Dynamic Coding Adaptation in a GPRS Network

Suzana de F. Dantas and Djamel H. Sadok
Federal University of Pernambuco, Recife, PE, Brazil

Abstract – **New TDMA overlay packet switching networks such as the General Packet Radio Service (GPRS) and its enhanced mode (EDGE – Enhanced Data Rate GPRS Evolution) are being deployed for data users. These standards define a number of coding schemes (CS) to be used in accordance with the prevailing radio environment conditions, i.e., interference, noise and packet loss. Firstly, this work presents a smart dynamic CS allocation technique.**

Throughout this study user throughput is analyzed. The results show traffic optimization, a lower packet loss, as well as an increase in system capacity.

Keywords: GPRS, mobile communications, coding schemes, simulator, Internet.

I. INTRODUCTION

Packet technologies such as GPRS and EDGE have been developed for second-generation digital mobile networks based on Time-Division Multiple Access (TDMA). GPRS may be seen as a packet-oriented (IP or X.25) extension of GSM and other TDMA technologies such as the north American TDMA standard IS-136/IS-41. GPRS is embedded in the physical channel structure of GSM and can be implemented using GSM's own cell structure [1] with data transmission rates of up to 170 kbits/s, as compared to the 9.6 kbits/s offered by GSM's switched circuits [2].

These new data networks represent an overlay to the already deployed GSM infrastructure. Additionally to GSM's Base Station System - BSS, the Home Location Register - HLR and the Visitor Location Register – VLR, two new nodes, known as GPRS Support Nodes (GSN), are introduced to support GPRS.

According to the standards themselves, the throughput of a user data transmission does not depend exclusively on the numbers of timeslots in use, but also on the adopted coding scheme. For example, the GPRS standard defines four separate coding schemes (CSs) where a given CS may be maintained throughout a call or modified as a response to the operating conditions.

This paper proposes an adaptable model based on the level of dropped blocks in a transmitted window as a result of carrier-to-interference ratio (C/I) information. Based on local parameters such as the level of dropped blocks, the system decides whether to continue or move to a new CS. This algorithm falls outside the scope of the ETSI standards.

The rest of this paper is organized as follows. The next section contains a rapid description of GPRS data networks. The GPRS simulator and the adaptable model are described in section III. Some simulation results are given and discussed in section IV. Finally, conclusions are drawn.

II. GENERAL PACKET RADIO SERVICE

II.A. Network Architecture

GPRS requires the introduction of two new network nodes or GSM Support Nodes (GSN) as depicted in Figure 1 [3].

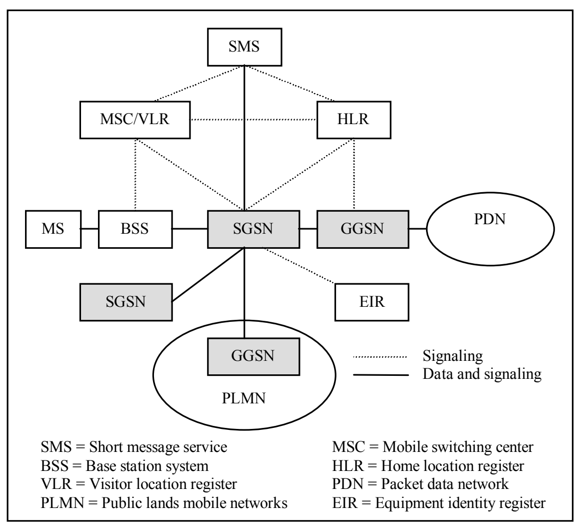

Figure 1: Network Architecture

The Serving GPRS Support Node (SGSN) is the interface responsible for the delivery of packets to the mobile station within its service area. The Gateway GPRS Support Node (GGSN) provides the interworking with external packet networks, such as the Internet, as well as to other GPRS/EDGE networks. The GGSN encapsulates and routes incoming packets to the appropriate SGSN for a particular mobile station.

A given GGSN would normally support more than one SGSN and may, optionally, require location information on mobiles from the Home Location Register (HLR) database. Similarly, the SGSN consults the HLR for the subscriber's data profile.

II.B. Coding Schemes (CS)

The payload in a radio link control (RLC) block depends on the applied CS. There are four CSs (CS1-CS4) specified by the ETSI for use by data traffic channels. Control channels, on the other hand, always use CS1. Table 1 shows a summary of some the main CSs for GPRS [4]. These values are related to the occupation of a single timeslot when transmitting.

Table 1: GPRS Coding Scheme

CS	Code Rate	Payload Bits	Data Rate
CS1	½	181	9.05
CS2	≈2/3	268	13.4
CS3	≈3/4	312	15.6
CS4	1	428	21.4

II.C. Overview of Physical and Logical Channels

The Packet Data Channel (PDCH) is the physical channel used for packet data traffic. A cell may be configured with one or more PDCHs, depending on factors such as data traffic forecast, cell channel capacity and network project specification. Usually, channel allocation is driven by demand [5] and is subject to quality of service (QoS) and the availability of physical resources at a cell.

Among the allocated PDCHs in a cell, there is at least one Master Packet Data Channel (MPDCH), responsible for the transport of both data and control logical channels. The other PDCHs, when used, are called Slave Packet Data Channels (SPDCH) and may only used for the transport of user data.

GPRS logical channels could be divided into four groups [4] responsible for broadcast, common control functions (such as access control, paging, timing and notification) and traffic. The latter is known as the Packet Traffic Channel (PTCH) and is in turn subdivided into two subgroups, namely, the packet data traffic channel (PDTCH) and the Packet associated control channel (PACCH).

III. THE GPRS SIMULATOR

In order to gain an in-depth understanding of GPRS traffic and study the impact of choosing among the CSs on capacity, a simulator has been developed using the SIMSCRIPT II.5 simulation language tool by CACI Products Company [6].

The network topology used is presented in Figure 2. For simplicity, the simulation scenario considers a single cell with a varying number of mobile users and traffic patterns. This scenario is seen as adequate for the type of study we want to undertake. The simulator may easily be extended to consider a more complex cell topology.

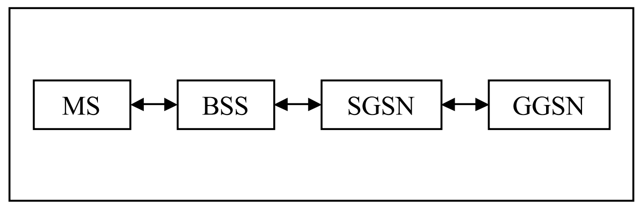

Figure 2: Adopted Simulation Topology

A mobile station is only allowed to use a single timeslot per TDMA frame to send/receive data. Packet loss at the fixed network (interfaces BSS, SGSN and GGSN) is assumed to be negligible. Three types of users have been considered according to their respective traffic patterns:

1. user type **SEND**: sends data on the uplink;
2. user type **REQUEST**: sends data requests on the uplink in order to actually receive data on the downlink;
3. user type **VOICE**: used to create background concurrent voice traffic that competes with data traffic for the acquisition of channels at a cell.

Furthermore, it is also assumed that all calls are mobile originated. A voice user merely requests a channel allocation and if successful an exponential distribution is used to determine the call duration. When this time is up, the resource is retrieved and the voice user rejoins the queue for a possible future call attempt.

All the Coding Schemes were implemented according to Table 1. The packet, frame and block segmentations that occur at the protocol stacks were also implemented according the ETSI standard layer specifications [7, 8].

III.A. The Error Model

The error model is based on a pre-simulated Block Error Rate (BLER) with curves produced from research work from Ericsson, see [9] for more details. These curves have also been adopted by a number of papers including research from [10] and [11]. It is also assumed, for simplicity, that all logical control channels are transmitted correctly to avoid their retransmission. For each PDTCH, a C/I value is selected according to a *Gaussian* distribution. Therefore, using the C/I and knowing the coding scheme, a BLER could be found by looking up the BLER curves. Next, a uniform distribution between 0 and 1 is used to decide, by comparing the value drawn with the BLER value, in order to decide whether the block will be accepted or rejected. Figure 3 shows the implemented error model.

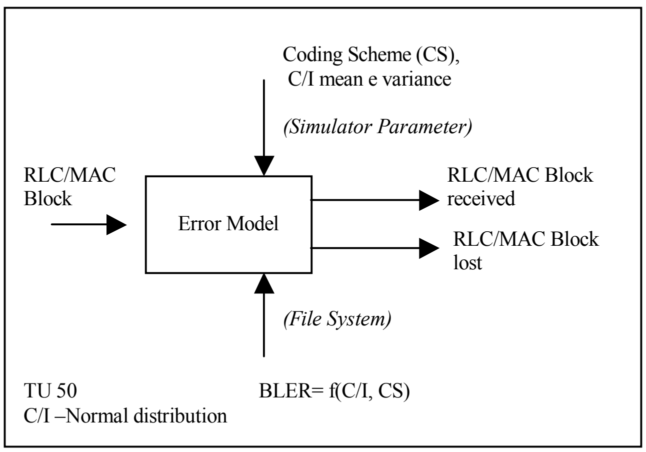

Figure 3: The Simulator's Error Model. See [10].

III.B. The Coding Scheme Adaptation Model

Once an initial CS is defined, the network operator could opt for maintaining the same CS during all the transmission or adapting to a new one according to some algorithms using criteria such as information loss and actual transfer conditions. We show in this work, that the proposed model optimizes traffic and increases network capacity.

CS1 is the default mandatory scheme that all GPRS systems must implement. It is the most reliable one as it introduces more redundancy. Furthermore, the user QoS profile or service agreement may also be seen as important parameters in choosing the initial CS scheme for a new data association. The impact of the use of this criterion is outside the scope of this paper.

In this paper an adaptable model based on the block dropping rate during a transmission window is considered. This window is limited to a maximum of 64 RLC blocks. At the end of a transmitted window, the loss rate is evaluated according to (1):

$$\%\ \text{Rejected blocks} = \frac{\#\ \text{Rejected blocks}}{\#\ \text{Transmitted blocks}} * 100 \qquad (1)$$

The proposed adaptation algorithm makes use of the monitored current block error rate to determine the transition parameters between the coding schemes. Figure 4 illustrates the state machine of the adaptation mechanism whereas Figure 5 presents the values obtained using simulation in order to guaranty efficient transitions in terms of system and user transmission capacities. The following process is applied in order to determine the transition thresholds: the C/I value has been assigned all possible values in the simulation and a plot of the throughput is made. Next, the resulting throughput curve is examined and we observe the C/I point where no further throughput improvement may be achieved.

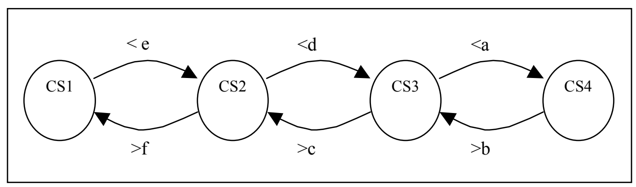

Figure 4: State Machine for transitions between coding schemes

III.C. Calculating CS Adaptation Thresholds

Figure 5 (5(i), 5(ii) and 5(iii)) shows transmissions starting with a relatively more reliable[1] CS and moving towards less reliable ones. For example, a user CS may be adapted from CS1 to CS2 or CS2 to CS3, etc. Note that the less reliable the CS is, the more "useful" data could be packed in its RLC block and sent, as shown in Table 1. Therefore, an algorithm seeking system optimization should strike a balance between reliability and the actual *trueput*[2].

On the one hand, for a lower C/I value, the used CS tends to remain stable or fixed. In this case, a migration could be forced by the selection of larger values of the

[1] The term reliable is used in this context to refer to coding schemes with large redundancy information.
[2] Unlike throughput, *trueput* refers to the actual user information transported across.

transition parameters (a, b, c, d, e and f in Figure 5). Furthermore, if a CS change occurs then the number of rejected blocks will probably increase due to the use of less user information redundancy and the likely occurrence of more errors. Typically, a service provider should draw a compromise between throughput and the number of rejected blocks.

On the other hand, for a higher C/I value, fewer blocks are likely to be dropped. Similarly, if the CS is altered as a result of a transition, the number of rejected blocks will probably increase as a result.

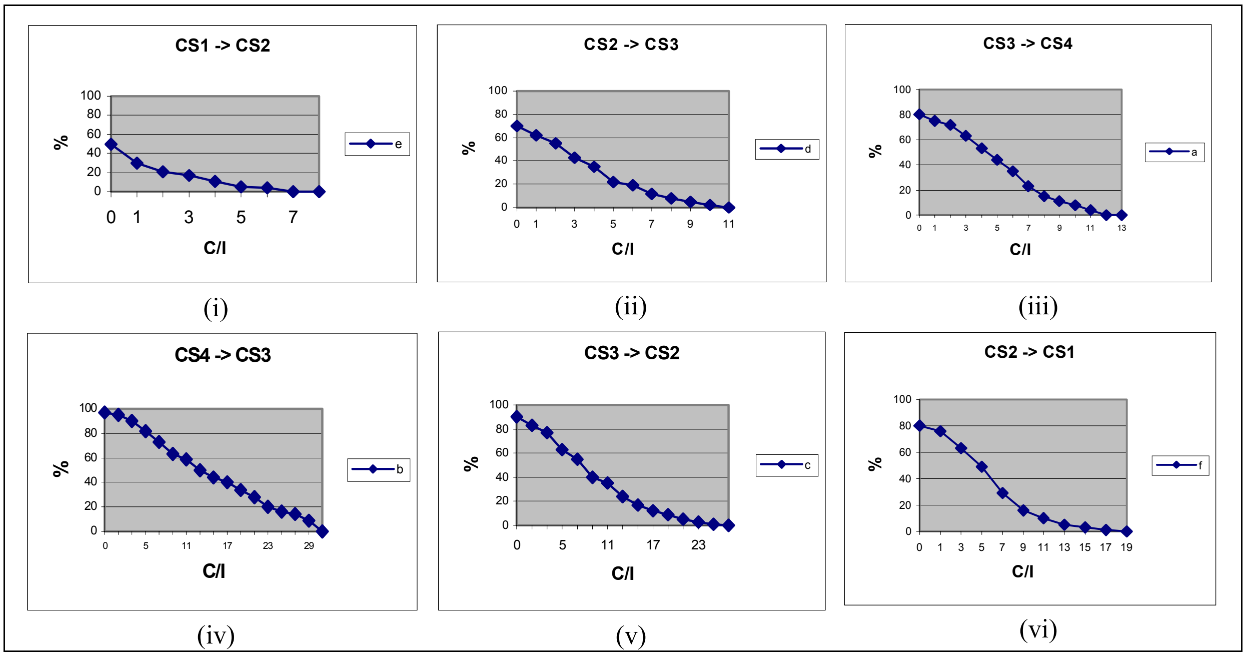

Figure 5: Migration parameters rates versus carrier-to-interference ratio. These possible rates for migration between coding schemes were empirically defined.

Figure 5 (5(iv), 5(v) and 5(vi)) also shows the system behavior when a transition between a less reliable CS to a more reliable one occurs such as when going from CS4 to CS3. In this case, there is a stronger tendency to have CS changes in an environment with low C/I values. When the C/I is high, almost no rejected blocks occur and migrations between CS only happen if the migration parameter values (transition thresholds) are selected by the provider to be relatively small.

CS transition is not suitable for more reliable CSs when a good environmental condition is dominant, for example, a change of CS4 to CS3 in a high C/I ratio. The number of RLC blocks will increase and consequently the throughput decreases. Similarly, it is not suitable to have a CS adaptation for less reliable CSs in bad environmental conditions for example an adaptation occurring between CS1 and CS2 while the CI ratio remains low. In this case, the number of retransmissions could compromise the transmission efficiency.

Using Figure 5 it is possible to select the transition parameters to start the proposed simulation. In this paper, these values were chosen using the BLER curves. Firstly, C/I values corresponding to error rates between 20% and 30% were collected from the BLER curves for all four CSs. With these C/Is, the transition parameters were extracted from Figure 5. Note that other techniques and error rate intervals may be used to determine the C/I threshold values. Furthermore, a cellular access provider could choose to be more or less flexible when setting up these migration parameters values. Figure 6 presents the values used in this work and obtained with a 95% confidence interval.

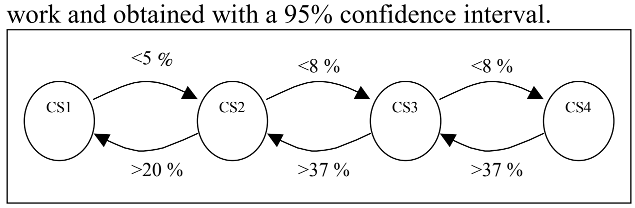

Figure 6: Adopted migration parameters values

IV. THE SIMULATION RESULTS

The simulation parameters shown in Table 2 were used.

Table 2: Simulation Parameters

Number of SEND user	20
Number of REQUEST user	20
Number of VOICE user	60
Application size – uplink (request):	30 bytes
Application size - uplink (send):	2048 bytes
Application size - downlink:	2048 bytes
Number of PDCH uplink channels	1
Number PDCH downlink channels	2
Number of frequencies uplink/ downlink	15 + 15
Simulation Duration	30 min
C/I mean	3 dB - 22 dB
C/I variance	3 dB

Thirty simulations were carried out and the results obtained represent a confidence interval of 95%.

The following figures represent the downlink user transmission. If there is no available channel, the PDCH information is scheduled in a FIFO queue. Voice users are simply included to create background concurrent traffic competing for the available channels at the cell.

Two initial conditions are defined to initialize the simulation when the adaptable model is used. The first one is where all users start their transmission using CS1 (the pessimistic approach) whereas the other one determines that all users begin transmission with CS4 as their default (the optimistic approach).

Comparing the scenarios where the CS is maintained fixed during all the transmission with those where the adaptable model is used for low C/I values (see Figure 7), it is shown that the adaptable model did not make new gains when compared with the fixed CS1 scenario. On the other hand, when compared with the use of the fixed CS4 scenario, the adaptable model became very efficient in both situations (adaptable CS1 and adaptable CS4) - further simulation results could be found in [12]. A low C/I value represents a bad environmental condition and the more reliable the used CS becomes the less retransmissions will be needed. Therefore, the use of CS1 seems to be the most suitable coding scheme for such a situation as it is the one with the largest redundancy.

When using intermediate C/I values such as those around 13dB (see Figure 8), the adaptable CS1 scenario shows clear gains over the non-adaptable one. This is due to the decrease in the BLER ratios corresponding to these intermediate C/I values hence allowing the transport of PDUs with less redundancy overhead (i.e. when using less reliable CSs).

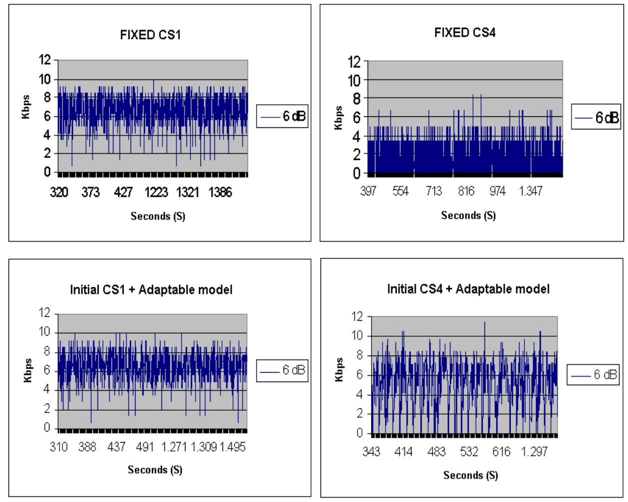

Figure 7: Comparing the use of the CS Adaptation Model with fixed CS in a low C/I condition.

When using C/I values between 7 and 18 dB, the results show that the intermediary CSs, namely CS2 and CS3, are more adequate. In this case, the use of the adaptable model allowed both CS1 and CS4 (the extreme CSs) to converge to the intermediate CSs (CS2 and CS3).

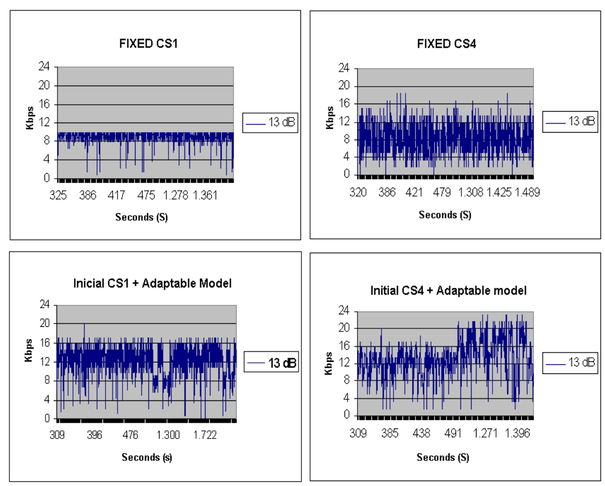

Figure 8: Comparing the use of CS Adaptation Model with fixed CS in an intermediate C/I condition.

For higher C/I values (see Figure 9) the use of a fixed CS almost does not cause any rejected blocks occurrences and the throughput becomes limited only by the payload of each coding scheme. On the other hand, the adaptable model was started at CS1 and CS4 as the initial (starting) coding schemes. The adaptations presented clear throughput gains over scenarios using fixed CS1, fixed CS2 and fixed CS3. Such an environmental condition was suitable for the use of CS4 as confirmed by the results of the simulations. This is due to the fact that when fewer blocks are rejected is a good indication for sending more data in the RLC block. Furthermore, no adaptation was made when starting with CS4 since no actual improvement was detected when compared with the use of a fixed CS4 scenario.

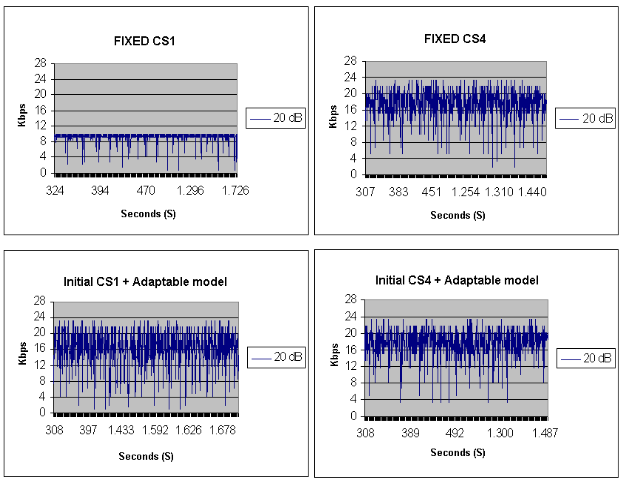

Figure 9: Comparing the use of the CS Adaptation Model with fixed CS in a high C/I condition.

V. CONCLUSION

This paper proposed an adaptable model that allows migration between coding schemes according to the level of observed rejected blocks in a transmitted RLC block window in real time. The ETSI does not specify any algorithm or guidelines for optimized use of the adopted coding schemes. Furthermore, a GPRS traffic simulator was developed to analyze some of the features of this complex technology.

The proposed adaptable model achieves its goal while it does not compromise good situations (for instance, using CS1 when there is a low C/I), and it also improves the transmission in bad situations (the use for example of CS4 with a low C/I).

The gain using the proposed adaptation may be highly dependent on the transition parameters being used (see Figure 6). Lower values may cause system instability and increase CS fluctuations without actual throughput gains.

This work has also shown that the selection of the initial CS for a data transmission may also be seen as an important strategy which highly depends on the operating environment. Such a decision may ultimately lie with the network operator and may de made on a cell basis.

Future work includes the use of this simulator to analyze delay and jitter variations using the proposed adaptation algorithm, the optimization of the proposed adaptation state machine, the introduction of QoS parameters, the use of multislot allocation, the introduction of new coding schemes from the Enhanced Data Rates of GSM Evolution (EDGE) and expanding the current topology to include multiple cells and other mobility models.

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Suzana de F. Dantas and Djamel H. Sadok can be contacted at the Centro de Informática, Universidade Federal de Pernambuco, P.O. Box Nº 7851, Cidade Universitária, 50732-970, Recife, PE, Brazil. Phone: +55 81 3271-8430 Fax: +55 81 3271-8954 E-mails: {sfd, jamel}@cin.ufpe.br