A New Bi-processor SmartPhone

Evaluation of the perfomance generating GPRS and UMTS data traffic

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Summary
In this work a new type of device is introduced: a bi-processor SmartPhone equipped with a network interface GSM/GPRS/EDGE/UMTS. The architecture used, with two processors, is introduced to take advantage of the subdivision of the functionalities provided by the two processors. A better distribution of various functions is achieved, all above when multi-media traffic is considered. Moreover, the interaction between the two processors has to be realized in a convenient manner to better take advantage of the sub-division. The form of communication established between the two processors chosen is serial communication based on the standard RS232 and ETSI 07.10. Before realizing the product we propose to evaluate the performance of a similar project. To do this a simulator was built focusing our attention on communication between the two processors and different types of data traffic (voice call, sms, etc.). were generated. Furthermore, GPRS and UMTS connections were considered on our simulator to verify that a similar architecture, when correctly developed and dimensioned, can support this type of data traffic. To be sure about the effectiveness of our simulator this was validated through a statistical method, associating a queue model both with GPRS data traffic and UMTS data traffic.

Key words:
SmartPhone, GPRS, UMTS, RS-232, 3GPP TS 07.10, Validation

1. Introduction

When The SmartPhone can be considered as the fusion of a small pc and a mobile telephone, the new type of device that can support new functions. It is different from the last mobile telephones. In this work the design of a new type of device with two processors is considered. It is based on a GNU/Linux platform. Two different entities can be characterized: an Application Processor (AP) and a Mobile Termination (MT). The final product will permit data traffic and voice traffic to be received and transmitted. To connect the AP and the MT a serial channel and a multiplexing protocol based on ETSI 07.10 [1] were considered. The multiplexer protocol 07.10 permits some sessions to be realized simultaneously over a normal asynchronous interface. It permits a connection between a Mobile Station (MS) and a Terminal Equipment (TE) to be realized. Each session consists of a byte flow transferring different types of "data information" such as voice, data traffic, fax, sms, GPRS traffic [2, 13], and UMTS traffic [5]. The multiplexer can be considered a mechanism through which it is possible to generate a virtual connection between a process on the TE and a similar process on the MS to transfer data between the TE and the MS. The channels created between the TE and the MS are called Data Link Connection (DLC). To better evaluate the performance of our design an *event-discrete* simulator was developed. Specifically, the serial channel through the standard protocols were simulated and GPRS data traffic and UMTS data traffic were generated to evaluate whether our work can support these types of traffic. The rest of this paper is organized as follows: in section II details about the SmartPhone are presented; in section III our simulator is introduced; in section IV our simulator is validated; in section V GPRS and UMTS traffic implementation are considered; in section VI GPRS and UMTS traffic generations to validate the simulator are considered; in section VII GPRS and UMTS campaign simulations are conducted. Finally, conclusions are drawn and future research directions are outlined in section VIII.

2. The Details of The SmartPhone

The design of a new type of SmartPhone with a network interface GSM/GPRS/EDGE/UMTS based on two processors was considered. The two processors are called: Application Processor (AP) and Mobile Termination (MT). Figure 1 shows the software and hardware components of our model representing the connection between the AP and the MT. Here we have two entities called Mobile Equipment (ME) and Terminal Equipment (TE) characterized by two different environments, with different processors and different operating systems connected through a serial channel. Similarly we use ME as AP and

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TE as MT. The first is used to represent the Application Processor and the last is used to represent the Mobile Termination. The Wireless Telephony Manager (WTM) is a software module permitting the application layer to communicate with the MT through a communication device that here is a serial device (we can also imagine a connection realized through a different type of device). Through the WTM the applications running over a terminal can communicate with other remote applications through the wireless network available (GSM/GPRS/EDGE/UMTS/Bluetooth..). Generally the applications running over a terminal need to communicate with remote applications using standard protocols such as the suite of protocols TCP/IP or legacy (files, data stream, etc.). Specifically, the WTM has to do: activation, opening, closing and controlling of the IPC channel (Inter Processors Communication), telephone calls management (data or voice), management of asynchronous events such as messages and error management, control and access to Toolkit SIM, emergency, SMS, EMS, MMS, FAX.

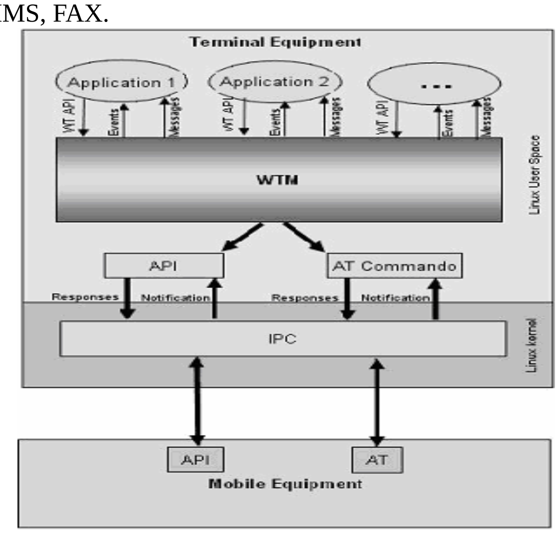


Figure 1. Connection between AP and MT

2.1 The serial channel between ME and TE

Different types of interfaces between ME and TE can be considered:

- AT commands (standard or legacy);
- the use of a serial multiplexing, the *Inter Processors Communication* (**IPC**) based on the ***3GPP TS 07.10*** standard;
- the use of an *Application Programming Interface* (API) for the MT and building an owner interface.

In this work we have chosen to develop the option with the IPC based on the standard 3GPP TS 07.10.

2.2 The RS232 Standard

The standard RS232 was designed to realize the connection between a communication device called Data Communications Equipment (DCE), i.e. a modem, and a terminal device called Data Terminal Equipment (DTE), i.e. a pc. More details about the RS232 standard can be found in [3]. The fundamental devices constituting a serial interface are the Universal Asynchronous Receiver/Transfer (UART), and the driver/receivers. The UART permit the conversion of the data in parallel from serial and vice versa in reception. The others (driver/receivers) permit adaptation of UART input/output signals in the typical signal value of the used interface. In the UART there is a FIFO (First In First Out) buffer to receive and to send data even if the CPU is busy. The dimension of the buffer depends on the serial connection speed.

2.3 The Multiplexer 07.10

The 3GPP TS 07.10 [1] establishes the communication rules between the *Mobile Station* (MS), called ME, and the *Terminal Equipment* (TE) and permits the creation of some number of sessions over an asynchronous serial channel. Each session can be used to transfer voice, fax, data, sms, GPRS, etc. In this way it is possible to execute more applications simultaneously. The multiplexer protocol is independent of the MS and the TE and it is designed for mobile terminals with battery and for this reason it includes *power saving* functions. The multiplexer creates a virtual channel between a TE process and a similar process in MS. Each channel is called *Data Link Connection* (DLC) and when a new connection has to be established the operation mode is chosen: *Error-Recovery Mode* (ERM), which considers the errors recovery, and the *non-Errors-Recovery Mode* (non-ERM), without errors recovery. The multiplexer considers different operating procedures: Basic, Advanced without errors recovery, Advanced with errors recovery; In the basic option the unavailable frames are dropped and no acknowledgement is sent to the sender. Naturally to implement our simulator each field of the frames was analyzed. Moreover , the frame types to make a selection of the frames that we have implemented were considered [2].

3. Our Simulator

An *event-discrete* simulator was developed to simulate our SmartPhone with two processors.

Specifically, our simulator focuses on the serial connection existing between the two processors. The parameters to use were analyzed to characterize the speed of the channel and the buffer dimension. Regarding speed typical speeds considered in a serial channel based on the RS232 standard are: 9600 bit/sec; 19200 bit/sec; 38400 bit/sec; 57600 bit/sec; 115200 bit/sec. Furthermore, other speed values were added typical of the more "modern" serial standards such as the *Universal Serial Bus* (USB): 230400 bit/sec, 460800 bit/sec.

3.1 Development Environment and 07.10 protocol Implementation

The simulator was developed using a well-known *object-oriented* programming language, the C++. To simulate communication between the AP and the MT it was chosen to implement the Inter Processors Communication (IPC) based on the 3GPP TS 07.10. The only base function without errors recovery was considered; this type of implementation requires a more simple frame structure, without flow-detection mechanisms and the context considered has to be error-free. The following frames were built: SABM (Set Asynchronous Balanced Mode); UA (Unnumbered Acknowledgment); DM (Disconnect Mode) ; DISC (Disconnect); UIH (Unnumbered Information with Header check); UI (Unnumbered Information); PSC (Power Saving Control); CLD (Multiplexer Close Down); Test (Test Command); Fcon (Flow Control On Command); Fcoff (Flow Control Off Command); MSC (Modem Status Command); NSC (Non supported Command Response); SNC (Service Negotiation Command). As we have seen the 07.10 protocol is based on the use of a multiplexer that can create some virtual channels, called Data Link Connection (DLC). In the standard the number of channels that have to be opened is not specified. This is a project parameter. In fact, the standard establishes only the maximum number of channels that can be opened: 63 and that channel 0 is the control channel. Furthermore, the protocol establishes the channels included between channel 1 and channel 7 have the same priority. Channel 0 has the maximum priority. To choose the number of virtual channels it was considered that each channel can be "associated" with an application (Voice calls, SMS, GPRS data connection, UMTS data connections, Video Call).It was decided to open 5 virtual channels and the control channel simultaneously. This is due to the GPRS modem currently in commerce that opens a variable number of virtual channels between 4 and 6 (the control channel is considered apart). Then 5 DLCs were considered. Each channel DLC has two associated buffers , one to consider the data transmitted from the AP to the MT, and the other to consider the data traffic transmitted from the MT to the AP. To dimension these buffers the requirements of a GPRS modem such as the Wavecom [4] that uses a dimension buffer of 1024 bytes and Ubinetics [14] that uses a dimension buffer of 3600 bytes have been considered. For this reason an intermediate value of 2048 bytes was considered. Really, to verify that dimensioning of the buffers is correctly chosen two or three CPU time-slice are needed. The time-slice is the time "dedicated" from the CPU to a specific process. When this time has elapsed the CPU is assigned to the stopped process or to a new process. Typically the modern CPUs have 8 ms of time-slice.

$$NTS = \frac{2048 byte}{(460800/8) byte/ms} = 35{,}5 ms \qquad (1)$$

In this way, if a dimensioning buffer of 2048 byte and a speed channel of 460800 bit/sec are considered it is possible to realize more than 4 time-slices through a modern CPU. In fact,

$$(4*8\text{ms}) <(35{,}5 \text{ ms}) <(5* 8\text{ms}) \qquad (2)$$

The dimension of the buffers as an input parameter was considered, enabling the dimension of the buffers to be chosen at each simulation run.

3.2 Bandwidth Management

Another important parameter to be fixed is bandwidth. In fact, it has to be established how the total speed of the physical channel has to be distributed among the virtual channels considered. In our SmartPhone bandwidth management is realized at the WTM, but the manner in which the bandwidth is assigned is reflected in the performances of our channel. For this reason two different algorithms to manage the bandwidth were considered here. Two different schedulers were used to manage the bandwidth:

1) Simple Scheduler,

2) Priority Dynamic Scheduler

In the first of the two mechanisms, the bandwidth is equally assigned to each open channel. Through the simulation of the results it can be seen that this bandwidth assignment is not optimal. For this reason the applications were classified in different classes based on the frame dimension of the different applications: 1)Class A: data traffic due to the voice calls and SMS; 2)Class B: data traffic GPRS and UMTS; 3)Class C: Real-time traffic (video calls).

A simple consideration is made: to have different applications in a simulation run (classified in different classes) such as a GPRS connection and a voice call, we have to "privilege" the data connection assigning more bandwidth to this application with respect to the voice call. This is due a scheduling mechanism based on the priority concept. In fact, it is possible to assign to each application a value between 0 and l and the sum of the priority assigned in a single run has to be l. If in a simulation run different applications belong to the same class, then the bandwidth is equally distributed among these applications. It has to be taken into account that on the first channel (channel 0), which is a control channel, the AT commands consume some bandwidth. In fact, in channel 0 is realized, for example, monitoring of the radio signal quality. The AT commands are used to inquire and to set parameters both of the modem and of the external network. On each command line more commands can be written [12]. One of the main activities of a cellular phone is that of voice calls. To implement these voice calls it was necessary to implement the AT commands related to [4].A SmartPhone user wants to use the SmartPhone as a normal telephone, but he wants to have the possibility of connecting to the Internet, send an e-mail, connect at the VPN etc. The traffic, required in these applications, travels along the channel between the AP and the MT. This traffic is dependent on the network interface considered, GPRS or UMTS. To implement the data traffic different steps were considered:

- to characterize the traffic sources;
- shaping of the traffic source;
- Implementation.

The SmartPhone realizing a GPRS call is based on the GSM infrastructure and creates a GPRS session representing a WWW session. This session is represented by a continuous succession of data packet sequences, called packet calls and sequences of reading time called reading times. A packet call is a download of a WWW document and for its duration different packets can be generated originating burst sequences . The reading time included between a packet call and the next packet call is the time required from the user to analyse the information required. To describe GPRS traffic the traffic model in Internet can be adopted as defined in [6]. The number of the packet calls can be represented as a random variable distributed as a geometric distribution. The reading time between the packet calls can be represented as a random variable distributed as an exponential distribution with a rate μ_R. Each packet call includes a packet number represented by a random variable geometrically distributed with average value N_p. The interval between the arrival of a packet and the next arrival of another packet (inter-arrival of the packets in a packet-call) is modelled as a random variable exponentially distributed with the average value μ_D. As was seen in [6] the average reading time $1/\mu_R$ equal to 41.2 sec and N_p equal to 25 packets is assumed. The value of μ_D depends on the data traffic. The interval between two packets is represented by a random variable exponentially distributed with parameter μ_C .

4. Validation Processing

4.1 Associated Model to Validation Processing

It has been seen that our simulator has to simulate the asynchronous communication between the AP and the MT. To model our system two different blocks were considered: the first one is represented by the MT and its transmission channel . The second is represented by the AP and its transmission channel . We cannot use the term channel in this context because it is only the transmission buffer of the AP and the transmission buffer of the MT and for this reason it seems two processors have two different channels because the communication is full-duplex. First the module constituted by the Mobile Termination (MT) is considered. The model studied for this module can be applied to the module constituted of the Application Processor due to the symmetry of the system. The validation has to be realized in both of the modules. Validation was conducted in this way. A data connection in download was considered. In this way the external IP traffic can be considered as an arrivals flow in our system. The MT receives these data packets, "tunneled" in the 07.10 frame, and it sends the data packets over the serial channel of the application processor. It seems, in this way, that the MT is a server and manages the arrival of the PPP packets. This behavior can be modeled as a classical queue system constituted of an arrival process, a buffer, a server and a service process. Our server is the MT. It captures the PPP data packets from the external GPRS network, it encapsulates these data packets in a 07.10 frame and it copies these in its transmission buffer. The server behavior is characterized by the service time distribution. Generally the service time is represented by a random variable and this indicates the interval time a PPP data packet remains in the server. Here this random variable is the transmission time of the packet in the MT buffer. In this case it can be imagined that the discipline queue is *First-In-First-Out* (FIFO). An external network GPRS was considered and for this reason there are no limits in the buffer dimension for our model. A GPRS data connection was introduced to verify the behavior of our simulator. Furthermore, the UMTS data traffic was introduced to complete our validation processing. The analysis conducted in the

Section III can be used to model this type of traffic in input. In fact, the Poisson model describes a burst communication and it is characterized by the consideration that the interval time between two renew consecutive is described by an exponential distribution. In each session it is possible to consider the last instant of the reading time as the renewal instant, or the reading time between two packet calls. Between two consecutive renewals there is an interval described by an exponential distribution. The probability of observing two or more renewals in the same time interval dt can be neglected if dt tends to zero. Another important property is the complete randomness of the renewal instants. In fact, if n renewals have been verified in the interval [0, t], based on a Poisson model, then the correspondent renewal instants will be located in the interval time [0, t] in the same way if a completely random observer systems these. A typical behavior of the birth-death process can be associated with our system. If the rate of births is constant, the only birth process is the Poisson process of arrivals to the service station with an unbounded queue. If the death rate is constant, the only death process in the system is the distribution service carried out by a server. In this way there is an M/M/1 queue system, because there is an exponential reading time, only a one server , and an exponential service time. Considering the module constituted of the AP the statistical model we can adopt is the same, that is, an M/M/1 queue system. In this case the Application Processor (AP) is the server and it copies the data packets uploading.

4.2 Parameters used (GPRS)

As far as the parameters used to model the GPRS data traffic are concerned a GPRS rate of 48 Kbit/s was considered and the rate of the serial channel is 57.6 Kbit/s. In this way the external traffic does not have a higher rate than the serial channel rate. In fact, to validate our simulator it is not necessary to “flood” the channel. The GPRS traffic was modeled as an ON/OFF source traffic. Hence, the peak dimension of the packet needs to be set. The PPP packet dimension is fixed to 1000 bytes. Naturally this choice is related to the validation processing. In fact, as said above, we do not “stress” our simulator to verify its validity. Finally, an average service time and an average arrival time were chosen. The first one is fixed at 16.16 sec. This value is due to the GPRS rate of 48 Kbit/s. The second parameter was chosen to 41.2 sec. (this value was established in the section below). The GPRS parameters chosen are the following: (1) GPRS speed = 48000 bit/sec (2) Channel Speed = 57600 bit/sec (3) Packet Dimension = 1000 byte (4) T_{ON} = 16.167 sec (5) T_{OFF} = 41.2 sec.

5. GPRS and UMTS Traffic Implementation

5.1 GPRS Traffic Model

To generate the GPRS traffic a traffic model had to be introduced to correctly shape the GPRS traffic [9, 10, 11, 13]. When the traffic model is established data traffic can be implemented in our simulator setting the parameters required. As will be seen the traffic generation used, requires fundamental parameters. To set these parameters a random number generator was used in which the number are distributed with opportunistic distributions.

A GPRS source can be modelled as an ON-OFF process, that is a renewal process with two states associated: activity state (ON), inactivity state (OFF). Each time the process changes state there is a renewal, that is, the past time is “forgotten “. The voice sources emitting this type of traffic are called bursty sources and the periods in which the source emits is called burst. Generally the burst has a bounded duration but the bit rate is high. A quantity, called burstiness, is introduced defined as the ratio between the peak of the bandwidth P and the average source bandwidth B. This is an index related to its activity; in fact, if this quantity is high the sources cannot be classified as CBR (Constant Bit Rate), in fact, the CBR sources have a burstiness of 1. The arrival processes from a single voice source ON-OFF can be characterized through an IPP (Interrupted Poisson Process). Hence, this process can be described through a continuous-time Markov chain (phase process) with two states $\{0,1\}$ in which the status 0 corresponds to the status OFF of the source, status 1 corresponds to the status ON, Q = 0 is the bit rate of emittance of the source in the status OFF, P (Peak of Bandwidth) is the bit rate of emittance of the source in the status ON, P = $1/T_{on}$ is the transition frequency OFF-ON, P = $1/Toff$ is the transition frequency ON-OFF, the stay in ON or OFF state is exponentially distributed. Hence, in the ON interval of time the source transmits packets with a time exponentially distributed with an average time P. The transition rate matrix Q is a 2X2 matrix in which the generic element Q[i,j] is the transition rate from status i to status j. Let p be the transition rate from status 0 to status 1 and q the transition rate from status 1 to status 0, the transition rate matrix is defined as:

$$Q = \begin{bmatrix} -p & p \\ q & -q \end{bmatrix}$$ in which

$$\sum_{j=0}^{1} Q_{[i,j]} = 0$$ (in the continuous time)

Hence the IPP is an interrupted Poisson process: the interruption is present when the Markov chain passes from status 1 to status 0. When the Markov chain is in the status OFF it does not emit, otherwise if it is in the status ON it emits with a constant bit-rate P. Finally, this model approaches a GPRS ON-OFF source in the continuous-time domain. The ON period is considered as the period to download a document, as a Web page, an e-mail, or a file; the OFF period is the reading time in the packet call. A session begins with an ON period. The average duration of a period is defined through the ratio:

$$T_{OFF} = \frac{1}{\mu_R} = 41.2 \qquad (3)$$

The average duration of an ON period is computed considering an average number of packets in a packet call equal to 25 and an average inter-arrival among the packets equal to 0.5 sec based on the formula:

$$T_{ON} = \frac{1}{\mu_C} = \frac{PN_p}{Speed} + \mu_D N_p \qquad (4)$$

The GPRS data are "tunneled " in a PPP packet. The PPP protocol (Point to Point Protocol) [7, 8] is used to transport IP traffic over a point-to-point connection. It uses an ISO standardized structure frame.

5.2 UMTS Traffic Model

The UMTS traffic model considers the generation of ON periods, in which information packets are generated, and some reading periods among these ON periods. As considered above it is possible to model the traffic as an ON-OFF source. The same considerations made about the GPRS traffic can be made about the UMTS traffic. The main difference between the GPRS and UMTS traffic is the parameters choice characterizing the traffic. In effect the UMTS traffic is related to a higher speed than the GPRS traffic. The average number of packets generated in a T_{ON} period is 25. To the contrary the reading time is lower and it is set to 30sec. The inter-arrival between two packets sent is distributed as an exponential distribution and the average value is 0.125 sec. By Analogy with the GPRS the T_{ON} value is computed using the (4) formula and substituting the UMTS speed.

6. GPRS and UMTS Traffic Generations to Validate the Simulator

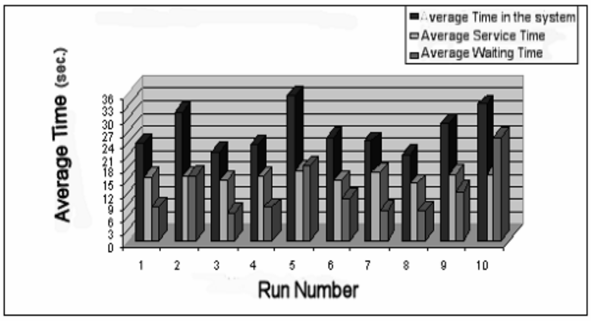
6.1 GPRS simulation campaigns and parameters used

Our simulations were conducted applying two different methods to analyze the output: batch means method and repetition method. As far as the two methods considered are concerned we have:

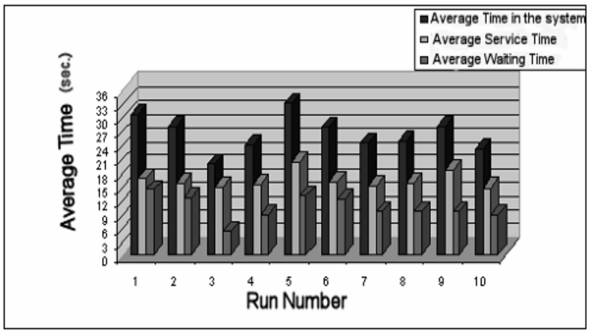
- *independent repetitions method*: 10 runs were made to simulate a GPRS connection whose duration is 15000 sec.;
- *batch means method*:, in this case, only one GPRS connection was considered whose duration is 200000 sec. In this way there is a high number of batches.

In both of the methods some important parameters were measured, considering the arrival of a new packet call as an event. The parameters considered are: the arrival instant to the service station (MT), the inter-arrival, the number of packets in a packet call, the service time, the departure instant from a station (MT), the waiting time in the queue and the time spent in the system (buffer and MT). Considering the independent repetition method the average values for each parameter were computed in each run and successively the average value was computed of the average values computed in each simulation run. When the batch means method was considered the average values for each parameter on each batch were computed and successively the average value was calculated of the average values of the batch. In both the methods the typical parameters of the queue systems were considered: throughput, utilization, delay in and number in. Considering some values:

- **time:** current value of the simulated time. The simulation is considered from 0;
- **sbt (sum of busy periods)**: sum of the activity periods of the server;
- **noc (number of completions)**: jobs number completed to the station;
- **stl (sum of time length)**: time spent by the job in the station.



a) MT (200 Observations)



b) AP (200 Observations)

Figure 2. a) MT (200 Observations), b) AP (200 Observations.)

Hence in a run or in a batch the following indexes can be considered:

- *utilization = sbt/time*: is the fraction of the time in which the server is distributing a service;
- *throughput (or productivity) =noc/time*: is the productivity of the server, that is, the jobs number completed at the station with respect to the total time;
- *number in = stl/time*: users number in the station;
- *delay in = stl/noc*: is the time spent by a job in the station.

In Figure 2 (a) and b)) the average value obtained for each parameter can be seen: service time, time spent in the station, and waiting time. The same type of analysis was conducted using the batch means method. Naturally in this latter case only one run was considered and the average values were computed on the different batch. The results are shown in the Figure 3 and Figure 4. To validate the results obtained through our simulator an already validated simulator used to model M/M/1 queue systems was considered .

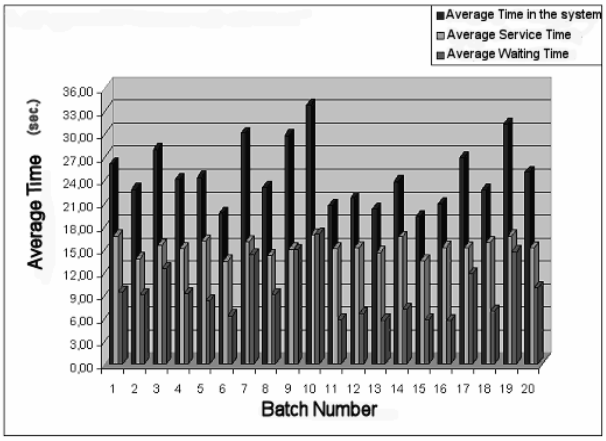


Figure 3. Average Times for MT Batch (150 Observations).

The simulator receives some parameters as input that are: the average rate of arrivals and the average rate of service. The parameters used in the M/M/1 simulator are: 1) Average Arrival Time $1/T_{OFF}$ = 0,02427 sec, 2) Average Service Rate $1/T_{ON}$ = 0,06185 sec, 3) Simulation Duration = 15.000 sec. The simulator considered (to validate our simulator) computes the confidential intervals of 90 % and the performances indexes of the system. In the Table 1 the output results of the simulator can be observed as far as the serial channel of the SmartPhone is concerned, evaluated both through the independent repetitions method and the batch means method. Similar results were obtained (also considering our simulator), hence our simulator can be considered validated. In fact, it was already demonstrated above how the M/M/1 queue is the correct model to represent the buffer of MT and the buffer of AP. In the Table 1 the values in the first column represent the confidential interval of 90% of M/M/1 simulator.

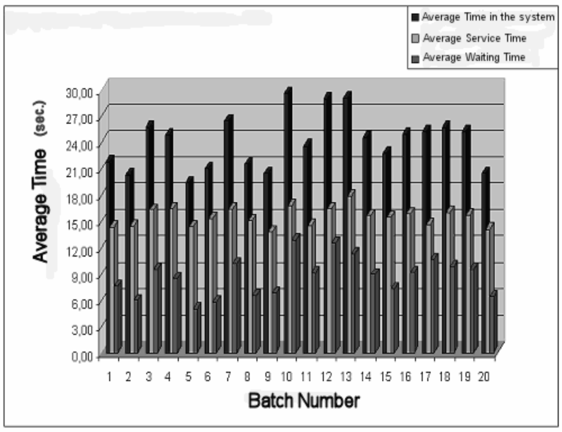


Figure 4. Average Times for AP Batch (150 Observations)

Table 1: Final Results (GPRS): IR is Independent Repetitions, BM is Batch Means

	M/M/1 Simulator	MT (IR)	MT (BM)	AP (IR)	AP (BM)
Productivity	[0.0235,0.0242]	0,0243	0,0246	0.0244	0,0249
Utilization	[0.3808,0.3985]	0,3807	0,3728	0,393	0,3846
Number in	[0.5893,0.6400]	0,6547	0,6126	0,6442	0,6005
Delay in	[24.848,26.637]	26,788	24,705	26,394	24,03

6.2 UMTS Simulation Campaigns and Parameters Used

In this case the parameters were used to model the UMTS data traffic executing the simulation runs. In each simulation run only one UMTS connection is executed. The connection rate used is 384 Kbit/sec and the serial channel rate is 230.4 Kbit/sec. The same as above the UMTS traffic was modelled as ON/OFF source. The parameters used are shown in the Table 2 .

Table 2: Validation Parameters (UMTS)

UMTS Rate	384000 bit/s
Channel Speed	230400 bit/s
Packet Dimension	1000 byte
Ton	3.521 s
Toff	30 s

Also in this case both the methods considered above were used: the independent repetitions method and the batch means method. In both important parameters were measure considering the arrival of a new UMTS packet call as an event. The parameters considered are: arrival instant to the service station (MT), the inter-arrival, number of packets in a packet call, the service time, the departure instant from the station (MT), the time spent in the system (buffer + MT) and the queue waiting time. To validate the simulator with the UMTS traffic, a comparison was made between the results obtained through both of the methods and the results obtained applying the same input parameters to another already validated M/M/1 simulator. This latter simulator receives the average arrival rate and the average service rate of the UMTS connection as input parameters. The parameters considered for M/M/1 Simulator are: 1) Average Arrival Time $1/T_{OFF}$ = 0,0333 sec, 2) Average Service Rate $1/T_{ON}$ = 0,06185 sec, 3) Simulation Duration = 15.000 sec. The same as above this simulator computes the confidential intervals of 90% and the performances indexes. In the Table 3 the results obtained are summarized. In the Table 3 the values in the first column represent the confidential interval of 90% of M/M/1 simulator. The results obtained through our simulator, considering both of the methods and the results obtained with the already validated simulator are similar. The same consideration made above is valid here: an M/M/1 queue model is a correct model to represent both the MT-buffer system and the AP-buffer system. The simulator designed therefore correctly represents the AP-Serial-Channel-MT system also in a UMTS connection.

Table 3: Final Results (UMTS): IR is Independent Repetitions, BM is Batch Means

	M/M/1 Simulator	MT (IR)	MT (BM)	AP (IR)	AP (BM)
Productivity	[0.0328,0.0338]	0,0317	0,0339	0.0336	0,0342
Utilization	[0.1142,0.1195]	0,1076	0,1112	0,1174	0,1148
Number in	[0.1282,0.1354]	0,6547	0,1269	0,134	0,1298
Delay in	[3.8881,4.0156]	0,3848	3,7416	4,0039	3,7872

7.GPRS and UMTS Traffic Generations to Validate the Simulator

Different simulation campaigns have been conducted to understand if our device can support either GPRS and UMTS data traffic and to understand the limits of the proposed architecture. For this reason different applications (a simple call, sms, GPRS connections, UMTS connections etc.) were considered. The average duration of the call was set to the value of 600 sec. As far as the sms is concerned it is not possible to set any value and we associated the generation of the sms with a random number generator. As far as the GPRS parameters are concerned we considered the parameters in the Table 4. Here the Priority Dynamic Scheduler introduced above is considered and we assigned the following values:

- priority assigned to the data connections: 0.80;
- priority assigned to the voice calls and SMS: 0.20

Naturally there is a portion of bandwidth assigned to the control channel that is equal to 250 bit/sec for each channel used. As far as the bandwidth required to maintain the voice calls or to send some sms is concerned there is a small quantity of bandwidth required by these services. The GPRS connections require more bandwidth. When the bandwidth requirement increases lost packets increase too. In fact, this parameter was measured in the Figure 5 . Naturally, applications such as those considered here (GPRS connections) cannot be characterized by the lost packets. Above all, this is not acceptable if only a single GPRS connection and the basic functionalities of the telephone are considered. In fact, in this way the error in the download phase happens in the communication phase between the two processors and not in correspondence to the external network. For this reason a similar error (due to the lost packets) is not acceptable.

Table 4: Parameters used in a GPRS campaign

Number of Connections	Variable between 1 and 3
GPRS Connection rate	48000 bit/sec
Maximum PPP dimension	1510
Inter-arrival among the packets in a TON	0.5 sec.
Duration of a data connection	15000 sec.
Average number of packets in a TON	1000 byte
Number of simulation run	10
Number of samples	500 in each simulation
Serial channel rate	115200 bit/sec
TON	16.167 sec
TOFF	41.2 sec

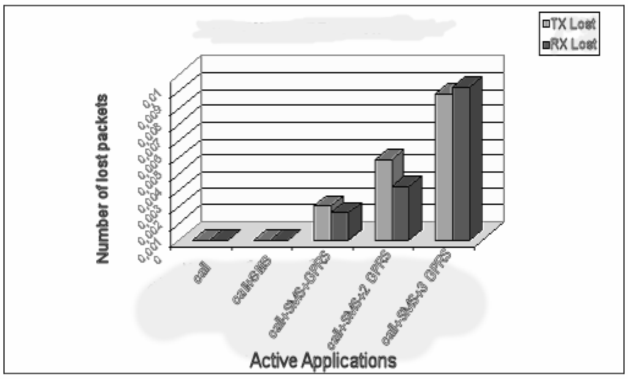


Figure 5. Average Number of Lost Packets (115200 bit/sec)

This implies that a higher serial channel rate of the channel between the AP and the MT, higher than 115200 bit/sec (i.e. 230400 bit/sec) has to be considered. For this reason identical simulation runs like the above considered were considered and 3 simultaneous GPRS connections were opened and the channel rate considered is 230400 bit/sec. In the Figure 6 the results are shown as far as the lost packets with the parameters considered are concerned. The lost packets have diminished but there are also some lost packets. In order to avoid this the parameters were analysed characterizing the previous simulation runs. A GPRS data traffic was introduced in which the dimension of the packets considered is the maximal dimension that can be considered as far as the PPP protocol is concerned, that is, 1510 bytes. In this way, a dimension buffer such as those considered, 2048 bytes, is sufficient to manage only a single PPP packet. It was decided to dimension the buffer dimension differently. Hence, a dimension buffer of 3600 bytes was considered (Ubinetics considers a similar dimension for its GPRS modem [14]). The other parameters were unchanged. It can be observed that there is not a substantial difference in the two cases considered. The more important results are the lost packets shown in the Figure 7. In fact, using a buffer of 3600 bytes the lost packets prove to be null. Naturally, a higher dimension buffer is necessary when a PPP dimension packet of 1510 bytes is considered. If 4 simultaneous GPRS connections and a dimension of PPP packets of 1000 bytes and the dimension buffer of 2048 bytes are considered there are no lost packets.

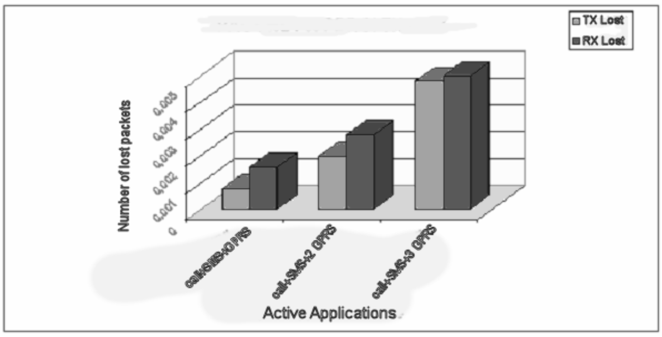


Figure 6. Average loss of packets (230400 bit/sec)

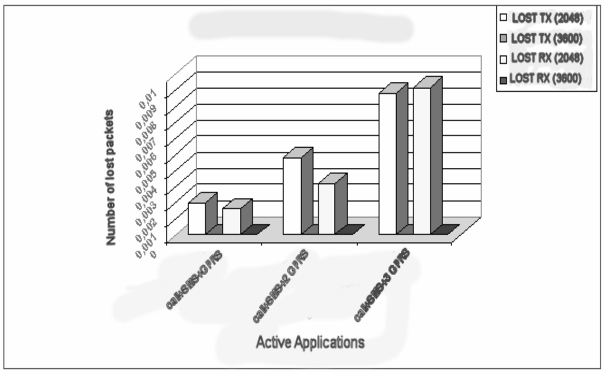


Figure 7. Differences between lost packets for each champion

7. Conclusions

The design of a new type of SmartPhone is presented in this paper: a bi-processor SmartPhone. A serial connection to make the communication between two processors based on different operating systems has been considered. After studying the standard to use for this type of connection an event-discrete simulator was developed to simulate the two processors, but above all, to simulate the serial channel. This is to evaluate whether the “modern” type of data traffic can be supported by this type of architecture. In fact, a model was developed to shape both the GPRS and UMTS traffic and based on this model the data traffic was developed and this was considered as input in our simulator. The results obtained can be used as guidelines to dimension the buffer in transmission and in reception and demonstrated that the architecture considered is extremely valid if the dimensioning are correctly made.

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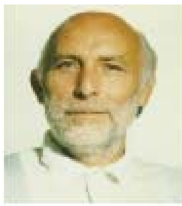

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