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Published in:
Lecture Notes in Computer Science

Link to article, DOI:
10.1007/978-3-540-89183-3_9

Publication date:
2008

Document Version
Early version, also known as pre-print

Teletraffic Performance Analysis of Multi-class OFDM-TDMA Systems with AMC

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Abstract. In traditional channelized multiple access systems, such as TDMA and FDMA, each user is assigned a fixed amount of bandwidth during the whole connection time, and the teletraffic performance in terms of time congestion, call congestion and traffic congestion can easily be obtained by using the classical Erlang-B formula. However, with the introduction of adaptive modulation and coding (AMC) scheme employed at the physical layer, the allocation of bandwidth to each user is no longer deterministic, but dynamically based on the wireless channel conditions. Thus a new connection attempt will be blocked with a certain probability depending on the state of the system and the bandwidth requirement of the connection attempt. In this paper, we present an integrated analytical model of multi-rate loss system with state-dependent blocking to evaluate the performance of multi-class OFDM-TDMA systems with AMC scheme.

Keywords: OFDM, AMC, performance evaluation, state-dependent blocking.

1 Introduction

Future mobile communication systems will provide not only speech and low-speed data services, but also high-speed data services such as wireless multimedia applications ranging from kilobits to megabits per second. This can be achieved by operating the air interface with Orthogonal Frequency Division Multiplexing (OFDM), which is immune to intersymbol interference and frequency selective fading, as it divides the frequency band into a group of mutually orthogonal subcarriers, each having a much lower bandwidth than the coherence bandwidth of the channel. Recently, OFDM-based systems have become a popular choice for such an endeavor. The IEEE 802.16 standard, for instance, has adopted OFDM-TDMA and OFDMA as two transmission schemes at the 2-11 GHz band.

The economical usefulness of a system is effectively measured by the Erlang capacity, which is generally defined as the maximum traffic load the system can support when the blocking probabilities at the call admission control (CAC)

L. Cerdà-Alabern (Ed.): Wireless and Mobility 2008, LNCS 5122, pp. 102–112, 2008.

level do not exceed a certain thresholds. Many models have been proposed at separate layers, e.g., Rayleigh, Rician and Nakagami fading models at the physical layer [8], and queuing models at the data link layer [6]. In traditional channelized multiple access systems, e.g., TDMA and FDMA, each user is assigned a fixed amount of bandwidth during the whole connection time, and the Erlang capacity can easily be obtained by using the well-known Erlang-B formula. One important assumption for applying the classical Erlang-B formula is that the capacity of each channel is constant and the capacity assignment for each connection is fixed. This is true in wired networks such as the traditional public switched telephone network (PSTN). However, in wireless networks, the channel capacity of a wireless link is time-varying due to multipath fading and Doppler shift. Thus the Erlang-B formula cannot directly be used to calculate the blocking probability. Furthermore, unlike wired networks, even if large bandwidth is allocated to a certain wireless connection, the QoS requirements may not be satisfied when the channel experiences deep fades.

In order to enhance the spectrum efficiency while maintaining a target packet error rate (PER) over wireless links, adaptive modulation and coding (AMC) scheme has been widely adopted to match the transmission rate to time-varying channel conditions. With AMC, the allocation of bandwidth to each user is no longer deterministic (i.e., a fixed amount of bandwidth), but depends on the channel conditions in a dynamic way. Outage is defined to occur when the total number of time slots required by the admitted users exceeds the total number of available time slots. Therefore, a new connection may be blocked with a certain probability depending on the state of the system and the bandwidth requirement of this new connection.

An analytical model to investigate the performance of transmissions over wireless links is developed in [1], where a finite-length queuing is coupled with AMC. However, the authors only concentrate on a single-user case. Reference [2] calculates the Erlang capacity of WiMAX systems with fixed modulation scheme, where two traffic classes, streaming and elastic flows, are considered. In reference [3], the authors evaluate the Erlang capacity of a multi-class TDMA system with AMC by separating the calculation of blocking and outage probabilities. Performance analysis of OFDM systems has so far been conducted primarily by simulations. An analytical framework to evaluate the teletraffic performance in terms of time congestion, call congestion and traffic congestion of multi-user multi-class OFDM-TDMA systems with AMC scheme is still missing. In this paper to achieve this goal, we propose an integrated analytical model of multi-rate loss system with state-dependent blocking.

The rest of the paper is organized as follows. In Section 2, we introduce the system model, which includes OFDM transmission with AMC and calculation of state dependent blocking probabilities. In Section 3, an analytical model of multi-rate loss system with state-dependent blocking is presented with relevant performance measures. Numerical results are given in Section 4. Finally, conclusions are drawn in Section 5.

2 System Model

We consider an infrastructure-based wireless access network, where connections are established between a base station (BS) and mobile stations (MSs). Several service classes with different data rate requirements are supported in the system. Users from each service class arrive at the cell in a random order. The call admission control (CAC) module decides whether an incoming call should be admitted or not, based on the current state of the system and the bandwidth requirement of the call. We assume that the BS has perfect knowledge of the channel state information (CSI) of each subchannel of each connection. We further assume that each subchannel is frequency flat and that the channel quality remains constant within a frame, but may vary from frame to frame.

2.1 OFDM Transmission with AMC

We consider an OFDM-TDMA system with M subchannels. At the physical layer, the time axis is divided into frames. A frame is further divided into K time slots, each of which may contain one or more OFDM symbols. Users transmit in the assigned time slots over all subchannels. Adaptive modulation and coding scheme is employed to adjust the transmission mode in each subchannel dynamically according to the time-varying channel conditions.

We assume that each subchannel follows a Rayleigh fading. For flat Rayleigh fading channels, the received SNR on subchannel m is a random variable γ_m with probability density function (pdf) [1]:

$$p_\gamma(\gamma_m) = \frac{1}{\overline{\gamma}_m} \exp\Big(-\frac{\gamma_m}{\overline{\gamma}_m}\Big) \tag{1}$$

where $\overline{\gamma}_m$ is the average SNR over subchannel m.

The design objective of AMC is to maximize the data rate by adjusting the transmission parameters according to channel conditions, while maintaining a prescribed packet error rate (PER) P_0. Let N denote the total number of transmission modes available (e.g., $N = 5$). Assuming constant power transmission, we divide the entire SNR range into $N + 1$ non-overlapping consecutive intervals with boundaries denoted as $\{\mathit{\Gamma}_n\}_{n=1}^{N+1}$. Specifically, mode n is chosen when $\gamma_m \in [\mathit{\Gamma}_n, \mathit{\Gamma}_{n+1})$. Therefore, with Rayleigh fading, transmission mode n will be chosen on subchannel m with probability:

$$\mathrm{P_m}(n) = \int_{\mathit{\Gamma}_n}^{\mathit{\Gamma}_{n+1}} p_\gamma(\gamma_m)\, d\gamma_m = \exp\Big(-\frac{\mathit{\Gamma}_n}{\overline{\gamma}_m}\Big) - \exp\Big(-\frac{\mathit{\Gamma}_{n+1}}{\overline{\gamma}_m}\Big) \tag{2}$$

Let $\overline{\mathrm{PER}}_{m,n}$ denote the average PER corresponding to mode n on subchannel m. It can be obtained in closed-form as [1]:

$$\overline{\mathrm{PER}}_{m,n} = \frac{1}{\mathrm{P_m}(n)} \int_{\mathit{\Gamma}_n}^{\mathit{\Gamma}_{n+1}} \alpha_n \, \exp(-g_n\gamma)\, p_\gamma(\gamma)\, d\gamma \tag{3}$$

where α_n, g_n are the mode dependent parameters shown in Table 1. The average PER of AMC can then be computed as the ratio of the average number of packets in error over the total average number of transmitted packets:

$$\overline{\mathrm{PER}} = \frac{\sum_{m=1}^{M}\sum_{n=1}^{N} R_n \, \mathrm{P_m}(n) \, \overline{\mathrm{PER}}_{m,n}}{\sum_{m=1}^{M}\sum_{n=1}^{N} R_n \, \mathrm{P_m}(n)} \tag{4}$$

where R_n is the number of bits carried per symbol in transmission mode n as shown in Table 1.

The algorithm for determining the thresholds $\{\Gamma_n\}_{n=1}^{N+1}$ with the prescribed $\overline{\mathrm{PER}} = P_0$ is described in details in [1].

Table 1. Transmission modes with convolutionally coded modulation [1]

	Mode 1	**Mode 2**	**Mode 3**	**Mode 4**	**Mode 5**
Modulation	BPSK	QPSK	QPSK	16QAM	64QAM
Coding rate	1/2	1/2	3/4	3/4	3/4
R_n (bits/sym)	0.5	1.0	1.5	3.0	4.5
α_n	274.7229	90.2512	67.6181	53.3987	35.3508
g_n	7.9932	3.4998	1.6883	0.3756	0.0900

Let $\mathcal{R}_m$ be a random variable with probability mass function f_m, denoting the number of bits that can be transmitted over subchannel m in one time slot.

$$\begin{aligned} \mathcal{R}_m &\in \{sR_0, sR_1, \cdots, sR_N\} \\ f_m(sR_n) &= \mathbb{P}(\mathcal{R}_m = sR_n) = \mathrm{P_m}(n) \end{aligned} \tag{5}$$

where s is the number of symbols per time slot.

Let random variable $\mathcal{R} = \sum_{m=1}^{M} \mathcal{R}_m$ denote the number of bits that can be transmitted over all subchannels in one time slot. Based on the assumption that the channel quality of each subchannel is independent identically distributed (`i.i.d.`), the probability mass function (pmf) of $\mathcal{R}$, denoted as $f_{\mathcal{R}}$, can be obtained by convolving the pmf of each subchannel f_m as follows:

$$f_{\mathcal{R}} = f_1 \otimes f_2 \cdots \otimes f_M \tag{6}$$

where $a \otimes b$ denotes discrete convolution.

2.2 State Dependent Blocking Probability

Assume the system accommodates L types of service classes, each of which requires a constant bit rate of r_i bits per frame. In multi-class systems, different service classes with different bit rate requirements need different channel bandwidth in terms of time slots. Thus it would be beneficial for teletraffic calculations if we could specify a common channel bandwidth which we may call a *unit channel*. The higher the required accuracy (i.e., bandwidth granularity), the smaller a unit channel we have to specify. Let us define r_{unit} be the constant bit rate of a unit channel and let $d_i = r_i/r_{\text{unit}}$ be number of unit channels needed to establish one connection of service class i. Due to the time-varying nature of wireless channels, the number of time slots occupied by a unit channel can be modeled by a random variable $\mathcal{D}_{\text{unit}} = r_{\text{unit}}/\mathcal{R}$, with probability mass function $f_{\mathcal{D}_{\text{unit}}}$, which can be easily obtained from $f_{\mathcal{R}}$.

In AMC scheme, the modulation and coding rate is chosen according to time-varying channel conditions. As a consequence, the number of time slots allocated to each user is varying on a frame by frame basis. Outage is defined to occur when the total number of time slots required by the admitted users exceeds the total number of available time slots. If outage occurs, the admitted users may not get the prescribed data rate and thus the QoS will be degraded. Suppose that the system is in state x (x unit channels are currently occupied by the admitted users), then the total number of time slots required by the x unit channels can be modeled by the sum of x `i.i.d.` random variables $\mathcal{D}_x = \sum_1^x \mathcal{D}_{\text{unit}}$ with probability mass function $f_{\mathcal{D}_x}$:

$$f_{\mathcal{D}_x} = \underbrace{f_{\mathcal{D}_{\text{unit}}} \otimes f_{\mathcal{D}_{\text{unit}}} \cdots \otimes f_{\mathcal{D}_{\text{unit}}}}_{x \text{ times}} \tag{7}$$

and the outage probability in state x can be calculated as:

$$\begin{aligned} \mathrm{P}_{\text{outage}}(x) &= \mathbb{P}(\mathcal{D}_x > K) \\ &= 1 - \mathbb{P}(\mathcal{D}_x \leq K) \\ &= 1 - \sum_{i \leq K} f_{\mathcal{D}_x}(i) \end{aligned} \tag{8}$$

When a new connection arrives at the system, it will be accepted with a certain probability under the condition that the acceptance of the new connection will keep the system outage probability below a predefined threshold. Therefore, the blocking probability of a new connection is a random variable depending on the state of the system and the bandwidth requirement of the new connection. Specifically, assume that a new single-channel call arrives at the time instance when the system is in state x. The call admission control module first checks the outage probability of current state $\mathrm{P}_{\text{outage}}(x)$. If it is larger than the threshold, the new call is always rejected. If it is smaller than the threshold, the CAC module estimates the outage probability of the next state $\mathrm{P}_{\text{outage}}(x+1)$. If it is smaller than the threshold, the new call is accepted with probability one, otherwise, the new call is accepted with probability $p \in (0, 1)$. The value of p is determined under the condition that the estimated system outage probability $\mathrm{P}_{\text{outage}}(x + p)$ is below the threshold. For example, if a new single-channel call arriving in state x is accepted with probability p, the estimated outage probability can be calculated as: $\mathrm{P}_{\text{outage}}(x + p) = 1 - \sum_{i \leq K} f_{\mathcal{D}_{x+p}}(i)$, where $\mathcal{D}_{x+p} = \mathcal{D}_x + \mathcal{D}_p = \sum_1^x \mathcal{D}_{\text{unit}} + p \cdot \mathcal{D}_{\text{unit}}$ is a random variable with probability mass function $f_{\mathcal{D}_{x+p}} = f_{\mathcal{D}_x} \otimes f_{\mathcal{D}_p}$, where $f_{\mathcal{D}_p}$ can be easily obtained from $f_{\mathcal{D}_{\text{unit}}}$.

Let us define the blocking probability of a new single-channel call in state x as $b_x = 1 - a_x$, where a_x is the acceptance probability in state x, determined as follows:

$$a_x = \begin{cases} 1 & \text{if } \mathrm{P}_{\mathrm{outage}}(x+1) \leq \mathrm{Out}_{\mathrm{Th}} \\ 0 & \text{if } \mathrm{P}_{\mathrm{outage}}(x) > \mathrm{Out}_{\mathrm{Th}} \\ \max\left\{p : \mathrm{P}_{\mathrm{outage}}(x+p) \leq \mathrm{Out}_{\mathrm{Th}}\right\} & \text{else} \end{cases} \tag{9}$$

where $\mathrm{Out}_{\mathrm{Th}}$ is the predefined outage probability threshold.

For a d-channel call, we have to choose the acceptance probability in state x as a function of the number of unit channels currently occupied and the bandwidth request as follows [5]:

$$\begin{aligned} a_{x,d} = 1 - b_{x,d} &= \prod_{j=x}^{x+d-1} (1 - b_j) \\ &= (1 - b_x)(1 - b_{x+1}) \cdots (1 - b_{x+d-1}) \end{aligned} \tag{10}$$

Notice that $b_x = b_{x,1}$. This corresponds to that a d-channel call chooses one unit channel d times, and it is accepted only if all d unit channels are successfully obtained. This is a quite natural requirement as we assume full accessibility. In the next section, we will see that this is a necessary and sufficient condition for maintaining the reversibility of the process.

3 Analytical Model

We evaluate the performance of the above mentioned multi-class OFDM-TDMA systems with AMC scheme by using the classical teletraffic model of multi-rate loss system with state-dependent blocking.

3.1 Traffic Model

We use the *BPP* (Binomial, Poisson & Pascal) traffic model in our analysis [6]. This model is insensitive to the service time distributions, thus it is very robust for applications. Each traffic stream i is characterized by the offered traffic A_i, the peakedness Z_i and the number of unit channels d_i needed for establishing one connection. The offered traffic A_i is usually defined as the average number of connection attempts per mean holding time. Peakedness Z_i is the variance/mean ratio of the state probabilities of a traffic stream when the system capacity is infinite, and it characterizes the arrival process. For $Z_i = 1$, we have a Poisson arrival process, whereas for $Z_i < 1$, we have a finite number of users and more smooth traffic (Engset case). Engset traffic can alternatively be characterized by the number of traffic sources S and the offered traffic per idle source β. We have the following relations between the two presentations [4]:

$$\begin{aligned} A &= S \cdot \frac{\beta}{1+\beta} & \qquad Z &= \frac{1}{1+\beta} \\ \beta &= \frac{1-Z}{Z} & \qquad S &= \frac{A}{1-Z} \end{aligned} \tag{11}$$

For $Z_i > 1$, the model corresponds to a more bursty arrival process, called Pascal traffic.

3.2 Algorithms for Calculating Global State Probabilities

The call-level characteristics of multi-class OFDM-TDMA systems with AMC described in Section 2 can be modeled by a multi-dimensional Continuous Time Markov Chain (CTMC). As an example, we consider a system supporting two traffic streams with different data rates. The arrival process of both streams follows a Poisson process with rate λ_1 and λ_2 respectively, and the service times are exponentially distributed with intensity μ_1 and μ_2 respectively. A call of stream one requires one unit channel and a call of stream two requires two unit channels. Fig. 1 shows the state transition diagram for a system with limited accessibility, where state (i, j) denotes the state of the system (i.e., i and j are the number of unit channels occupied by stream one and two respectively), and $1 - b_{x,d} = \prod_{j=x}^{x+d-1}(1 - b_j)$ is the state dependent acceptance probability derived above. From the figure, we can see that the diagram is reversible as the flow clockwise is equal to the flow counter-clockwise (Kolmogorov's criteria), but there is no product form. Due to reversibility, we can apply the local balance equations to calculate the relative state probabilities, all expressed with reference to state $(0, 0)$, then normalize the relative state probabilities to obtain the absolute state probabilities and the relevant performance measures.

Delbrouck [7] developed a general algorithm for calculating the global state probabilities for multi-rate loss systems with BPP-traffic, which is insensitive to

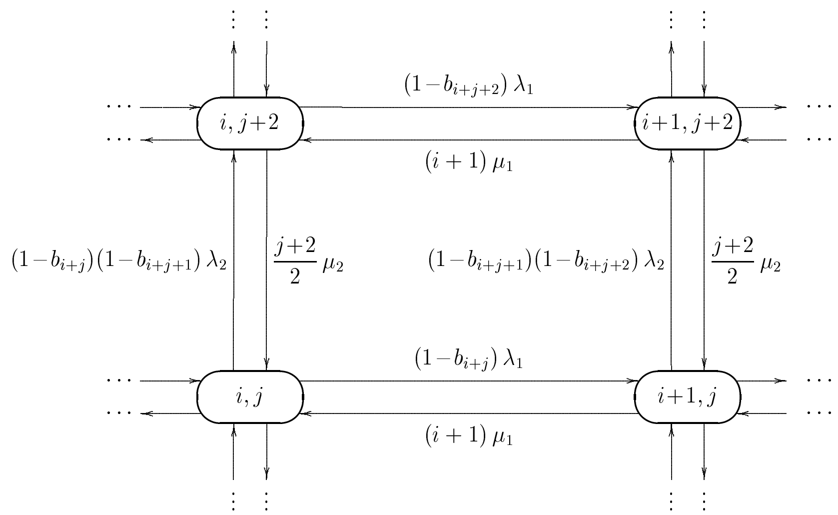

Fig. 1. State-transition diagram with state-dependent blocking probabilities for a multi-rate loss system. The process is reversible as the flow clockwise equals the flow counter-clockwise [5].

service time distribution, i.e., the state probabilities of the system only depend on the holding time distribution through its mean value. Reference [5] extended the Delbrouck's algorithm to allow for evaluating individual performance measures for each service and include state-dependent blocking as shown in Fig. 1. If we consider a system with C unit channels and L traffic streams, the relative global state probabilities $q(x)$ for multi-rate loss systems with state-dependent blocking can be calculated by a generalized recursion formula expressed as follows [5]:

$$q(x) = \begin{cases} 0 & x < 0 \\ 1 & x = 0 \\ \sum_{i=1}^{L} q_i(x) & x = 1, 2, \cdots, C \end{cases} \tag{12}$$

where

$$q_i(x) = \left\{ \frac{d_i}{x} \cdot \frac{A_i}{Z_i} \cdot q(x - d_i) - \frac{x - d_i}{x} \cdot \frac{1 - Z_i}{Z_i} \cdot q_i(x - d_i) \right\} \cdot (1 - b_{x-d_i,d_i}) \tag{13}$$

In the above equations, $q_i(x)$ is the contribution from traffic stream i to the global state $q(x)$. The initialization values of $q_i(x)$ are $\{q_i(x) = 0, x < d_i\}$. The absolute global state probabilities $p(x)$ and $p_i(x)$ can be obtained after normalization.

$$\begin{aligned} p(x) &= \frac{q(x)}{\sum_{j=0}^{C} q(j)} \qquad 0 \le x \le C \\ p_i(x) &= \frac{q_i(x)}{\sum_{j=0}^{C} q(j)} \qquad 1 \le x \le C \end{aligned} \tag{14}$$

A numerically stable algorithm is obtained by normalizing in each step of the iteration [4].

3.3 Performance Measures

Based on the global state probabilities derived above, we are able to get the performance measures of the system in terms of time congestion, call congestion, and traffic congestion.

Time Congestion is by definition equal to the proportion of time the system is blocked for new call attempts. In multi-class OFDM-TDMA systems with AMC scheme, a call attempt of stream i will experience congestion with probability b_{x,d_i} if the system is in state x. Thus the time congestion E_i of stream i is calculated as follows:

$$E_i = \sum_{x=0}^{C} b_{x,d_i} \cdot p(x) \qquad i = 1, 2, \cdots, L \tag{15}$$

Traffic Congestion is by definition equal to the proportion of offered traffic which is blocked. It should be noticed that traffic congestion is the most important performance measure. The carried traffic Y_i of stream i measured in unit channels is given by [5]:

$$Y_i = \sum_{x=1}^{C} x \cdot p_i(x) \qquad i = 1, 2, \cdots, L \tag{16}$$

The offered traffic of stream i measured in unit channels is $d_i \cdot A_i$. Thus the traffic congestion C_i of stream i becomes:

$$C_i = \frac{d_i \cdot A_i - Y_i}{d_i \cdot A_i} \qquad i = 1, 2, \cdots, L \tag{17}$$

Call Congestion is by definition equal to the proportion of call attempts which are blocked. It is said in reference [4] that the call congestion B_i of stream i always can be obtained from the traffic congestion C_i as follows:

$$B_i = \frac{C_i}{Z_i + (1 - Z_i)C_i} \qquad i = 1, 2, \cdots, L \tag{18}$$

where Z_i is the peakedness of stream i.

4 Numerical Results

In this section, we present some numerical results based on the analytical models developed above. We consider the downlink of an OFDM-TDMA system with time division duplex (TDD) operation. The total bandwidth is set to be 5 MHz, which is divided into 5 subchannels with the assumption that the average SNR $\overline{\gamma}_m$ of each subchannel is the same. The duration of a frame is set to be 1 ms so that the channel quality of each connection almost remains constant within a frame, but may vary from frame to frame. The total number of time slots used for downlink data transmission within a frame is set to be 200, each of which contains 4 OFDM symbols. We set the number of transmission modes to 5 with the target PER at 10^{-4}. Three types of service classes with traffic parameters shown in Table 2 are considered. The target outage probability is set to be 2%, which is usually considered to be an acceptable QoS requirement.

The results are shown in Table 3. From the table, we can see that for Poisson traffic ($Z = 1$), the time congestion, call congestion and traffic congestion are identical. This is in accordance with the PASTA property. For Engset ($Z < 1$) and Pascal ($Z > 1$) traffic, there is a difference between the three performance

Table 2. Traffic parameters for the case considered. A is the offered traffic in Erlangs, Z is the peakedness, and d is the bandwidth requirement in unit channels.

Class i	**Offered traffic** $d_i \cdot A_i$	**Peakedness** Z_i	**Req. bit rate** r_i	**Channel bit rate** r_{unit}	**Channels** d_i
1	20	1	50 bits/frame	50 bits/frame	1
2	30	0.5	100 bits/frame	50 bits/frame	2
3	30	2	50 bits/frame	50 bits/frame	1

Table 3. Performance measures for the parameters given in Table 2 with state-dependent blocking

Class	Time cong. E	Call cong. B	Traffic cong. C	Carried Traffic Y
1	0.0557	0.0557	0.0557	18.8854
2	0.1132	0.1040	0.0549	28.3544
3	0.0557	0.0604	0.1139	26.5824

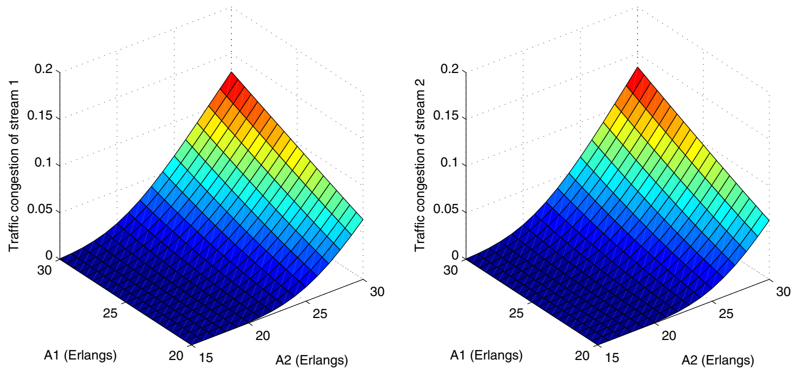

Fig. 2. Traffic congestion versus different traffic loads

measures. From traffic engineering point of view, the traffic congestion is the most important one.

Next, we analyze how different traffic characteristics and various traffic loads will affect the performance metrics. The first two traffic classes in Table 3 are considered, which correspond to voice and data traffic respectively. Fig. 2 shows the traffic congestion versus different traffic loads in Erlangs of the two classes. From the figure, We can see that the increase of the traffic load in class two will increase the traffic congestion in both classes more sharply than class one does. This is because class two requires higher data rate than class one, thus occupies more system resources in terms of time slots.

5 Conclusions

Adaptive modulation and coding (AMC) has been widely used to match transmission parameters to time-varying channel conditions. In this paper, we have developed an analytical model of multi-rate loss system with state-dependent blocking to evaluate the teletraffic performance of multi-class OFDM-TDMA systems with AMC scheme. With state-dependent blocking, the process is still reversible, but the product-form is lost. It has been shown that a d-channel call has the same blocking probability as d consecutive single-channel calls. So the blocking probability of a call attempt depends both on the state of the system and the bandwidth required. Due to reversibility, we have local balance and may calculate the global state probabilities by an effective algorithm. The performance measures in terms of time congestion, call congestion and traffic congestion are derived with numerical examples.

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