Analysis of Voice Traffic Performance over OFDM Wireless Access Systems with AMC

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Abstract – **Orthogonal Frequency Division Multiplexing (OFDM) is an attractive technology developed for future wireless and mobile communications systems.**

In this paper we evaluate the voice traffic performance of a multi-class OFDM wireless access system with specific modulation and coding schemes, employed at the physical layer to match transmission parameters to time-varying channel conditions. Numerical results demonstrate an approach for efficient radio resource management, which can improve the system performance.

Keywords – **performance evaluation, adaptive modulation and coding (AMC), OFDM, VoIP**

I. INTRODUCTION

In order to satisfy customer demands for access to a wide variety of multimedia applications over wireless communication networks, significant research efforts are being put in development of next generation mobile networks (NGMN), also referred to as 4G, based on all-IP environment. This can be achieved by employing the OFDM technique at the air interface, which is immune to intersymbol interference and frequency selective fading, due to the time-varying nature of the wireless channel. As a consequence, OFDM-based systems have become a popular choice, and have been adopted in several standards towards NGMN [1], [2].

In contrast to the traditional channelized multiple access systems, where each user is assigned a fixed amount of bandwidth during the whole connection time, in wireless networks the channel capacity of a wireless link is time-varying, and thus the quality of service (QoS) requirements may not be satisfied, even though a large amount of resource (i.e. bandwidth) is allocated to a certain connection. This is especially true when a mobile station (MS) is located at the cell edge area. In order to maintain a target packet error rate (PER) over wireless links, meeting QoS requirements, adaptive modulation and coding (AMC) schemes have been widely adopted to match the transmission parameters to time-varying channel conditions [3], [4].

The interaction of analytical framework for evaluation of teletraffic performance characteristics of the system under consideration at the data link layer with AMC at the physical layer provides a base for interesting and realistic design work. This topic is covered by a number of excellent papers describing in depth the essence of the underlying problems. Reference [5] investigates the performance of transmission over wireless links, in case an interaction between a finite-length queuing and different AMC schemes are taken into consideration. The Erlang capacity of WiMAX systems with fixed modulation schemes is calculated in [6], considering two traffic classes – streaming and elastic. An interesting approach for evaluation of Erlang capacity of a multi-class TDMA system with AMC by separating the calculation of blocking and outage probabilities is proposed by the authors of [7].

In this paper, we evaluate the teletraffic performance in terms of traffic congestion of multi-rate OFDM wireless access systems with AMC, under specific circumstances, attempting to find the optimal allocation of available resources.

The rest of the paper is organized as follows. We first introduce the system model, which takes into account the OFDM transmission with AMC. Section 3 represents an analytical model of a multi-rate loss system with relevant performance measures. Numerical results illustrating the dependences of system performance on various parameters are given in Section 4. Finally, concluding remarks have been drawn in Section 5.

II. SYSTEM MODEL

We consider an OFDM infrastructure-based wireless access network, where connections are established between a base station (BS) and mobile stations (MSs) inside the cell area. The access to the system resources is based on time-division multiplexing (TDM), and thus at the physical layer, the data stream is organized in frames of fixed length (duration). Since the data transmission is based on OFDM technology, a frame is divided into n time slots (TS). Users can transmit their data in the assigned time slots over all available subchannels. An AMC scheme is employed at the physical layer to match the transmission rate to the time-varying channel conditions as well as to maintain a target PER over wireless link. This results in splitting up the overall cell area on several regions (rings), corresponding to the available modes of AMC scheme, each of which represents a pair of specific modulation format and a forward error correcting (FEC) code.

From network operator's point of view, offered calls from MSs, belonging to the cell area working with the modulation scheme of highest order (closest to the BS), are most profitable, compared to the other ones, since they are allocated the smallest amount of resources (i.e. bandwidth) in order to be served. For this reason, to prioritize this class of calls, a mechanism for reservation of fixed amount of resources can be applied, and it allows the other classes of calls to compete for the remaining bandwidth (class limitation). If n denotes

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the total number of time slots in a frame for data transmission, Δ denotes the number of time slots that can be only reserved for incoming traffic flow of so called “profitable” class.

In this paper, we consider that a voice service is supported by the system. Since the voice is not tolerant to packet delay, it requires constant bit rate, which means that each voice user is allocated a number of time slots per frame, depending on its location in the cell (current modulation and coding scheme used).

III. Performance Analysis

Following the adoption of AMC scheme, we assume that the call attempts, following Erlang case, offered from the MSs belonging to a particular region (ring) of the cell generate a traffic stream. Each traffic stream (class) *i* is characterized by the offered traffic A_a^i (call attempts traffic), maximum number of allocated time slots n_i in the frame (class limitation) and the number of time slots d_i required for establishing one connection (supporting a voice call with a constant bit rate). For the *i*-th arrival process the arrival rate is $\lambda_i(x_i)$, for state $x_i \cdot d_i$, that is, when x_i simultaneous calls of type *i* are being served, and hence, the following restrictions shall be fulfilled:

$$0 \le x_i \cdot d_i \le n_i \le n, \qquad i = 1,\ldots,N \tag{1}$$

and

$$0 \le \sum_{i=1}^{N} x_i \cdot d_i \le n. \tag{2}$$

where number of classes *N* (traffic streams) is equal to the available transmission modes, supported by the AMC scheme.

We conduct the performance analysis of the system, described in Section 2, by applying the classical teletraffic model of multi-rate loss systems, which is based on flexible channel/slot allocation [8]. The call-level characteristic of the system can be modeled by a multi-dimensional Continuous Time Markov Chain (CMTC), which is reversible and has a product form. Due to reversibility, we can apply the local balance equations to calculate the relative state probabilities and to find performance measures of interest. Instead, the numerical evaluations are performed in an efficient way by using the convolution algorithm [8].

The state probabilities of each traffic stream *i* as if it is alone in the system are following Erlang distribution and can be represented as:

$$p_i(j \cdot d_i) = \frac{\dfrac{(A_a^i)^j}{j!}}{\sum\limits_{k \in \Omega_a^i} \dfrac{(A_a^i)^k}{k!}} \quad if \quad j \in \Omega_a^i \qquad p_i(\cdot) = 0 \quad else \tag{3}$$

$$\Omega_a^i = \left\{0,1,\ldots,\left\lfloor \frac{n_i}{d_i} \right\rfloor\right\} \tag{4}$$

where Ω_a^i are the sets of possible number of users in each service class *i* (*i* = 1,…,N) that can be served by the system and $\lfloor k \rfloor$ denotes the largest integer not exceeding *k*. Eq. (3) takes into account multi-slot traffic ($d_i > 1$) offered to a system.

Based on the state probabilities, we are able to get the performance measure of the system in terms of overall traffic congestion C_{tot}, obtained by using the convolution algorithm. The aggregated state probabilities for the system $Q_{N/i}$ is calculated by successive convolutions of state probabilities vectors of each traffic stream, excepting traffic stream *i*. The convolution of aggregated states probabilities and excluded traffic stream results in a state probabilities vector, which defines the probability of a given number of TS in the frame being occupied by calls offered by all traffic streams:

$$Q_N(j) = \sum_{x=0}^{j} Q_{N/i}(j-x) \cdot p_i(x) = \sum_{x=0}^{j} p_x^i(j), \tag{5}$$

where for $p_x^i(j)$, *i* denotes the traffic stream number, *j* is the total number of busy resources (time slots), and *x* is the number of occupied TS by stream number *i*.

The carried traffic Y_c^i of stream *i* measured in busy channels (TS) is:

$$Y_c^i = \sum_{j=0}^{n_i} \sum_{x=0}^{j} x \cdot p_x^i(j). \tag{6}$$

Since we are interested in determine the carried traffic Y_a^i of stream *i* measured in call attempts, it can be expressed as:

$$Y_a^i = \frac{Y_c^i}{d_i} = A_a^i \cdot (1 - C_i). \tag{7}$$

The traffic congestion C_i is equal to the ratio of offered traffic A_a^i which is blocked. It should be noted that both offered traffic A_a^i and carried traffic Y_a^i of stream *i* are measured in Erlangs (erl).

Due to its attractive performance characteristics, AMC has been adopted at the physical layer of several standards, e.g. 3GPP, 3GPP2, IEEE 802.11, IEEE 802.16 [1], [2], [9]. Based on channel state information (CSI) estimated at the BS, the set of available transmission modes, which are most commonly used, are based on 64-QAM, 16-QAM and QPSK modulation techniques.

As we stated above, the AMC scheme is applied to maintain a target PER over wireless link and therefore both the modulation scheme and coding rate are chosen according to time-varying channel conditions. As a consequence, the number of time slots d_i allocated to users of traffic stream *i* is varying and can be determined in accordance with the constant bit rate R_V bits per frame, required for voice users as well as the transmission mode i chosen.

$$d_i = \left\{ \left\lceil \frac{R_v}{s \cdot R_i} \right\rceil, \quad i = 1, \ldots, N \right\} \tag{8}$$

where s is the fixed number of OFDM symbols per time slot, and R_i denotes the number of bits carried per OFDM symbol in transmission mode i. Notation $\lceil k \rceil$ denotes the smallest integer larger than k.

We consider the following group of transmission modes i, as they are adopted in [9], [10] (Table I).

TABLE I
TRANSMISSION MODES WITH MODULATION AND CODING SCHEMES

i	Mode 1	Mode 2	Mode 3
Modulation	64-QAM	16-QAM	QPSK
Coding rate	3/4	3/4	3/4
R_i (bits/sym)	4.5	3.0	1.5
Target SNR for 1% PER (dB)	24.4	16.2	8.2

IV. NUMERICAL RESULTS

The current sections deals with numerical results based on the traffic model presented in Section 3.

We consider a single-cell OFDM system with time-division duplex (TDD) operation. The duration of a frame is set to be 1ms. The total number of time slots used for data transmission within the frame is set to be 200, each of which contains 4 OFDM symbols [7]. The modulation order and coding rate in the AMC scheme is determined by the instantaneous SNR of each user in the cell. We follow the AMC schemes shown in Table I, which specifies the minimum SNR required in order to maintain a target packet error rate of 1 %. Thus, the cell area is split up in three non-overlapping areas, at which MSs are served by the BS at a particular modulation and coding scheme. In this reason, the call attempts offered from MSs of specific area can be represented as three independent traffic streams, having an access to the common pool of resources in terms of available time slots n in the frame. The consecutive number of a traffic stream (shortly Stream) corresponds to the transmission mode i, serving data flows offered from a particular cell area.

Since the calls offered from the cell area closest to the BS (Stream 1) are seemed to be most profitable for the network operator, we prioritize them by reservation of a fixed number of time slots Δ in the frame, by restriction of accessibility of the remaining traffic streams to the resources in the frame (a class limitation for Stream 2 and Stream 3 is used).

From traffic engineering point of view, the traffic congestion is the most important measure, and hence the research work is oriented towards finding the optimal allocation of available resources (time slots) for both traffic stream 2 (call attempts served by the BS in transmission mode 2 – 16QAM) and traffic stream 3, so that a minimal value of overall traffic congestion C_{tot} to be obtained.

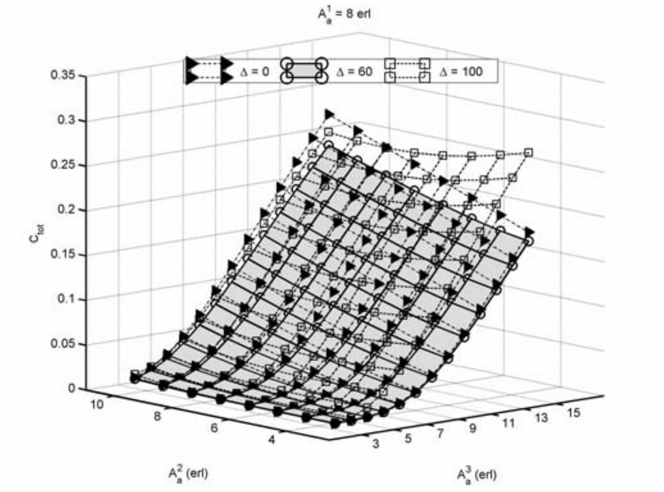

Fig. 1. Overall traffic congestion versus different traffic loads and limited accessibility of both Stream 2 and Stream 3

At the first point, we analyze how various traffic loads of Stream 2 and Stream 3 will affect the overall traffic congestion C_{tot} of the system, assuming a traffic volume of 8 erl. for Stream 1. We have conducted the analysis for three levels of reservation of time slots Δ in the frame for Stream 1 (Δ = 0, 60, 100 TS). Results are depicted on Fig. 1 and demonstrate that minimal levels of the overall traffic congestion, in case of high traffic loads, can be realized by using reservation scheme with Δ = 60 TS, in comparison with results for Δ = 0 (all three traffic flows can access to entire resource space of each frame). It is interesting to notice, that a similar trend can be also observed for reservation mode with Δ = 100 TS, but only if the offered traffic volume of Stream 2 is greater than 8 erl. If this rule is not properly fulfilled, the current reservation scheme that is applied is strongly inefficient (Fig. 1). We can also see that the increase of the traffic load in Stream 3 will increase the traffic congestion more sharply than Stream 2 does, since it requires and occupies more time slots in the frame for establishing a connection than Stream 2 does.

Following the assumptions, stated in Section 2, concerning user distribution into non-overlapping cell areas, at the second point of the analysis, we investigate the overall traffic congestion introduced by the system, versus the class limitation of both Stream 2 and Stream 3. Results depicted on Fig. 2 demonstrate this for different traffic volumes for each traffic stream (A_a^1 = 5 erl, A_a^2 = 8 erl, A_a^3 = 12 erl). We have shown two scenarios, taking into account the case where the traffic volume of Stream 1 is increased as well as the offered traffics of three streams increase their values with approx. 30%. These results are logical, since the class limitation of Stream 2 and Stream 3 leads to reservation of time slots exclusively for traffic Stream 1, resulting in decreasing of its traffic congestion, in case of increased traffic loads generated by the MSs in the cell, being served by the transmission mode employing the available modulation technique of highest order (64-QAM).

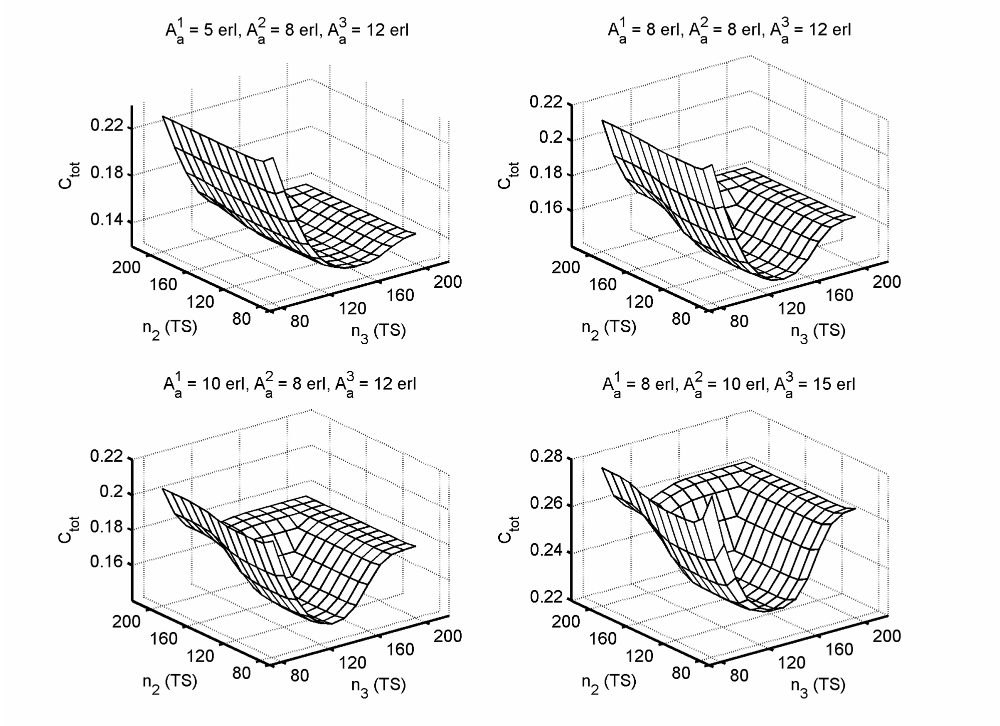

Fig. 2. Overall traffic congestion versus class limitation of Stream 2 and/or Stream 3, over different traffic loads

V. Conclusion

In this paper, we have investigated the performance of an OFDM system with AMC that carries voice traffic. Based on the model of multi-rate loss systems, we have determined the overall traffic congestion in an efficient way by applying the convolution algorithm. We have defined several service classes, depending on the AMC scheme used, related to data transmission with constant bit rate from MSs inside a cell area. It has been shown that the traffic congestion of a call attempt of a particular service class depends on the state of the system, bandwidth required as well as the optimal allocation of available resources by applying a class limitation policy (limited accessibility to common resources in the frame of Stream 2 and Stream 3). The analysis, we have conducted, demonstrates that this approach results in finding the minimal overall traffic congestion, which led to improvement in system throughput.

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