URAE: A TOOL TO ANALYSE PACKET SERVICES BASED APPLICATIONS IN UMTS

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Abstract - Advanced strategies must be developed to provide packet services based applications in 3G mobile systems. This article presents a tool to evaluate the radio access impact in packet services based applications within a UMTS context. The tool architecture is described in detail and the trials that can be performed are explained. Some results are also shown.

Keywords – Radio Interface, UMTS, Packet Services.

I. INTRODUCTION

The Internet application's attractive supply and the large amount of information existing in the Internet have promoted the demand on the Internet access from/to mobile wireless users. Moreover the transmission of high-quality images as well as the multimedia data transmission demand has increased in the recent years. Besides it is foreseeable that the variety of wideband services will continue to increase. The convergence of mobile and multimedia services over Internet is one of the main features of third generation (3G) mobile systems. Data and multimedia capabilities improvement is one of the most important targets in 3G mobile communications systems, apart from the improvement in spectral efficiency.

The Universal Mobile Telecommunications System (UMTS) is the 3G mobile communications standard proposed by the European Community. It is expected to provide a high data rate services (up to 2 Mbps) and to use these rates to support a wide range of services including multimedia applications. The UMTS universality is given by the expected terrestrial (T-UMTS) and satellite (S-UMTS) coverage, and by the operation from outdoor to indoor and from stationary to vehicular mobility.

Moreover to take advantage of frequency spectrum, the 3rd Generation Partnership Project (3GPP), who specifies UMTS, has described two transmission duplex modes: Frequency Division Duplex (FDD), for operating with paired band, and Time Division Duplex (TDD), for operating in unpaired band.

UMTS must be a multiservice platform, and consequently high and low rate users will coexist in the network. Besides, asymmetric data transmission and variable rate services can also be considered. To achieve that target packet data services and efficient radio resources management algorithms are essential.

The easy way to import the services already present in fixed networks to the mobile network would be to analyse the drawbacks of the protocols used in those networks and try to improve them so that they can be used in mobile networks. Even though that work can not be quickly, to develop advanced strategies starting from scratch will surely be a harder task. Thus a contribution to make easy that work has been done.

This paper presents a detailed description of URAE (UMTS Radio Access Emulator); a tool made to evaluate the radio access impact in packet services based applications. It also presents the study of what could be a real case. Although this tool is devoted to FDD mode it could be easily adapted to TDD mode. Starting from the results of this evaluation, appropriate advanced strategies should be defined. This last step is not included in this text.

The organisation of the paper is as follows. In section II there is a detailed description of the URAE. In section III the trials that can be performed are explained, and some results are shown in section IV. Finally some conclusions can be found in section V.

II. TOOL DESCRIPTION

A tool to evaluate packet services based applications behaviour in terrestrial UMTS has been developed. This tool, called URAE, reproduces the radio interface lower layers, that are the data link layer (L2) and the physical layer (L1), and also the digital radio channel through the radio interface (Uu) (see Fig. 1).

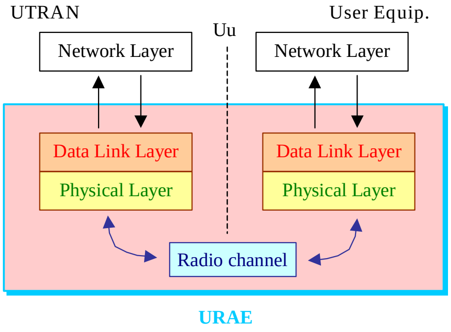

Fig. 1. Radio Interface Emulator.

To use the URAE, it is necessary to have Network Layer Packet Data Units (PDUs) at the input. These packets are processed as they were transmitted through the air interface, and they are delivered to the output (other end of transmission). For each layer the input packet data is called Service Data Unit (SDU). Thus from the emulator point of view the Network Layer PDUs are the SDUs.

Network layer protocols as Internet Protocol (IP) can be directly studied with this tool. However, to study higher layer protocols as TCP an intermediate element must be introduced to reproduce the network layer.

A. The physical layer & the radio channel

The physical layer is in charge of sending and receiving the information to the air interface, that is to the radio channel. To build the physical layer blocks and the radio channel a Wideband Code Division Multiple Access (WCDMA) scheme has been simulated following the 3GPP technical specifications [1].

To emulate these parts an entity based on Hidden Markov Models has been made. This entity has been called Radio Channel Emulator, see [2]. To train the HMM a statistical system behaviour is needed, so that, off-line simulations have been made. This emulator based on HMM has been extensively checked and validated. The validation procedure consists of analysing the features of the off-line simulated channel and comparing them with the features of the HMM emulated channel. In Fig. 2 a validation result example is shown.

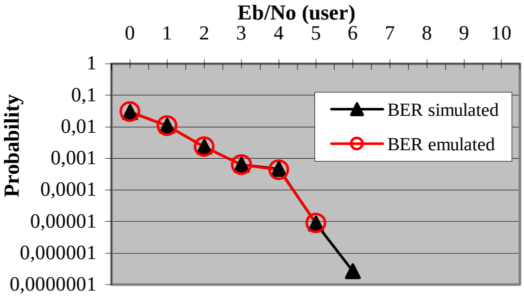

Fig. 2. BER simulated vs. BER emulated.

This sample shows the mean Bit Error Rate simulated by means of off-line simulations, and the mean Bit Error Rate emulated by the HMM emulator. The HMM huge level of accuracy is clearly shown. Apart from mean BER, the error probability distribution, the error burst length, and the Frame Error Rate (FER) have been large validated for the different emulated channels. The use of HMM allows to the emulator to operate in real time.

As regards channel coding, two different performances have been considered for this Emulator. The first one does not consider the use of channel code (apart from scrambling and channelisation codes, of course), and the second one considers the use of the convolutional codes proposed by 3GPP in Technical specification 25.212 [3]. The second option is obviously better than the other. The first option has been developed just to demonstrate the importance of channel coding within WCDMA systems. Moreover, the first option of URAE can be used to determine performance by using different channel codes. In section IV some of these results are shown.

B. The data link layer

On the other hand, the data link layer has an important role in the radio resources allocation, and data flow control. To fulfil all the corresponding functions the data link layer has been split in four sublayers: the Medium Access Control (MAC), the Radio Link Control (RLC), the Packet Data Convergence Protocol (PDCP) and the Broadcast/Multicast Control (BMC) as illustrated in Fig. 3 [4].

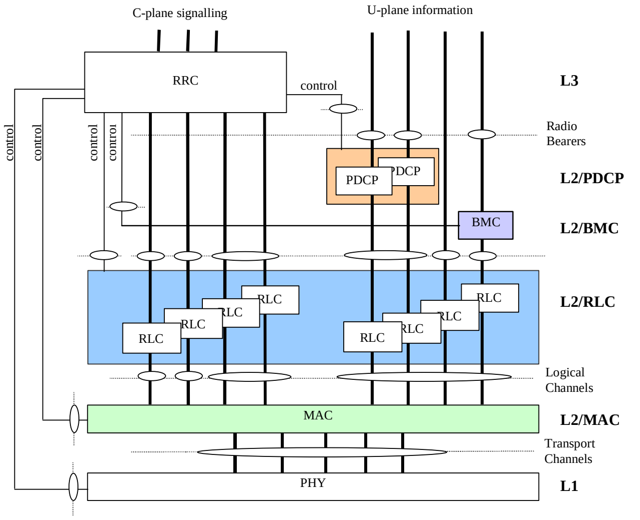

Fig. 3. Radio Interface Protocol architecture.

For the tool presented in this paper not every data link sublayer has been studied to emulate that layer. In fact, in phase 1, the made emulator will only allow to analyse information in dedicated mode from a single user, thus the Broadcast/Multicast Control (BMC) block has been left to one side.

As regards Packet Data Convergence Protocol (PDCP), it has been partially emulated. The non compression mode has been reproduced as a first step, and currently two header compression methods are under study to improve channel efficiency.

Due to its implication in dedicated mode operation, the Medium Access Control (MAC) and the Radio Link Control (RLC) have been developed in more detail.

The MAC layer is charged to manage the medium access. To that end different scheduling algorithms have been studied. To allow real time operation in the tool the radio resources management strategies have not been implemented directly, but statistics. These statistics have been obtained with off-line simulations that take into account different users load in different environments.

The RLC layer must carry out segmentation and retransmissions services. It can be configured to operate in one of the three data transfer modes:

- Transparent Mode (TM): transmits PDUs without adding any information, only limited segmentation/reassembly functionality.
- Unacknowledged Mode (UM): transmits PDUs without guaranteeing delivery to the peer entity. No retransmission protocol is used and only sequence numbers are added to correctly concatenate the RLC PDUs.

- Acknowledged Mode (AM): transmits PDUs and guarantees delivery to the peer entity. In case the RLC is unable to deliver the data correctly, the user of RLC at the transmitting side is notified.

Although 3GPP has proposed a relationship between these modes and the types of services (see table 1), this tool allows the whole combinations. For example, an interactive service can be analysed over RLC AM, over RLC UM or over RLC TM.

Table 1. Type of traffic and RLC operation mode.

Traffic type	RLC operation mode
streaming	TM
conversational	UM
interactive	AM
background	AM

III. TRIALS

Several trials can be performed with this tool according to the selected parameters. Really, the emulator has different parameters to be selected, and the combinations of their values allow to analyse different services, or the same service under different conditions.

The parameters can be separated in different levels as follows:

1.- Environment parameters: these are parameters related with the environments where the emulator can operate. They are:

- Mobile speed: 120 km/h or 50 km/h (both outdoors)
- System load: High (there are lots of active users), Medium or Low (there are few active users).

2.- System parameters: these are parameters related with the internal system operation such as:

- RLC sublayer data transfer mode: AM, UM or TM.
- Channel coding use in physical layer: On or Off.
- Closed loop power control: Yes or No.

3.- Service parameter: it is a parameter related with the studied service. It defines four traffic classes and it is called:

- Traffic type:
  - conversational (voice service,…),
  - streaming (video or continuous data transfer),
  - interactive (web browse,…) or
  - background (e-mail service,…).

Although the use of channels with closed loop power control can be selected in each trial, the relationship between traffic type and channels shown in Table 2 has been considered. That is, for real time services as voice services or videoconference, and for services that generate a huge amount of information as video streaming, web browse or continuous data transfer, a dedicated channel will be assigned, whereas for applications that generate few information as could be a simple e-mail, data packets will be sent through common channels Random Access Channel (RACH) in the uplink and Forward Access Channel (FACH) in the downlink.

Table 2. Type of traffic and channels relationship.

Traffic type	Channel
conversational	DCH
streaming	DCH
interactive	DCH
background	RACH/FACH

RACH/FACH channels do not use closed loop power control, and Dedicated channels (DCH) do.

To define a trial, each one of these parameters must take a specific value. For example to compare the behaviour of an e-mail application for a user travelling at 120 km/h and travelling at 50 km/h, a value must be set for the other parameters. A right option could be: System Load = Medium; RLC mode= Acknowledged Mode (AM); Channel coding = On; Traffic type = background.

The system load medium in this situation means around 25 e-mail users in the downlink and 50 e-mail users in the uplink, plus a similar number of interactive users and 1 streaming user. The traffic models used are described in [5].

Apart from the data standard output, all the trials provide another output including statistics of the current execution. These statistics are:

- Bit Error Rate (BER) = error bit / total transmitted bit. Mean, Instantaneous and Delta are computed.
- BLoc Error Rate (BLER) = error bloc/total trans. bloc. Mean and Instantaneous are computed.
- Frame Error Rate (FER) = error frame/total trans. frame. Mean and Instantaneous are computed.
- Delay = time elapsed from the instant that the SDU is transmitted until the instant that it has completely received. This delay does not include the latency. Mean, Instantaneous and Delta are computed.
- PDU correctly transmitted / PDU erroneous
- SDU correctly transmitted / SDU erroneous
- Bits correctly transmitted / Bits erroneous
- Spreading Factor distribution
- Eb/No distribution

Where Frame refer to the SDU (1500 bytes IP packet), Block refers to the radio block, that is the amount of information sent within 10 ms, and PDU is the packet sent between Data Link layer and Physical layer. In next section some of these values are shown.

It is important to point out that the URAE can accept UP and DOWN data simultaneously, and different radio bearers. If real time execution is required, the number of bi-directional radio bearers is limited to three. That means that a trial with a radio bearer carrying interactive traffic, a radio bearer carrying streaming traffic, and a radio bearer carrying background traffic, can be made.

IV. RESULTS

As it is explained in previous section a wide range of studies can be performed with the URAE, in this section two studies are presented. The first one makes reference to the need to use channel codes, and it compares the two mobile speed channels. The second one is a comparison between two traffic types under what could be real conditions.

Analysis 1:

One interesting result obtained through the trials is the study of the Frame Error Rate (FER). In packet services the FER is an important Quality of Service (QoS) indicator. For example, a FER lower than 10% is desirable for an application using streaming traffic. With the URAE it is easy to check that channel codes are essential to fulfil that requirement.

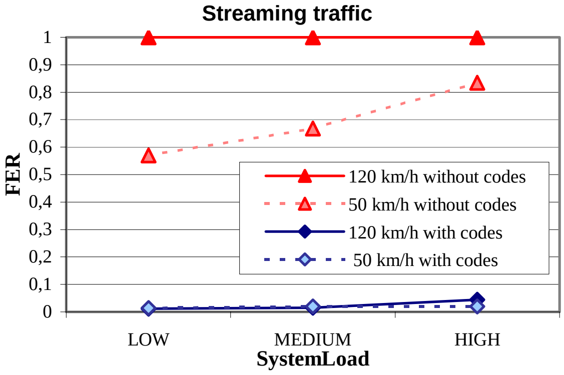

Fig. 4. FER with/without codes for uplink streaming traffic.

In Fig. 4 the results belonging to streaming traffic in the Uplink are shown for 120 km/h and 50 km/h environments. A streaming data rate of 100 kbps has been considered in this analysis.

When the mobile speed is higher the radio channel changes faster, so there are more errors. From Fig. 4, and watching the "…without codes" lines, this fact can be observed. Really it is impossible to transmit streaming traffic without channel codes at 120 km/h, the FER is always 1. However, it is possible at 50 km/h, but in not very good conditions. So, it can be reassert that channel codes are essential in UMTS radio channels as it is well known.

Although this analysis does not directly contribute to UMTS development, it has been included just to show that similar analysis can be done in order to compare different channel codes behaviour such as Turbo codes and convolutional.

Analysis 2:

Let us go to compare interactive traffic behaviour with background traffic behaviour. Take for example mobile speed 120 km/h, using channel codes, with RLC acknowledged mode and the Downlink.

To perform this analysis the same data rate has been applied to the two traffics, even though background traffic should be lower than interactive traffic in a realistic situation.

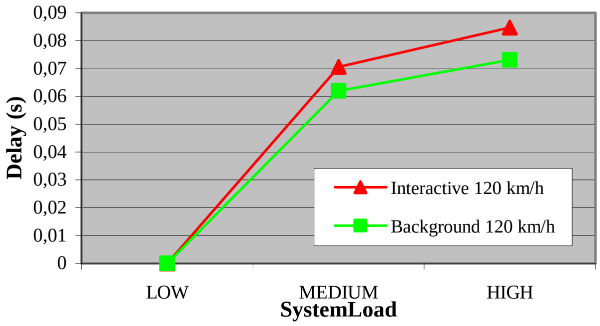

Fig. 5. Downlink delay

In Fig. 5 the delay is presented. Notice that it is higher for interactive traffic than for background traffic. Although this seems erroneous, or almost not logical this result is correct and can be justified looking Fig. 6 to 8.

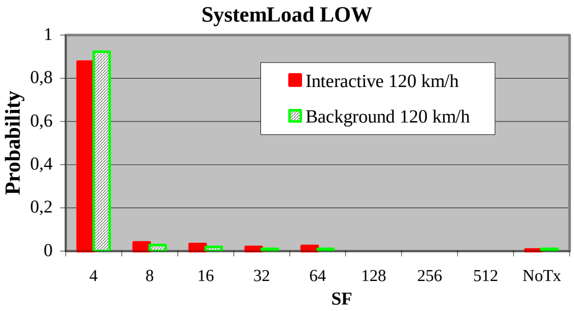

Fig. 6. SF distribution for System Load Low.

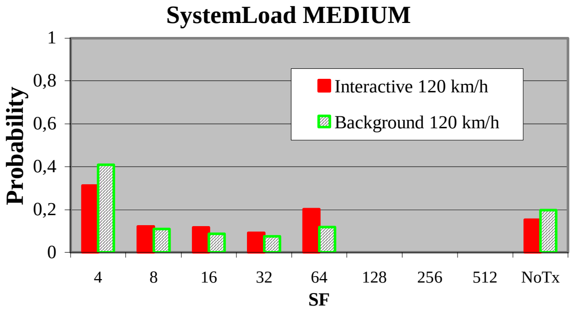

Fig. 7. SF distribution for System Load Medium.

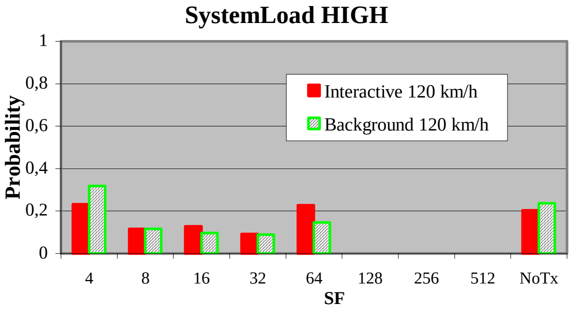

Fig. 8. SF distribution for System Load High.

In these last three figures the SF distribution for the reference user is shown in the different system load levels. The interactive distribution and the background distribution are similar, but the background has always a higher probability in SF=4. That means higher data rate for background traffic than for interactive traffic. This justifies the delay graphic.

In fact, the average bit rate can be calculated through these SF distributions. The results are shown in table 3.

Table 3. Average bit rate

Bit rate (Kbps)	Interactive	Background
LOW	465,685	483,095
MEDIUM	218,959	257,601
HIGH	178,354	214,821

Still with the same example, if we look at the FER background case is worst than interactive case, see Fig. 9.

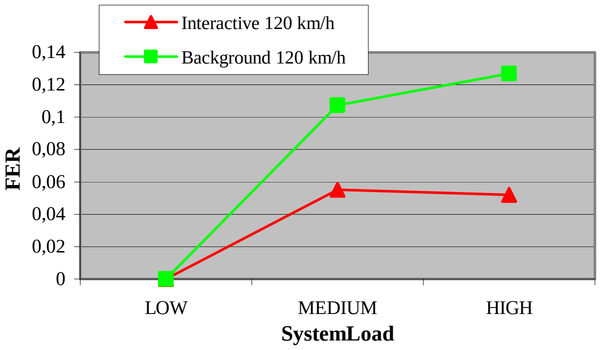

Fig. 9. FER comparison

The key is the power control. We have considered that background traffic, for example an e-mail application, is transmitted through RACH/FACH channels, whereas interactive traffic is transmitted through dedicated channels (DCH). The former channels have not closed loop power control, and thus they introduce more errors than the DCH channels that have closed loop power control. So, if the BER is higher the FER is as a consequence also higher.

V. CONCLUSIONS

In this paper a tool to analyse packet services based applications in UMTS context has been presented. This tool is called URAE, which stands for means UMTS Radio Access Emulator. A detailed description about the URAE has been done, including the trials possibilities, and two analysis examples of what could be a real case have also been presented.

The URAE has been conceived as a key tool within the convergence of mobile and multimedia services over Internet. There is no doubt about the importance of Internet access through wireless connections. It is clear that advanced strategies must be developed to combine these two markets. To design new strategies the drawbacks must be studied, so this tool has been developed.

An important feature of the URAE is the possibility to operate in real time. Apart from the URAE description, the configuration parameters have also been mentioned and the different trial that can be performed have been explained.

Finally in order to show some results two examples with the URAE have been analysed. The first analysis is devoted to physical layer, and in particular to the necessity of channel codes in applications that manage streaming traffic. From the results it is clear that channel codes are essential to use that kind of applications over a wireless connection, otherwise the frame error rate (FER) is so worse that application does not work.

The second analysis is devoted to compare the behaviours of interactive traffic and background traffic. The delay, the FER, and the SF distribution have been compared and commented. A higher delay and a lower FER have been obtained for the interactive traffic as they are justified.

In short, a flexible tool that operates in real time and emulates the UMTS radio access for the FDD mode has been presented. This tool is called URAE and it serves to analyse packet data services under emulated real conditions.

VI. ACKNOWLEDGEMENTS

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