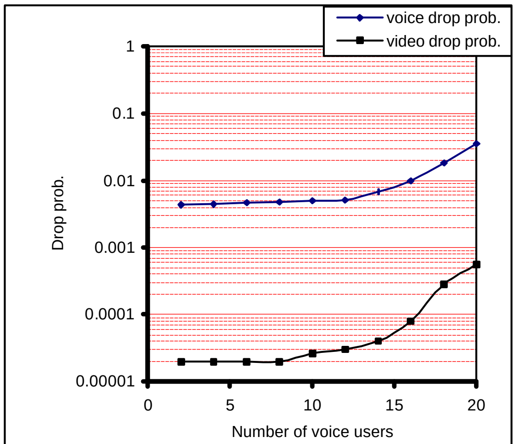

Figure 3. Variation of voice and video packet drop prob. with the number of voice users

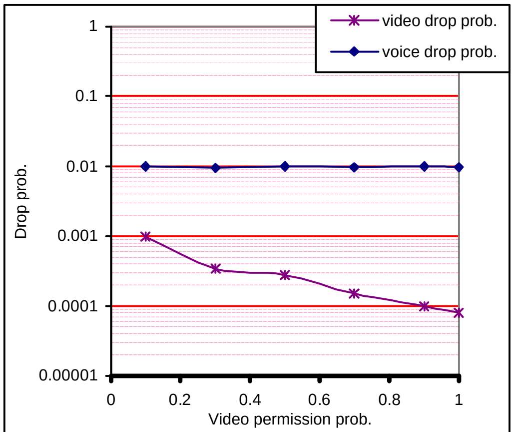

Figure 4. Variation of voice and videopacket drop prob. With video permission prob.

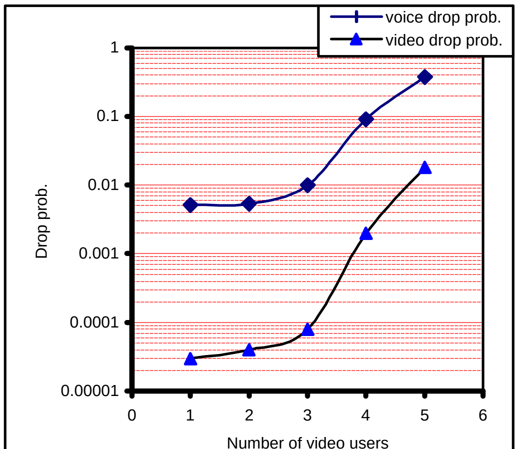

Figure 5. Variation of voice and video packet drop prob. with the number of video users

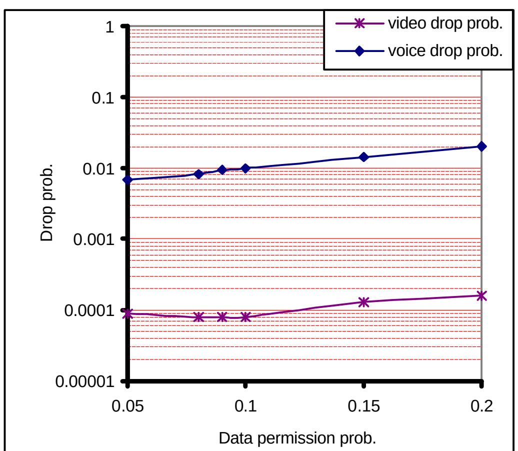

Figure 6. Variation of voice and video drop prob. With the data permission prob.

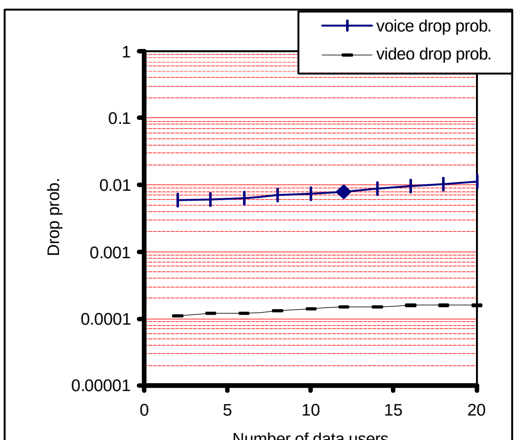

Figure 7. Variation of voice and video packet drop prob. with the number of data users