Packet Data Services over Wireless Connections Study

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***Abstract* – This paper presents a study[1] of packet data services over wireless connections in the context of Universal Mobile Telecommunication System (UMTS). To perform the study an emulator of the radio access part has been made which includes the mobile terminal and the UMTS Terrestrial Radio Network (UTRAN). The main features, the validation procedure and some results are contained.**

I. INTRODUCTION

It is well-known that Internet growth has been linked together with packet data services. These services have been large developed through years, and currently there is a huge amount of applications that use packet technology as bearer service. Moreover new applications that use packet data services appear every day. This situation gives good future expectative to packet data services.

Parallel to the mentioned situation, mobile systems expansion has produced a new added value to the communications market: the availability to connect with anyone, from anywhere and at any time.

It is foreseeable that next phase includes the fusion of these two giants. In fact, the support of Internet applications over mobile systems will offer connection to anyone, from anywhere, at any time and to any information. This gives a wide range of facilities to the users, and some technological challenges to the engineers, as it could be the use of packet data services on third generation mobile systems.

Our research work has been focused on this challenge in the context of the Universal Mobile Telecommunication System (UMTS), the European standard for third generation mobile systems [1]. Thus packet data services over wireless connections has been studied in order to determine their behaviour within the third generation mobile systems. The aim of this study is to be a contribution to the new services and applications integration within UMTS.

In this paper this study is explained. First of all the research work and the study usefulness are presented in section II. Then the different configuration options are detailed in section III. Next some validation results are shown in section IV, called the validation procedure. After this several analysis examples are included in section V. And finally the conclusions close the paper.

[1] This research was supported by CICYT project TIC 2001-2222.

II. RESEARCH WORK

Packet data services use a set of well-defined protocols when work over fixed networks. These protocols have been developed for fixed links and thus they do not work properly over wireless connections. Packet loss in fixed networks is interpreted as a sign of congestion, since errors are infrequent, and gives rise to congestion control mechanisms. However in the radio interface, packet loss can be produced by an error independently of network congestion. Problems arise since protocols do not distinguish between congestion and packet loss due to radio channel errors.

The first research challenge has been to study the radio access influence on protocols operation, and to achieve that target, a tool that emulates this access has been developed.

Concerning the Internet protocol stack, the packet management throughout the network is allocated in the Network layer, where Internet protocol (IP) is applied. So this study has been performed from Network layer point of view, it means that the Data Link layer (Medium Access Control: MAC, Radio Link Control: RLC…) and the Physical layer, as well as the digital radio channel have been emulated.

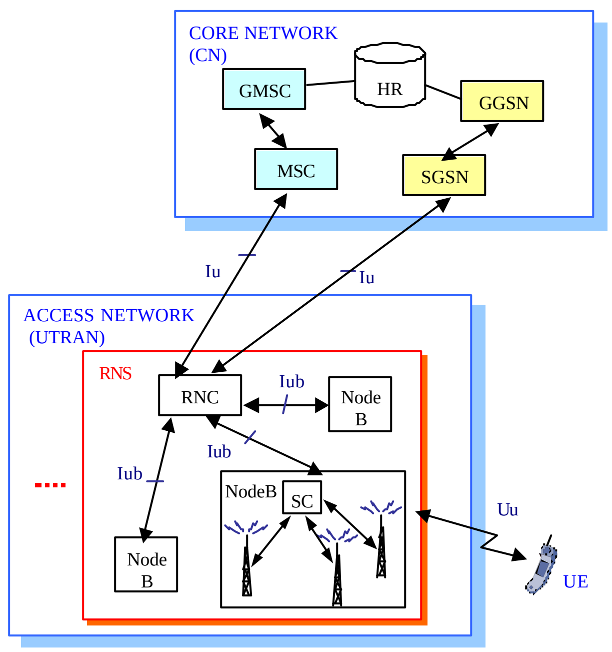

Fig. 1. Context Architecture.

The packet service behaviour when it works over wireless UMTS connections has been observed through the analysis of different parameters such as delay, frame error rate (FER) or average bit rate.

The present study has been done over the radio interface part of the UMTS Terrestrial Radio Access (UTRA) corresponding to the Frequency Division Duplex (FDD) mode, and in particular, between the two entities that borders the radio interface (Uu), see Fig. 1.

The main target of this study is to build an architecture that allows analysing different UTRAN configurations and their impact on the system performance. So the study and work on packet data services over wireless connections has several usefulness.

On the one hand, it serves to analyse the radio interface impact on different higher layers protocols (i.e. IP, Transport Control Protocol-TCP,..), as it was mentioned.

On the other hand it serves to study radio resources management (RRM) algorithms. In fact, due to the importance of Quality of Service (QoS) in UMTS future services, the use of an optimised RRM algorithm is a key aspect under study. The developed emulator has been prepared to work with whatever RRM algorithm.

Moreover the use of retransmission modes could be analysed for different services, only the configuration parameters must be properly selected.

The UMTS operation modes suitability depending on the amount of information could be also checked. That is, send information through a physical dedicated channel or through a common physical channel.

Next section describes the different options supported in this study. Apart from the research tool flexibility, one important feature is the real time operation, which allows to study real applications such as they work over the real world.

III. Configuration Options

Inside the research framework, a wide range of possibilities can be set so that several scenarios and situations can be studied.

Concerning the environment two radio channels have been emulated, one with mobile speed 120 km/h and the other with mobile speed 50 km/h, and three different system loads have been considered: LOW when there are few active users, MEDIUM, and HIGH when there are lots of active users.

As regards system operation, a retransmission mechanism can be selected if required at RLC level. Thus the transparent mode (TM), the unacknowledged mode (UM) and the acknowledged mode (AM) defined by 3GPP [2] can be reproduced.

Likewise, the use of a convolutional channel encoding, or a specific transport channel used to send the user information can be chosen. The implemented codes are: a convolutional code with rate 1/2 for the common random access channel (RACH), and a convolutional code with rate 1/3 for the dedicated and fast associated channels (DCH & FACH) [3].

TABLE I
QUALITY OF SERVICE (QoS) PARAMETERS

Traffic Class	System Load	FER IP packet	Delay IP packet	Average Bit rate
Streaming	LOW	1%	60 ms (DL) 120 ms (UL)	200 Kb/s (DL) 100 Kb/s (UL)
	MEDIUM	5%	60 ms (DL) 120 ms (UL)	200 Kb/s (DL) 100 Kb/s (UL)
	HIGH	10%	60 ms (DL) 120 ms (UL)	200 Kb/s (DL) 100 Kb/s (UL)
Interactive	LOW	<5%	<100 ms	<200 Kb/s
	MEDIUM	<5%	<250 ms	<60 Kb/s
	HIGH	<5%	<400 ms	<25 Kb/s
Background	------	Best	Effort	-------

DL: downlink, UL: uplink

Moreover several types of packet traffic are supported: streaming traffic, background traffic and interactive traffic. The former is assumed to work without any retransmission mechanism, whilst the two remaining use a selective repeat mechanism. Background traffic is considered as a best effort service used for example by an e-mail application. On the other hand, interactive traffic has a fixed target Frame Error Rate (FER) value, and streaming traffic have a fixed target delay and a fixed target mean bit rate values.

The selected quality of services parameters are shown in Table I.

IV. The Validation Procedure

In order to validate the data link layer of the tool used to carry out this study, two different strategies have been made:

1) For streaming traffic validation, off-line simulations have been performed. These simulations use the radio resources allocation algorithms used to build the emulator MAC part. Their values have been compared with those obtained from the emulator.

2) For Interactive traffic, the results obtained from the tool have been compared with the maximum values specified. And for Background traffic due to the little information about its behaviour, the obtained results have been compared with Interactive traffic.

As it is shown in Fig. 2 to Fig. 4 results obtained by applying both strategies are quite good.

Notice that the FER presented in all the figures of this paper refers to IP packet, that is FER equals to number of erroneous IP packets over the total number of IP packets sent.

On the other hand the delay presented in the results count the time elapsed from the instant that the IP packet is transmitted until the instant that the IP packet has completely

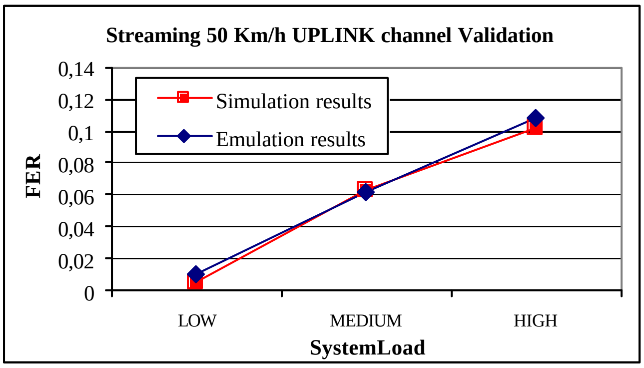

Fig. 2. Streaming Traffic FER validation.

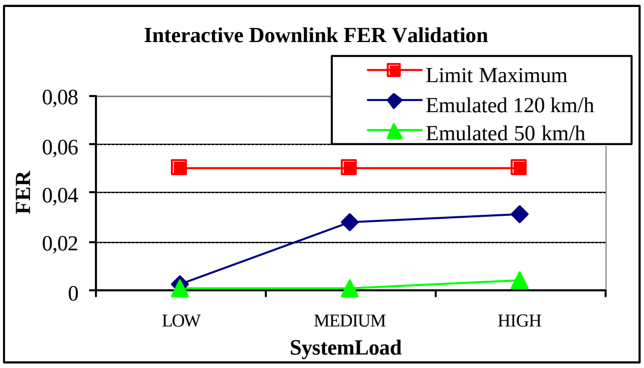

Fig. 3. Interactive Traffic FER validation.

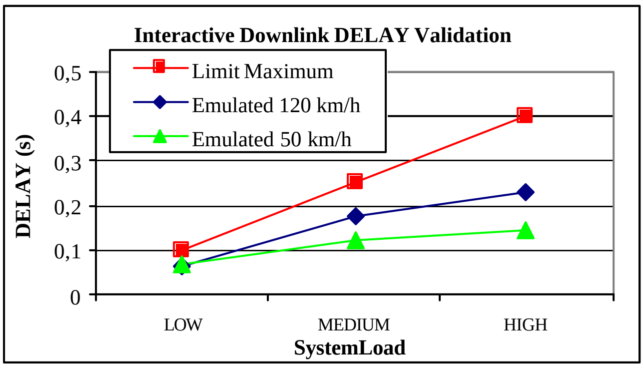

Fig. 4. Interactive Traffic delay validation.

received. So this delay does not include the latency, which describes the delay of a transmission from the time it enters the network until the time it leaves.

The results shown in Fig. 2 belong to streaming traffic FER in an uplink 50 Km/h channel. As it is specified in Table I this value should be 0.01 when the system load is low, 0.05 when the system load is medium, and 0.1 when the system load is high, as it is more or less.

On the other hand in Fig. 3 and Fig. 4 a proof of interactive traffic validation is shown. In these figures the FER and the delay for both environments (50 Km/h and 120 Km/h) have been compared with the limits specified in Table I. It could be shown that maximum limits are kept always in both environments.

The physical layer and the radio channel have been separately validated. For further information see [4].

V. Analysis Examples

In this section several analysis examples are shown in order to present different aspects that could be analysed within this study.

A. Transparent Mode versus Acknowledgment Mode

Two different RLC operation modes are compared in this subsection in order to check the retransmission use consequences. The FER versus delay of this comparison are shown in Fig. 5, the results belong to a 50 Km/h mobile speed channel in the downlink direction.

The interactive traffic printed in Fig. 5 uses the RLC AM, which implements a selective repeat retransmission mode. Thus, in order to guarantee a specific FER value more packet retransmissions are needed when there are more active users in the system, this implies a higher delay to correctly transmit an IP packet.

Otherwise streaming traffic in Fig. 5 uses the RLC TM, which do not use retransmissions. Thus the delay is almost constant, whereas the FER increases when the number of active users increases.

Therefore for applications that require a given constant packet delay retransmissions could not be applied, whilst for applications that have a maximum FER value retransmissions should be applied at the expense of delay.

B. Packet size impact

In this subsection the importance of packet size is commented. In particular 1500 byte IP packet size test have been compared with 500 byte IP packet size test.

As it is shown in Fig. 6 large packets have more difficulties for being correctly transmitted through the air interface. In fact, the probability to have some erroneous bit in a packet

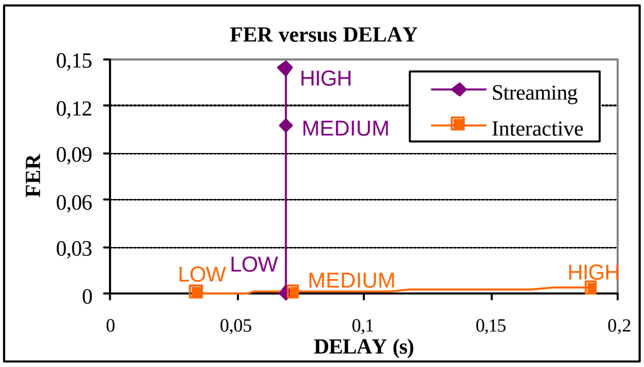

Fig. 5. Retransmission use versus non-retransmission use.

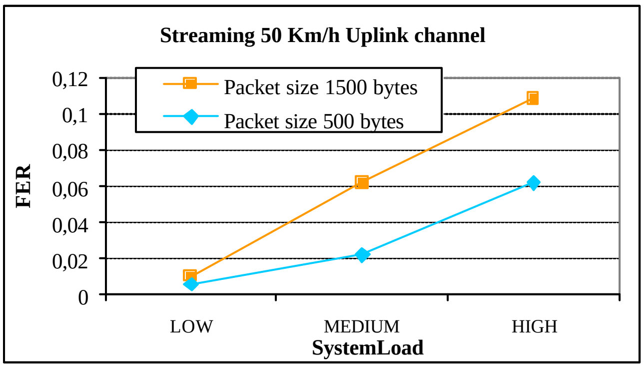

Fig. 6. Packet size comparison.

increases with the packet number of bits.

So applications that generate large packets could have difficulties to be used over a mobile terminal. Several test performed in this study indicate that packets higher than 500 bytes (i.e. 4000 bits) have difficulties to be sent correctly through a wireless connection. To add an extra segmentation element must be a possible solution in order to provide these applications within the UMTS environment.

C. *Mobile speed comparison*

The two different mobile speed channel emulated have been analysed in this subsection. In Fig. 7 the average radio channel Eb/No has been drawn for traffic class type streaming in the uplink channel, and for the three system load situations in the two channels.

Notice that when mobile speed is 120 Km/h the average Eb/No is approximately 3 dB higher than the averaged value in the 50 Km/h channel, so the "fast" speed channel requires an Eb/No higher than the "moderate" speed channel.

In 120 km/h channel fast power control is not able to compensate the channel fading so the performance is worst an a higher Eb/No is required, whereas in 50 Km/h channel fast power control compensates the channel fading and a lower Eb/No is enough.

Moreover it can be observed that Eb/No decreases when traffic increases since the FER requirements are less restrictive.

D. *System Load effect*

Another parameters that may be analysed in this study are the Spreading Factor (SF) and the required Eb/No distributions. In Fig. 8 the SF distribution for a streaming service in a 120 km/h mobile speed downlink channel using convolutional code with rate 1/3 is shown for the three system loads considered.

Obviously when the system has few active users (LOW), the probability to use low spreading factors is the highest, as well as when the system has lots of active users (HIGH), the probability to not transmit (No_Tx) is the highest.

E. *Channel codes need*

It is well-known the need of channel codes when sending information through a radio channel. Really the channel code gives protection to the information sent so that the transmission throughput improves. This improvement is essential to transmit on wireless connections.

In the present study the use of these codes has been set as a parameter. Indeed it is possible to test applications without channel code, or using convolutional codes.

The operation without channel code gives rise to another usefulness of the developed tool: the analysis of channel codes. Certainly, the made emulator serves to check and to determine the performance features of different channel codes. Thus various channel code schemes can be compared and test under different conditions in order to find the most suitable.

Fig. 9 presents the FER obtained in both situations (with and without channel code) for streaming traffic in a 50 Km/h uplink channel. From the provided results we observe that FER cannot be lower than 0'3 when no channel codes are used. Moreover when there are lots of users generating traffic it is not possible to transmit without codes since FER is always 1.

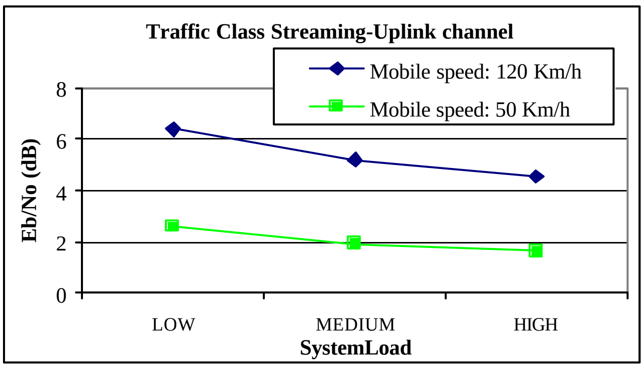

Fig. 7. Average Eb/No for different mobile speed and system load.

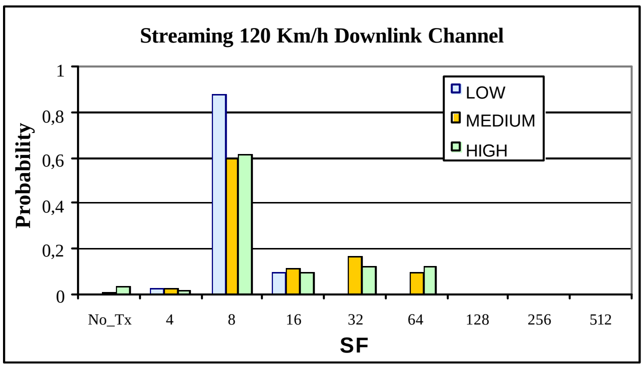

Fig. 8. SF distribution in a 120 km/h mobile speed downlink channel.

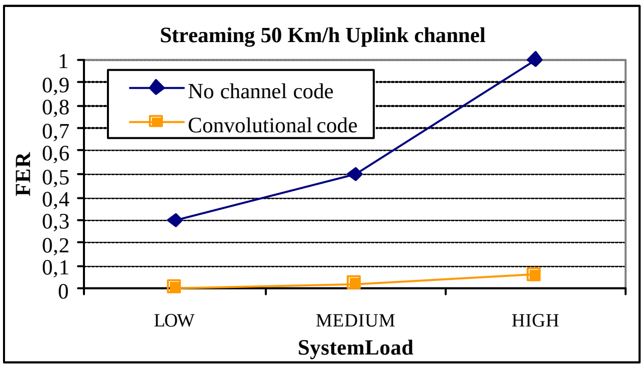

Fig. 9. FER comparison using or not convolutional codes.

In that example the convolutional code with rate 1/3 proposed by 3GPP for dedicated channels [3], has been implemented.

VI. CONCLUSIONS

In this paper a study of packet data services over wireless connections within UMTS environment has been presented. To perform this study a configurable emulator of UTRAN and mobile terminal has been made. Thus it is only necessary to set the corresponding parameters to reproduce different UTRAN architectures.

The main target of this study is to analyse the behaviour of packet data services in the future third generation mobile systems in order to improve the protocols or the applications that should be provided to mobile users.

The accuracy and the correctness of the constructed tool is shown through some validation results.

Moreover the paper presents several analysis examples to show different things that have been analysed within this study.

In the context of third generation mobile systems, packet data services are though to be very important since new and present internet services are packet oriented. However radio channel introduces more errors than fixed connections and thus to study data packet services over wireless connections is a key task.

ACKNOWLEDGMENT

This work was part of the WINEGLASS project (IST-1999-10699), partially funded by the European Commission under the Information Society Technologies (IST) framework.

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