Extended Erlang-B law for performance evaluation of radio resources sharing in GSM/(E)GPRS Networks

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***Abstract*— This paper is focused on the performance analysis of a particular cell where a commonly implemented resource sharing scheme, namely Partial Partitioning (PP), is employed to distribute downlink radio channels among GSM and (E)GPRS traffics. The aim of this work is to derive simple and accurate Erlang-like law that could be used to dimension partitions sizes.**

I. INTRODUCTION

Over the last years, (E)GPRS networks have been successfully deployed in a vast number of networks worldwide. The forthcoming generalizations of multimedia mobiles with advanced capabilities, along with the deployment of 3G networks, are expected to trigger an exponential soaring of packet-based data traffic. Now as UMTS is becoming a commercial reality, GSM/(E)GPRS technology is still expected to play a key role in its successful deployment. As a fact, well dimensioning of GSM/(E)GPRS networks is still the central concern of wireless operators.

In order to maximize data throughput, the Radio Resource Manager (RRM) usually seeks to allocate the largest possible number of available time-slots to each (E)GPRS mobile. This number depends on how the cells Radio Resource Manager (RRM) manages the total cell capacity in order to serve simultaneously GSM and (E)GPRS traffics with respect to some performance criteria. To achieve this goal, the RRM has to employ a particular resources sharing scheme.

In the literature, one could distinguish three main static resources sharing schemes [6]. In the first one, called Complete Sharing (CS), all radio channels are accessible for both data and voice traffics, whereas in the second scheme, often known as Complete Partitioning (CP), time-slots are divided into two sets and each type of traffic is allowed to use only its dedicated set (i.e. reserved channels). These two strategies could be mixed to form a third hybrid scheme where a set of channels is shared between voice and data traffics, whereas the remaining time-slots are partitioned into two sets, each one being reserved for strict usage of its dedicated type of traffic. This scheme is often referred in literature as Partial Partitioning (PP).

PP combines the advantages of both CS and CP schemes. First, reserving a set of time-slots for each type of traffic intend to guarantee, as in CP, at least a minimum QoS level for data and voice traffics. Second, since CP is not suitable for maximizing the radio utilization ratio, especially, in the case of highly varying demands, then thanks to the shared partition, PP could performs a better radio efficiency. Because of these advantages, PP is widely implemented in a number of actually operating GSM/(E)GPRS networks. The only difficulty it induces relies on the correct partitions' size dimensioning.

In this paper, we are focusing on the downlink radio interface of a cell where resources are shared between GSM and (E)GPRS demands according to the PP scheme. The goal of this work is to derive the average system performance parameters using formulas that have the simplicity of Erlangs laws but which apply to voice and data traffics. By deriving these close-form expressions, our aim is to provide an easy and effective tool for the correct dimensioning of GSM/(E)GPRS systems (i.e. PP partitions size).

The rest of the paper is organized as follows: after a discussion on related works in Section II, we present in Section III the GSM/(E)GPRS system and the basic modeling assumptions. The system performance is derived through a continuous-time bi-dimensional Markov chain model described in Section IV. The approximation of this model by an equivalent conditional product-form model that avoids any numerical resolution is investigated in Section V. Validation of these models is discussed in Section VI. Finally, summary of this work is provided in Section VII.

II. RELATED WORKS

Dimensioning the radio interface of a GSM/(E)GPRS system has typically been performed using simulations (see e.g. [8], [9], [14], [17]). Performance statistics with high degree of accuracy could be obtained after that the detailed behavior of the system is coded and simulated. However, as a counter part, one data point could take hours of computing. This prohibitive time consuming, along with the need to gain insight on the comprehension of the GSM/(E)GPRS system, have motivated many research works to provide mathematical models for GSM/(E)GPRS radio engineering.

Most existing mathematical models belong to two classes corresponding to different modeling assumptions on (E)GPRS data traffic. In the first class, the input traffic is supposed to be independent of the actual load of the cell. MMPP [6], [7], [11] or MMAP [18] are often used as an input process for the RLC/MAC queuing model or to reproduce the aggregated traffic originated form several data sources. These models rely on a number of relevant hypothesis which holds when focusing on an (E)GPRS core network link or when studying the uplink radio interface. We believe however that some of these modeling assumptions are not realistic for the downlink, considered as the main bottleneck (because of packet-based data traffic asymmetry). More rational for the downlink modeling are the models of the second class [2]–[5], [12], [15]. They are in conformity with the Internet traffic model defined by the 3GPP [10]. In this models class, each active (E)GPRS session is modeled as an alternating sequence of transfer period (ON) and reading times (OFF). However, contrarily to the first class of models, input traffic is dependent on the actual load of the cell. Precisely, each reading period can start only after the end of the preceding transfer period which has the particularity of being dependant: (1) on the data size and (2) on the number of ongoing transfers. Most of the time, the data size within the ON period is described by a geometrically distributed random variable, while the OFF period is traditionally assumed to be exponentially distributed.

Note that by considering CP scheme many of these works have addressed GSM and (E)GPRS separately [15], [18]. Others have considered CS or PP, but with assuming a preemptive priority to voice calls [7], [15]. The resulting models could leads to product form solutions, in particular when (E)GPRS input traffic is supposed to be independent of the actual load of the cell. In this case, the simple M/M/C/C model proposed by Erlang is often used to derive voice performance parameters. Erlang formula may also be employed to obtain (E)GPRS performance parameters [13]. But most of times, numerical resolution of multi-dimensional continous-time Markov chains are necessary [6], [7], [11], [15].

In summary, none of the previous studies, have fully allowed us to reach our objective, which is: (1) to focus on the downlink radio interface of a cell where resources are shared between GSM and (E)GPRS demands according to the widely implemented PP scheme, (2) to develop an analytical model which is based on realistic traffic assumptions and that are simple enough to derive accurate Erlang-like close-form expressions that could easily be used for traffic engineering of the GSM/(E)GPRS systems. To achieve this goal, we extend in this paper our works presented in [2]–[5] by integrating GSM voice traffic and considering PP scheme. The original contribution of our work in [2]–[5] is the detailed description of the GPRS/EDGE downlink radio interface in a discrete-time model with a realistic ON/OFF traffic performed by a finite number of users over the cell.

III. SYSTEM MODELING

In this study, we discern only two types of traffics: GSM voice calls and (E)GPRS data flows. Moreover, taking into consideration packet-based data traffic asymmetry, we assume that downlink is the critical resource. Our study is thus focused on the radio allocator which distributes the downlink cells radio channels among GSM voice calls and (E)GPRS data flows according to the PP scheme.

A. (E)GPRS system description

The radio part of an (E)GPRS system is connected to the core network by the Gb interface linking SGSN and PCUSN over Frame Relay. The PCUSN manages radio transmission of data traffics. It fragments the LLC frame into smaller RLC/MAC blocks. The size of each radio block depends on the coding scheme used when transmitting the block. GPRS standard defines 4 Coding Schemes (CS) whereas EDGE allows 9. Regardless of the coding scheme used, every block is segmented into 4 bursts. Each burst needs one time-slot to be transmitted. In the 52 (E)GPRS multi-frames, 12 RLC/MAC blocks could be transmitted. Therefore, every RLC/MAC block has a duration t_B equal to 20 ms.

Each per mobile downlink flow has its own RLC/MAC buffer which in actual implementations is large enough to allow us to consider that there is no buffer dimensioning. The PCUSN distributes radio resources among all active RLC/MAC buffers.

Since (E)GPRS traffics use the same radio interface as GSM calls, radio resources available in the cell have to be shared among GSM and (E)GPRS traffics. These resources consist of T time-slots (or channels) where T depends on the number of TDMAs in the cell.

A GSM voice call needs the assignment of a single time-slot for its entire duration whereas in the (E)GPRS case, each time-slot could be shared by several simultaneous users by assigning different temporary flow identities (TFI) to the mobiles. Up to 32 TFIs could be defined per TDMA. Multiplexing several data flows could be achieved using an operator dependant scheduling algorithm (e.g. Round Robin). Moreover, when radio resources are available, (E)GPRS allows multi-slot assignment per mobile. Depending on its capability, a mobile downloading a data flow could receive up to d time-slots simultaneously. Hence every 20 ms it may receive up to d RLC/MAC blocks. Today, most mobiles have $d = 3$ or 4.

B. Modeling assumptions and notations

In summary, our GPRS/EDGE system is characterized by the following parameters:

- t_B: the system elementary time interval equal to the radio block duration, i.e. $t_B = 20$ ms;
- x_B: the number of data bytes that are transferred during t_B over one time-slot. $\frac{x_B}{t_B}$ is the throughput offered by the RLC/MAC layer to the LLC layer. The value of x_B depends on the radio coding scheme ;
- T: the number of time-slots of the TDMA that are partitioned into a contiguous set of T_V time-slots dedicated to voice calls, a contiguous set of T_D time-slots dedicated to GPRS, and a contiguous set of T_{VD} time-slots shared between voice and data; all these time-slots are supposed to use a single TDMA.

We also make the following assumptions:

- All (E)GPRS mobiles have the same reception capability. They are (d+u), where d is the number of time-slots that can be simultaneously used for the downlink traffic and u is the number of time-slot that can be simultaneously used for the uplink traffic;
- For seek of simplicity we assume, as in many similar studies [15], that the RRM continuously performs a time-slot rearrangement in order to maintain a perfect repacking (i.e. time-slots used by phone calls, and as a consequence by (E)GPRS flows are contiguous);
- Conforming to a number of actual implementations, we suppose that on the lack of an available channel, a voice call will be lost on arrival;
- Since GSM voice calls are still generating the largest amount of revenue we will consider, as in most actually operating GSM/(E)GPRS networks [11], [13], that voice calls have a preemptive priority over data flows on the shared part of the TDMA. In other words, if all the T_V time-slots dedicated to phone calls are occupied and all the T_{VD} shared time-slots are in use with at least one of them allocated to data traffics, then one time-slot assigned to (E)GPRS traffics in the shared part of the TDMA will be reallocated to voice call on the arrival of any GSM request.

Consequently, if we define $T_{max}\left(t\right)$ as the maximum number of time-slots that can be used for (E)GPRS mobiles when there are t voice mobiles in communication, one can easily see that:

$$T_{max}\left(t\right) = T_D + T_{VD} - \max\left(0, t - T_V\right), \, t \leq T_V + T_{VD} \quad (1)$$

C. Traffic model

In this study, we do not distinguish new GSM phone calls from handoff. Then, as in similar case studies [12], we assume that:

- New voice calls arrive according to a Poisson process with rate λ_V;
- Call durations are exponentially distributed with mean $t_{call} = 1/\mu_V$ seconds.

Data traffic is modeled as in [4]. There is a fixed number N of (E)GPRS mobiles in the cell. Each of them is doing an ON/OFF traffic with an infinite number of pages:

- ON periods correspond to the download of an element (a WAP, a WEB page, an email, a file, etc.). Its size is characterized by a discrete random variable X_{on}, with an average value of x_{on} bytes; We then define $\mu_D = \frac{x_B}{x_{on} t_B}$ as the average data rate by time-slot, i.e. the number of data bytes that a time-slot can transmit by unit of time;
- OFF periods correspond to the reading time, which is modeled as a continuous random variable T_{off}, with an average value of $t_{off} = 1/\mu_D$ seconds.

Let $N_{max}\left(t\right)$ be the maximum number of (E)GPRS mobiles that can simultaneous be in active transfer, when there are t voice mobiles in communication. Because of the GPRS system limitations on the signaling capabilities (no more than 7 USFs per uplink time-slot) and by introducing an additional settable parameter m which describes a minimum throughput per mobile if an admission control scheme is used (no more than m mobiles per time-slot), $N_{max}\left(t\right)$ can be derived from $T_{max}\left(t\right)$ as:

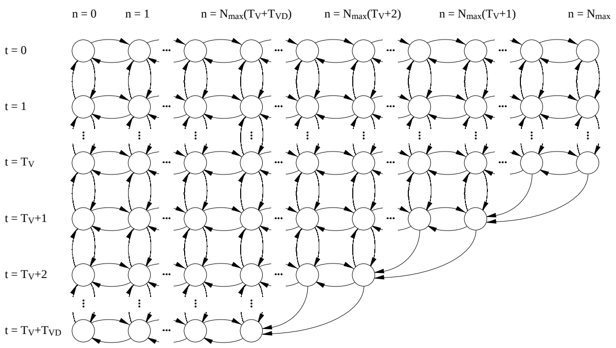

Fig. 1. Bi-dimensional Markov-chain model

$$N_{max}\left(t\right) = \min\left(N, \, 7T_{max}\left(t\right), \, mT_{max}\left(t\right)\right), \, t \leq T_V + T_{VD} \quad (2)$$

We finally define N_{max} as the maximum number of (E)GPRS mobiles that can be in active transfer, whatever the number of voice mobiles in communication. Obviously, this maximum is obtained when there are less than T_V voice mobiles in communication in the cell (i.e. when voices do not occupy the shared part of the TDMA). Thus, $N_{max} = N_{max}\left(t\right)$, $\forall t \in \left[0, T_V\right]$. According to equations (1) and (2) one can see that:

$$N_{max} = \min\left(N, \, 7\left(T_D + T_{VD}\right), \, m\left(T_D + T_{VD}\right)\right) \quad (3)$$

IV. BI-DIMENSIONAL MODEL

A. Markov chain

A direct extension of the Erlang-B model for voice and the Erlang-like model developed in [4] for GPRS/EDGE systems, consists in the bi-dimensional Markov chain illustrated in Fig. 1. A state of this Markov chain is a couple $\left(t, n\right)$ where $t = 0, \ldots, T_V + T_{VD}$ is the number of voice mobiles in communication and $n = 0, \ldots, N_{max}\left(t\right)$ is the number of (E)GPRS mobiles in active transfer. The transitions out of a generic state $\left(t, n\right)$ (with $0 < t < T_V + T_{VD}$ and $0 < n \leq N_{max}\left(t+1\right)$) are given in the left part of Fig. 2. "Vertical" transitions correspond to classical Erlang-B model transitions. "Horizontal" transitions can be considered as the "continuous" extension of the discrete-time model developed in [4]. When the system is in state $\left(t, n\right)$, (E)GPRS mobiles can share $T_{max}\left(t\right)$ time-slots for data transmission. Now, because of the maximum downloading capacity d of each (E)GPRS mobile, if $nd < T_{max}\left(t\right)$, each mobile receives d time-slots and the part of the TDMA associated with data transmission is not fully utilized. The transition rate from state $\left(t, n\right)$ to state $\left(t, n-1\right)$ corresponding to one mobile ending up its transfer, is thus equal to $nd\mu_D$. On the other hand, if $nd \geq T_{max}\left(t\right)$, the allocator has to share the $T_{max}\left(t\right)$ time-slots between the n (E)GPRS mobiles, and the transition rate from state $\left(t, n\right)$ to state $\left(t, n-1\right)$ is thus equal to $T_{max}\left(t\right)\mu_D$. Now,

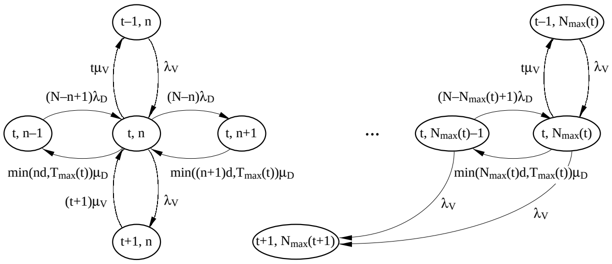

Fig. 2. Transition out a generic state (t,n)

because of the ON/OFF traffic assumptions, when there are n (E)GPRS mobiles in active transfer, $N-n$ are in a reading state. Each of these $N-n$ (E)GPRS mobiles are likely to start a new transfer after an exponential time of rate λ_V. As a consequence, the transition rate from state (t,n) to state $(t,n+1)$ is equal to $(N-n)\,\lambda_V$. It is important to understand that, as voice has preemptive priority over data, if a new voice communication starts when the number of existing voice communications is greater than T_V (and lower than T_V+T_{VD}), this new communication preempts a time-slot that could be in use by (E)GPRS mobiles. As a consequence, the allocator has to rearrange the (E)GPRS mobiles over the remaining time-slots. This is not always possible, and can thus result in (E)GPRS rejections, when the (E)GPRS limiting capacity is reached. At most $\min(m,\,7)$ rejections may occur. These rejections are represented in the Markov chain by transitions going from any state (t,n) where $t \geq T_V$ and $n = N_{max}(t+1)+1,\ldots,N_{max}(t)$ to state $(t+1, N_{max}(t+1))$. This is illustrated in the right part of Fig. 2 that shows transitions out of a limiting state $(t, N_{max}(t))$.

B. Average performance

The Steady-state probabilities $p(t,n)$ of the bi-dimensional continuous-time Markov chain can be obtained using any numerical technique (see [16] for a list of possible methods). From these probabilities the average performance parameters for data and voice users can easily be derived.

1) (E)GPRS performance : First, the mean number $\bar{Q}$ of (E)GPRS mobiles in active transfer can be directly derived from the stationary probabilities as:

$$\overline{Q} = \sum_{t=0}^{T_V+T_{VD}} \sum_{n=1}^{N_{max}(t)} np(t,n) \tag{4}$$

From the Markov chain we can obtain the mean number of (E)GPRS mobiles ending up their data transfers per unit of time as:

$$\overline{X_d} = \sum_{t=0}^{T_V+T_{VD}} \sum_{n=1}^{N_{max}(t)} np(t,n)\min(nd,\,T_{max}(t))\,\mu_D \tag{5}$$

From Little's law, we can thus derive the average duration of ON periods:

$$t_{on} = \frac{\overline{Q}}{\overline{X_d}} \tag{6}$$

Then, we can calculate the average throughput obtained by each mobile in active transfer as:

$$\overline{X} = \frac{x_{on}}{t_{on}} \tag{7}$$

Finally, in order to derive the last important performance parameter, namely the data transfer rejection probability P_r, one needs to note that rejection may happen because of two reasons. First, a new data transfer is rejected whenever the system is in a state (t,n) where $n = N_{max}(t)$. This event occurs with a rate λ_b given by:

$$\lambda_b = \sum_{t=0}^{T_V+T_{VD}} p(t, N_{max}(t))\,(N - N_{max}(t))\,\lambda_D \tag{8}$$

Second, the arrival of a voice call will lead to the interruption of at least one data transfer whenever the state of the system induces the preemption of the allocated time-slot from data users to the phone call. This event occurs with a rate λ_p:

$$\lambda_p = \sum_{t=T_V}^{T_V+T_{VD}-1} \sum_{n=N_{max}(t+1)+1}^{N_{max}(t)} p(t,n)\,(n - N_{max}(t+1))\,\lambda_V \tag{9}$$

We can thus obtain the data transfer rejection probability P_r as:

$$P_r = \frac{\lambda_b + \lambda_p}{\lambda_e} \tag{10}$$

where λ_e denotes the average number of requests for new data transfers per unit of time. We can compute this last quantity as:

$$\lambda_e = \sum_{t=0}^{T_V+T_{VD}} \sum_{n=0}^{N_{max}(t)} p(t,n)\,(N-n)\,\lambda_D \tag{11}$$

2) Voice performance: As voice has a preemptive priority over data, voice performance can obviously be obtained by Erlang-B formulas, with a number of resources equal to $T_V + T_{VD}$.

V. CONDITIONAL PRODUCT-FORM MODEL

A. Markov chain

In order to avoid the complexity of the numerical resolution of a bi-dimensional Markov chain, we have developed an approximation conditional product-form model. The basic idea of this approximation comes from the fact that voice has preemptive priority over data. As mentioned before, voice performance can be derived by classical Erlang-B model with a total number of resources equal to $T_V + T_{VD}$. Now, conditioned by the fact that there are currently t voice mobiles in communication, the performance of (E)GPRS can be obtained by the Erlang-like model developed in [4] with a number of time-slots that can be used by (E)GPRS mobiles equal to $N_{max}(t)$. This decomposition of the original bi-dimensional Markov chain into several one-dimensional Markov chains (birth-death processes) is illustrated in Fig. 3. The transitions out of a generic state t of the classical Erlang-B model are illustrated on the left part of Fig 4. From the birth-death structure of this model, we can easily derive the voice probabilities $p_V(t)$ as:

$$p_V(t) = \frac{\rho_V^t}{t!} p_V(0)\,,\ 0 \leq t \leq T_V + T_{VD} \tag{12}$$

where $\rho_V = \frac{\lambda_V}{\mu_V}$ and $p_V(0)$ is obtained by normalization. Similarly, the transitions out of a generic state $(n|t)$ of the

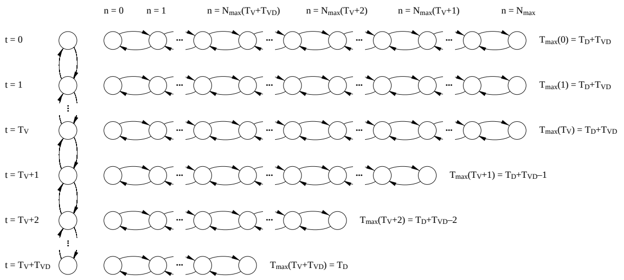

Fig. 3. Conditional product-form model

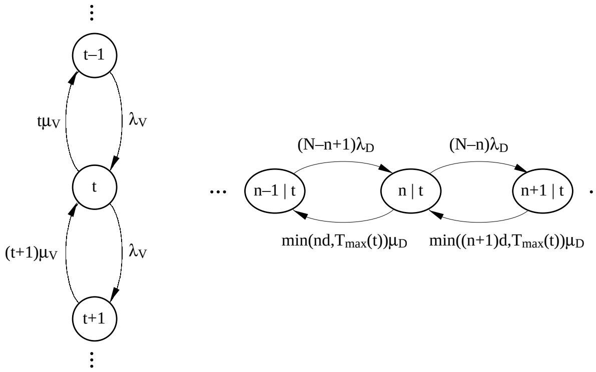

Fig. 4. Transitions out a generics states t and $n|t$

Erlang-like model for (E)GPRS networks are illustrated on the right part of Fig. 4. Transitions out of a generic state $(n|t)$ of any of these linear models are the same as the horizontal transition of the bi-dimensional model developed in Section IV. The steady-state probabilities of (E)GPRS mobiles $\forall n \in [0, N_{max}]$, are thus given by (see [4] for derivations):

$$p_D\left(n|t\right) = \frac{N!\, x^n p_D\left(0|t\right)}{\min\left(d^n,\, d^{n_0(t)} T_{max}\left(t\right)^{n-n_0(t)}\right)(N-n)!} \tag{13}$$

where $x = \dfrac{\lambda_D}{\mu_D} = \dfrac{t_B x_{on}}{x_B t_{off}}$, $n_0\left(t\right) = \left\lfloor \dfrac{T_{max}\left(t\right)}{d} \right\rfloor$ and $p_D\left(0|t\right)$ is obtained by normalization.

B. Average performance

The approximation consists in estimating the detailed probabilities $p\left(t,n\right)$ as the product of the marginal probabilities:

$$p\left(t,n\right) = p_V\left(t\right) p_D\left(n|t\right) \tag{14}$$

Now, we can derive from theses approximation probabilities all the performance parameters of (E)GPRS mobiles by applying relations (4) to (11). Note that by using this conditional product-form approximation, we avoid the numerical resolution of the bi-dimensional Markov chain developed in Section IV. Instead, we obtain instantaneously approximations of the steady-state probabilities by using close-form expressions (12), (13) and (14). Let us finally emphasize that performance of voice mobiles are directly obtained from Erlang-B model without any approximation.

VI. VALIDATION

Our mathematical models were validated against an event driven simulator which we have developed using the CNCL library [1]. Our simulator is based on the same modeling assumptions as those described in Section III.

To compare our mathematical models against the simulator we have considered several scenarios. For a complete validation, we have covered a large set of the system parameters space by varying the value of the model inputs. Because of the lack of space we can give here only a part of the obtained results.

Figures 5 and 6 show respectively the (E)GPRS average throughput and the average (E)GPRS transfer blocking probability obtained by the bi-dimensional model, the conditional product-form model and the simulator for a large range of N values. For each N value, we have considered three GSM load cases: lightly loaded, moderately loaded and heavily loaded. The values of ρ_V associated to these loads are chosen so as to have respectively an average of 0, 1 and 2 time-slots used by the phone calls in the shared partition. Note that in all these comparison scenarios we have taken: GPRS Coding Scheme CS2 ($x_B = 30$ bytes), $T_V = 4$, $T_{VD} = 2$, $T_D = 2$, $t_{off} = 7$ seconds and $x_{on} = 1000$ bytes.

All these curves show: (1) the perfect match between bi-dimensional and conditional product form models and (2) an excellent agreement between the simulator and these mathematical models. Less than 3% error has been observed in the worst case of Fig. 5. This very good agreement between simulation and our mathematical models was obtained for all the test scenarios that we have executed. The highest difference that could be observed between the simulator and the mathematical models is less 5% and could be observed only for the average throughput in the case of small x_{on} values (e.g. less than 1000 bytes) under light load of GSM traffics.

Average throughput and blocking probability obtained by our two mathematical models are almost identical, especially for high values of x_{on} (e.g. $x_{on} > 6000$ bytes) or for heavily or lightly GSM loads. Curves of Fig. 7 show clearly that this observation holds also for moderate GSM loads. In this figure we have compared the bi-dimensional model against the conditional product-form under the case of a moderate GSM load (i.e. $\rho_V = 1.5$). The chosen metric is the average (E)GPRS throughput. The comparison scenario is similar to the one detailed in the previous paragraph, except that we have considered the following x_{on}values: 1KB, 4KB, 46KB and 64KB.

Note that, all these results were obtained instantaneously by the conditional product-form model. When N is large, few seconds were needed by the bi-dimensional model in order to compute the average performance parameters whereas each simulation point took a processing time of several minutes.

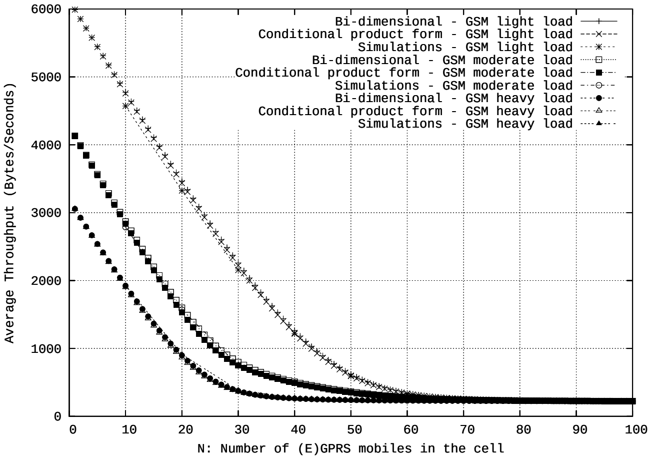

Fig. 5. Average data throughput when varying N and ρ_V

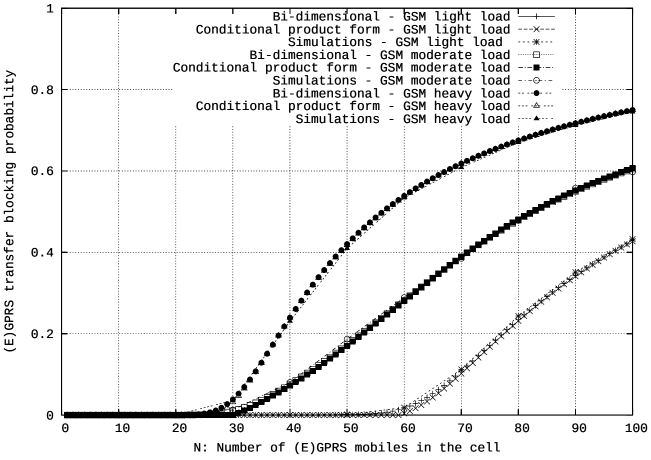

Fig. 6. Data blocking probabilities when varying N and ρ_V

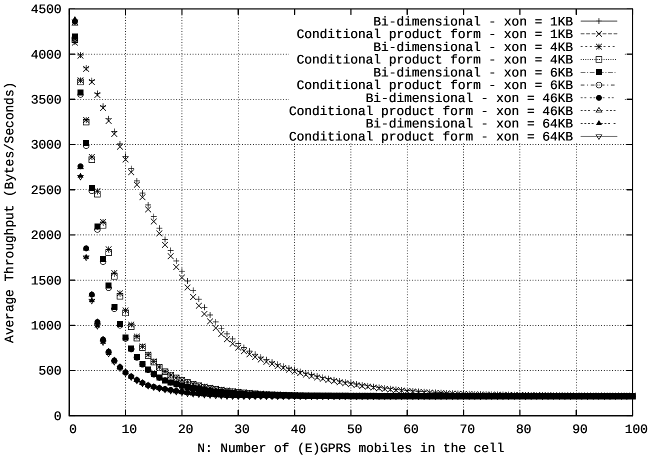

Fig. 7. Average data throughput when varying N and x_{on}

VII. CONCLUSION

In this paper, we have proposed an extension of Erlang-B law which could be used for dimensioning the downlink radio interface of a cell. We have considered the PP scheme, as it is widely implemented in actually operating GSM/(E)GPRS systems. We have successfully derived a conditional product form model, where steady-states probabilities are characterized by Erlang-like formulas. This model was validated against a simulator and compared to a more accurate bi-dimensional model. In all our experiments we have obtained a perfect match between both mathematical models and the simulator. Moreover, the conditional product form model, allowed us to obtain average system performances instantaneously.

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