Dimensioning Rules Regarding Radio Resources in GSM/GPRS Networks

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Abstract: - Many works have studied traffic modeling in GSM/GPRS networks trying to establish different performance parameters needed for dimensioning purposes. An important element of these networks is the radio resources. Sharing resources between different users and different services is a key concept of radio resources dimensioning in GSM/GPRS networks. In this paper we focus on the problem of performance evaluation in GSM/GPRS networks. We consider different resources allocation strategies such as: CP (Complete Partitioning) and PP (Partial Partitioning) and we study the influence of different operational details concerning TS (time-slots) assignment: FR (full rate), HR (half rate) and packing. Finally, we establish dimensioning rules based on traffic evaluation and quality of service level for GSM/GPRS users. For voice users the quality of service is given by the blocking probability calculated according to the proposed packing rule and for GPRS users the individual throughput and the blocking probability represent the performance parameters. We consider also the preemption probability of voice over data users.

Key-Words: - GSM/GPRS, modeling, Erlang-B law, packing, blocking probability, Engset law, throughput

1 Introduction

Due to the Internet popularity a large number of services are available online. These include, World-Wide-Web browsing, E-Mail,... The world wide integration of General Packet Radio Service and Enhanced GPRS into the GSM system raises many problems. For networks operators, equipment vendors and system integrators dimensioning rules have to be developed to plan and estimate the radio capacity needed for the predicted amount of data users when the radio resources are shared between circuit and packet switched services.

It is well known that GSM operators have dimensioned their networks for voice service in terms of offered voice traffic and blocking probability. The reference model for this system is the Erlang-B formula [1], [2]. This formula gives the proportion of calls that are blocked as a simple function of system capacity and voice traffic intensity.

A GSM voice call needs the assignment of a single circuit, also called time-slot (TS*)*, for its entire duration because it is a time-division-multiplexing scheme.

The GPRS network is designed for supporting several types of data traffic such as Wap, Web, E-Mail, etc. Therefore characterizing the GPRS traffic process is a complex task.

Usually the data traffic is characterized by bursts and is application dependant. A communication session may last for an extended period of time with intermittent packet transmissions.

On the other hand in GPRS service each time-slot can be shared between several users by assigning different Temporary Flow-Identities (TFI) to the mobiles. Up to 32 TFI's can be allocated per TDMA. A mobile can identify it's its own blocks and decode them by monitoring the TFI of each radio block. Data flows are multiplexed by a PCU-scheduling algorithm.

This traffic behavior is coupled with time-slot aggregation: a mobile can be allocated up to *d* time-slots simultaneously, depending on its capability.

Another major problem of dimensioning GSM/GPRS networks is the choice of strategy to partition the available cell capacity between traditional GSM and new GPRS services.

The Radio Resources Manager (RRM) is in charge of optimizing the usage of radio resources, based on a specific resource sharing algorithm.

Three main static resources sharing schemes can be distinguished [3]:

- In the first one, all radio channels are shared between voice and data, as shown in Fig.1. It is called Complete Sharing (CS*)*.

TS (Voice + Data)

Fig.1 Time slots allocation with CS scheme

-In the second one, called Complete Partitioning, time-slots are divided into two sets as indicated in Fig. 2 and each type of traffic is allowed to use only its dedicated set.

TS_V	TS_D

Fig. 2 Time-slots allocation in CP scheme

In this case we have the relation:

$$TS = TS_V + TS_{VD} \tag{1}$$

- The third scheme (Fig.3), known as Partial Partitioning, contains the following channel sets: one set shared between voice and data traffic and two sets each one being reserved for strict usage of its dedicated traffic: voice or data.

TS_V	TS_{VD}	TS_D

Fig.3 Time-slots allocation in PP scheme

According to Fig.3 we have the relation:

$$TS = TS_V + TS_{VD} + TS_D \tag{2}$$

This scheme, offers many advantages: first, reserving a set of time-slots for each type of traffic allows guaranteeing, as in CP a minimum QoS for each type of traffic. Second, the PP scheme provides a better efficiency than CP which is not suitable for maximizing radio utilization, especially when dealing with a highly varying demand.

Due to these advantages, the PP resource sharing algorithm is widely implemented in a large number of deployed GSM/GPRS networks.

Several papers have been published on traffic modeling and performance evaluation in GSM/GPRS networks. The major works in this field are based on analytical models using queuing theory and continuous-time Markov chains, and assuming an infinite number of users in the cell [1], [4] - [7].

In [8]-[10] analytical models based on discrete-time Markov chains have been proposed and a single type of traffic is considered. It is assumed to be generated by a finite number of users and modeled by an Erlang-like law.

In [3] Dahmouni et al. present an approach for dimensioning GPRS networks based on the modified Engset model.

Recently published papers address the problem of improving the quality of service and performance parameters for GSM/GPRS systems [11], [12].

In our study performance evaluation is done based on voice and data traffic models presented. For voice traffic we also studied different strategies applied in resources (time-slots) assignment in order to improve the users number that the network can manage. A new strategy for voice calls assignment is proposed and evaluated. For GPRS networks we have introduced some performance parameters and then we have used them for dimensioning purposes.

2 GSM/GPRS System Description

Our paper considers a single cell which supports two types of traffic: GSM voice calls and GPRS data flows.

In traditional circuit-switched GSM networks, on each frequency carrier a 200 kHz bandwidth is shared between 8 voice calls. In the initial specifications for GSM system [13], each voice call is given a circuit, also called time-slot (TS) because it is a Time-Division multiplexing scheme (TDMA). Each voice call needs the assignment of a single time-slot for its entire duration.

GPRS data traffic uses the same radio interface as GSM voice calls, hence, radio resources available in the cell have to be shared among GSM and GPRS traffics. As a reminder, GPRS is a packet switching technology over circuit-switching based GSM system.

In the GPRS technology a mobile station can use several time-slots simultaneously for one application, in order to perform its transmission with a higher throughput. Each time-slot can be shared among several users by assigning different Temporary Flow Identities (TFI) to the mobile phones. Each TFI identifies a GPRS physical connection called Temporary Block Flow (TBF). Up to 32 TFI’s can be allocated per TDMA frame. Data flows are multiplexed by a PCU- based scheduling algorithm. In addition to time-slot partitioning, the GPRS system also allows for time-slot aggregation: for a single mobile user the system can allocate up to d-time-slots simultaneously for downlink and up to u-time-slots simultaneously for uplink, depending on mobile station capability class ($d+u$).

The choice of the number of TBF's that a PDCH can have in uplink and downlink depends on the operator's choice. For example, in the Alcatel-Lucent technology up to 6 uplink and 10 downlink TBF's are allocated per PDCH.

Our study is focused on the radio allocator which distributes the downlink radio channels among voice calls and GPRS data flows.

When modeling our system we consider the following parameters:

- TS : the number of time-slots of the TDMA partitioned into a contiguous set of TS_V time-slots dedicated to voice calls, TS_{VD} time-slots shared between voice and data and TS_D time-slots dedicated to GPRS; time-slots used by data $TS_D + TS_{VD}$ are on a single TDMA which has a total number of 8 time-slots.

- d (resp. u): is the number of time-slots that can be used simultaneously for downlink (resp. uplink) traffic. All GPRS mobiles have the same radio capability, denoted $d+u$.

- The RLC radio block size represents an important parameter for the GPRS system. In the downlink, IP packets are fragmented and encapsulated into LLC frames by the SGSN. The payload size of each radio block depends on the coding schemes, i.e., the applied radio error protection. The GPRS standard defines four Coding Schemes. The corresponding sizes of the RLC block radio are indicated in TABLE 1.

- We have also mentioned the data rate associated with each coding scheme.

TABLE 1

GPRS Coding Schemes	CS-1	CS-2	CS-3	CS-4
RLC block radio (*bytes*)	23	33	39	53
Data rate: $\mu_{GPRS}(kbits/s)$	9.05	13.4	15.6	21.4

- We suppose that on the lack of an available channel, a voice call will be lost on arrival.

- Voice calls have a preemptive priority over data flows on the shared part of the TDMA due to the fact that they generate the largest amount of revenue in most actual operating systems. As a consequence, if all TS_V time-slots dedicated to voice are occupied and all TS_{VD} time-slots are in use with at least one of them allocated to data, then one time-slot assigned to GPRS traffic in the shared part of the TDMA will be reallocated to voice on the arrival of a GSM request.

3 System Model

3.1 Cells with Complete Partitioning Scheme

Complete Partitioning strategy allows two separate sets of time-slots dedicated to voice respectively to data. As a consequence each system can be analyzed separately.

3.1.1 Voice Traffic Model

We apply the classical Markov chain model for voice. It is based on the birth-death structure shown in Fig.4 and on the following assumptions:

-New voice calls arrive according to a Poisson process with rate λ_V .

-Call durations are exponentially distributed with mean μ_V .

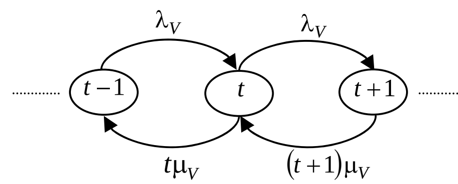

Fig.4 The birth-death model applied to voice traffic process

Base on this model, we have computed the steady-state probabilities given by relation (3) indicating the probability of having t voice calls in active transfer as function of voice traffic intensity $\rho_V = \frac{\lambda_V}{\mu_V}$ and system capacity TS_V :

$$p_V(t) = \frac{\frac{\rho_V^t}{t!}}{\sum_{i=0}^{TS_V} \frac{\rho_V^i}{i!}}, \quad t \in [0, TS_V] \tag{3}$$

It is assumed that for each call a full time slot is given. This strategy is also known as full rate assignment.

Based on relation (3) we have computed the Erlang-B [1] formula that gives the call blocking probability necessary to dimension the cell in order to guarantee a minimum QoS for voice traffic:

$$B_{V,CP} = \frac{\frac{\rho_V^{TS_V}}{TS_V!}}{\sum_{i=0}^{TS_V} \frac{\rho_V^i}{i!}} \tag{4}$$

In order to improve the number of voice users that the network can manage, different strategies could be applied in the time-slot assignment. A common strategy, called half-rate assignment is to share the same time slot between two users. It increases the number of calls in the system in cases when there are few time slots available. The strategy implementation needs to define a threshold *HRTh*. When the number of free time slots is less than *HRTh* the calls are assigned at half rate.

The problem associated with an HR assignment is that it can force the system into a state in which many slots are assigned at half-rate, but each slot to only one user. Because this time-slots are already allocated to voice calls and due to the priority of voice over data they are not available to data. This could produce inefficiencies in the system when the *PP* strategy is implemented.

The solution to avoid this problem is to define a second threshold *PACKTh*. When the number of free time-slots is less then *PACKTh* the system will reallocate the half–rate TS packing isolated calls in pairs sharing the same time slot. The result is a reduced number of slots occupied by voice.

In order to describe this implementation we consider a tri-dimensional Markov process $n(t)=(n_1(t),n_2(t),n_3(t))$, as proposed by [14]. The first parameter $n_1(t)$ represents the number of *TS* assigned in full-rate mode, the second one $n_2(t)$ denotes the number of *TS* occupied by two users in half-rate mode and $n_3(t)$ is the number of *TS* with only one user in half-rate mode.

Three regions can be identified:

Region 1. Is defined by coordinates (i, j, k), having the property:

$$z = i + j + k < TS - HRTh \tag{5}$$

Because the number of free time-slots is greater than *HRTh* the incoming calls are assigned in full rate-mode. Fig.5 shows the possible transitions: three death transition corresponding to the three possible call endings and a birth transition that increases the first coordinate *i*.

Region 2. Includes states which coordinates satisfy the relation:

$$TS - HRTh < z = i + j + k < TS - PACKTh \tag{6}$$

As indicated by Equation (6) the incoming calls are assigned half rate as shown in Fig.6 but there is no packing of half rate calls.

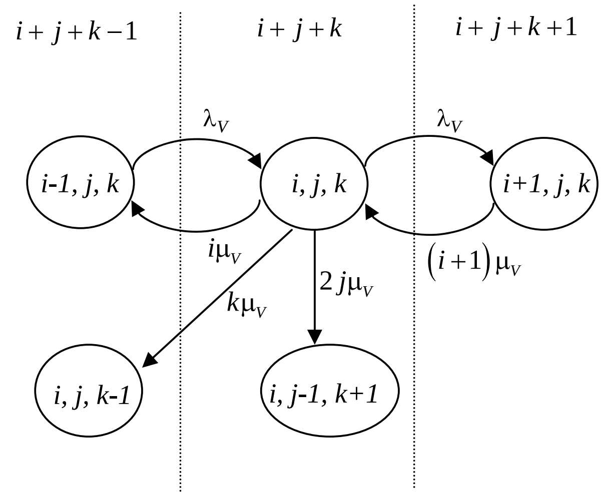

Fig.5 States evolution due to FR assignment of incoming calls

The birth transitions increase the second coordinate when $k > 0$ and the incoming call is combined with an existing one (half-rate assigned) or it increases the third component if $k = 0$ (no isolated half rate- calls).

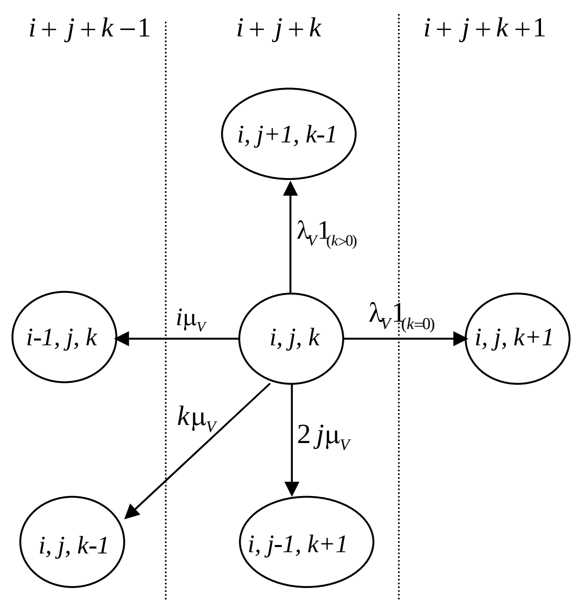

Fig.6 States evolution due to HR assignment of incoming calls

Region 3. Includes states satisfying the relation:

$$TS - PACKTh < z = i + j + k \tag{7}$$

In this case, the system starts packing the half-rate calls if there are two or more slots assigned in half-rate.

Due to this, k can only be equal to 0 or 1. State transitions are shown in Fig.7.

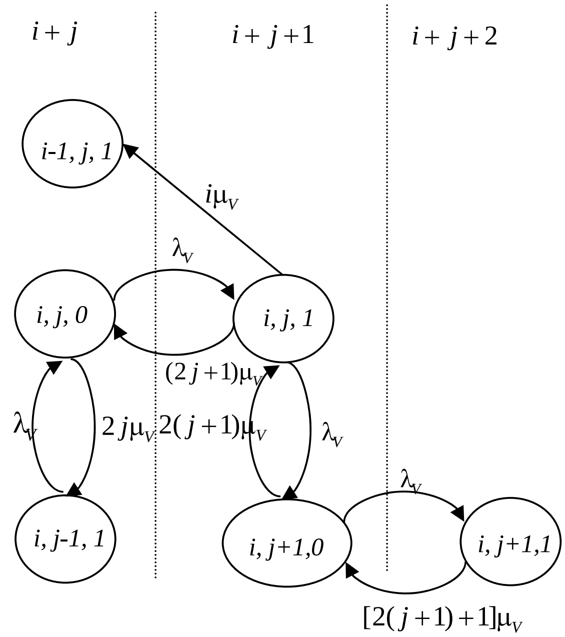

Fig.7 States evolution due to packing HR assigned calls

In the following we propose a scheme to allocate half-rate capable mobiles to full–rate or half-rate channels and packing partially allocated time-slots according to the existing traffic situation in the cell.

3.1.2 Data Traffic Model

Data traffic can be considered at different timescale levels: (i) the packets level where the elementary quantities are carried by the network, (ii) the flow level which is a concept close to the application and (iii) the session level which is a succession of flows that belong to the same application. In our paper we have considered the model at the flow level timescale. A session can be modeled as a series of flows (page download) separated by inactivity periods (think times) with no data transfer. The traffic generated by the users represents an ON/OFF process as represented in Fig.8 [14].

Data traffic is modeled under the assumption that there is a fixed number N of data mobiles in the cell. Each mobile is doing an ON/OFF traffic with an infinite number of pages:

- ON periods correspond to the download of an element like 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 $E[\sigma]$.
- OFF periods correspond to the reading time of the last downloaded element, which is modeled as a random variable T_{off} with an average value of $E[\tau]$ seconds.

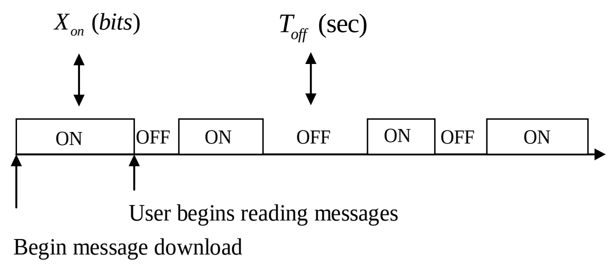

Fig.8 ON/OFF process associated to a session level

- The maximum number of GPRS users in active transfer is given by:

$$n_{\max}(TS_D) = \min(N, 32, mTS_D) \tag{8}$$

m- is the maximum number of users that can use a single time-slot.

The system model is based on the Engset model [2] and includes particular specifications as indicated in Fig.9. This stochastic process describes the number of active users at any point in time and represents a finite state space.

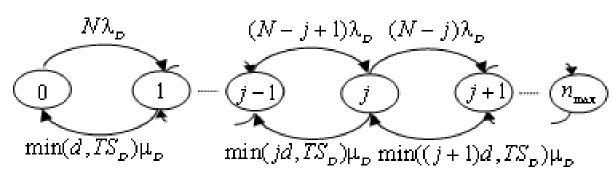

Fig.9 The Engset model applied to data traffic process

It follows from Fig.9 the transition rate from state j to j+1, λ_j :

$$\lambda_j = (N-j)\lambda_D = (N-j)\frac{1}{E[\tau]}, \quad for\ j = 0,1,\ldots n_{\max}-1 \tag{9}$$

We can express the transition rate of death process as:

$$\mu_j = \min(jd, TS_D)\mu_D = \min(jd, TS_D)\frac{\mu_{GPRS}}{E[\sigma]}, \quad for\ j = 1,\ldots,n_{\max} \tag{10}$$

As indicated in Fig.9, the state *j* of the Markov chain corresponds to the number of the data mobiles that are simultaneously in active transfer (i.e., in the *ON* state). The maximum bandwidth capacity they can use is TS_D. Because of the maximum downloading capacity *d* of each GPRS mobile, two situations can be distinguished:

(1) If $jd < TS_D$, the available bandwidth is not fully utilized by data mobiles. As a consequence the transition rate from state *j* to state $j-1$, given by the generated transfer of one mobile, is $jd\frac{\mu_{GPRS}}{E[\sigma]}$;

(2) If $jd \geq TS_D$ the allocator has to share the TS_D time-slots among the *j* data mobiles and the transition rate from state *j* to state $j-1$ is $TS_D\frac{\mu_{GPRS}}{E[\sigma]}$;

Let $p_D(j)$ be the steady-state probability that *j* users are in active transfer. According to the Engset model it modeled by the closed form below:

$$p_D(j) = p_D(0)\frac{C_N^j}{\prod_{i=1}^{j}\min(d, \frac{TS_D}{i})\frac{i\mu_{GPRS}}{E[\sigma]}}\frac{1}{E[\tau]^j} \tag{11}$$

We can express the steady-state probability in terms of data traffic ρ_D, defined by relation (12):

$$\rho_D = \frac{E[\sigma]}{E[\tau]}\frac{1}{\mu_{GPRS}} \tag{12}$$

$$p_D(j) = p_D(0)\frac{C_N^j}{\prod_{i=1}^{j}\min(d, \frac{TS_D}{i})}\rho_D^j \tag{13}$$

We can observe that the steady-state distribution depends only through the ratio $\frac{E[\sigma]}{E[\tau]}$ on data traffic parameters $E[\sigma]$ and $E[\tau]$.

Based on this distribution we have computed the average performance of the system. We consider the following performance measures: the average downlink throughput per user (X_u), the average downlink total throughput (X) and the blocking probabilities (B) for data are considered. All these parameters are function of the load ρ_D, the available cell capacity $TS_D;TS_{VD}$, the user capability *d* and the total number *N* of users.

The average total throughput is determined using the expression bellow:

$$X_{CP} = \sum_{j=1}^{n_{\max}} p_D(j)\,jr(j) \tag{14}$$

where *r*(*j*) represents the effective bandwidth received by each user:

$$r(j) = \min(d, \frac{TS_D}{j})\mu_{GPRS},\ for\quad j = 1,\ldots,n_{\max} \tag{15}$$

From formula (14) we can derive the average throughput per user as:

$$X_{u/CP} = \frac{X_{CP}}{E[j]} = \frac{\sum_{j=1}^{n_{\max}} p(j)\min(jd, TS_D)}{\sum_{j=1}^{n_{\max}} jp(j)}\mu_{GPRS} \tag{16}$$

The data blocking probability can be expressed based on Engset model as follows:

$$B_{CP} = p(0)\frac{C_{N-1}^{n_{\max}}}{\prod_{i=1}^{n_{\max}}\min(d, TS_D / i)}\rho_D^{n_{\max}} \tag{17}$$

and represents the probability that TS_D time-slots are being used by $n_{\max}$ users among the other $(N-1)$ users.

3.2 Cells with Partial Partitioning Strategy

In this case, the available time slots (TS) of the TDMA are partitioned into TS_V time-slots dedicated to voice, TS_D time-slots dedicated to data and TS_{VD} time slots shared between voice and data with a total preemptive priority of voice over data on the shared part as mentioned before in Section 1.

3.2.1 Classical voice traffic model

The voice traffic process is modeled as mentioned before by relation (3). Due to the fact that voice has a preemptive priority over data, the blocking probability for voice can be obtained by Erlang-B formula (because data traffic is transparent to voice), with a number of resources equal to $TS_V + TS_{VD}$:

$$B_{V,PP} = \frac{\dfrac{\rho_V^{(TS_V + TS_{VD})}}{(TS_V + TS_{VD})!}}{\sum_{i=0}^{TS_V + TS_{VD}} \dfrac{\rho_V^i}{i!}} \quad (18)$$

3.2.2 Combined FR and HR voice traffic model – the proposed implementation

The model applied to this system has to deal with two traffic processes: voice and data, sharing the same air interface and using the same physical channels.

In our previous work [16] we have implemented the bi-dimensional model proposed in [8] and have computed the blocking probability formulas according to the PP strategy.

In this paper we apply the modified Engset model proposed in [3] to analyze the GPRS performances and to establish dimensioning rules. The basic idea in constructing the model relies on two assumptions: the voice calls are independent of GPRS connections and the voice and data traffic evolve at different time scales. The time required to transfer data is around several seconds and should be shorter than the mean call duration which is about several minutes. As a consequence between two variations of the number of voice calls, the number of data transfers reaches its stationary regime.

The proposed scheme allocates half-rate capable mobiles to full – rate or half-rate channels according to the existing traffic situation in the cell. If the number of time slots (traffic channels) in the cell is above a predefined threshold, half–rate capable mobiles are allocated to full–rate channels. Otherwise half-rate capable mobiles will be given a half-rate time-slot.

The transitions between the possible states are illustrated in Fig.10. Once a mobile has been allocated to a full/half rate time-slot, the mobile will operate in this mode until the call is terminated.

The procedure drawback is the creation of so-called partially allocated time-slots which are time slots occupied by only one half-rate call. To make an optimal uses of resources and to avoid the rejection of a call from a mobile that has only full rate capabilities, a re-packing procedure is applied.

In this case, it is necessary to repack two time-slots, each occupied by only one half-rate call, into a single time-slot with two half-rate calls.

For voice calls in the classical voice traffic model, the steady-state probabilities are given by Equation (3) with the maximum number of available resources equal to: $TS - TS_D$. Among the TS_{VD} time slots, those not used by the voice calls may be used for data traffic with a probability equal to the probability that $TS - TS_D - s$ are used by GSM users: $p_V(TS - TS_D - s)$.

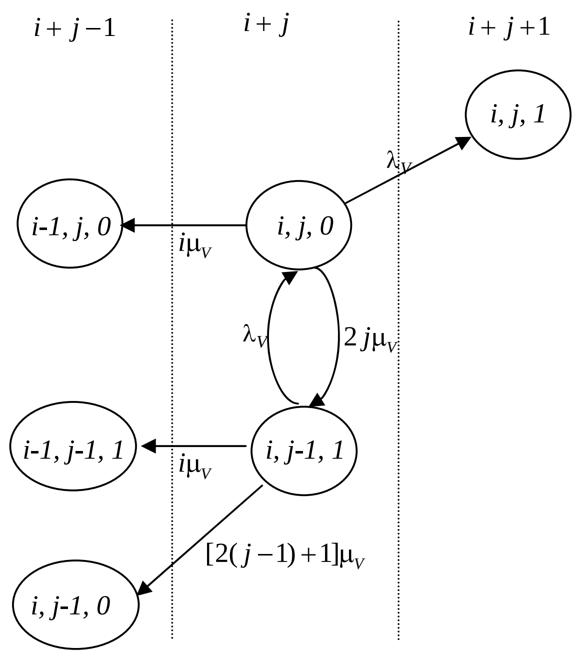

Fig.10 States evolution according to the proposed packing algorithm

For the proposed repacking model it is not possible to find out an analytic expression for the stationary distributions of the process. Having all the sates of the system, the possible transition between these states and their transition rates, it is possible to construct the ***Q***-matrix of the process and find the stationary distribution solving numerically the equation $\mathbf{P}_V\mathbf{Q} = 0$ with the normalization condition $\sum_n P_V(n) = 1$. We denoted by $P_V(n)$ the probability of the current state of our model, where n depends on tree coordinates: $n(t) = (n_1(t), n_2(t), n_3(t))$.

Based on $P_V(n)$, it is possible to compute the steady-states probabilities $p_V^*(s)$, where s represents the number of time-slots occupied by voice calls in the system, by summing $P_V(n)$ over all the states included in the macrostate *s*.

$$p_V^*(s) = \sum_{\{i,j,k\}|i+j+k=s} P_V(i,j,k) \tag{19}$$

The data traffic model is constructed with respect to the Engset model and tacking into account the characteristics imposed by the resources sharing capability as presented above.

The total number of available resources is given by:

$$n_{\max} = \min(N, 32, mTS_D^*) \tag{20}$$

We denoted by TS_D^* the available cell capacity for data traffic.

The average performance parameters are determined as follows:

- the average throughput per user can be expressed as [3]:

$$X_{u/PP} = \sum_{s=0}^{TS-TS_D} p_V(s) X_{u/CP}(\min(TS-TS_V, TS-s)) \tag{21}$$

- for the total average throughput we propose the formula:

$$X_{PP} = \sum_{s=0}^{TS-TS_D} p_V(s) X_{CP}(\min(TS-TS_V, TS-s)) \tag{22}$$

- the data blocking probability is computed similarly as presented in [3]:

$$B_{PP} = \sum_{s=0}^{TS-TS_D} p_V(s) B_{CP}(\min(TS-TS_V, TS-s) \tag{23}$$

- finally we propose a formula for estimating the preemption probability due to voice calls priority over data users:

$$B_{P,PP} = \sum_{s=TS_V}^{TS-TS_D-1} p_V(s) B_{CP}(\min(TS-TS_V, TS-s)) \tag{24}$$

We compute the average performance parameters for the repacking model using the same Equations (21) - (24) and replacing $p_V(s)$ with the steady-state probabilities $p_V^*(s)$ calculated using Equation (19).

4 Experiments for systems performances evaluation

We have implemented the models by simple programs written in Matlab and various scenarios were experimented. For the data traffic we have adopted the following parameters: $E[\sigma] = 5KB$, $E[\tau] = 12s$, GPRS mobile class: 4+1 and CS2 coding scheme ($\mu_{GPRS} = 13{,}4kbits/s$). In the first scenario we have considered a cell with CP strategy, equipped with a single TRX that provides: $TS = 8$, $TS_V = 7$, and $TS_D = 1$.

In the second scenario we have considered a cell with PP strategy, equipped with a single TRX that provides: $TS = 8$, $TS_V = 3$, and $TS_D = 1$. We have considered three loading situations, according to different offered voice traffic values: $\rho_V = 0.602$, $\rho_V = 2.94$ and $\rho_V = 20$. The first two values correspond to the different occupancies of the time-slots that can be used by voice: 3 and 7, considering in that case a 2% for the lost voice traffic according to Erlang-B formula. The third traffic value corresponds to 7 voice TS occupancy but with a blocking probability closed to 90%.

In Fig. 11 we have plotted the total average throughputs X_{CP} and X_{PP} , the average throughputs per user $X_{u,CP}$ and $X_{u,PP}$ as well as the blocking probabilities B_{CP} and B_{PP} .

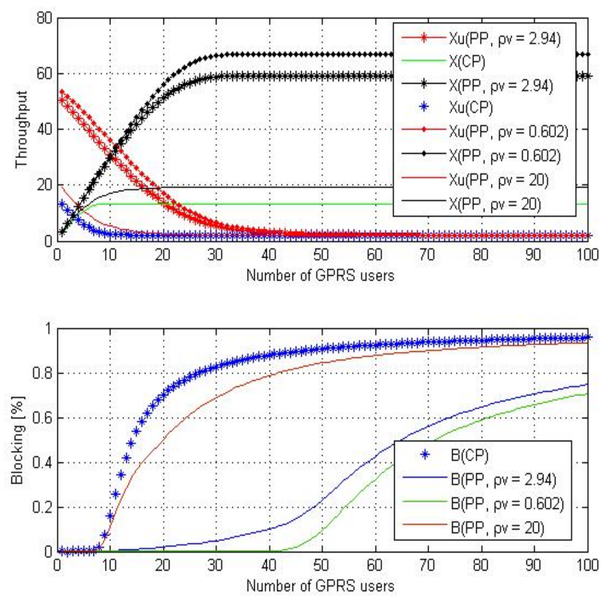

Fig.11 Performance parameters considering the classical voice model

The *PP* scheme performs better regarding the throughput and the blocking probability comparing with *CP* scheme.

The same scenario as those considered before for PP scheme was used for evaluating the performance parameters of the proposed model with repacking. We have considered two loading situations, according to different offered voice traffic values: $\rho_V = 5$ and $\rho_V = 10$ Erlang. For the throughput per user the considered voice traffic values are: $\rho_V = 2$ and $\rho_V = 10$ Erlang.

The half-rate threshold was set at 4. We have build the corresponding $\mathbf{Q}$ matrix of the model, we have solved the equation $\mathbf{P}_V\mathbf{Q} = 0$ and finally we have computed the steady-states probabilities $p_V^*(s)$. The performance parameters: throughput per user, total throughput and blocking probability, computed according to the Equations (21), (22) and (23) are shown in Fig.12, Fig.13 and Fig.14 respectively.

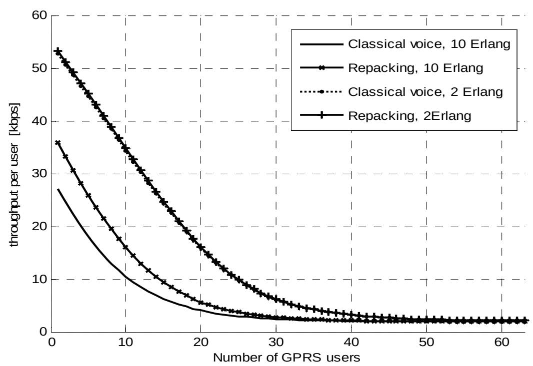

Fig.12 Performance parameter: throughput per user considering the packing voice model

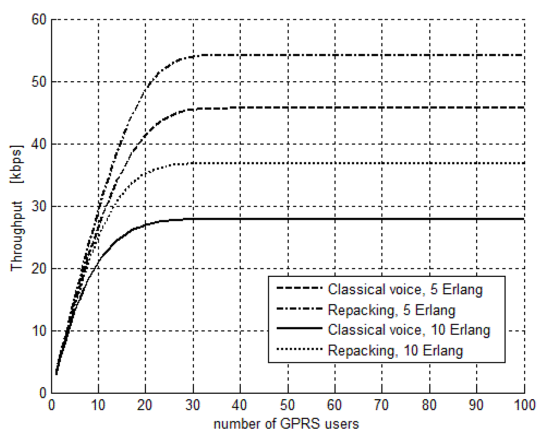

Fig.13 Performance parameter: total throughput considering the packing voice model

As can be observed in Fig.12, Fig.13 and Fig.14, for small voice traffic values, the performance parameters of the two models are comparable, meanwhile for high voice traffic values, the performance parameters of the packing voice model are superior to classical voice model.

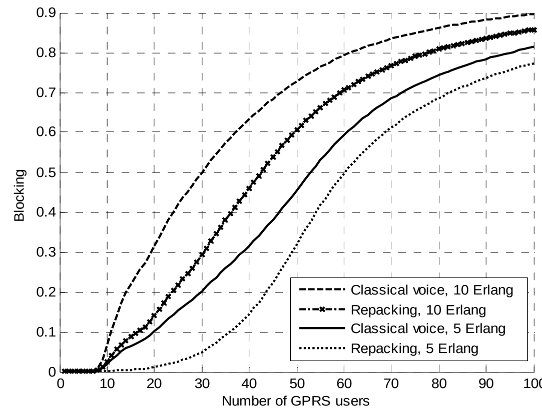

Fig.14 Performance parameter: blocking probability considering the packing voice model

The typical network dimensioning problem is to compute how many time-slots to allocate for data traffic. Currently used dimensioning rules need to estimate the number of active users per cell [17]. Our dimensioning method avoids this estimation using dimensioning criteria based on the throughput per active user combined with blocking probabilities in an interactive algorithm. Blocking probabilities in commercial GPRS networks must be small enough in order to avoid interruption caused by TBF rejection. As we can see in Fig.11 for a reasonable number of users in the cell the minimum accepted throughput is reached, that means that the limiting factor for dimensioning is the throughput rather then the blocking probability.

To illustrate the proposed rule, in Fig.15 we have plotted as a detail of Fig.11 the throughput per user and the blocking probability as function of number of GPRS active users in the cell, for two different total time-slots values.

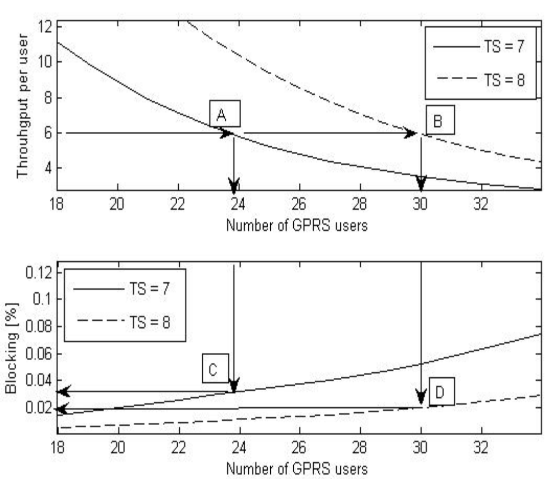

Fig.15 Dimensioning algorithm illustration

The same previous PP scenario was used for dimensioning algorithm illustration, except that the initial value of TS number was set to 7 (3 dedicated for voice, 1 dedicated for data and 3 shared between voice and data).

Point A gives the number of active users that can obtain a desired throughput value. According to this number point C indicates the blocking probability that is reached. If that probability is greater that an imposed value, an increment of the number of time slots gives the new users number (point B) and the corresponding smaller blocking probability (point D).

The algorithm will be repeated until the desired value for blocking probability is reached.

5 Conclusion

In this paper we have presented a new voice traffic model for cell with PP allocation strategy. It is based on a scheme that allocates half-rate capable mobiles to full–rate or half-rate channels and packing partially allocated time-slots according to the existing traffic situation in the cell. We have compared the performance parameters of this model with those of the classical voice model. The results show a better utilization of resources for both voice and data traffic when using the packing model, especially at high voice traffic values.

We have also introduced a dimensioning algorithm for GSM/GPRS networks with PP strategy. The proposed algorithm is based on performance parameters such as the average downlink throughput per user (X_u), the average downlink total throughput (X) and the blocking probabilities (B). All these parameters are function of the cell load (voice and data), the available cell capacity, the user capability d and the total number of users. As quality criteria we have considered the blocking probability, according to Erlang-B law, for voice and the average downlink throughput per user for data.

References:

[1] L. Kleinrock, *Queuing Systems: Vol. I – Theory*, New-York Wiley, 1976

[2] S. Sa Esteves, "Algorithms for Higher-Order Derivatives of Erlang-C Function", *Proceedings of the 13th WSEAS International Conference on COMMUNICATIONS*, Rodos Island, Greece, July 23-25, 2009, ISBN 978-960-474-098-7, ISSN1790-5117, pp. 72-77

[3] H. Dahmouni,. B. Morin, S. Vaton, "Performance Modelling of GSM/GPRS Cells with Different Radio Resource Allocation Strategies", *IEEE Wireless Communications and Networking Conference*, March 2005, Volume 3, pp: 1317- 1322

[4] S. Pedraza, J. Romero, J. Munoz, "(E)GPRS Hardware Dimensioning Rules with Minimum Quality Criteria", *Proc. of IEEE VTC* Spring, pp. 391-395, May 2002

[5] C. Lindemann, A. Thummler, "Performance Analysis of the General Packet Radio Service", *Computer Network*, 41, pp. 1-17, Jan., 2003

[6] S. Ni, S. Haggman, "GPRS Performance Estimation in GSM Circuit-Switched Services and GPRS Shared Resources Systems", *WCNC*, Vol.3, pp.1471-1421, 1999

[7] M. Mahdavi, R. Edwards, P. Ivey, "Performance Evaluation of Data Subsystem in GSM/GPRS Using Complete Sharing, *Proc. of London Communications Symposium*, Univ. College of London, 2001

[8] B. Baynat, K. Boussetta, P. Eisenmann, N. Ben Rached, "Discrete-Time Markov Model for EDGE/GPRS Radio Engineering with Finite-length Sessions", *Int. Symp. On Performance Evaluation of Computer and Telecommunication Systems* (SPECTS'2004), San Jose, California, USA, July, 2004

[9] B. Baynat, K. Boussetta, P. Eisenmann, N. Ben Rached, "Towards an Erlang-like Law for the Performance Evaluation of GPRS/EDGE Networks with Finite-Length Sessions", *Proc. of 3rd IFIP-TC6 Networking Conference*, May 2004, pp. 1288-1293

[10] B. Baynat, P. Eisenmann, "Towards an Erlang-like law for GPRS/EDGE network engineering", *Proceedings IEEE ICC,* June 2004

[11] P.M. Papazoglu, A. Karras, R.C. Papademitriou, "Improved Integral Channel Allocation Algorithms in Cellular Communication Systems Enabling Multimedia QoS Services", *WSEAS TRANSACTIONS on COMMUNICATIONS*, ISSN 1109-2742, Issue 10, Volume 7, October 2008, pp. 1014 -1023

[12] R. Dobrescu, D. Hossu, S. Mocanu, M. Nicolae, "New algorithms for QoS performance improvement in high speed networks**",** ISSN: 1109-2742, *WSEAS TRANSACTIONS on COMMUNICATIONS*, Issue 12, Volume 7, December 2008, pp. 1192 -1201

[13] A. L. Armenta-Vilches & al "Blocking Probabilities in FDMA-TDMA Cellular System",

Proceedings of the 13th WSEAS International Conference on COMMUNICATIONS, Rodos Island, Greece, July 23-25, 2009, ISBN 978-960-474-098-7, ISSN1790-5117, pp. 157-160

[14] Copca, "An improved model for GSM/GPRS/EDGE performance evaluation", ACM/IFIP LANC 2007, San Jose, Costa Rica

[15] B. Baynat, K. Boussetta, P. Eisenmann, „Performance evaluation of GPRS/EDGE networks: A novel discrete - time Markov model", 5th World Wireless Congress (WWC), San Francisco, USA May, 2004

[16] G. Budura, C. Balint, E. Marza, "Blocking Probabilities in GSM/(E)GPRS Cells with Different Radio Resources Allocation Strategies", *Scientific Bulletin of "Politehnica" University of Timisoara*, *ETc series*, Tom 53(67), Fasc.2, 2008, pp. 85-92

[17] P. Stuckmann, O. Paul, "Dimensioning GSM/GPRS Networks for Circuit- and Packet-Switched Services", *Proceedings of the 10th Symposium on Wireless Personal Communications*, Aalborg, Denmark, 2001, pp. 169-174