Computer Communications 31 (2008) 3978–3983

Contents lists available at ScienceDirect

Computer Communications

journal homepage: www.elsevier.com/locate/comcom

# A new framework for multiple access and call admission control in wireless cellular networks

Polychronis Koutsakis *

*Department of Electrical and Computer Engineering, McMaster University, Hamilton, Canada L8S 4K1*

A R T I C L E I N F O

*Article history:*
Received 11 October 2007
Received in revised form 4 August 2008
Accepted 5 August 2008
Available online 13 August 2008

*Keywords:*
Multiple access control
Call admission control
H.26 video traffic
Wireless cellular networks

A B S T R A C T

In recent work, we have introduced MI-MAC (Multimedia Integration Multiple Access Control), a new access control protocol for wireless cellular networks. MI-MAC was shown to be a good candidate for next generation wireless cellular networks, due to its superior performance in comparison to other (TDMA and WCDMA-based) protocols in the literature when integrating various types of multimedia traffic. In this paper we propose the combination of MI-MAC with a new efficient call admission control (CAC) mechanism, which will prevent bursty H.264 users from entering the network if system stability is not guaranteed. We proceed to evaluate the ability of our framework to efficiently integrate streams from latest technology video encoders with other types of packet traffic over noisy wireless networks, especially in the case of significant handoff loads. To the best of our knowledge, this is one of the first works in the literature to study the integration of H.264 streams with other types of multimedia traffic over wireless cellular networks.



## 1. Introduction

In recent work [1], we introduced and evaluated a new multiple access scheme which was shown to efficiently integrate voice (Constant Bit Rate, CBR, On/Off Traffic), email and web traffic with MPEG-4 and H.263 video streams (Variable Bit Rate, VBR) in high capacity picocellular wireless systems with burst-error characteristics. In this work we continue the performance evaluation of the scheme. The new elements of this work, in comparison to [1], are:

(a) the integration of streams from the latest technology video encoding (H.264) with voice and WAP (Wireless Application Protocol) traffic;
(b) the use of a new CAC mechanism which prevents bursty H.264 video users from entering the network if network stability is not guaranteed;
(c) the use of a different channel error model than the one used in [1];
(d) in [1], in order to facilitate the comparison with other protocols of the literature and given that the protocols were evaluated over one cell of the network, no traffic was considered to be arriving from other cells (handoff traffic). This assumption is waived in the present work, where a significant portion of the traffic in our simulations is considered to be handoff.

We focus on the uplink (wireless terminals to base station) channel, where a MAC scheme is required in order to resolve the source terminals contention for channel access. We compare the results of our framework with those of DPRMA, a well-known MAC protocol for wireless networks.

The paper is organized as follows. In Section 2 we analyze our proposed scheduling and CAC schemes, and present the various traffic types used in our study. Section 3 includes the details of the adopted channel error model. The system parameters are presented in Section 4. In Section 5 we discuss our simulation results, and in Section 6 we present our conclusions.

## 2. Multiple traffic type integration

### 2.1. Channel frame structure

The uplink channel time is divided into time frames of fixed length. The frame duration is selected such that a voice terminal in talkspurt generates exactly one packet per frame (packet size is considered to be equal to the ATM cell size for reasons of comparison with DPRMA [2]; however, the nature of our results remains the same, regardless of the packet size, therefore the scheme could be used in any GSM-type network). As shown in Fig. 1 (which presents the channel frame structure), each frame

* Tel.: +1 9055259140.
*E-mail address:* polk@ece.mcmaster.ca

0140-3664/$ - see front matter © 2008 Elsevier B.V. All rights reserved.
doi:10.1016/j.comcom.2008.08.003

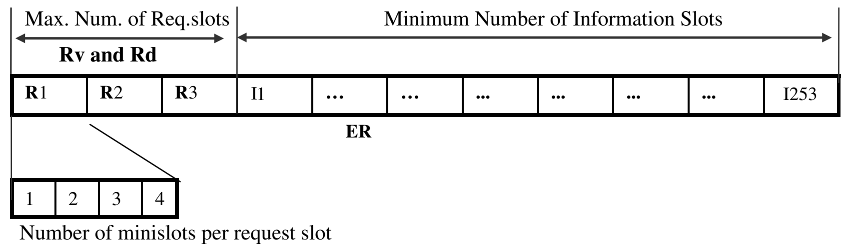


**Fig. 1.** Dynamic frame structure for the 9.045 Mbps channel, frame duration 12 ms.

consists of two types of intervals. These are the *voice and data request* interval (by data, we refer to WAP traffic), and the *information* interval.

Since we assume that all of the voice sources state transitions occur at the channel frame boundaries (this assumption will be explained in Section 2.2.1), we place the voice and data request interval at the beginning of the frame, in order to minimize the voice packet access delay. Request slots can be shared by voice and data terminals, in this priority order. No request slots are used for the video terminals. Since video sources are assumed to "live" permanently in the system (they do not follow an ON–OFF state model like voice sources) and the duration of our simulation study is long, we assume without loss of generality that they have already entered the system at the beginning of our simulation runs; thus, there is no need for granting request bandwidth to the video terminals (this assumption is again made in order to facilitate our scheme's comparison with the DPRMA protocol [2], which makes the same assumption). Regarding handoff video terminals, it is assumed that their current bandwidth requirements are known to the new Base Station (BS) through interaction with the last BS that serviced the video call.

The frame structure parameters have been chosen as follows:

(a) For all the examined scenarios of system load (a vast number of scenarios has been studied), we tried to find a *maximum* request bandwidth which would suffice for voice and data terminals. This was found, via simulations (both in [1] and in the present work), to be equal to three request slots.
(b) We design the protocol so that we can enforce *a fully dynamic mechanism* for the use of the request bandwidth: the number of request slots is variable *per channel frame* (between 1 and 3, which is the maximum number, as explained above), and depends on the total voice and data channel load in each frame. In the cases when less than three request slots are needed for the end of the voice and data terminals' contention, the Base Station signals all user terminals for the existence of additional information slots in the current frame.Also, *any free information slot of the current channel frame can be temporarily used as an extra request slot (ER slot)*[1] (the use of a slot as an ER slot is conveyed to the terminals by the BS after the end of the request interval in each channel frame).

## 2.2. Traffic types and models

### 2.2.1. Voice traffic model

Our primary voice traffic model assumptions are the following:

1. The speech codec rate is 32 Kbps [1]. The output of the voice activity detector (VAD) is modeled by a two-state discrete time Markov chain. The mean talkspurt duration is 1.0 s and the mean silence duration is 1.35 s.
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 s.
3. Reserved slots are deallocated immediately.

The allowed voice packet dropping probability is set to 0.01, and the maximum transmission delay for voice packets is set to 40 ms [2].

### 2.2.2. WAP traffic model

We adopted the WAP traffic model presented in [3] (corresponding to the WAP release 1.2.1) in our work. WAP sessions consist of requests for a number of decks, performed by the user. The number of decks is modeled by a geometric distribution with mean equal to 20 decks and the packet size by a log2-normal distribution. To cover the influence of different applications, four different types of user profiles are introduced: email, news, m-commerce and common (referring to mixed traffic traced from a WAP server in real operation. The size of a wap request message in [3] ranges on average between 82 and 112 bytes, depending on the specific user profile, i.e., it ranges between 2 and 3 ATM packets in size. The standard deviation of the size of a wap request message ranges between 16.5 and 84.7 bytes, i.e., between 1 and 2 ATM packets.

The arrival process of WAP sessions is chosen to be Poisson with rate $\lambda$WAP sessions per second, with an upper limit on the average WAP request transmission delay equal to 2 s. Given that the average size of a WAP request is quite small in terms of number of packets, it is clear that we adopt the widely accepted assumption that data traffic is delay-tolerant. Still, if we take into consideration that estimations of GSM networks' SMS transmission delays refer to delays of 2–30 s [4] (SMS messages have a payload of 140 bytes [5], i.e., similar to a WAP request), the upper bound set in this work for WAP request transmission is quite strict.

### 2.2.3. H.264 video streams

H.264 is the latest international video coding standard. It was jointly developed by the Video Coding Experts Group (VCEG) of the ITU-T and the Moving Picture Experts Group (MPEG) of ISO/IEC. It uses state-of-the-art coding tools and provides enhanced coding efficiency for a wide range of applications, including video telephony, video conferencing, TV, storage (DVD and/or hard disk based, especially high-definition DVD), streaming video, digital video authoring, digital cinema, and many others [6].

The 3rd Generation Partnership Project (3GPP) standardizing the Universal Mobile Telecommunications System (UMTS) has approved the inclusion of H.264/AVC (Advanced Video Coding) as an optional feature in release 6 of its packet oriented mobile multimedia telephony [7] and streaming service [8] specifications.

In our study, we use the trace statistics of actual H.264 streams from the High Definition (HD) Video Trace Library of [9]. The video streams correspond to videoconference traffic; they have been ex-