A Modified PRMA Protocol for Joint Voice Data Packet Wireless Networks

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Abstract. This paper discusses the possibility of using the Packet Reservation Multiple Access (PRMA) protocol to provide spatially dispersed voice and data user terminals wireless access to a base station over a common short-range radio channel. The case of ideal radio channel propagation conditions as well as that of non-ideal and non-stationary radio channel propagation conditions is considered. A modified PRMA (MPRMA) protocol is proposed in order to enhance system performance under heavy traffic load conditions. A technique to break off the possibility of accessing the common radio channel by a data user terminal is also discussed to avoid unsuccessful transmission of data packets during poor channel conditions. Performance comparisons with the standard PRMA protocol clearly point out a better behavior for the proposed MPRMA protocol.

1 INTRODUCTION

Wireless access to fixed telecommunication networks represents the main topic of research activities by researches, developers, manufactures and service providers throughout the world. To support a large population of users, with a limited radio spectrum, frequency reuse with low spatial separation (microcells) has been proposed [1]-[5]. A simple and efficient multiple access scheme for speech and data communications, named Packet Reservation Multiple Access (PRMA), has been proposed by Goodman *et al.* [2]. PRMA can be considered as a modification of the reservation ALOHA protocol [2]. Its main feature is the implementation of the principle "*listen-before-you-send*" which in particular makes it possible to distribute control to the user terminals. Note that this is extremely important in future third-generation wireless networks. The PRMA protocol permits seamless handovers for mobile users with minimal base station (BS) intervention and to significantly improve the bandwidth efficiency with respect to fixed assigned TDMA techniques [4], [5]. Although voice communications will have a dominant role in mobile communication systems, new important applications have been foreseen. Examples are wireless personal computers, mobile officers and electronic funds transfer, as well as road transport telematics, field service business, fleet management, and remote telematics. Therefore, interest in the integration of voice and data communications in wireless personal networks has been strongly stimulated. In [5] it is demonstrated that PRMA can only support sporadic data traffic (e.g., signaling traffic) without loosing the quality constraints for the voice service, i.e., to keep low the probability, P_{drop}, that a voice packet is discarded because it has encontrated an excessive delay in accessing the shared channel. This behavior can be explained by taking into account that PRMA suffers from a serious problem of variable access delay, which is enhanced in the presence of data traffic.

This paper investigates the possibility of using the PRMA protocol to support voice and data traffic in a wireless network. Moreover, a modified PRMA protocol (MPRMA) is proposed and analyzed in order to support more heavy data traffic than the classical PRMA without loosing the quality constraints for voice services. The MPRMA mainly differs from the standard PRMA in that a suitable scheduling procedure is used to increase data throughput and to lower P_{drop}. Performance of the PRMA and MPRMA protocols have been derived under the assumption of ideal and non-ideal, non-stationary radio channel propagation conditions.

This paper is organized as follows. Section 2 is a description of the wireless communication environment assumed herein and of the standard PRMA in a joint voice, data packet wireless network. In section 3 the MPRMA protocol is presented. In section 4, the performance of the MPRMA protocol is shown and compared with that of the standard PRMA protocol under different assumptions for the radio channel propagation conditions. Concluding remarks are given in section 5.

2 SYSTEM DESCRIPTION

This section presents the joint voice, data packet wireless network considered in this paper and discusses a possible approach based on the PRMA protocol in order to provide spatially dispersed voice and data UTs wireless access to a central BS. A more efficient scheme based on the MPRMA protocol will be decribed and analyzed in the next Section. All the acronyms and symbols used in this Section and in the next one are listed below for fast reader's reference.

UT: user terminal;
BS: base station;
τ: slot duration;
N_s: number of slots in a frame;
N_v: number of voice UTs;
N_d: number of data UTs;
ID: identification sequence of voice/data UT;
D_{max}: maximum acceptable speech delay at voice UTs;
VFB: finite capacity, First In First Out (FIFO) buffer for speech packets;
p_v: permission probability for contending voice UTs;
p_d: permission probability for contending data UTs;
p_a: access probability for waiting data UTs;
λ: data packets mean arrival rate per second;
L_d: mean number of packets in a data message;
DFB: infinite capacity FIFO buffer to store data packets;
HUT: hold on line (data) UT;
ρ: data traffic load at each data UT;
ρ_d: overall network data traffic load;
ρ_{dM}: maximum possible value of ρ_d;
ρ_l: highest value of ρ_d for which the constraint on P_{drop} is satisfied;
P_{drop}: speech packet dropping probability;
P_{dropM}: maximum acceptable value of P_{drop};
$\overline{X}$: mean data packet delay (slots).

We consider here a two way wireless communication network with a star topology, in which the spatially dispersed (mobile or fixed) UTs transmit voice/data packets to a central BS (uplink channel). The case of the classical PRMA protocol is first considered and analyzed. The uplink channel is slotted. Each slot has an equal duration τ and, N_s slots are grouped together to form a frame. The frame rate is the same as the arrival rate of speech packets [3],[4]. At the end of each slot, the BS broadcasts to all UTs short feedback packets on the downlink containing information concerning the outcome of the reservation for UTs or data transmission attempts and about the status (idle/reserved) of the next slot. We assumed that feedback packets are available to all UTs instantaneously (i.e., just before the beginning of the next slot) and error free, independently of the radio channel (downlink) propagation conditions [1]-[5]. In the remainder of this paper we will focus on the multiple access uplink only. Each speech packet to be transmitted is provided by a header, which contains the identification sequence (ID) of the sending UT and the specification of the traffic type. A slow speech activity detector is assumed at each UT during a voice call to distinguish when the speaker is silent or talking [4],[5],[9]. During talkspurts, the speech information generated in $N_s\tau$ seconds is organized in a packet of duration τ seconds. Conversational speech usually gives rise to a number of packets during a talkspurt greater than one. A voice UT is in the silent state (*SIL*) until it has no speech packet to send out. As soon as the first packet within a talkspurt is generated, the voice UT leaves the *SIL* state and moves to the contending state (*CON*) [3]-[5]. A voice UT in the *CON* state achieves permission to contend for reservation by transmitting a packet in an idle slot with probability p_v independently of other UTs in CON state. Parameter p_v is set to 0.3 to allow terminals to contend more frequently, to decrease the voice UTs access time and, therefore, to lower P_{drop} [4]. If no collision with other UTs arise, the UT attains reservation for the slot in subsequent frames. A successful reservation is attained if only one UT in the *CON* state achieves permission to access the channel (no collision) and the packet transmission is error free. In such a case, the BS broadcasts a positive acknowledgement (ACK) message immediately and a busy message before the beginning of the reserved slot in successive frames. The BS grants the successful voice UT a reservation for exclusive transmission on the same slot in successive frames. At the end of the talkspurt, the UT stops sending packets and moves back into SIL state. The first reserved slot left idle causes an interruption of reservation. The BS stops broadcasting the busy message before the beginning of the reserved slot as an indication that the slot is once again idle. If the BS is unable to decode the header of an arriving packet, a negative acknowledgment (NACK) message (usually in the form of a "*null*" message [4]) is broadcast to inform all UTs of this result.

Speech packets tolerate only a small delay for delivery. Hereafter, we denote with D_{max} the maximum acceptable speech delay at voice UTs. In this paper, as in [4],[5], we have assumed D_{max} equal to 2 frames. Each voice UT is provided with a finite capacity First In First Out (FIFO) buffer (VFB) to store incoming speech packets. During a talkspurt, a new speech packet is generated at every frame. Speech packets that wait longer than D_{max} in the VFB of a voice UT are dropped. A long delay in accessing the radio channel for voice UTs in the *CON* state (congestion) increases P_{drop}. The speech quality is strongly influenced by packets dropping. For practical applications P_{drop} values lower or equal to 0.01 are considered acceptable because they introduce an acceptable degradation on speech quality [4],[5],[10].

As to data communications we assume for all data UTs bulk packet arrivals according to independent Poisson processes with equal mean arrival rate per second (λ). The probability distribution of the number of packets arrived together at a data UT is geometric with mean value equal to L_d (packets). Packet duration time is τ (i.e., equal to the slot duration time). Packets arrived at data UTs are stored in data FIFO buffers (DFBs), which are assumed as having infinite capacity. Packets arrived together are randomly selected for storing. A data UT may be in the *SIL* state if its DFB is empty, otherwise it is in the *CON* state. A data UT in the *CON* state achieves permission to transmit a packet on an idle slot (i.e., not reserved by any UTs) with probability p_d. Taking into account the requirement of a prompt delivery of speech packets, it is suitable to set parameter p_d to a value lower than p_v. Whenever a data UT attains permission to access the channel, the packet at the head of its DFB is transmitted. As for speech UTs, permission to transmit a packet on an idle slot is generated independently of all other data UTs in the *CON* state. A data UT successfully transmits a packet on an idle slot if it has been the sole UT to access the channel on that slot (no collision) and the packet is received error free at the BS. In the case of a successful (unsuccessful) reception of a packet the BS immediately broadcasts an ACK (NACK) message to inform the sending data UT of the outcome of the transmission attempt. Each NAKed packet needs to be retransmitted. A packet is dropped from a DFB only when an ACK message is received. A successful data packet transmission on an idle slot provides reservation of that slot in successive frames to the sending data UT. As for speech packets transmission, the first reserved slot left idle causes an interruption of reservation. Nanda [5] has proposed a slight different approach by assuming that a successful data packet transmission does not provide the sending data UT with a reservation. The successful data UT must start again the contention procedure for transmission of other packets (if any) stored in its DFB on successive idle slots. This approach is efficient only under the assumption of sporadic data packet transmissions (i.e., very low data traffic load conditions). Conversely, it is not suitable when data traffic builds up. In this case, the number of UTs in the *CON* increases and UTs encounter a longer delay in gaining a reservation, therefore enhancing P_{drop} (the reservation procedure is the main bottleneck of the PRMA protocol [11]).

In deriving the performance of the PRMA protocol in the case of joint voice and data transmissions we have made the following assumptions:

a) number of slots per frame N_s equal to 20;
b) number of voice UTs (all involved in contemporary voice communications), N_v, equal to 20;
c) number of data UTs, N_d, equal to 20;
d) mean number of packets arrived together (bulk), L_d, equal to 50[1];
e) mean talkspurt duration equal to 100 slots;
f) mean silence duration equal to 135 slots.

In such a system, the speech activity factor (i.e., the portion of the time that the speech detector reports active speech [4]) results to be 0.43. The data traffic load ρ at each data UT can be defined as:

$$\rho = \lambda \tau L_d \tag{1}$$

Therefore, the overall network data traffic load is:

$$\rho_d = N_d \rho \tag{2}$$

The maximum possible value of ρ_d for a joint voice, data PRMA wireless network with N_v voice UTs contemporaneously involved in voice communications results to be:

$$\rho_{dM} = 1 - \frac{N_v\, 0.43(1-P_{dropM})}{N_s} \tag{3}$$

Parameter P_{drop} and the mean packet delay $\bar{x}$ (in slots), defined as the mean time elapsed between the arrival of a packet at the DFB of a data UT to its final departure (i.e. successful transmission), are shown in figures 1 and 2, respectively, as a function of parameter ε (i.e. ρ_d normalized with respect to ρ_{dM}) for different values of p_d. From these figures we can highlight that:

1. The highest value of ρ_d (ρ_{dl}) for which the constraint on P_{drop} is satisfied is lower than ρ_{dM}. This is due to the mutual interaction between voice and data traffic.
2. Suitable values of p_d depend on the desired trade-off between P_{drop} and $\bar{x}$. High (low) values of p_d lower (increase) $\bar{x}$ and increase (lower) P_{drop}.
3. Parameter P_{drop} exhibits a trend (limited to the interval [0,1]) which is close to exponential (figure 1). This means that under high traffic load conditions small variations of ρ_d lead to wide variations of P_{drop}. Note that P_{drop} should be as constant as possible and always below the maximum acceptable value P_{dropM}.
4. Parameter $\bar{x}$ results to be excessively high even under small values of ρ_d. This drawback is mainly due to the fact that the PRMA protocol does not make an efficient use of idle slots.

Hereafter, we set p_d to 0.15 as a good tradeoff between ρ_{dl}, $\bar{x}$ and P_{drop}. In particular, under this assumption we obtain ρ_{dl} equal to $0.6\rho_{dM}$ (Figs. 1,2), meaning that 60% of the slots not used for delivering speech packets are used for transmission of data packets.

[1] The number of packets arrived together has a geometric distribution.

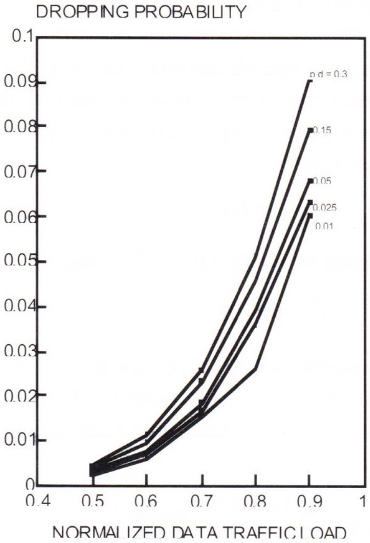

Figure 1 : Packet dropping probability for the PRMA protocol for different values of p_d (p_v=0.3, L_d=50 pkts).

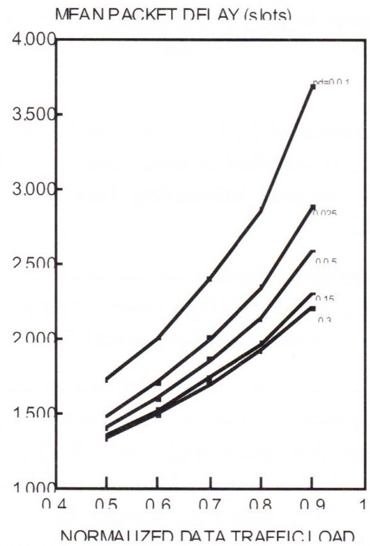

Figure 2 : Mean data packet delay for the PRMA protocol for different values of p_d (p_v=0.3, L_d=50 pkts).

3 Modified PRMA protocol

A better performance for a voice, data PRMA wireless network is possible through modifications of the standard protocol [12]. The aim of this paper is to propose and analyze a modified PRMA (MPRMA) protocol suitable for applications in joint voice, data packet wireless networks. The main characteristics of the MPRMA protocol are:

1. efficient use of idle slots for data transmissions;
2. increase of the highest value of ρ_d for which we have $P_{drop} \leq P_{dropM}$;
3. linear trend for P_{drop} as a function of ρ_d (instead of exponential);
4. reduction of the probability of collision for UTs in *CON* state;
5. low implementation complexity.

As for the classical PRMA scheme (see [4] at page 585), in the MPRMA protocol, the BS broadcasts feedback packets (downlink) to UTs, each containing service information, such as ACK/NACK messages concerning the outcome of reservation or data transmission (data UTs) attempts and notification about the status (idle/reserved) of next slot. In order to make the results presented in the paper consistent with those given in [4],[5] we have assumed that all messages broadcast on the downlink channel are correctly received. In many actual applications, this condition can be met by resorting to appropriate transmission power levels and/or by the use of an error correcting codes in order to guarantee a very low probability of an erroneous reception for a wide range of channel propagation conditions.

In the MPRMA protocol, the header of each packet to be transmitted includes the ID of the sending UT, the specification of the traffic type (voice or data) and, the request of reservation for slots in successive frames. Note that in the case of data transmission, slots reservation is not requested if there is only a packet stored in the DFB of the sending data UT.

The MPRMA protocol under consideration performs like the standard PRMA for speech communications. Hence, parameter p_v has been kept to 0.3 [4]. Conversely, the MPRMA protocol differs slightly from the standard PRMA in handling data communications. Each data UT stays in the *SIL* state if its DFB is empty. As soon as a bulk arrival occurs, the data UT moves to the *CON* state. In the MPRMA protocol, a data UT contends for successive slot reservations by transmitting the packet at the head of its DFB with probability p_d ($<p_v$). Through computer simulations we have found that system performance is slightly influenced by p_d and that a good tradeoff is again to set p_d to 0.15. A successful packet transmission is attained if no collision with other voice or data UTs in the *CON* state arises and the received packet is errors free. In this case, the BS immediately broadcasts an ACK message; otherwise, a NACK message is sent out. A data UT leaves the *CON* state and moves to the waiting (*WAIT*) state after receiving an ACK message.

In the MPRMA protocol, the BS manages transmission of data UTs in the *WAIT* state by means of a simple scheduling procedure. The BS stores all the IDs associated with correctly received packets sent by UTs in the *CON* state in a FIFO queue, named hereafter access queue. Only the data UT with ID at the head of the access queue (hold on line data UT (data HUT)) may obtain permission to transmit data packets. An ID message remains in the access queue until a slot assigned to the associated data UT remains idle (i.e., the DFB of the data HUT has been emptied). In this case, the BS drops the ID of the data HUT from the access queue and moves the subsequent ID (if any) at the head of the queue (renewal of the data HUT). It is important to note that with this scheduling policy the data HUT may obtain reservation of more than one slot per frame. An idle slot (not used by any voice UT) is assigned to the data HUT with probability p_a, while with probability $1-p_a$ the slot remains idle. This "*random*" assignment procedure permits to relax the congestion situation lowering the radio channel access delay for voice or data UTs in the *CON* state. This statement will be demonstrated in the next section by means of performance comparisons with the classical PRMA protocol. It is important to point out that in this paper we have neglected the capacity waste due to the extra-overhead interval within each slot needed to implement the access (polling) policy for data UTs in the WAIT state. However, by taking into account that this extra-overhead period may be limited to few bytes duration (e.g., 2-3) the waste of the available capacity for data transmissions results to be very small (even negligible in specific applications characterized by medium and long data packet length in bytes). Therefore, the numerical results given here for the data subsystem performance may be considered a good approximation of actual values.

Each data UT in the *WAIT* state compares its ID with that marking each slot devoted by the BS to a data transmission. When an exact matching is highlighted, the data UT is enabled to transmit a packet. After transmission completion the data UT waits for an ACK/NACK message. The ACK/NAK delay is assumed negligible (i.e., ACK/NAK messages arrive just after packet transmission completion). The data UT moves to the *SIL* state as soon as it leaves idle a reserved slot (i.e., the DFB is empty).

It is straightforward to note that the performance of the MPRMA protocol is influenced by parameter p_a. In particular, low (high) p_a values lower (increase) P_{drop} but increase (lower) $\bar{x}$. We have carried out the design of p_a by means of computer simulations. Specifically, we have found that the best tradeoff is to set p_a to 0.8 in the case of ideal radio channel propagation conditions and to 0.75 in the case of non-ideal, non-stationary radio channel propagation conditions.

4 PERFORMANCE EVALUATION AND COMPARISONS

In this section numerical results[2] concerning the performance of the MPRMA protocol are presented and compared with those of the standard PRMA protocol. The case of an ideal radio channel propagation conditions and non-ideal, non-stationary radio channel propagation conditions have been considered. System parameters values used in performing our computer simulations are those specified in sections 2 and 3.

a) Ideal Radio Channel Propagation Conditions

A performance comparison between the MPRMA and the standard PRMA protocols assuming error free all packets transmitted (ideal radio channel propagation conditions) is shown in figures 3 and 4. In particular, figure 3 highlights that the MPRMA protocol permits to achieve higher values of ρ_{dl} (i.e. $\rho_{dl} \approx 0.82\rho_{dM}$) with respect to the standard PRMA protocol ($\rho_{dl} \approx 0.6\rho_{dM}$). Figure 4, showing parameter $\bar{x}$ as a function of ρ_d, points out that the MPRMA protocol permits to attain a lower $\bar{x}$ with respect to the standard PRMA protocol for all the ρ_d values ($0<\rho_d \leq 0.6\rho_{dM}$) for which P_{drop} is below the specified P_{dropM} (i.e., 0.01 in our case).

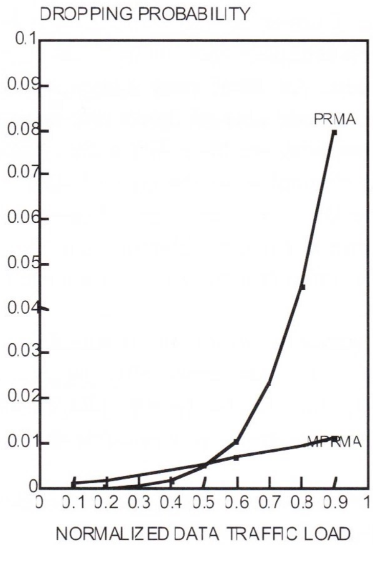

Figure 3: Packet dropping probability comparison (p_d=0.15, p_v=0.3, L_d=50 pkts, p_a=0.8).

b) Non-ideal and Non-stationary Transmission Channel Conditions

[2] Derived by means of computer simulations.