Performance of a link adaptive system in UMTS networks

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ABSTRACT: In the UMTS network, the radio link is neither reliable and nor stationary. In order to provide the Quality Of Service (QoS) and to optimize the radio resource consumption, we propose to dynamically adapt the parameters of the Radio Access Network. We first simulate the behavior of the network. Those results let us predict if a set of parameters provides or not the required service. At every Transmission Time Interval, and with respect to the expected channel conditions, we will select among the sets of parameters that provide the service, the one that will optimize the efficiency. We provide tables that indicate the set of parameters to select with respect to the required QoS. We show that in favorable cases, adaptive coding could often increase the efficiency by 10%, or even 40 % in the most favorable case.

KEYWORDS: UMTS, RLC, adaptive link, convolutional code, turbo code.

1 Introduction

The UMTS network provides services for applications with various QoS requirements on a radio link. The radio channel is not as reliable as the wired channel, 4 types of error correcting codes are proposed: turbo code, convolutional code of rate ⅓ and ½ and no coding. The performance of these schemes strongly depends on the channel condition.

Adaptive coding algorithms have already been proposed for GPRS [1]. We propose here an algorithm for the UMTS network.

In this work, we first evaluate the UMTS physical link behavior to select parameters that provide the required QoS to the applications. We calculate the efficiency to select the set of parameters which maximize the efficiency while satisfying the QoS requirements. Then, we propose tables for some applications that indicate which code to select with respect to the expected channel state. Then we estimate the benefits of a link adaptive policy.

2 The UMTS Architecture

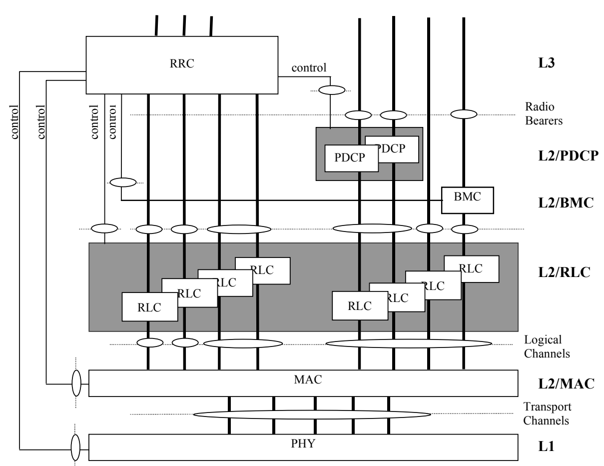

Figure 1: Radio Interface protocol architecture [2]

The Radio Access Network is divided into the following layers: RLC (Radio Link Control), MAC (Medium Access Control) and physical layer. The RLC receives packets (SDU, Service Data Unit) from the upper layer. It segments them into PDU (Protocol Data Unit). The PDU are transmitted across the UTRAN. The RLC Receiver detects when PDU are missing, and the RLC Transmitter retransmits the missing PDU. The MAC layer manages the priority between the services of a user. The physical layer is in charge of coding, and error correction and detection. The physical layer also transmits the packets on the radio medium.

3 Results

We make our experiences on a Gaussian channel. We generate packets of 1280 bits at 64 kbit/s at the Base Station. The RLC segments the SDU into 4 PDU of 336 bits (320 bits of data and 16 bits of header (we neglected the SDU header compression and the length indicator of the PDU)). The TTI is of 10 ms. We add a CRC of 16 bits on the physical layer. We use the following coding schemes as described in [3]: turbo code, convolutional code of rate ½ and ⅓ and no coding. At the User Mobile, the erroneous PDU are discarded. We allow, according to the experience, 0, 1, 2 or an infinite number of retransmissions for the missing PDU. The SDU are delivered in sequence to the upper layer. The erroneous or incomplete SDU are discarded.

3.1 Influence of the number of transmission on the SDU Error Rate

We first want to provide the SER (SDU Error Rate). We measure it with respect to the selected coding scheme, maxdat (the number of allowed retransmissions for a PDU) and Eb/N0. The SER is defined as:

$$SER = 1 - \frac{r}{g}$$

where r is the number of SDU received at the RLC receiver entity and g is the number of SDU transmitted by the RLC. *Maxdat* is set to 0, 1, 2, or is unbounded (+∞).

Figure 2 shows the SER with different coding schemes. (t⅓ represents the convolutional code ⅓, t½ the convolutional code ½, tc the turbo code, and noc, the no coding option).

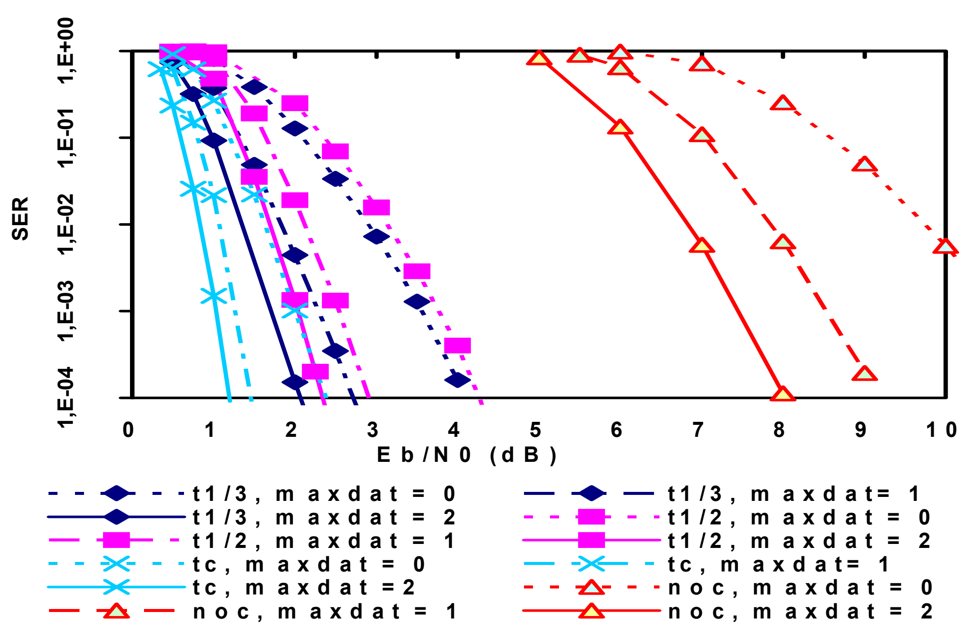

Figure 2 : SDU Error Rate with respect to Eb/N0, the correcting code and maxdat.

The results show the influence of the error correcting code with respect to the transmission without coding. The retransmissions increase the reliability.

3.2 *Influence of the correcting code upon the transfer delays.*

The SDU Transfer delay is the time interval from the generation of a packet to be transmited by the mobile, until the successful reception by the base station on the IP layer.

We want to determine the maximum delay for the delivery of 95% of the issued SDU.

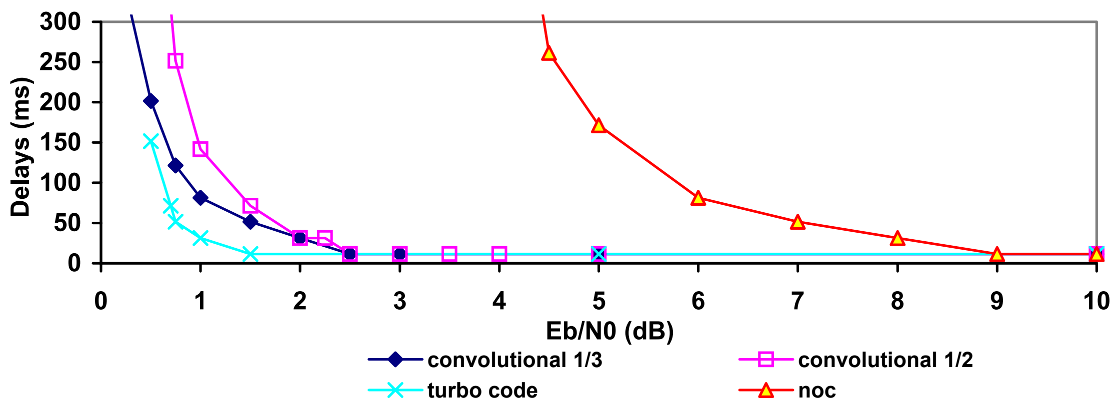

Figure 3: Transfer delay for 95% of the received SDU according Eb/N0 and the error correcting code with *maxdat* = +∞.

For a channel of poor quality, delays are very important and decrease quickly with the channel amelioration. The strongest codes (turbo code and convolutional ⅓) always give the best delays. At the opposite, no coding yields the worst delays. The causes of delays are the round trip time, the retransmission of a PDU, and the wait for the order delivery. The delays are highly correlated to the SER. The loss of a PDU causes further delay for a SDU of at least 4 TTI (loss detection, send of a status, reception of the status and retransmission).

4 Efficiency and adaptive link.

To set the UTRAN parameters, the system must satisfy the QoS criteria set by the application. An error-correcting code will be selected only if it satisfies the SDU Error Rate and the delay criteria (if the application has delay requirements). Table 1 gives typical values of error rate and delays for standard applications. By comparing the required values of table 1 to figures 2 and 3, we can decide if a set of parameters is valid or not. For a given range of Eb/N0, if the SER (figure 2) of an error correcting code is less to the required SER, and if the delay of this correcting code is less to the required delay, then this configuration is valid.

QoS requirements	Traffic class	File transfer	Interactive	Non Interactive
	SDU error rate (SER)	minimal	10^{-1}	10^{-2}
	Transfer delay (ms)	N/A	100	250

Table 1: Quality of service for application carried by the UMTS.

We notice that for the lower Eb/N0, no error correcting code satisfies the constraints, but for the range where Eb/N0 is high several choices are possible. When none of the settings satisfies the QoS requirements, the service must be refused or the system must work under the power control (which is out of the scope of this topic). When several parameters are possible, then we will use another criterion to make our decision. The parameters that will maximize the efficiency will be selected.

The Efficiency E is given by:

$$E = \frac{(1-t)*p}{r*f}$$

where t is the SER, p is the size (in bits) of the data field in a PDU, r is the average number of retransmission for a PDU and f the number of symbols in a radio frame.

Figure 4 shows the efficiency with respect to the error-correcting codes and Eb/N0.

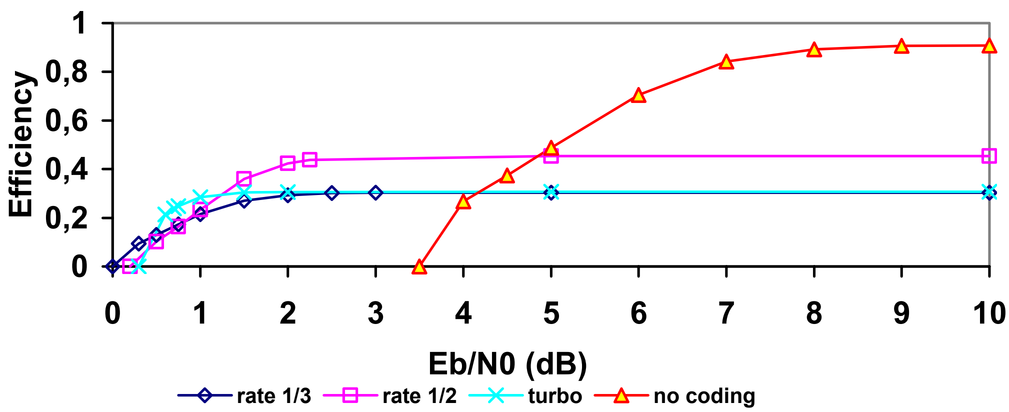

Figure 4: Efficiency with respect the corrector code and Eb/N0.

The Efficiency curves show two regions. At first, the Efficiency increases quickly with Eb/N0, and then the efficiency tends to an asymptotic value. The asymptotic value depends on the code rate and the PDU size. For each code, the limit at high Eb/N0 corresponds to no retransmissions.

In a given range of Eb/N0, the figures 2 and 3 indicate the allowed set of parameters: we will only select codes that have lower SER and lower delays than required. For every range, we will select in the allowed set, the one that maximize the efficiency (according figure 4). For the applications described in table 1, we get the following selections (table 2 -4): Table 2 gives the code with respect to Eb/N0 for a file transfer, table 3 for a real time interactive application and table 4 for a non interactive application.

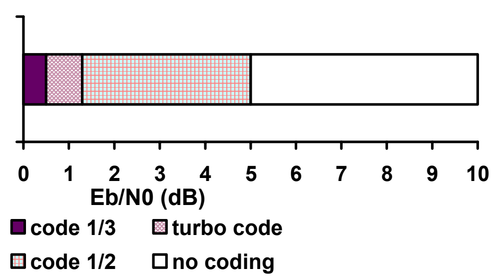

Table 2 : Error correcting code to select with respect to Eb/N0 for a file transfer application.

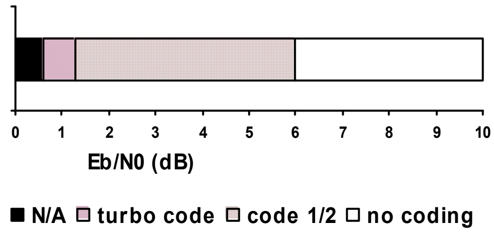

Table 3 : Error correcting code to select with respect to Eb/N0 for an interactive application.

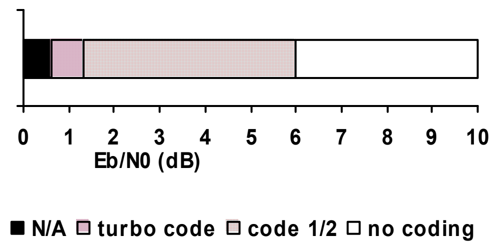

Table 4 : Error correcting code to select with respect to Eb/N0 for a real time non interactive application.

At every TTI, the system will forecast the next Eb/N0 according to measurements. For every application, the system will select the correcting code indicated by the appropriate table (from table 2 to 4).

Benefits of link adaptation.

To compare the adaptive and static coding, we examine several scenarios where the radio quality channel evolves linearly during a communication.

- In the first scenario, Eb/N0 is in the range of 0 to 3 dB. The radio channel is slightly unstable, and has a poor quality.
- In the second, Eb/N0 is in the range of 3 to 6 dB. The radio channel is slightly unstable, and has a medium quality.
- In the third, Eb/N0 is in the range of 6 to 9 dB. The radio channel is slightly unstable, and has good quality.
- And in the fourth, Eb/No is in the range of 0 to 9 dB. The quality of channel is very unstable.

We assume that the system is able to compute the Eb/N0 value in real time. To compare the adaptive and static coding, we compute the average efficiency

$$\frac{\int_a^b eff_{code}(x)dx}{b-a}$$

where eff_{code} is the efficiency function of the code, and a and b are the limits of the considered Eb/N0 range.

		File transfer				interactive				Real time non interactive			
scenario		1	2	3	4	1	2	3	4	1	2	3	4
Adaptive	Eff.	0.32	0.50	0.84	0.55	0.30	0.45	0.84	0.53	0.30	0.50	0.84	0.55
Code ⅓	gain	39.1 %	67.3 %	180 %	99.9 %	-	-	-	-	-	-	-	-
Turbo	gain	33.3 %	61.9 %	171 %	94.1 %	27.0 %	46.6 %	171 %	88.7 %	27.0 %	61.6 %	171 %	94.3 %
Code ½	gain	10.3 %	11.5 %	86.7 %	39.4 %	27.0 %	0.0 %	84.8 %	40.8 %	13.8 %	11.3 %	86.7 %	41.6 %
No coding	gain	∞	32.1 %	0.00 %	35.9 %	∞	∞	0.0 %	89.0 %	∞	152 %	0.0 %	57.6 %

Table 5 : **Gain (%) of adaptive coding with respect to the static coding and the required QoS**

The calculus of the efficiency of the adaptive system is done by taking point by point the highest value of the efficiency among the efficiency values of the different codes (The efficiency curve of the adaptive system is the superior envelop of all the efficiency curves).

The benefit of adaptive coding is given in percent with respect to the efficiency of the considered code on the considered scenario. The real gain of adaptive coding for a scenario is the minimum value among the gain, because if a non-adaptive system is used, we consider that the system will use the set of parameters that is statistically the most efficient.

The adaptive system is more efficient when the channel is bad (scenario 1) and unstable (scenario 4). The gain is more important if the applications require low delays. The gain of the adaptive system is around 10 % for the first and second scenario. For the third scenario, where the channel is good, adaptive coding does not yield a gain. In the fourth scenario, the range of Eb/N0 for the channel is more important. So, in this scenario, the benefit ranges between 35 and 40 %.

5 Conclusion

Network simulations show that adaptive coding is a promising method to provide the QoS on the UMTS. If the next channel state could be accurately forecasted, the most efficient set of parameters could be selected.

We have simulated the behavior of the correcting code on an UMTS network. We have provided a table presenting the appropriate code to select with respect to the channel condition. We have shown a link adaptive system enables to increase the efficiency of 10%, or even 40% in the most favorable cases.

6 Bibliography

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