RESOURCE ALLOCATION FOR INTERACTIVE TRAFFIC CLASS OVER GPRS

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ABSTRACT

The General Packet Radio Service (GPRS) is a new bearer service for GSM that greatly simplify wireless access to packet data networks, e.g., to the Internet, to the corporate LAN or to the mobile portals. It applies a packet radio standard to transfer user data packets in a well-organized way between Mobile Stations (MS) and external packet data networks. The introduction of guaranteed performance services in GPRS networks requires detailed studies of the resource allocation and service integration issue. This paper proposes some different schemes of allocating the physical channels to mobile stations. We consider the integration of voice and data over Time Division Multiple Access wireless cellular networks. We describe different radio resource allocation algorithms, and describe their similarities and differences in the context of GSM and GPRS networks. However these algorithms can be used for diversified types of wireless networks. In this paper we consider the interactive best effort traffic class as one of the most important traffic classes in GPRS.

1. INTRODUCTION

The General Packet Radio Service (GPRS) is an overlay to the circuit switched GSM-networks. GPRS increases the possible bandwidth for data transmission by introducing packet switching. The radio interface of GPRS system is defined in [7] and an overview is given in [3]. The highest sub-layer is the SubNetwork Dependent Convergence Protocol (SNDCP), which maps network level characteristic onto one of the underlying network. Under the SNDCP is the Logical Link Control (LLC) sub-layer, which provides a highly reliable logical link between two entities. Under LLC sub-layer is the Radio Link Control (RLC) sub-layer, which performs the segmentation and reassembly of LLC Protocol Data Unit (PDU) into RLC data blocks. Under the RLC sub-layer is the Medium Access Control (MAC) sub-layer, which enables Mobile Stations (MS) to share one or several Physical Channels (PhCH). Under the MAC sub-layer is the Physical layer that provides services for information transfer over physical channels between Mobile Stations and the GSM/GPRS network, as is shown in Table 1.

SubNetwork Dependent Convergence Protocol (SNDCP)	Network Layer
Logical Link Control (LLC)	Data Link Layer
Radio Link Control (RLC)	
Medium Access Control (MAC)	
Physical Layer PhCH	Physical Layer

Table 1: Layers in GPRS/GSM Networks

On the physical layer, GSM uses a combination of Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA). Two frequency bands 45MHz apart have been reserved for GSM/GPRS, namely 890 to 915 MHz for the uplink and 935 to 960 MHz for the downlink. Each of these 25 MHz bandwidths is divided into 124 single carrier channels of 200 kHz. A certain number of these frequency channels are allocated to a Base Transceiver Station i.e. to a cell. Each frequency channels carries eight TDMA channels, called time slots and these slots form a TDMA frame. Each time slot of TDMA frame has duration of 576.9 μs, so a TDMA frame has duration of 4.613 ms and every time slot defines a physical channel. In conventional GSM, a channel is permanently allocated for a particular user during the entire call period (whether or not data is transmitted). In GPRS channels are only allocated when data packets are sent or received, and they are released after the transmission. A cell supporting GPRS may allocate physical channels for GPRS traffic. Such a physical channel is denoted as Packed Data Channel (PDCH). The PDCH are taken from the common pool of all channels available in the cell. The allocation of physical channels to Circuit Switched (CS) services in GPRS is done dynamically according to the "capacity on demand" principle. It means that GPRS does not require permanently allocated

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PhCHs. When the PDCH are congested but more resources are available in the same cell, the network can allocate more channels for PDCHs. Dynamic allocation of PDCHs can be done by the allocation of unused channels as PDCHs to increase the overall GPRS QoS and by the de-allocation of PDCHs upon resource demand for other services with higher priority. Physical channels available in the cells could be shared between different traffic types, either in a fixed boundary or in a movable boundary way [9]. In a fixed boundary scheme, each traffic type has the same number of physical channels allocated to it. This policy is not efficient, because resources are not fully utilized. The one type of traffic wastes the resources, while the other one is suffering from congestion. The movable boundary policy overcomes this disadvantage by allowing a limited sharing of resources. The double movable boundary policy considers that the total of resources is separated into three parts. One part is reserved for the Circuit Switched traffic, second for Packet Switched traffic, and the third part contains a number of channels that are dynamically assigned, either to Circuit Switched traffic or to the Packet Switched traffic. The number of fixed Circuit Switched traffic channels depends on the input load of Circuit Switched calls in the cell while the number of fixed Packet Switched traffic channels assures a minimum QoS for Packet Switched services [9].

In this paper we focus on Physical Channels sharing to Circuit Switched and Packed Switched transmissions assignment to GPRS Mobile Stations. The basic radio packet in GPRS is the RLC/MAC block. It uses a sequence of four time slots on a PDCH. A Temporary Block Flow (TBF) is a physical connection used to support the transfer of a number of blocks and identified by Temporary Flow Identifier (TFI). The TFI is included in every transmitted block, so that multiplexing of blocks originated from different Mobile Stations on the same PDCH is possible.

The next section proposes different schemes in order to assign physical channels to GPRS Mobile Stations. The third section presents the traffic model of interactive traffic class used for the simulation, shows the input parameters and simulations results and finally conclusions are drawn.

2. RESOURCE ALLOCATIONS ALGORITHMS

In GPRS systems two scheduling algorithms have to be chosen, depending on where the algorithms are implemented.

- **A.** Concerns the PDCHs allocation to Mobile Stations and it is executed at the connection establishment
- **B.** Concerns the distribution of block periods belonging to the same PDCH among Mobile Stations assigned to this PDCH [9] and is executed within the transmission

A: CONNECTION ESTABLISHMENT

Increase by M – "MULTIM": This favours the allocation of small indexes of physical channels to new arrival Mobile Station. When a Mobile Station whose multi-slot class is *x*, wants to establish a Temporary Block Flow, the algorithm assign *x* physicals channels to it while the PDCHs are assigned to no more than *M* Mobile Stations. If all the PDCHs are assigned to *M* Mobile Stations the parameter *M* is increased by *M.* The parameter *M* can depend on the users multi-slot class.

Minimise the loading – "MIN": The algorithm knows the number of Mobile Stations assigned to each PDCH. When a Mobile Station, whose multi-slot class is *x*, want to establish a Temporary Block Flow, it is assigned to the least loaded *x* PDCHs.

SAFE: It gives the same priority to both the loss rate and to the delay parameter of QoS. It divides the potential number of PDCHs *Y* for two groups, first (*Y*-[*Y*/2]) risky physical channels (high index) and second ([*Y*/2]) safe physical channels (small index) ([ ] means the integer part). The probability that the risky physical (high index) channel is occupied by a Circuit Switched calls is higher than the one of a safe (small index) physical channels. When a Mobile Station, whose multi-slot class is *x*, wants to establish a Temporary Block Flow, then the network assigns for it the ([x/2]) least loaded safe physical channels and (*x*-[x/2]) risky least loaded physical channels.

B: DURING CONNECTIONS

First Come First Served (FCFS): This is the first proposed scheduling algorithm for distribution of block periods within the transmission. It avoids the interference of blocks originated from different Mobile Station on the same PDCH. An arriving set of blocks is transmited together until the end. New blocks arriving from other users or retransmitted blocks have to delay in a FIFO queue according to when they arrive. When the current user releases the PDCH, the next set of blocks is served. If some blocks are negatively acknowledged, they are retransmitted as a new set of blocks. This algorithm is very simple and easy to implement, but is more suitable for small Temporary Block Flow than for a large flow, because the number of retransmitted blocks increases rapidly when they have to wait until the end of the transmission of a larger set of blocks, originating from other users.

FCFS with Priority: This is very similar to the FCFS algorithm. The difference is that this one gives a transmission priority to the set of blocks that depends only on the number of block retransmissions. The FCFS algorithm with priority is also very simple but is more efficient than FCFS.

Round Robin: It divides the block periods of PDCH between all the users of this PDCH. The resource assignment is updated at each new user arrival and at each block period. The algorithm operates like a set of FIFO queues with a Round Robin server whose allocation cycle is one block period. This algorithm is more protective and flexible than the FCFS, but is also more complicated to implement.
Round Robin with Priority: This Round Robin with priority gives a transmission priority to the set of blocks depends of the number of block retransmissions. This algorithm is efficient for large TBF transmission, but is more complex to implement than the other algorithms.

3. SIMULATION

3.1 Traffic model

In our model we have used four voice sources with strong mobility behaviour, three voice sources with week mobility characteristics and 15 data sources that simulate www sessions. Voice source generate calls according to Poisson process with mean value of active time 120 seconds for mobile stations which remain in one cell and 45 seconds for mobile stations which move to another cell before finishing the call. We assume that each voice source generates traffic equal 0.4 Erlang. Data sources in our model try and simulate the behaviour of www sessions. This can be modelled as a non real time service proposed in [8], [9].
Data sources are modelled by burst traffic with ON-OFF characteristics, where the OFF time represents the time during which a user is reading or browsing a Web page. The ON time represents the time for the download of the files that belongs to particular Web page as shown in Figure 1. Of course it is impossible to find the number and the size of these files, as shown in Figure 2. This is because these variables are generated statistically, so we concentrate on the average throughput of data, which we model to be 5 kb/s. In the network model, the Web server and backbone network are ideal except for a propagation delay. We assume the use of HTTP v1.1 where one TCP connection is used for all HTTP requests for a page [10].

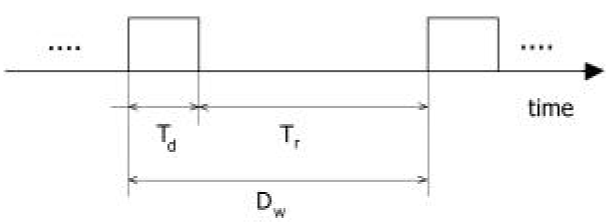

Figure 1: Data Source ON Time

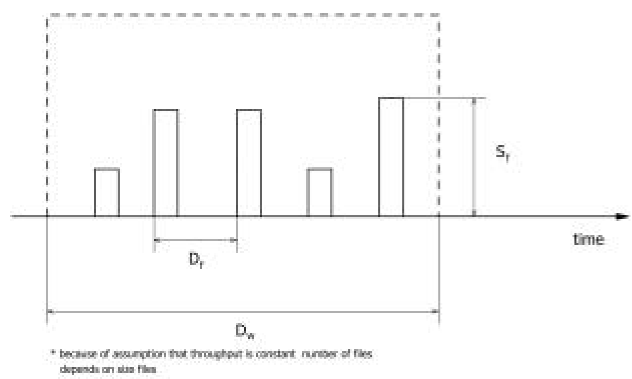

Figure 2: Data Source Files Distribution

T_d Time of downloading web page [sec.]
T_r Time of reading/ browsing web page [sec.]
D_w Inter arrival time between two web pages. D_w is truncated gamma distributed random variable with:
Mean = 60 seconds
min. = 5 seconds (this is minimum time need to read information required to make hop to next page)
standard deviation = 6 seconds
max. = 300 seconds
D_f Inter arrival time between two files from the same web page. It is truncated exponential distribution variable with:
Mean = 0.062 seconds [9]
min. = 0.04 seconds
max. = 1 second
S_f - size of the file, it is assumed to be a Poisson distribution with:
mean = 45 blocks (where "blocks" represents 268 bits of data)

3.2 Simulation parameters

In this paper we consider an interactive traffic class that is typically used by traditional Internet applications like WWW. GPRS systems will support principally best effort services and therefore we take into account only best effort parameters. The first performance parameter is the average throughput of one Mobile Station, which is the quantity of data received correctly in a unit of time and normalised for one Mobile Station. The second performance parameter is the average delay requested to transmit a TBF. The third performance parameter is the packet loss. The throughput, delay and packet loss parameters consider only the TBFs correctly received. When a TBF is not correctly transmitted all blocks related to this TBF are considered failed. In our simulation we consider only a single carrier frequency because a GPRS system allows users to send packets only within a single carrier frequency. It is not possible to use multi-slot class transmission within multi-carrier frequency [7].

The simulator input is the data load in the single carrier frequency, which increases by increasing the Mobile Stations number per carrier frequency. To simulate a GPRS system we used SES/*Workbench* simulation tool. We ran a number of simulations for three hours duration.

The simulation parameters for the model that we used are as follows:

- SNDCP header length = 2 bytes
- LLC header length = 7 bytes
- Length of LLC frame = 1520 bytes
- Mode Ack L_{imit} = 60 seconds
- Coding schemes CS-2 used and BLER = 10%
- Average load per Mobile Station = 5kb/s, multi-slot class = 2, 3, 4
- Parameters of PDCH sharing:
    - PDCH assigned for Circuit Switched = 2
    - PDCH assigned for Packet Switched = 1
    - Number of PDCHs dynamically assigned = 5

3.3 Performance of different scheduling algorithms within the transmission

As we evaluate performance parameters for scheduling algorithms for the distribution of block periods within the transmission. As the packet loss rate values are insignificant in the case of these simulations, those loss values are not taken into account. The scheduling algorithm executed at the connection establishment used by default in this case of simulation is the "MIN" one.

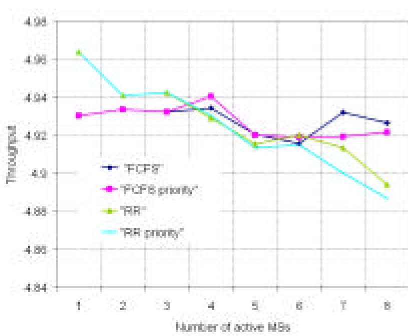

Figure 3. The Throughput – Scheduling During Transmission

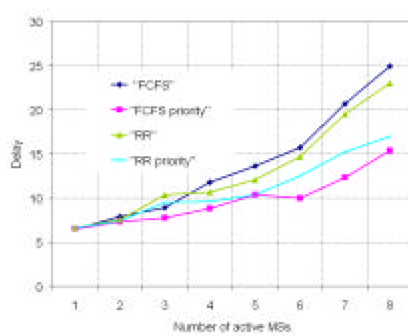

Figure 4. The Delay – Scheduling During Transmission

Figure 3 displays transmission throughput versus the number of Mobile Station assigned to the single carrier frequency. In the case of a small number of Mobile Stations better throughput performance have algorithms "RR" and "RR priority", but if we consider larger number of Mobile Stations in the single carrier frequency better throughput performance has the algorithm "FCFS". When we compare algorithms "FCFS priority" and "RR priority, the algorithm "FCFS priority" has slightly better performance in case of larger number of Mobile Stations and smaller throughput in case of smaller number of Mobile Stations.

Figure 4 contains the transmission delay of a TBF versus the number of Mobile Stations in the single carrier frequency. The "FCFS with priority " scheduling algorithm is the most efficient one. Next, the "RR with priority" is slightly less efficient. Algorithms without priority have largest delay that algorithm with priority. The worst delay performance has scheduling algorithm "FCFS".

3.4 Different scheduling algorithms executed at the connection establishment

Here we evaluate the performance parameters for scheduling algorithms that are executed at connection establishment. The three algorithms that are considered in these simulations are: “MULTIM”, “MIN” and “SAFE”. Parameter *M* is set to (*M*=M_Cl). The scheduling algorithm for distribution of block periods within the transmission used by default in this case of simulation is the “FCFS priority” one and the multi-slot class is equal four.

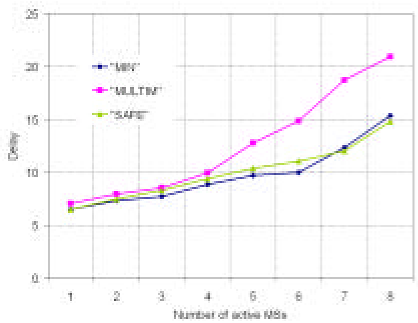

Figure 5. The Delay Versus Number of Mobiles

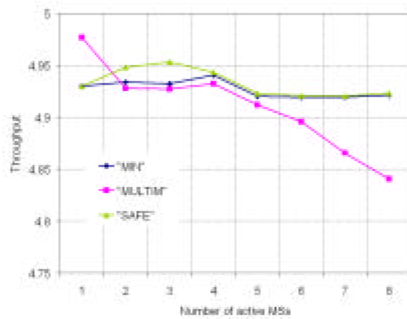

Figure 6. The Throughput Versus Number of Mobiles

We consider normal voice traffic load scenario, i.e. the voice traffic load is not much higher than the number of PhCHs assigned only for Circuit Switched. We assume voice traffic load equal 2.8 Erlang per single carrier frequency. Figures 5 and 6 display the performance comparison of the different proposed scheduling algorithms. Those figures contain the transmission delay of a TBF and throughput versus the number of Mobile Stations in the single carrier frequency. The packet loss rate values are insignificant, and hence, those values are not taking into account. Algorithms “MIN” and “SAFE” are more effective for throughput and delay performance than the algorithm “MULTIM” one. The algorithms “MIN” and “SAFE” differs only slightly in case of low voice traffic load. The difference between algorithms “MIN” and “SAFE” is more significant when we consider a larger voice traffic load and a larger data traffic load.

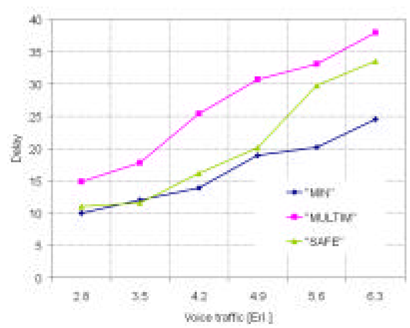

Figure 7. The Delay Versus Voice Traffic

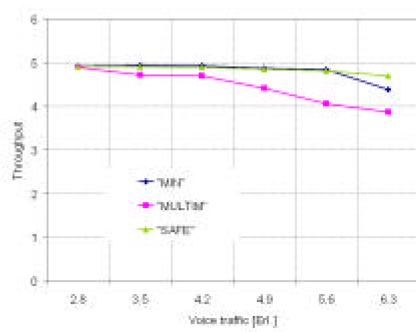

Figure 8. The Throughput Versus Voice Traffic

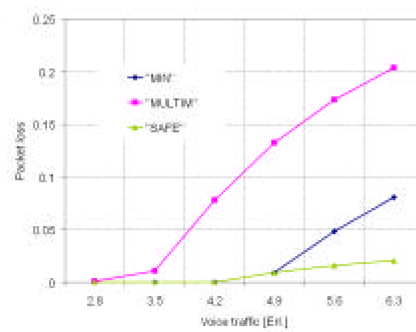

Figure 9. The Packet Loss Versus Voice Traffic

Figures 7, 8 and 9 display the performance comparison of the different proposed scheduling algorithms for different voice loads. The number of Mobile Stations in the single carrier frequency is equal to 6. In the case of large voice traffic loads the algorithm “MIN” is the most effective one for delay performance, but the algorithm “SAFE” is more effective for packet loss performance. The algorithm “MULTIM” is less effective than the others one for delay, throughput and packet loss performance.

3.5 The influence of multi-slot class

Figures 10, 11 and 12 give the performance using the “SAFE” scheduling algorithm. A comparison of the performance for different values of M_CL is shown (assumed that the number of Mobile Stations in the single frequency carrier equal 6). These figures display the amelioration of performance when the users M_CL increases.

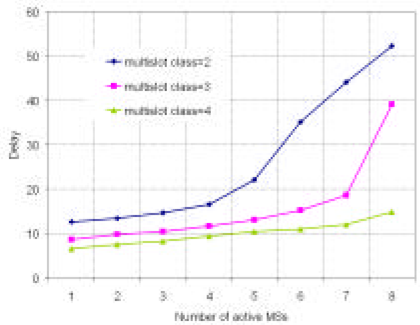

Figure 10. The Delay (SAFE)

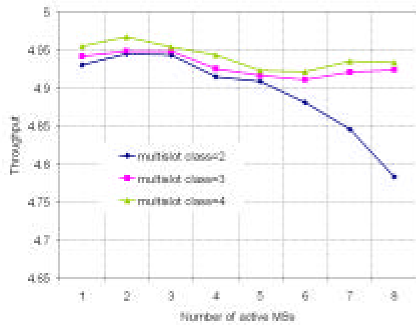

Figure 11. The Throughput (SAFE)

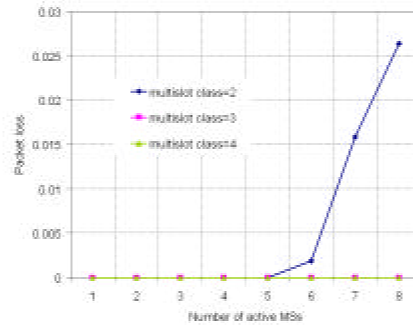

Figure 12. The Packet Loss (SAFE)

The differences between multi-slot classes 2, 3 and 4 are more significant when we consider large voice and data traffic loads, as in shown in Figures 13, 14 and 15. If we consider the low voice load equal 2.8 Erlang, the differences between different multi-slot classes are more significant than in larger data load conditions. The performance differs only slightly for multi-slot class 3 and 4, but for multi-slot class 2 the performance decreases significantly.

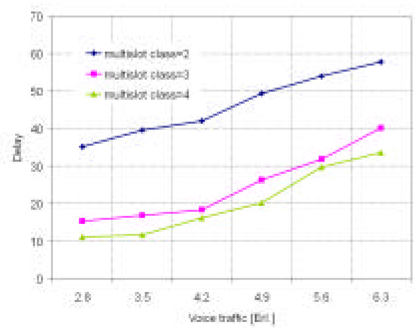

Figure 13. The Delay (SAFE)

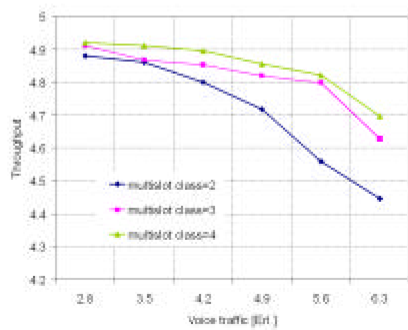

Figure 14. The Throughput (SAFE)

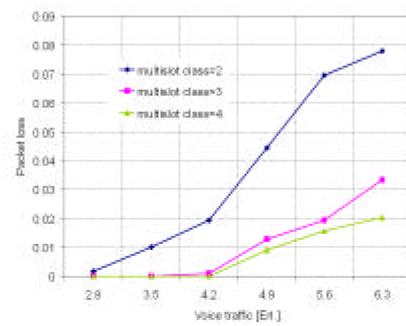

Figure 15. The Packet Loss (SAFE)

4. CONCLUSION

In this paper, we have discussed various algorithms for resource allocation in GPRS and their similarities and differences. We have proposed different strategies for scheduling the distribution of block periods within the transmission, and for scheduling at the connection establishment. Our simulation results consider different issues in the proposed scheduling algorithms and show the benefits of using different policies in radio resource allocation. In the simulations we have considered different performance parameters like throughput, delay, loss rate of the Temporary Block Flow (TBF) in a number of different traffic load scenarios. The performance of the algorithms is evaluated for the interactive traffic class and best effort service. Finally the proposed algorithms in this paper are in the context of GSM and GPRS, but they might be useful in any wireless packed data network.

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