UMTS Layers Parametrisation for Real-Time Flows

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ABSTRACT

This paper describes the emulation of a UMTS $Rel'99$ radio access network on a Linux-based testbed. The impact of design choices at transport (TCP vs. UDP), data link - RLC (Acknowledged vs. Transparent Mode) and physical (spreading factor 4 vs. 8) layers on the user experience at the reception of Real-Time flows is demonstrated for both synthetic and real life Conversational and Streaming traffics. Over a typically impaired wireless channel, it appears that Transparent Mode is not suited for Streaming flows, as it results in significant PER on UDP or significant delays on TCP, whereas it is acceptable for Conversational flows.

1. INTRODUCTION

The evolution of mobile communications and the Internet has lead to the third generation of cellular networks (3G), known as Universal Mobile Telecommunications System (UMTS). With the launch of this new generation of mobile communication, users are now truly able to access the Internet from mobile terminals, allowing them to enjoy a large variety of added value services.

In this context, the Real-Time (RT) flows are no longer limited to Conversational sessions. Streaming requests are capturing a growing share of the traffic, forcing operators to a clever planning and a more efficient operation of their radio network.

This paper addresses the Quality of Experience (QoE) of Voice over IP (VoIP) and streaming RT flows in UMTS $Rel'99$. Section 2 describes the motivations of this project. Section 3 introduces UMTS traffic classes. The next section presents the testbed designed to emulate the UMTS Radio Access Network (UTRAN). With the help of that testbed, results on the QoE of RT flows under various assumptions are presented in Section 5. Finally, conclusions are drawn in Section 6.

Qshine'07 August 14-17, 2007, Vancouver, Canada.

2. MOTIVATIONS

Researchers investigating new architecture designs and protocols in cellular networks can hardly rely on real life networks. Running experiments on a real life network is very expensive, and the experiment is hardly reproducible, due to the varying nature of the aggregated traffic.

They then turn to time- and event-driven computer simulators. Simulators can really help to figure out a number of issues to be dealt with before even thinking of a real deployment. But simulators have their limitations too. As the system to simulate becomes very large and complex, scalability issues can appear. On the other hand, simulators are usually based on simplifying assumptions that do not match real life conditions.

In this perspective, a testbed can be a complementary tool for researchers. Indeed, testbeds are cheaper than a real deployment while more faithfully mimicking the behaviour of the system under study than simulators would do, especially with respect to hardware limitations. Contrary to experiments on real life systems, experiments on testbeds are reproducible, since the whole system is under control. The testbeds can then be seen as a good trade-off between computer simulations and experiments on a real life network.

The first objective of this project has therefore been to design a modular Linux-based testbed mimicking a $Rel'99$ UTRAN. With the help of this emulator, it has been possible to stream multimedia content so as to assess the QoE of a mobile user connecting to the Internet through such a wireless access network.

3. TRAFFIC CLASSES

In [1], the 3^{rd} Generation Partnership Project (3GPP) has defined four classes of services that need to be supported in UMTS. In order to characterise them, it might be useful to focus ourselves on four representative applications (Table 1).

Table 1: Traffic classes and applications.

Traffic Classes	Representative Applications
Conversational	VoIP
Interactive	Web browsing
Streaming	Video streaming
Background	E-mail

As we are only focused on RT flows, we will only consider the Conversational and the Streaming flows in the rest of this paper.

4. EMULATION OF A UTRAN SEGMENT

The objective of the testbed is to emulate the UTRAN over an Ethernet local network. Details on the implementation are given in [2]. The following paragraphs highlight some design choices, namely the handling of UMTS transport channels, the emulation of time-correlated errors at Layer 3, the emulation of two of the three RLC modes and finally, the derivation of the BER of a mobile user.

4.1 Transport Channels

Each of the data flows presented in the Section 3 is mapped to one of the three available UMTS transport channels following the guidelines given in [3]:

- The Dedicated CHannel (DCH): this is the only dedicated transport channel in the UMTS standard, it carries the service data of a single user at variable bit rate,
- The Dedicated Shared CHannel (DSCH): this transport channel carries signalling and service data of several users at variable bit rate, or
- The Forward Access CHannel (FACH): this last transport channel carries also the service data of several users but at fixed bit rate.

The RT sessions are directly allocated to a DCH. This is natural since this kind of traffic can not tolerate the significant packet delay or the delay variation likely to appear when using shared channels. Since there is no bandwidth reservation mechanism into shared channels, RT sessions are guided to dedicated channels only.

In the opposite way, Non Real Time (NRT) sessions (Interactive and Background) may support a much longer delay. So, these traffics accept to be guided to either dedicated or shared channels since the packet scheduling of the shared channels does not affect the NRT sessions as it does to the RT ones.

4.2 Time-Correlated Errors

The emulation of the Radio Link Control (RLC) and Medium Access Control (MAC) layers is achieved using the Linux Traffic Controller (TC) which copes with packet shaping, scheduling, policing and dropping. More precisely, we use the TC NETwork EMulation component (NETEM) [4] which is a waiting queue able to delay, drop, corrupt, duplicate or even reorder packets.

The NETEM packet corruption functionality is the most appropriate to approach what packets really undergo over the UMTS radio link. Indeed, it introduces a single bit error at a random position in the packet. Instead of being dropped, this corrupted packet is transmitted to the UE, hence consuming a part of the bandwidth allocated to the flow it belongs to. Eventually, this corrupted packet is dropped, because of the CRC error detected by Ethernet.

It has been proven that the typical bit errors over a radio link are not totally decorrelated and tend to appear in bursts. One of the analytical models trying to describe this property is the Gilbert-Elliot model [5, 6]. It uses a two-state (error and error-free) time-homogeneous discrete time Markov chain to describe the channel variations. This model is a popular one for wireless bit errors since it is complex enough to capture burstiness and simple enough to be treated analytically.

Moreover, recent results have shown that an enhanced version of the Gilbert-Elliot model approaches even more the UMTS radio channel [7]. The error state in this new model is the same than the one in the simple two-state Markov model. However, the error-free state of the new model has been modified. The duration of the stay at that state is now taken from a Weibull distributed random variable which correctly fits the measurements over real networks.

We have then modified the NETEM module to respect to these specifications. We will consider the two states of the Markovian chain at the Transmit Time Interval (TTI) level: if we are in the error-free (resp. error) state, all the transmit RLC Packet Data Units (PDUs) transiting over the air interface during the TTI are considered as well received (resp. corrupted). We may then define a *burst* as a number of subsequent erroneous TTIs (Fig. 1) whereas a *gap* is a number of subsequent error-free TTIs (Fig. 2).

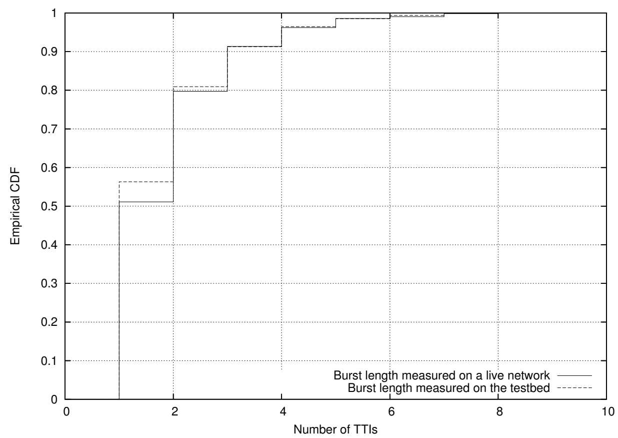

Figure 1: Bursts length comparison.

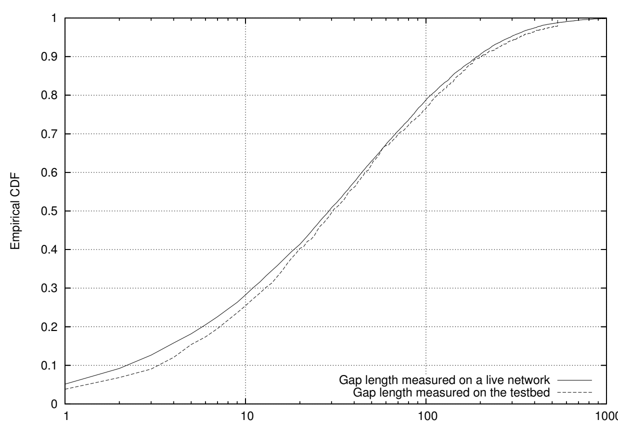

Figure 2: Gaps length comparison.

4.3 RLC modes

The RLC layer is able to transmit RLC PDUs in three different modes: Acknowledged Mode (AM), Unacknowledged Mode (UM) and Transparent Mode (TM). Note that the Automatic Repeat Request (ARQ) mechanism is only available in the AM.

We decided to focus on two of them, namely the TM likely to transmit the delay-sensitive RT traffic. On the other hand, we use the AM for NRT traffic to lower the packet error rate. The ARQ mechanism is able to retransmit an erroneous PDU a fixed number of times before declaring this PDU as lost. The number of retransmissions is known as maxDAT and is fixed at 3 in the emulation following the proposition made in [8, 9].

Those two modes have been adapted to an Ethernet link and included in the enhanced version of the NETEM module as described on Fig. 3-4. Focusing on the Fig. 3, when a packet P arrives, NETEM computes the numbers of TTIs needed to transmit P over the air interface. This number is referred to as m. This division is mainly based on P's length and on the SF allocated to the flow P belongs to. All these m TTIs are then injected into the two-states Markovian chain modified with the Weibull distribution presented in Section 4.2 to calculate the number of error-free TTIs referred to as TTI_score. If P does not have any erroneous TTI (meaning m is equal to TTI_score), P is correctly transmitted. However, if P has at least one erroneous TTI, P is corrupted before being transmitted to its final destination.

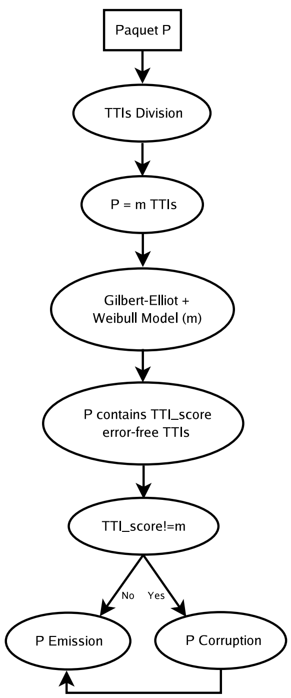

Figure 3: RLC TM using NETEM.

From the AM point of view (Fig. 4), the computation is just like in the TM case but if P contains erroneous TTIs, we reinject these $(m - TTI\_score)$ TTIs into the Markovian chain until all the TTIs needed to transmit P are error-free or until the number of attempts (referred to as Nb_try) exceeds maxDAT. If all TTIs are error-free, P is transmitted with a $Nb\_try \times 2 \times TTI\_length$ delay including the ARQ delay due to the retransmissions. On the other hand, if there is still at least one erroneous TTI at the end of the computation, P is corrupted, then transmitted with a $(maxDAT+1) \times 2 \times TTI\_length$ delay. Note that in a $Rel'99$ UMTS network, the TTI_length is fixed to $10ms$.

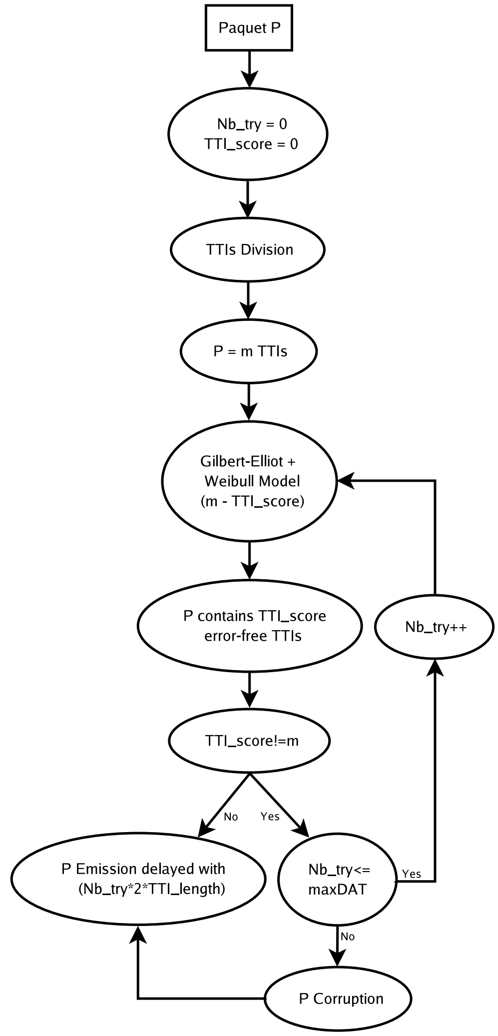

Figure 4: RLC AM using NETEM.

4.4 Received Power

As described in [2], the emulated mobile users are moving around in a wrap-around world. Periodically, the received power of each User Equipment (UE) is computed, so as to derive the quality of the on-going transmission. The quality of a transmission between a NodeB and a UE can be evaluated by the received power the UE is getting. This power is a direct function of the NodeB emitted power and is also influenced by a number of factors. At the receiver side, the carrier to interference ratio is calculated as followed [10]:

$$\left(\frac{C}{I}\right)_{CH} = \frac{Sh\ .\ Pl\ .\ (P_t)_{CH}}{\alpha I_{intra} + I_{inter} + P_N} \quad (1)$$

where Sh is shadow fading affecting a communication at a precise location in the emulated world. Pl is the path loss the communication undergoes at the same position. $(P_t)_{CH}$ is the transmitted power the NodeB allocates to the transport channel. I_{intra} is the intracell interference which is generated by those users that are connected to the same base station that the observed user. I_{inter} is the intercell interference which is generated from the other cells. α is the orthogonality factor which takes into account the fact that all the downlink communications are not perfectly orthogonal due to multipath propagation. An orthogonality factor of 0 corresponds to perfectly orthogonal intracell users while with the value of 1 the intracell interference has the same effect as intercell interference. Assumed value for the orthogonality factor is 0.4 in the macrocellular case. P_N is thermal noise power which is assumed to be equal to -99 dBm.

At the receiver side, from the $(C/I)_{CH}$ evaluated as in (1), the normalised energy per information bit is obtained as followed [11]:

$$\left(\frac{E_b}{N_0}\right)_{CH} = \frac{1}{2R_{CH}} \cdot SF_{CH} \cdot \left(\frac{C}{I}\right)_{CH} \quad (2)$$

where R_{CH} is the coding rate of the channel and SF_{CH} its Spreading Factor (SF).

On the DCH, the target E_b/N_0 setpoint is permanently adjusted and aims at a constant quality, usually defined as a certain target BER, depending on the traffic class. Following specifications presented in [1], we have fixed the target BER for the RT traffic at 10^{-4}. Assuming independent bit errors despite correlated errors at TTI level for simplification purposes, this BER enables to approximate the BLock Error Rate (BLER) as follows:

$$BLER = 1 - (1 - BER)^l \quad (3)$$

where l is the PDU length which has been fixed to 336 bits [12]. This BLER is then injected into our modified version of NETEM with two other informations: the allocated SF and the RLC mode used.

Conversely, the associated DSCH benefits from the same transmit power. However, as a different spreading factor can be allocated to the DSCH, the resulting BLER can vary. Finally, the FACH is transmitted at constant power, such that its BLER degrades when the mobile user moves away from the NodeB.

5. USER EXPERIENCE

With the help of the emulator described in the previous sections, we have investigated the QoE of Real-Time services, in UMTS Streaming and Conversational classes. In both cases, we have first run some emulations with synthetic traffic, to measure the impact of the testbed settings on the service under study. Performance have been evaluated with respect to the following Quality of Service (QoS) metrics: mean packet delay, jitter and packet loss rate. We have then set up a real life experiment, with actual content being delivered through the testbed so as to figure out the QoE at the end-user when s/he connects through a UTRAN. It should be mentioned here that these quality experiments did not involve a significant number of end-users, and hence do not claim to be consistent with ETSI recommendation on usability evaluation [13].

In the case of synthetic traffic, its characteristics are described in [2] in terms of inter-arrival times and packets lengths with the help of statistical distributions. To generate the packets we have installed the open-source application TG [14] which is a traffic generator program able to create one-way UDP or TCP streams between two computers. We have then generated Conversational and Streaming flows adopting parameter settings based on the options available in the emulated layers and on earlier results [15]:

- Transport layer: TCP vs. UDP
- Data link layer (RLC): error-free Ethernet, Acknowledged Mode with errors and Transparent Mode with errors
- Physical layer: SF = 4 vs. SF = 8

5.1 Streaming

5.1.1 *Synthetic traffic*

The results obtained for the synthetic streaming flows are presented in Table 2. All possible settings have been taken into account. It seems that streaming in TCP connections over a RLC layer in AM is the best option, despite the longer end-to-ed delay experienced by the TCP segments.

5.1.2 *Real traffic*

We have installed a Real Network Helix streaming server, with a client connecting to it through a RealPlayer 10 application. We have tested the QoE of all possible settings, and reached the same conclusion: TCP over AM is the best choice, thanks to the ability to buffer content before playing it back at the receiver. The reader can self assess the user experience of streaming a video file over a DCH by connecting to `http://www.info.fundp.ac.be/~hvp/rech/doc/streaming/`.

The quality of the streamed flow may be degraded due to a high packet loss rate. This can be the results of two different principles. First, the bandwidth available between the client and the server can't meet the requirement of the fully detailed flow. The server has to lower the video and/or audio quality to decrease the bandwidth consumption. On the other hand, streaming a video over the UMTS air interface involves a higher BER which inevitably involves a higher packet loss rate.

5.2 Conversational

5.2.1 *Synthetic traffic*

For the synthetic conversational traffic, the setting choices are reduced. Due to the delay sensitivity, retransmissions at transport layer are ruled out. Hence, UDP is to be used. Moreover, we limited ourselves to SF = 8, since the typical VoIP bandwidth requirement do not exceed 70 kbps, which is far below the 240 kbps ideally achieved with SF = 8. The results (Table 3) show that TM suffers from significant packet losses, but the communication remains acceptable. In this case, we would recommend to use UDP over a RLC layer in TM

5.2.2 *Real traffic*

We have tested VoIP over UTRAN using two BudgeTone-100 IP phones configured to use the G.720 vocoder with silent suppression. It appeared that, even with a PER of 10%, the communication was excellent.

Table 2: Synthetic streaming flow results.

Transport Layer	Data Link Layer	Physical Layer	Delay[s]	Jitter[s]	Packet Lost[%]
UDP	Ethernet	SF=4	0.003	0.011	0.00
	WCDMA-AM		0.005	0.013	0.35
	WCDMA-TM		0.003	0.011	17.73
TCP	Ethernet	SF=4	0.003	0.011	0.00
	WCDMA-AM		0.011	0.019	0.00
	WCDMA-TM		0.682	1.120	0.00
UDP	Ethernet	SF=8	0.001	0.005	0.00
	WCDMA-AM		0.003	0.011	0.31
	WCDMA-TM		0.001	0.005	16.07
TCP	Ethernet	SF=8	0.001	0.006	0.00
	WCDMA-AM		0.009	0.020	0.00
	WCDMA-TM		1.114	0.700	0.00

Table 3: Synthetic conversational flow results.

Transport Layer	Data Link Layer	Physical Layer	Delay[s]	Jitter[s]	Packet Lost[%]
UDP	Ethernet	SF=8	< 0.001	< 0.001	0.00
	WCDMA-AM		0.001	0.005	0.39
	WCDMA-TM		< 0.001	< 0.001	12.81

6. CONCLUSION

This paper has presented a testbed emulating a UTRAN segment over a wired infrastructure, including the various UMTS traffic classes and transport channels, and the time-correlated impairments of a wireless radio channel. The testbed has been shown to enable the evaluation of the QoE of services delivered through a cellular network. It helped in identifying the best parameter setting of the emulated UMTS layers.

Acknowledgements

The authors would like to thank Wolfgang Karner from the Institute of Communications and Radio-Frequency Engineering of Vienna for his valuable help.

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