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Performance analysis of a modified CDMA/PRMA MAC protocol

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Abstract

This paper analyzes the performance of a protocol proposed to improve the performance of Joint CDMA/PRMA protocol under heavy data traffic load condition. The proposed protocol features a demand-based assignment scheme for data transmission. Its performance is evaluated using two analysis methods: a Traditional Markov Analysis and a Transient Fluid Analysis. Simulation results are also given to validate the assumptions made in the mathematical models developed. Our work shows that demand-based assignment scheme is suitable for random data traffic transmission, especially when there are a large number of active data terminals with short random messages.

Keywords: Code division multiple access; Medium access control; Packet reservation multiple access; Traditional Markov analysis; Transient fluid analysis

1. Introduction

Today's mobile communication systems are designed primarily to provide cost efficient wide area coverage for a large number of users with moderate bandwidth demands (voice + low rate data). The consumer of telecommunication services of tomorrow will expect to receive the same services in a wireless fashion as he receives from a fixed network. To meet this demand, third generation wireless systems are currently under development around the globe and are likely to be deployed with limited capabilities by the end of this year with major deployment at least a couple of years later. However, from the air interface point of view, the goal of providing higher bit rates is still valid. In addition to applications calling for higher bit rates, users will also want to use multiple services simultaneously. Emerging requirements for higher data rate services and better spectrum efficiency are the main drivers identified for the future mobile radio systems. In a wireless system consisting of a number of mobile terminals that transmit traffic of any type on a shared medium to a centralized base station (BS), a procedure must be invoked to distribute packet transmission among all users, which is known as a Medium Access Control (MAC) protocol.

Many MAC protocols have been developed to support speech and data traffic and can be found in Refs. [1,2,3–7]. Other MAC protocols that have considered multimedia traffic can be found in Refs. [8–12]. Among these protocols, a transmission protocol for packet voice terminals in a cellular system proposed by Goodman and Wei called PRMA [2], deserves mentioning. It is a frequency division duplex based multiple access protocol that allows a group of spatially dispersed terminals to transmit packet voice and low bit rate data over a common channel. The key feature of this protocol is the utilization of user transmission to gain access to the radio resources. Once the radio resource has been acquired, it is up to the transmitter to release the reservation. Following PRMA, many other modified PRMA protocol were published, which extend this mechanism to many other areas. Dynamic Packet Reservation Multiple Access (DPRMA) is a MAC protocol developed for wireless ATM networks [13]. The Centralized PRMA (CPRMA) MAC protocol was designed for a micro cellular environment [14]. In Ref. [8], as an extension of classical PRMA protocol proposed by Goodman and Wei [2] to CDMA, Joint CDMA/PRMA was proposed by Brand and Aghvami. Access to the radio channel is controlled such that interference variance is reduced and throughput is increased compared to the random access CDMA.

In Ref. [15], a modified protocol of Joint CDMA/PRMA was proposed, which is for the integrated voice/data services scenario, employing slotted packet CDMA and

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PRMA. A mechanism for combining the random access scheme and the demand-based assignment scheme was introduced. A modified scheme of Ref. [15] is introduced in this paper, which increases the number of simultaneous calls admitted to the system while still maintain almost the same performance. The detailed modifications will be described in Section 2. Moreover, to evaluate the performance of the modified protocol, two analytical methods are introduced: a Traditional Markov Analysis (TMA) and a Transient Fluid Analysis (TFA). TMA is employed to evaluate the voice call blocking probability while TFA is used to evaluate the average data packet delay.

2. The proposed protocol

In Ref. [15] a protocol featuring both controlled random access and demand-based assignment is proposed. This protocol deals with both voice and data traffic, with each single voice call being broken down to a number of voice spurts and gaps by Voice Activity Detector (VAD). A voice spurt is then defined by one of the active periods and a voice gap by one of the inactive periods within a voice call. More information related to VAD can be found in Ref. [2]. This protocol operates on a frame-by-frame basis. A frame is divided into L slots, with the first one being further divided into several mini-slots. The mini-slots are only used to transmit request packets while the other normal slots are used to transmit information packets, including both voice and data packets. The radio resource is divided in both time and code domains hence a logical -channel is defined by a pseudo noise (PN) code within a slot (normal slot), just as that in Joint CDMA/PRMA. The maximum allowed number of logical channels within one slot is K_{opt}. Upon a voice call is initiated by a user, it uses its first voice packet generated to contend in the normal slots in order to obtain a reservation of a logical channel. Once a logical channel is reserved by indication of BS acknowledgement, it is under this voice user's control until the call ends. Data traffic is handled by a demand-based assignment procedure, with their requests sent within those mini-slots in the first slot of each frame and then wait for the BS to allocate a logical channel for them. What saves the radio resources as claimed in Ref. [15] is the mechanism introduced as 'semi-reserved channel', that is a logic channel which is reserved by a voice user but is undergoing a period when there is no voice packets being transmitted through the channel due to the bursty nature of voice traffic. A semi-reserved channel can then be assigned to any data users to transmit their packets until the next talk spurt comes back. This procedure repeats until the end of this on-going call, as long as there are more talk gaps come along. This protocol tries to make use of the radio resource previously being wasted in the talk gaps. By employing this mechanism, on one hand the reservation of a particular voice call is guaranteed, and on the other hand the gaps between the voice spurts are 'recycled' to contribute to the total system resource.

Following the creation of the above mentioned protocol in Ref. [15], a modified version of it is proposed in this paper. The modification is based on the observation that since even if there are K_{opt} terminals in the slot, they are not transmitting all at the same time, as VAD suggests. Therefore there are chances that if we let more than K_{opt} voice users reserve the logical channels in the same slot, they can still be successfully received by the BS. For data traffic, a first-in-first-out (FIFO) buffer is used to store request packets, 200 packets/user on average.

A system is said to be operating in a stable condition when no buffer overflow occurs. When buffer overflow occurs, the system will run with very low efficiency. To maintain the stability of our system, a buffer control function is employed to avoid overflow. A parameter, th_{buffer}, is set as the upper threshold of the buffer. When the buffer is full, the buffer control function starts discarding the packets beyond the threshold, th_{buffer}, which are the oldest packets at the end of the queue. Though allowing the buffer overflow naturally is also a kind of control, the difference is that our control function discards the oldest packets first and the natural overflow discards the latest packets first.

3. Traditional Markov analysis model

We define $P_{\mathrm{gc}}^{\mathrm{f}}$ as the probability that a voice call is generated during a frame and $P_{\mathrm{ec}}^{\mathrm{f}}$ as the probability that an ongoing voice call is ended during a frame. We assume the inter-arrival time and call duration are both exponentially distributed as in Eqs. (1) and (2):

$$P_{\mathrm{gc}}^{\mathrm{f}} = 1 - \mathrm{e}^{-\lambda_c L} \tag{1}$$

$$P_{\mathrm{ec}}^{\mathrm{f}} = 1 - \mathrm{e}^{-\mu_c L} \tag{2}$$

where

λ_c = The arrival rate of a voice call
μ_c = The leaving rate of a voice call
L = the number of slots in each frame

The following parameters are also defined:

C_n = the total number of voice packets contending for transmission in frame n
V_n = the total number of voice ongoing calls in the system in frame n
W_n = the total number of voice spurts ending in frame n.

Since voice is given a priority over data transmission in the system, the voice status is independent of data process. M_V is the total number of voice terminals in the system, and K_{opt} is the optimum number of codes that can be assigned to simultaneous transmissions in a slot. In general, $M_V >> K_{\mathrm{opt}}$.