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REPORT

DIPLOMA WORK 98

IP APPLIED TO MOBILE NETWORKS

Project-Type:	**Diploma Work**
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Deadline:	**18th December 1998**

Table of Contents

1. Abbreviations 4

2. Abstract 4

3. Introduction 5

4. The GSM Technology 5

4.1 General Overview 5

4.2 The GSM Architecture 6

4.3 Radio link aspects 6

4.4 Data Services in GSM 7

5. The GPRS Technology 10

5.1 Network architecture 10

5.2 Service capabilities 11

6. Measurements in the GSM Network (Layer 2) 11

6.1 Principle 11

6.2 Definitions 11

6.3 Platform 11

6.4 Locations 12

6.5 Time recovery 12

6.6 Investigated data 13

6.7 Evaluations 13

7. Behaviour of the transport protocol 23

7.1 Assumption PPP & IP 23

7.2 TCP 24

8. Measurement in the GSM Network (higher Layers) 30

8.1 Principle 30

8.2 Definitions 31

8.3 Measurement results / Verification of the estimations (see appendix 4) 31

9. Conclusion 32

10. References 33

Table of Figures

Figure 4-1 General architecture of a GSM network 6
Figure 4-2 Organisation of bursts, TDMA frames, and multiframes for speech and data 7
Figure 4-3 Data transmission planes 8
Figure 4-4 Comparison transparent and non-transparent mode 10
Figure 5-1 GPRS network entities 10
Figure 6-1 IDA platform 12
Figure 6-2 Composition of the RTT 14
Figure 6-3 Mean RTT and retransmission levels 14
Figure 6-4 Delay during 5 periods of one minute (BTS) 16
Figure 6-5 Histogram of the delay (BTS) 16
Figure 6-6 Throughput variation in % (BTS) 17
Figure 6-7 RxLev and RxQual during 5 periods of 1 minute (BTS) 17
Figure 6-8 Delay during 5 periods of one minute (Nationalbank) 18
Figure 6-9 Histogram of the delay (Nationalbank) 18
Figure 6-10 Throughput variation in % (Nationalbank) 19
Figure 6-11 RxLev and RxQual during 5 periods of 1 minute (Nationalbank) 20
Figure 6-12 Delay during 5 periods of one minute (Kantonalbank) 20
Figure 6-13 Histogram of the delay (Kantonalbank) 20
Figure 6-14 Throughput variation in % (Kantonalbank) 21
Figure 6-15 RxLev and RxQual during 5 periods of 1 minute (Kantonalbank) 21
Figure 6-16 Delay during 5 periods of one minute (Kongresshaus) 22
Figure 6-17 Histogram of the delay (Kongresshaus) 22
Figure 6-18 RxLev and RxQual during the period of the delay measurements 23
Figure 7-1 Behaviour of the Transport Protocol on a time-out 26
Figure 7-2 Composition of the delay while sending short TCP packets 27
Figure 7-3 Behaviour of TCP while sending short packets 27
Figure 7-4 Behaviour of the TCP congestion window size while sending short packets 28
Figure 7-5 Segmenting of the PPP Packets 29
Figure 7-6 Composition of delay while sending long TCP packets 29
Figure 7-7 Behaviour of the Timeout with short and with long TCP segments 30
Figure 8-1 Variation of the transmission time using an FTP tool 31
Figure 8-2 Variation of the throughput using an FTP tool 32

Table of Tables

Table 6-1 Comparison between sent and received chars (BER) 15
Table 6-2 Mean throughput of each measured period (BTS) 17
Table 6-3 Mean throughput of each measured period (Nationalbank) 19
Table 6-4 Mean throughput of each measured period (Kantonalbank) 21

1. Abbreviations

AC	Authentication Center
ACCH	Associated Control Channel
BCH	Broadcast Channels
BER	Bit Error Rate
BSC	Base Station Controller
BTS	Base Transceiver Station
CBT	Computer Based Training
FDMA	Frequency Division Multiple Access
FTP	File Transfer Protocol
GPRS	Packet Radio Service
GSM	Global System for Mobile Communications
HLR	Home Location Register
HSCSD	High Speed Circuit Switched Data service
IDA	Internet Direct Access
IMEI	International Mobile Equipment Identity
IP	Internet Protocol
ITU	International Telecommunication Union
IWF	Inter Working Function
ME	Mobile Equipment
MS	Mobile Station
MSC	Mobile Service Switching Center
PPP	Point to Point Protocol
PSTN	Public Switched Telephone Network
RFC	Radio Frequency Channels
RLP	Radio Link Protocol
RTT	Round Trip Time
TAF	Terminal Adaptation Function
TCH	Traffic Channel
TCP	Transmission Control Protocol
TDMA	Time Division Multiple Access
UMTS	Universal Mobile Telecommunication System
VLR	Visitor Location Register

2. Abstract

This diploma work was part of the project 'IP applied to Mobile Networks'. The goal was firstly to define and to perform a measurement campaign on the existing Swisscom Global System for Mobile Communications (GSM) network, secondly to analyse and interpret the results of the measurements, thirdly to investigate theoretically the use of the Internet protocols on the top of the mobile network and fourthly to validate the theoretical results through a second phase of measurements.

The analyses of the problem have been subdivided into three parts in order to answer to the three following questions :

1. How is it possible to carry out measurements delivering the required parameters (delay, throughput) without using a TCP/IP protocol stack ?

2. Which parameters of the protocols on the top of the GSM network are important and how can the theoretical aspects be investigated and correlated with the measurement results ?

3. How can be confirmed (or not) the theoretical conclusions by measurements ?

To measure the relevant data (delay, throughput) without using a TCP/IP stack a c++ program was developed. It allows on one hand to make a connection on layer 2 and on the other hand to measure the needed parameters. For the theoretical investigations the focal point was on the behaviour of the TCP if short or long TCP segments are sent over the GSM network. The behaviour of the other protocols (PPP, IP) is less critical for a successful engagement of the TCP/IP protocol stack over a GSM network. Two typical applications (Telnet for sending short TCP segments and FTP for sending long TCP segments) were chosen to make the verifications with the estimations.

The measurement results obtained by the programmed script have provided very interesting information about the delay and the throughput over the Radio Link Protocol (RLP). The measured parameters were used to make modulations and estimations of the behaviour of the transport protocol. The use of Telnet and FTP have allowed to demonstrate that the theoretical conclusions were fully valid.

I recommend that SWISSCOM use the results of this diploma work for further studies concerning data transmission on the GSM network. The results could be useful to compare the behaviour of data transmission with new technologies like High Speed Circuit Switched Data service (HSCSD), the General Packet Radio Service (GPRS) or the Universal Mobile Telecommunications System (UMTS).

3. Introduction

The diploma work was a fully separated part of the IAM project.

A Computer Based Training (CBT) about GSM has provided me an insight in the GSM technology. Part of the measurement equipment was at my disposal and was directly ready to be engaged. Some other parts of the equipment had to be prepared and installed. The measurement environment was not ready and had to be prepared. The know-how of some specialists by Swisscom was very useful to define, prepare and analyse the measurements.

4. The GSM Technology

1 General Overview

Today GSM is not only the abbreviation *for Global System for Mobile Communications,* GSM stands for an extraordinary successful development of the progressive Information Technology. GSM means for millions of users a new dimension of the personal communication. GSM is installed in more than 120 different countries even outside of Europe with about 70 million users world-wide. The main use of GSM today is the cellular telephony but the GSM data communication will become in a near future more important. The evolution of GSM will be on one hand the General Radio Packet service (GPRS) Technology and on the other hand UMTS which is a core network.

This modern digital system for mobile communication is based on standards which were acquired and putted into action in Europe.

GSM stands also for very high complexity. The GSM is probably one of the most complex communication system which has ever been developed.

2 The GSM Architecture

A GSM network is composed of several functional entities, whose functions and interfaces are specified. Figure 4-1 shows the layout of a generic GSM network. The GSM network can be divided into three broad parts. The Mobile Station is carried by the subscriber. The Base Station Subsystem controls the radio link with the Mobile Stations. The Network Subsystem, the main part of which is the Mobile services Switching Centre (MSC), performs the switching of calls between the mobile users, and between mobile and fixed network users. The MSC also handles the mobility management operations. Not shown is the Operations and Maintenance Centre, which oversees the proper operation and setup of the network. The Mobile Station and the Base Station Subsystem communicate across the Um interface, also known as the air interface or radio link. The Base Station Subsystem communicates with the Mobile services Switching Centre across the A interface.

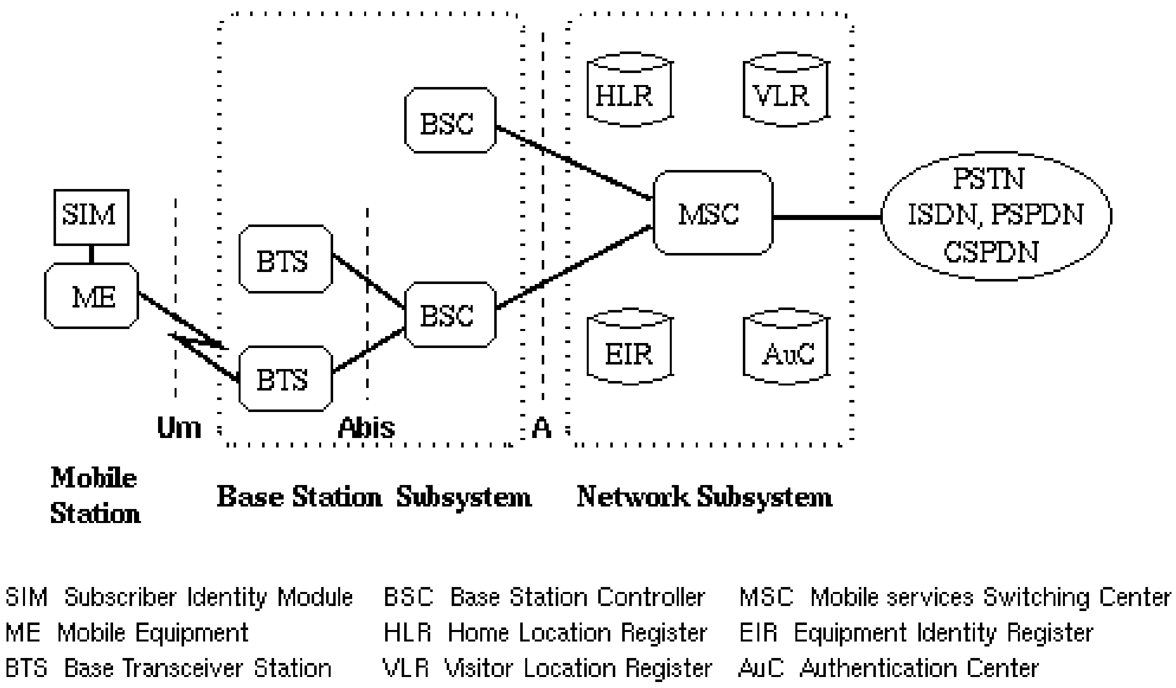


Figure 4-1 General architecture of a GSM network

3 Radio link aspects

The International Telecommunication Union (ITU), which manages the international allocation of radio spectrum (among many other functions), allocated the bands 890-915 MHz for the uplink (mobile station to base station) and 935-960 MHz for the downlink (base station to mobile station) for GSM networks in Europe. The success of GSM made it necessary to allocate additional bands for uplink (1710-1785 MHz) and downlink (1805-1880 MHz).

.1 Multiple access and channel structure

Since radio spectrum is a limited resource shared by all users, a method must be devised to divide up the bandwidth among as many users as possible. The method chosen by GSM is a combination of Time- and Frequency-Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the (maximum) 25 MHz bandwidth into 124 carrier frequencies spaced 200 kHz apart. One or more carrier frequencies are assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme. The fundamental unit of time in this

TDMA scheme is called a *burst period* and it lasts 15/26 ms (or approx. 0.577 ms). Eight burst periods are grouped into a *TDMA frame* (120/26 ms, or approx. 4.615 ms), which forms the basic unit for the definition of logical channels. One physical channel is one burst period per TDMA frame.

Channels can be divided into *dedicated channels*, which are allocated to a mobile station during an achieved communication, and *common channels*, which are used by BTS to broadcast information.

.2 Traffic channels

A traffic channel (TCH) is used to carry speech or data traffic. Traffic channels are defined using a 26-frame multiframe, or group of 26 TDMA frames (See Figure 4-2). The length of a 26-frame multiframe is 120 ms. Out of the 26 frames, 24 are used for traffic, 1 is used for the Slow Associated Control Channel (SACCH) and 1 is currently unused. TCHs for the uplink and downlink are separated in time by 3 burst periods and in frequency by 45 MHz, so that the mobile station does not have to transmit and receive simultaneously, thus simplifying the electronics.

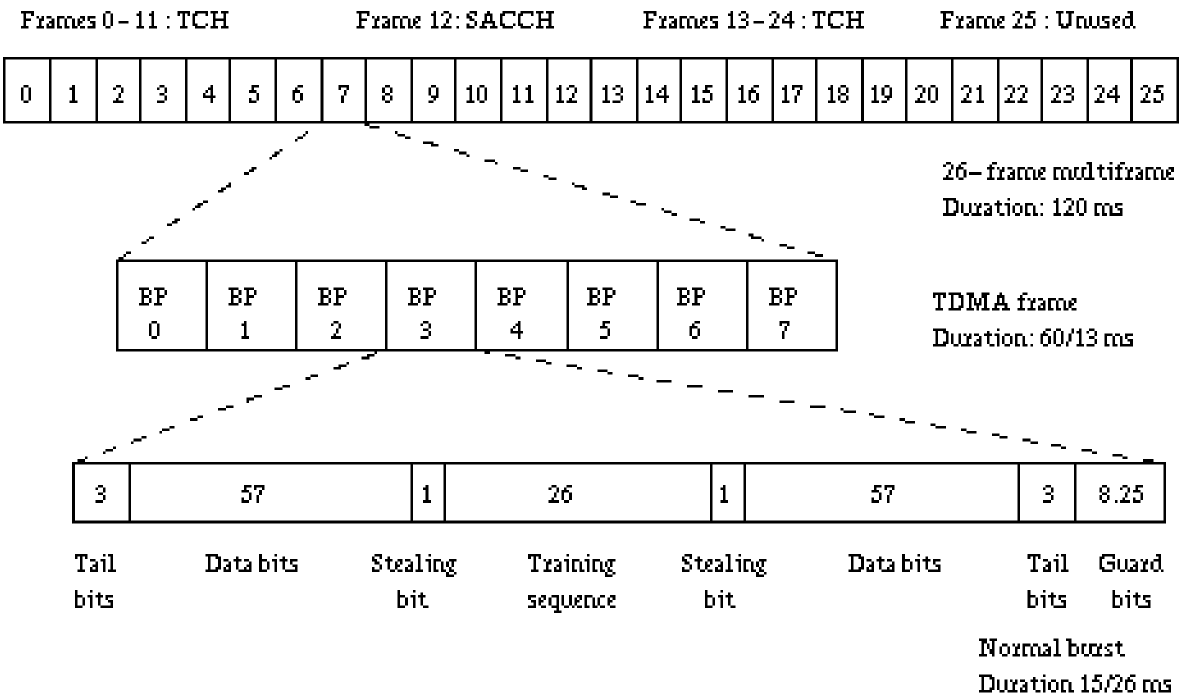


Figure 4-2 Organisation of bursts, TDMA frames, and multiframes for speech and data

4 Data Services in GSM

.1 Principle

Data Services cover the exchange of a lot of different types of information from one Mobile Station to another Mobile Station or to a server in the network. Data transmission encompasses the exchange of text, drawings, computer files, animated images, messages and so on or data from real-time applications like video or audio. An important part of the information processing is done at the two extremities, in a machine, called "terminal equipment", most often outside the scope of the specifications.

.2 Data transmission

An important point in the transmission path is the boundary between GSM and the external network. GSM can be connected to a variety of external networks: Public Switched Telephone Network (PSTN, which is still the principal carrier of data transmission) or Packet Switched Public Data Networks (PSPDNs).

The existence of an external network divides the transmission path into two segments. The segment between the GSM user terminal and the boundary point in entirely within GSM. But the other segment, from the boundary point to the terminal, is entirely outside the control of GSM, and follows transmission rules that are specific to the external network.

To reduce the number of cases dealt with by transmission equipment within GSM, despite the variety of interworking cases, two generic functions are inserted on each side of the GSM segment, as shown in Figure 4-3. These functions enable GSM to deal with a small amount of internal transmission modes, and still accommodate the various interworking needs. The adaptation function at the boundary between GSM and the external network is called the *network interworking function*, most often reduced to the last two terms, and often further reduced to “IWF”. On the GSM user side, the functional part of the mobile station which performs the adaptation between a specific terminal equipment (TE) and the generic radio transmission part is called the Terminal Adaptation Function, or TAF.

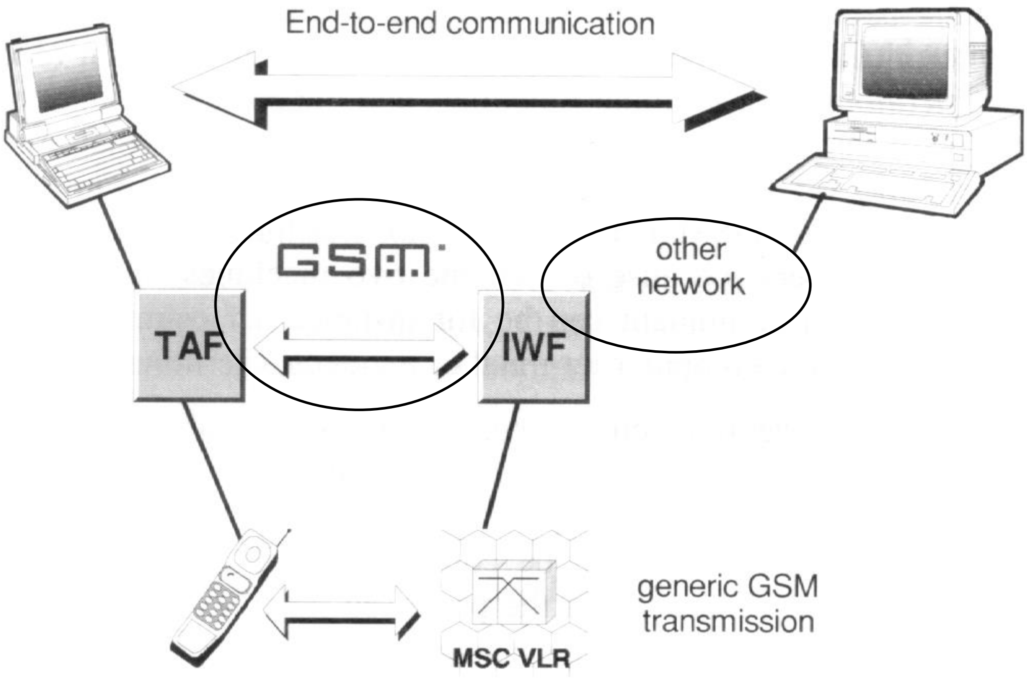


Figure 4-3 Data transmission planes

The TAF on one side and the IWF on the other side act as entry points into the GSM world. Their functions depend on the type of end-to-end service. Conversely, GSM entities in-between TAF and IWF are not concerned with the end-to-end service, but solely with the bearer capabilities required to transport the corresponding data flow. Figure 4-3 shows how data transmission can be looked at on three different levels, the end-to-end level, the TAF-IWF level and the generic GSM transmission level.

As already noted, most of the end-to-end domain is out of the scope of the specifications. But the adaptation functions (in TAF and IWF) are direct concern to GSM. For most of the data services, the tasks fulfilled by the adaptation functions can be inferred from the single knowledge of the bearer capability.

So the adaptation functions and the general configuration of the transmission paths depend mainly on the bearer capability and on the external network. It appears that the key factor, at the origin of most of the differences, is the external network. There are three different Network types:

- Public Switched Telephone Network
- Integrated Services Digital Network
- Packet Switched Public Data Network

There are mainly two adaptation functions:

- Conversion from synchronous to asynchronous data flows
- Bit rate adaptation

.3 Error Correction

Since the data has to be transmitted over a radio interface, the data may be corrupted during transmission. To avoid too high error rates two techniques have been implemented. The first one is FER, Forward Error Correction on Layer 1. This technique introduces some redundancy in the data stream and is used to correct errors if they occur. Since sometimes the error rate is to high to correct all errors at bit level using this technique, a protocol has been developed which retransmits erroneous data. This protocol is called RLP, Radio Link Protocol (Layer 2). It spans from the mobile user to the Inter Working Unit (IWU) in the MSC. If a service uses this protocol, the service is in *non-transparent* mode. If a service only relays on FER it is in *transparent* mode.

.1 *Transparent vs. non-transparent mode*

In the *transparent* mode no RLP is used. This means that there are more errors to be expected than in non-transparent mode. The advantage is that there is a constant delay on the data. In this mode the IWU just translates the digital data into the appropriate format for the other network (e.g. modulates the digital signal to an analogue signal). Another advantage is that standard compression protocols like the V42.bis protocol may be used.

In the *non-transparent* mode the problem of high error rate is reduced but due to retransmissions the delay may vary very much. This is a problem for real time applications if internal timers expire.

The next picture gives an overview over the differences between the transparent and the non-transparent mode:

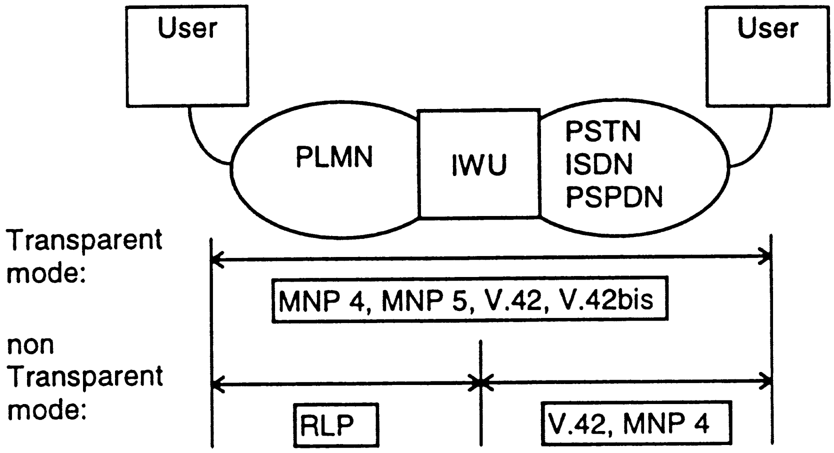


Figure 4-4 Comparison transparent and non-transparent mode

5. The GPRS Technology

This service is currently being standardised and its implementation will be probably available in two or three years. This new technique provides the possibility to send and receive data packets based on a packet circuit switched technology. It is a very flexible service. On one hand it supports applications with high bandwidth needs on the other hand very narrow band applications are supported. The resource is only used if required (statistical multiplexing). This saves much radio resources for the network operator and money for the user. Because of this GPRS is very useful for burst data applications like e-mail or accessing the WWW.

1 Network architecture

To implement the GPRS network new entities must be installed in the existing GSM network. The two most important entities are the SGSN (Serving GPRS Support Node) and the GGSN (Gateway GPRS Support Node). The task for the SGSN and the GGSN is to route the data packages through the network. Beside these new entities some changes are also to be implemented in the BTS, the Base Station controller (BSC) and the Mobile Station (MS) to adapt to the new service. To introduce this new service in the network the operator might start with a limited number of SGSN an GGSN. If the service becomes more popular more GSN can be easily introduces in the network.

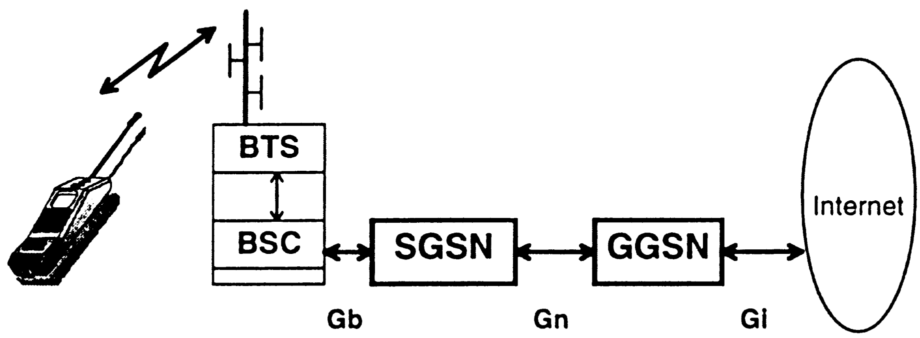


Figure 5-1 GPRS network entities

2 Service capabilities

Since GPRS is not implemented yet only speculations about the characteristics of the service can be done.

In the ETSI recommendation 3.60 and 2.60 some service classes are defined but some of them will not be possible to provide because of radio resource problems.

The *service characteristics* have cover a large range of capabilities:

- mean throughput: 0.22 bit/s to 111 kbit/s
- peak throughput: 8 kbit/s to 2048 kbit/s
- mean delay for 1024 bytes: from 2 s to 75 s to unlimited
- data loss probability: 10^{-7} % to 1 %

It may be observed that the service capabilities instead of the bearer services have been standardised, in order to encourage the competition also on the service level.

6. Measurements in the GSM Network (Layer 2)

1 Principle

The goal was to carry out measurements with GSM data without using the TCP / IP protocol stack. This has the advantage that we know afterwards exactly the delay and the throughput on Layer 2 which allows us to make considerations and modulations about the behaviour of the higher protocols.

2 Definitions

The definitions and the planning of the measurements is documented in *Definition of Measurements in phase 1, version 1.2* (Appendix 1).

3 Platform

All the measurements in phase 1 (measurements without the TCP/IP stack) have to be done on the IDA platform. This platform is a direct access to the Internet and not avoiding the passage through a public network (PSTN, ISDN) via a modem. With IDA a better throughput and better access conditions are expected.

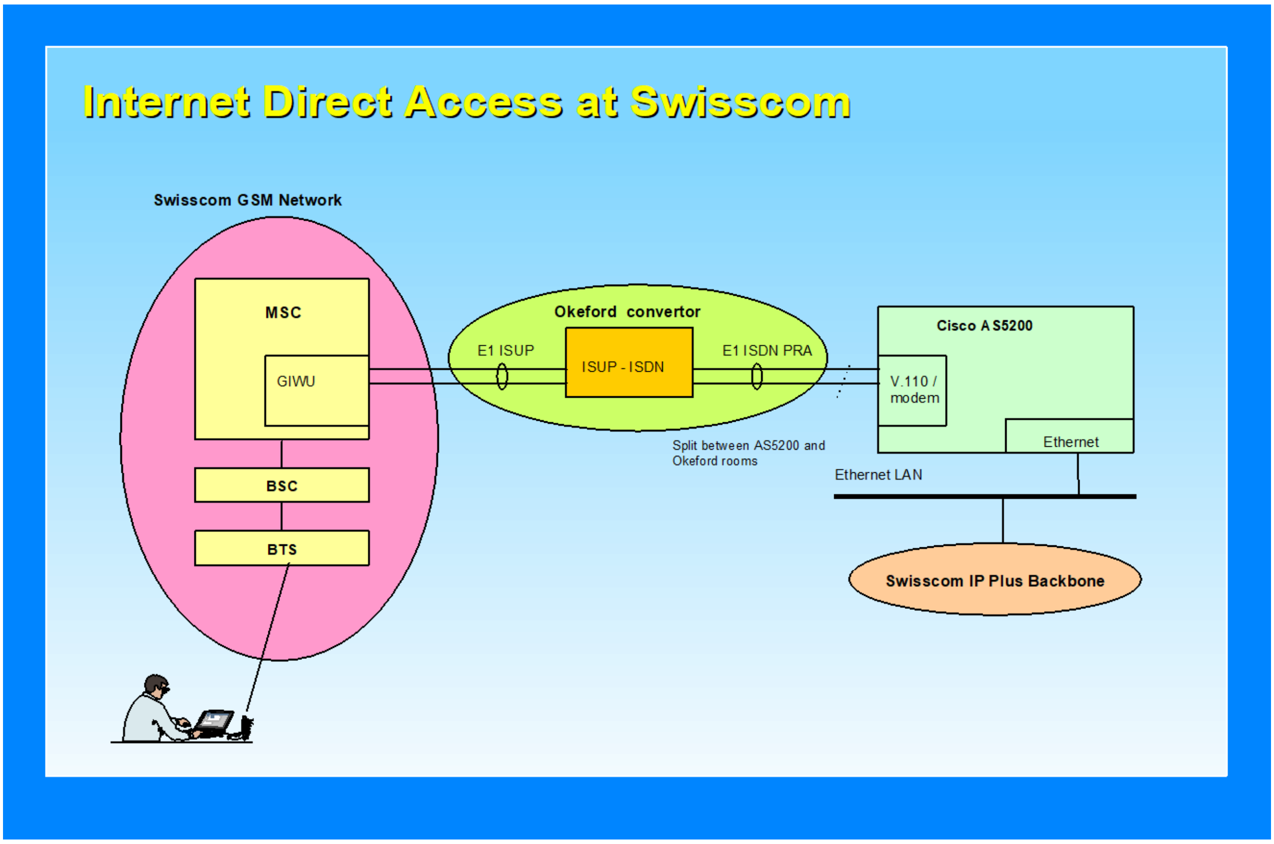


Figure 6-1 IDA platform

4 Locations

- Bern, Bahnhofunterführung
- Bern, next to the BTS BEA
- Bern, Berner Kantonalbank, ground floor
- Bern, Nationalbank, outdoor
- Bern, Swisscom building Ostermundigenstrasse 93, ground floor
- Bern, Swisscom building Ostermundigenstrasse 93, outdoor

- Zürich, Bahnhof, Treffpunkt
- zürich, Bahnhof, Shopville
- Zürich, Bahnhofstrasse, Planet Hollywood, first floor
- Zürich, Kongresshaus, ground floor
- Zürich, Kongresshaus, outdoor

- In the train between Lenzburg and Olten

5 Time recovery

Time matching: Because of the measurements were made on two computer systems at the same time, it was very important that the two system times matched together. The matching has been tested several time on each 'measurement day'. The difference between the two system times was about 4 seconds every time, so it synchronise both data sources.

6 Investigated data

Delay: The delay to transmit a character is calculated in the MS by the difference between the two time stamps (emission of char; reception of char). The measurement tool guaranties that only identical characters are compared.

Throughput: To measure the throughput a counter in the measurement tool counts the arrived characters each second. While looking at the measurement results it was obvious that the throughput is too small. Because all measurements were based on the same tool the same effect is present in all measurements.

After detailed studies of the counter function a part of the problem has been solved: During the throughput measurements more than 1 character are read from the COM interface at the same time but the counter is incremented only by one.

Another parameter which was recorded on each time receiving a file of characters is the transmission time. One the other the number of transmitted characters is known. A mean throughput can be calculated.

CER: We know on one hand the number of sent characters and on the other hand the number of arrived characters. These two numbers make it easy to calculate the character error rate.

7 Evaluations

The evaluations of the measurements are based on the log files produced by our measurement tool and by the measurement recordings produced by MTR [1] which is managed by the Operation and Maintenance System of the Swisscom GSM platform. All the measurement results with all the statistics are attached in appendix 2.

The next chapter presents a collection of typical results obtained during the measurement compaign.

.1 General comments

Based on the ensemble of the collected results, different observations have been made. The data have been analysed (see next chapters) and interpreted. The following comments on delay, throughput, BER, RXLEV and RXQUAL are also related to all measurements (attached in Appendix 2).

Delay: The delay measurements are based on the script as described in 'Definitions of Measurements in Phase 1'. The delay is divided on one hand into the delay between the TAF and the IWU and on the other hand on the delay between the IWF and the Linux Host on the IDA Network. All the measurement results and the statistics are based on the whole Round Trip Time.

[1] Mobile Traffic Recording (Ericsson system)

Figure 6-2 shows the composition of the RTT during a data transmission. A char is sent and mirrored at the Linux host. These values have been calculated as shown below and are based on the worst case assumption that all the retransmissions are on the same link (either on uplink or on downlink).

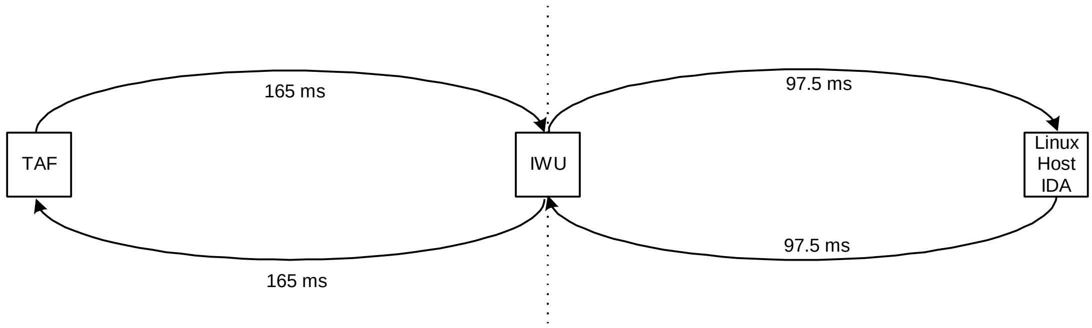


Figure 6-2 Composition of the RTT

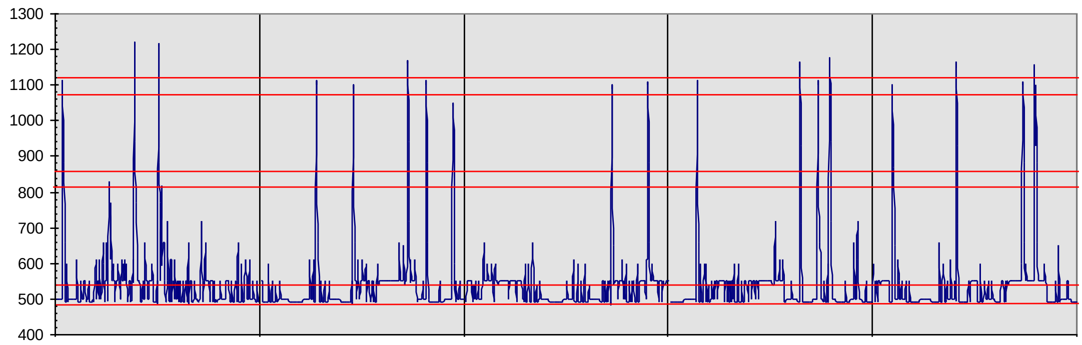


Figure 6-3 Mean RTT and retransmission levels

Looking at the measurement results more than 80 percent of the measured delays are between 490 and 720 ms. Other delay levels can be noticed about every 330 ms. Figure 6-3 shows this behaviour of the delay in a typical environment. The identified steps (of about 330 ms, of about 650 ms) can be assumed as retransmissions of RLP blocks on the layer 2 of GSM. Based on the example of Figure 6-3 considering that the delay is composed by the elements of Figure 6-2, the following estimations are computed:

mean RTT: *525 ms*
Retransmission time: *330 ms*
delay IWU-IDA = mean RTT - retransmission time = 525 ms - 330 ms = 195 ms
retransmission on one link = retransmission time / 2 = 330 ms / 2 = 165 ms

The two red lines next to another indicate that the values are not exactly on the same level. This effect can be looked at measurement noise which can have a lot of different reasons (synchronisation, buffering).

The conclusion is that retransmission time due to most of time to interleaving (22 burst) which is confirmed by the information provided by ETSI (see GSM 03.05 specifications).

Throughput: The throughput measurements are based on the script as described in 'Definitions of Measurements in Phase 1'. As described previously, the throughput measurements are not directly usable, but they illustrate tat least the qualitative aspects of the throughput as perceived by the end user. A mean throughput of 7.4 Kb/s can be calculated by using the transmission time and the number of transmitted bites. The maximum throughput for data transfer in the GSM network is 9600 bit/s (192 bits data in a RLP packet, RLP packets are transmitted every 20 ms).

CER: The CER (Char Error Rate) measurements are based on the sent and the received characters as described in 'Definitions of Measurements in Phase 1'. A typical raw BER on the radio channel with a user rate of 9600 bit/s is about 0.3 percent. A part of the logfiles (the throughput file is sent five times in each measurement) of the following cases had been compared with the sent file on the Linux host. The comparison delivered the following results:

place	sent char	received chat	BER
Bern, Nationalbank, outdoor	49980	49980	0
Berner Kantonalbank, indoor	49980	49980	0
Bern, Basisstation BEA, outdoor	49980	49980	0

Table 6-1 Comparison between sent and received chars (BER)

These results confirm that in the non-transparent mode there are no bit errors on the RLP layer.

RxLev / RxQual: RxLev and RxQual have been recorded by MTR (Mobile Trace Recording) during the measurement campaign. The correlation between RXQUAL and the delay is not obvious. However, the different measurements show a basic trend : good quality indicator (RXQUAL small) lead to a small number of retransmissions; A bad quality indicator (RXQUAL high) produces more retransmissions of the RLP blocks. The small correlation is probably imply by the averaging nature of RXQUAL (one value each 0.5 second), which does not enough illustrate the kinds of variations and peaks of the BER. The GSM decoder is very sensitive to the kind of BER variations on one burst and this behaviour is not reported in the RXQUAL value (one RXQUAL value is an average value over 102 bursts).

There is a also connection between RxLev and RxQual. If the RxLev becomes bad the BER on the channels raises which means that the RxQual raises too. A small RXLEV value means that the received power is closed to the receiver sensitivity or noise level. The reduction of SNR (Signal to Noise Ratio) degrates obviously the raw BER.

The reason for the differences between RxLev on downlink and RxLev on uplink is that on the uplink a power control is installed. I.e. that if a MS is close to a BTS the MS reduces the transmission level to a minimum which means that RxLev on the uplink is smaller than on the downlink. If a MS is located far away from the BTS the transmission level will be increased an the difference between up- and downlink reduces and may disappear.

.2 Reflections on some special cases

This chapter presents a set of selected measurements representative of the measurement campaign and illustrate what the GSM specialists consider as typical cases.

.1 Bern, Base Station BEA, outdoor

The measurements close to a base station had been carried out in order to obtain results in a very good GSM environment (from a radio channel point of view).

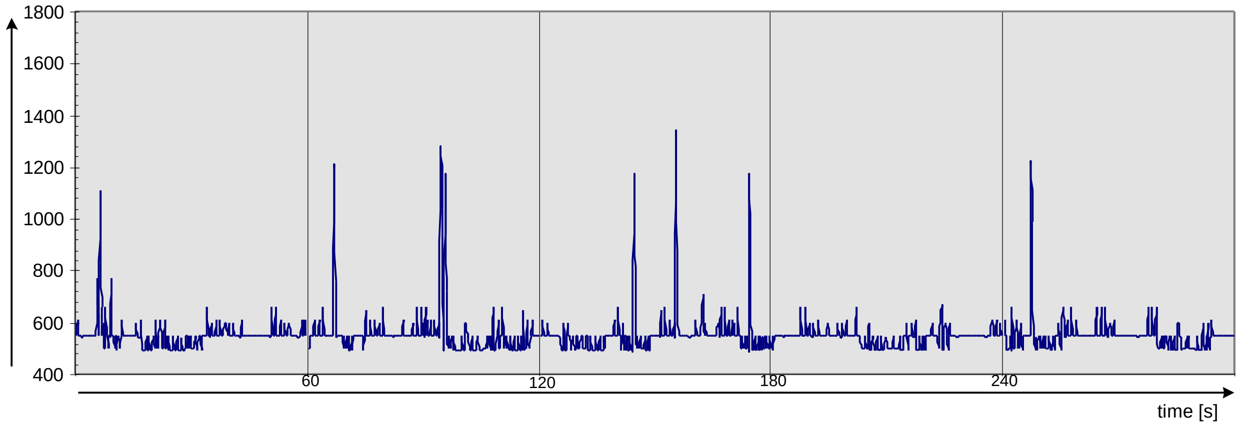


Figure 6-4 Delay during 5 periods of one minute (BTS)

The figure above shows that the delay is very stable. The mean delay is at 550 ms. Investigations have shown that more than 98 percent of the measured delays are between 490 and 720 ms, i.e. 98% of the data are not retransmitted leading to a high throughput (the difference can be looked as measurement 'noise' because the minimal time for a retransmission is about 330ms). Figure 6-5 shows the distribution of the absolute delays.

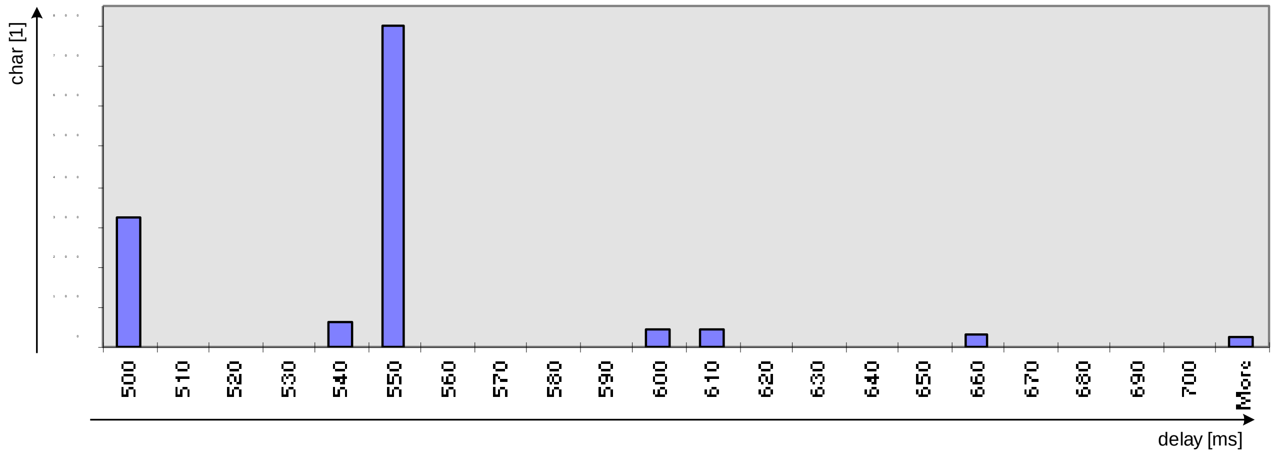


Figure 6-5 Histogram of the delay (BTS)

The number of delays larger than 1050 ms (if we take into account the assumption in the chapter 'General comments' and the inaccuracy of the measurements (noise) this corresponds to 2 retransmissions). The assumption in the chapter 'General comments' is based on the worst case (all retransmissions on the same link). Here in the real measurements it is possible that one retransmission is on the uplink and the other retransmission is on the downlink. The number of delays larger than 720 ms (1 retransmission) is 23. The total of retransmissions is 2 * 9 + 14 = 32. Correlated with the total of measured delays (1333) this corresponds to 2.4 percent which is a very good result, because the throughput can be kept as maximum.

As shown in Figure 6-6 the throughput is very constant. The maximum variation is about 20 percent but in most cases the variation is not higher than 5 percent.

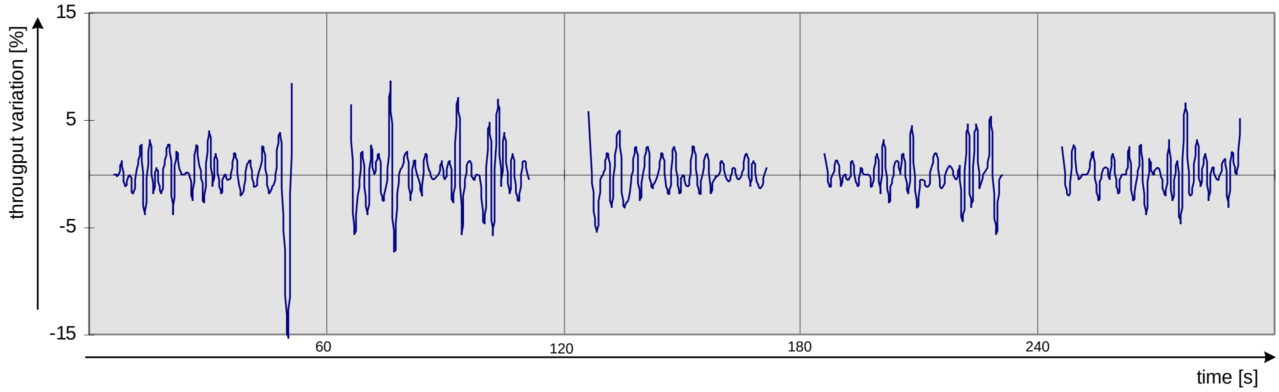


Figure 6-6 Throughput variation in % (BTS)

The mean throughput for each of these five transmission phases can be calculated:

throughput = received chars / transmission time

received chars [1]	transmission time [ms]	throughput [char/ ms]	througput [bit / s]
49980	53390	0.93613	7489
49980	53330	0.93718	7497
49980	53270	0.93824	7506
49980	53280	0.93806	7505
49980	53330	0.93718	7497

Table 6-2 Mean throughput of each measured period (BTS)

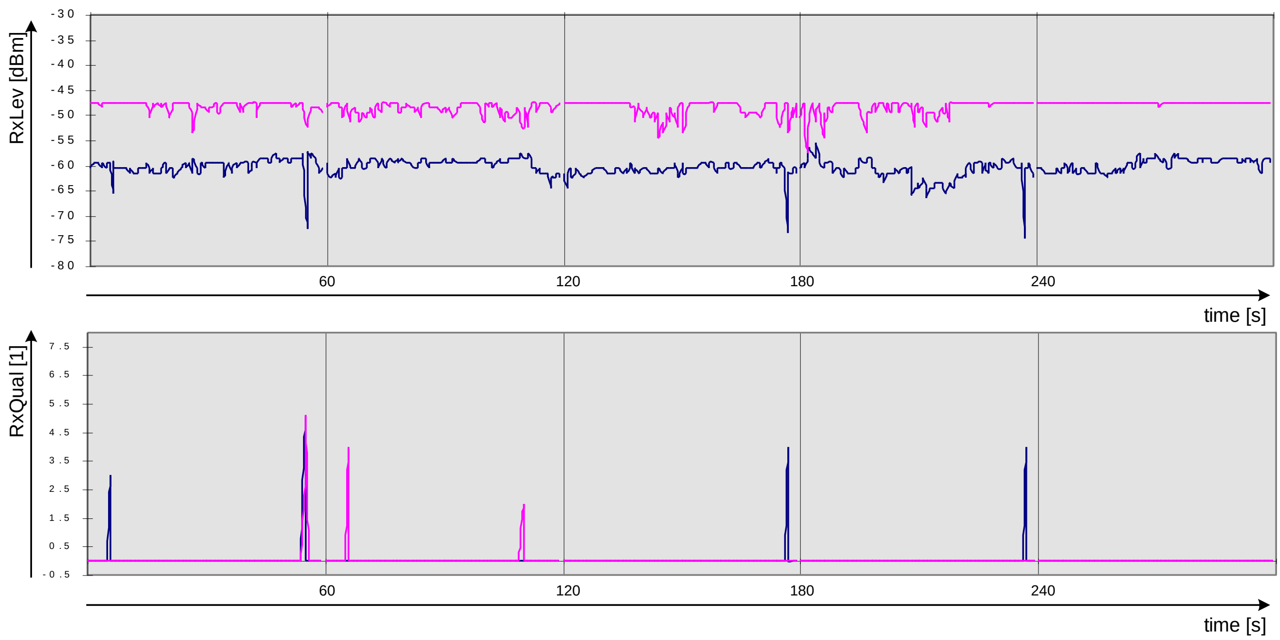


Figure 6-7 RxLev and RxQual during 5 periods of 1 minute (BTS)

The two graphs above show that the measurements were carried out in a GSM environment with a very high Quality. There are some high RxQual values in the graph. This could have different reasons. On one hand this could happen because the GSM environment has changed for a short moment (e.g. a car past in front of the MS). On the other hand it is possible that the DTX feature (Discontinuous Transmission Mode) induces wrong results to the MTR tool (this was already observed for speech during other measurement campaigns).

.2 Bern, Nationalbank, outdoor

The outdoor measurement results mode of the Nationalbank, Bern can be looked as results for a 'normal or typical urban' GSM environment. The results vary not very much from the measurement results presented previously.

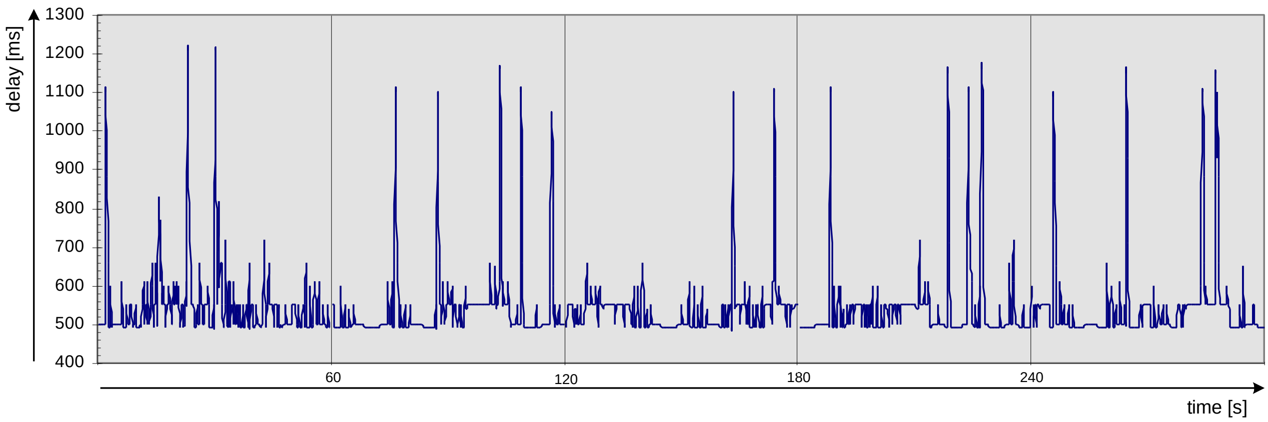


Figure 6-8 Delay during 5 periods of one minute (Nationalbank)

The figure above shows that the delay is very stable also. The mean delay is 537 ms. Investigations have shown that more than 96 percent of the measured delays are between 490 and 720 ms. Figure 6-9 shows the distribution of the absolute delays.

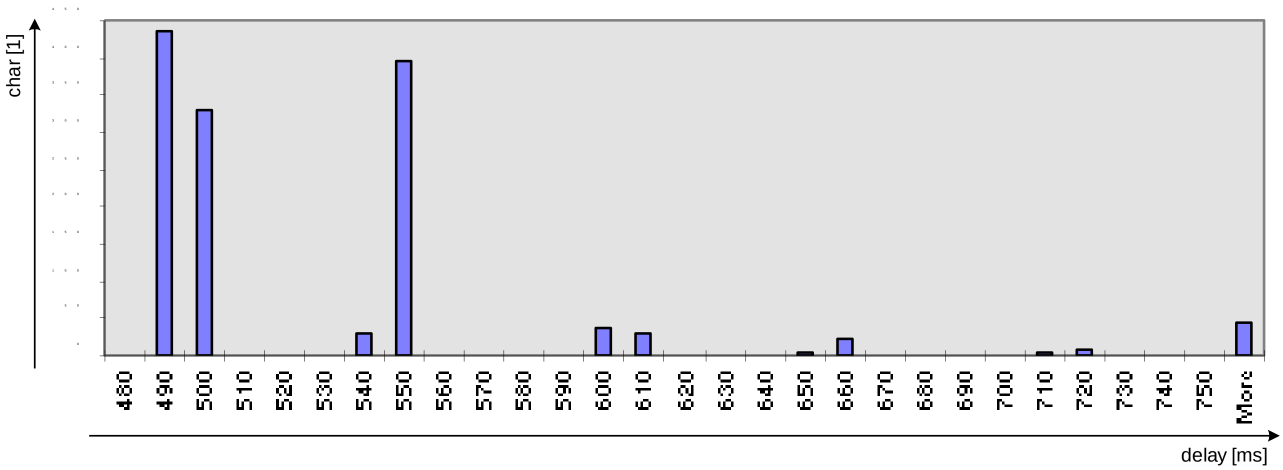


Figure 6-9 Histogram of the delay (Nationalbank)

The number of two retransmissions in this case is 18. The number of 1 retransmission is 25. The total of retransmissions is 2 * 18 + 25 = 61. Correlated with the total of measured delays (1332) the 61 retransmissions correspond to 4.6 percent which is also a good result and it is nearly the double of the retransmissions measured closed to the BTS. A reason of this doubling is not directly explainable with the parameters measured in the GSM system (RxLev, RxQual). Although this statistical doubling of the re-transmission is not quantitatively very important, the difference is explained by the difference in the nature of the radio channel environment. The system may be limited by noise or by interference. The capacity to the standardised GSM decoder to cope with the errors depend on the characteristics of the radio channel, which is not obviously illustrated by the Figure 7-13.

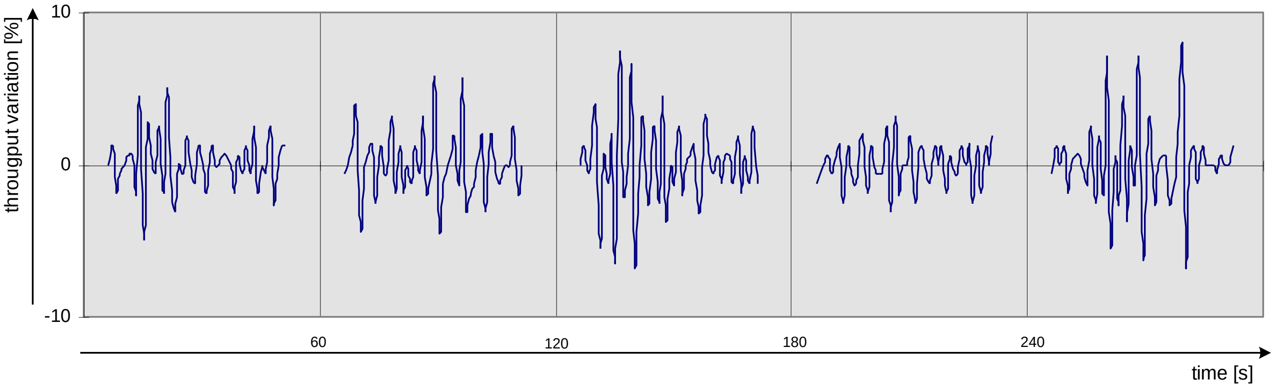


Figure 6-10 Throughput variation in % (Nationalbank)

The figure above shows that the throughput is again very constant. The maximum variation is less than 18 percent and in most cases the variation of the throughput is not higher than 5 percent.

The calculations of the mean throughputs deliver the following results:

received chars [1]	transmission time [ms]	throughput [char/ ms]	througput [bit / s]
49980	53230	0.93894	7512
49980	53110	0.94107	7529
49980	53270	0.93824	7506
49980	53280	0.93806	7505
49980	53270	0.93824	7506

Table 6-3 Mean throughput of each measured period (Nationalbank)

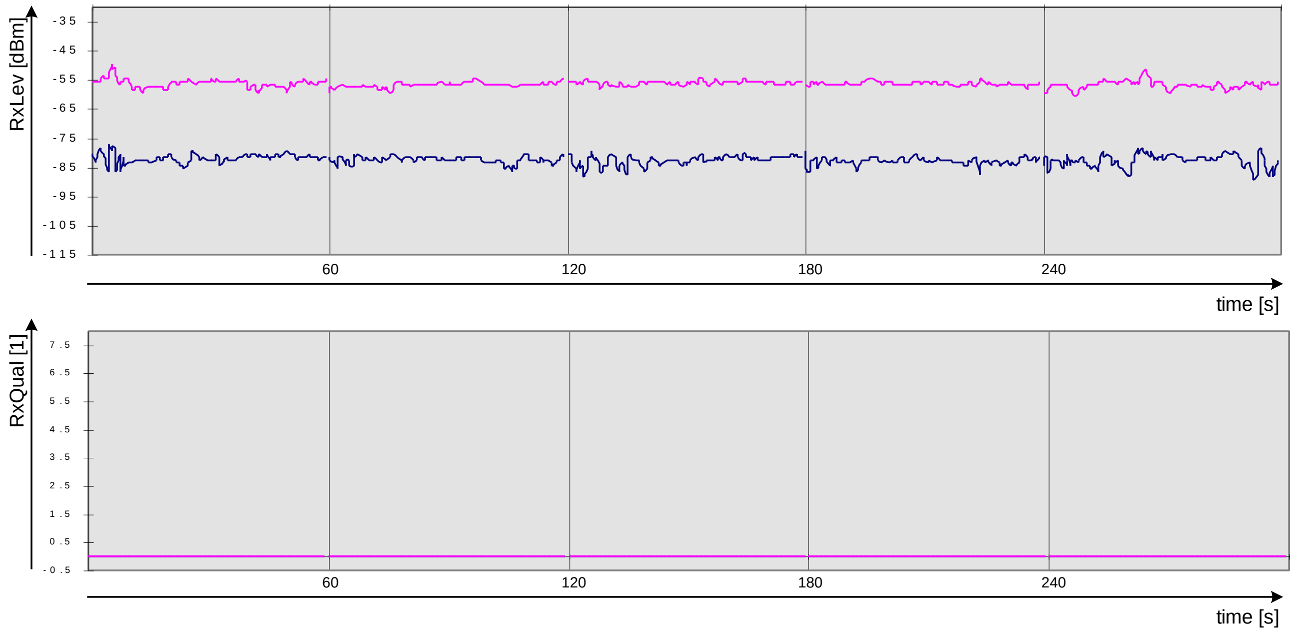


Figure 6-11 RxLev and RxQual during 5 periods of 1 minute (Nationalbank)

The two graphs above show that the measurements were carried out in one a priori very good GSM environment. The big difference between RxLev on the downlink and uplink could be an indication that a BTS was not far away from the measurement location. The RXLEV differences are produced as previously by the power control applied only in the uplink direction.

RxQual always on 0 indicates that the raw BER is not higher than 0.2 percent. I.e. that RxQual = 0 proves not that they are no retransmissions and indicates only an averaged BER.

.3 Bern, Berner Kantonalbank, indoor

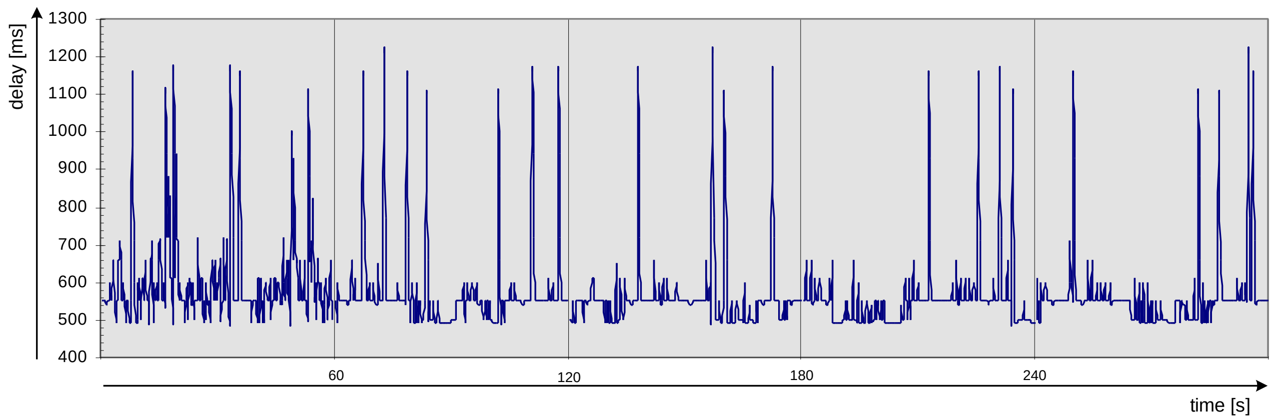


Figure 6-12 Delay during 5 periods of one minute (Kantonalbank)

The figure above shows that the delay varies more than in the two cases presented previously. The mean delay is at 569 ms. Investigations have shown that more than 81 percent of the measured delays are between 490 and 720 ms. Figure 6-13 shows the distribution of the absolute delay.

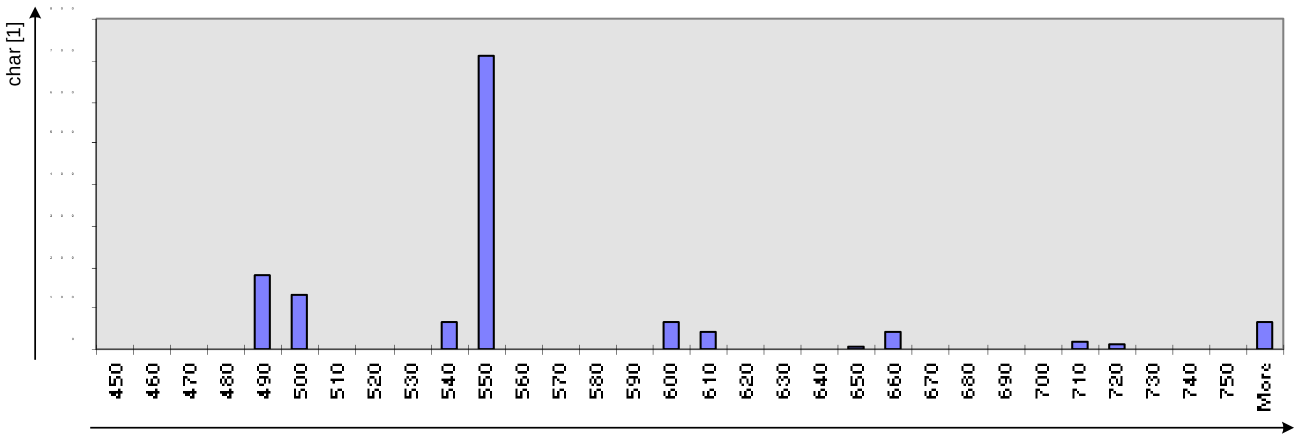


Figure 6-13 Histogram of the delay (Kantonalbank)

The number of two retransmissions is 26. The number of one retransmission is 37. The total of retransmissions is 2 * 26 + 37 = 89. Correlated with the total of measured delays (1335) the 89 retransmissions correspond to 6.7 percent which is a good result but it is almost three times the retransmissions close to a BTS. There are different parameters influencing the increase of the delay. The signal strength (indicated by RXLEV) is smaller, the raw BER (RXQUAL) increases due to the reduction of SNR and probably also due to the fact that the radio environment is more complex (multiple signal reflections and diffractions).

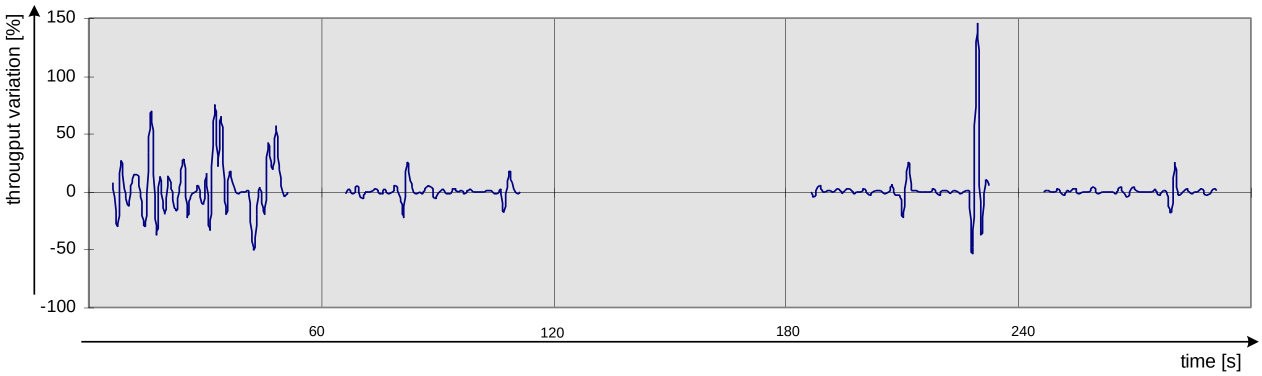


Figure 6-14 Throughput variation in % (Kantonalbank)

The calculations of the mean throughputs deliver the following results:

received chars [1]	transmission time [ms]	throughput [char/ ms]	througput [bit / s]
49980	53390	0.93613	7489
49980	53500	0.93421	7474
49980	53660	0.93142	7451
49980	53330	0.93718	7497

Table 6-4 Mean throughput of each measured period (Kantonalbank)

The figure above shows that the throughput variation in this case is very high (at least in two cases) i.e. that it goes up to more then 100 percent. The interpretation of the high variance lies again on the particular nature of the indoor radio environment, which may produce a lot of retransmissions.

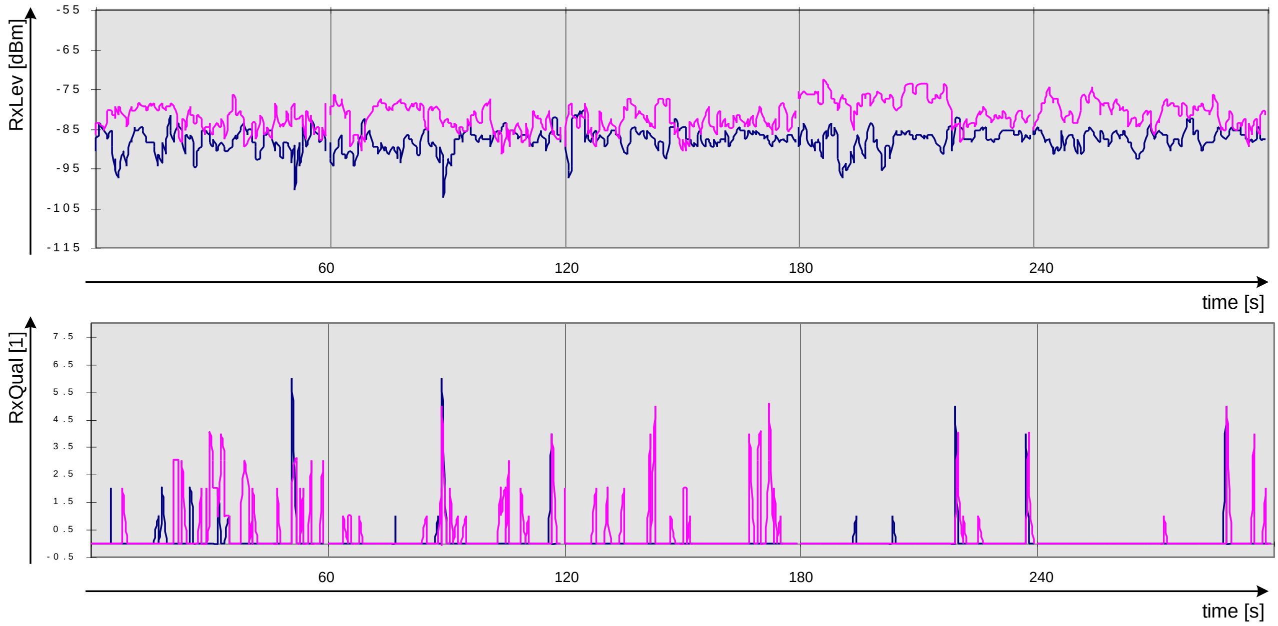


Figure 6-15 RxLev and RxQual during 5 periods of 1 minute (Kantonalbank)

The two graphs above show that the measurement environment is more bad that in the first two cases. Between RxLev on the downlink and uplink there is no big space i.e. that the power control on the uplink was not used because the BTS was far away from a BTS. To be more pragmatic, the BTS can be quite close to the MS, but the resulting radio pathloss is important due principally to material attenuations.

If RxLev is as bad as shown in the figure above the consequence is that RxQual is also bad. The first period corresponds to the first period of the throughput measurements (due to measurements alternance). Assuming that the radio environment was quite similar, the important number of re-transmissions can obviously explain the observed throughput variation.

.4 Zürich, Kongresshaus, indoor

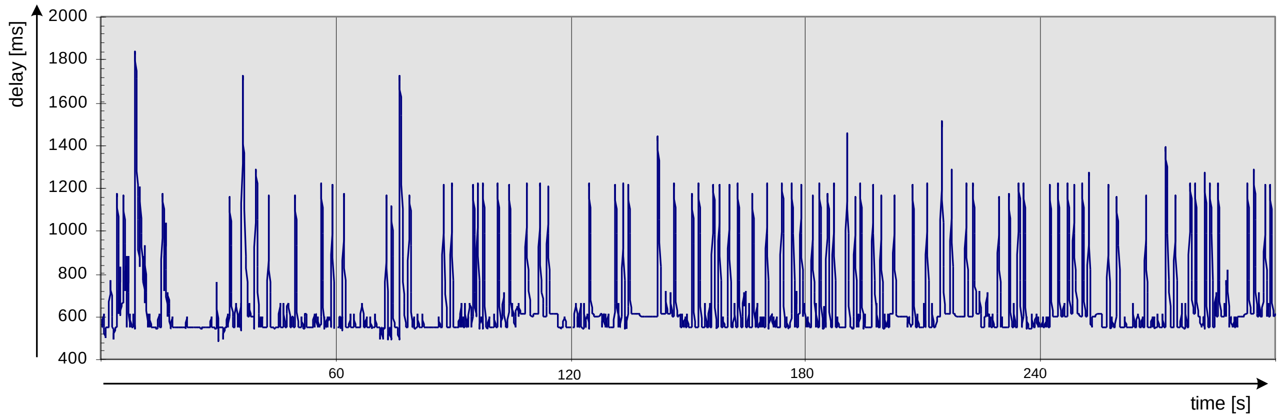


Figure 6-16 Delay during 5 periods of one minute (Kongresshaus)

The figure above shows that the delay varies very much. The mean delay is at 664 ms. Investigations have shown that about 80 percent of the measured delays are between 490 and 720 ms.

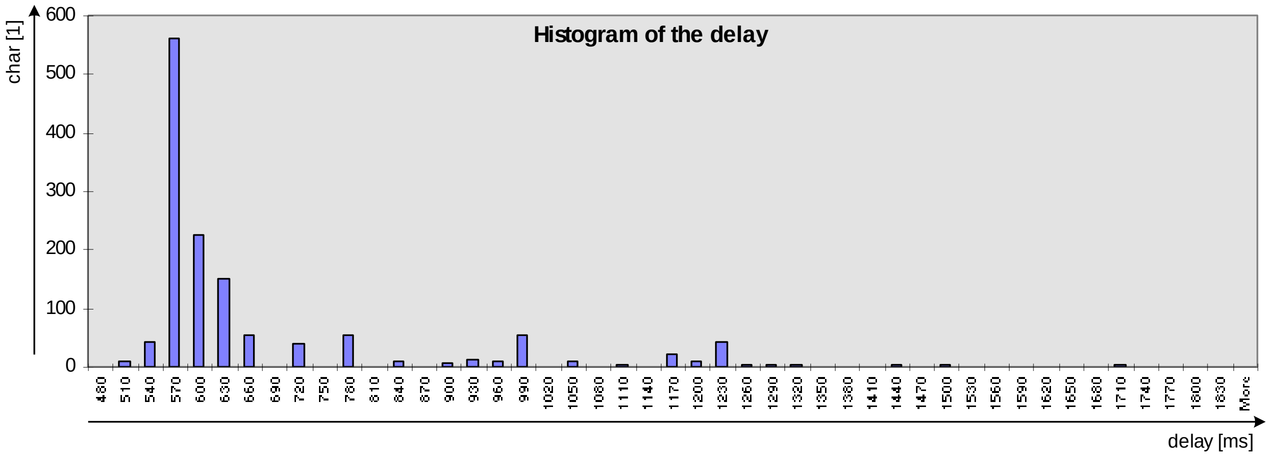


Figure 6-17 Histogram of the delay (Kongresshaus)

The number of 4 retransmissions is 2, the number of 3 retransmissions is 8, the number of 2 retransmissions is 92 and the number of one retransmission is 149. The total of retransmissions is 4 * 2 + 8 * 3 + 2 * 92 + 149 = 365. Correlated with the total of received chars (1328) this corresponds to 27.5 percent which is a bad result.

In this case no statements about the throughput can be made. Because of the very bad network quality at this location the command to send the file did not arrived on the IDA side and it was sent manually. The experienced troubles were not more investigated, considering that such problem is out of scope of this study.

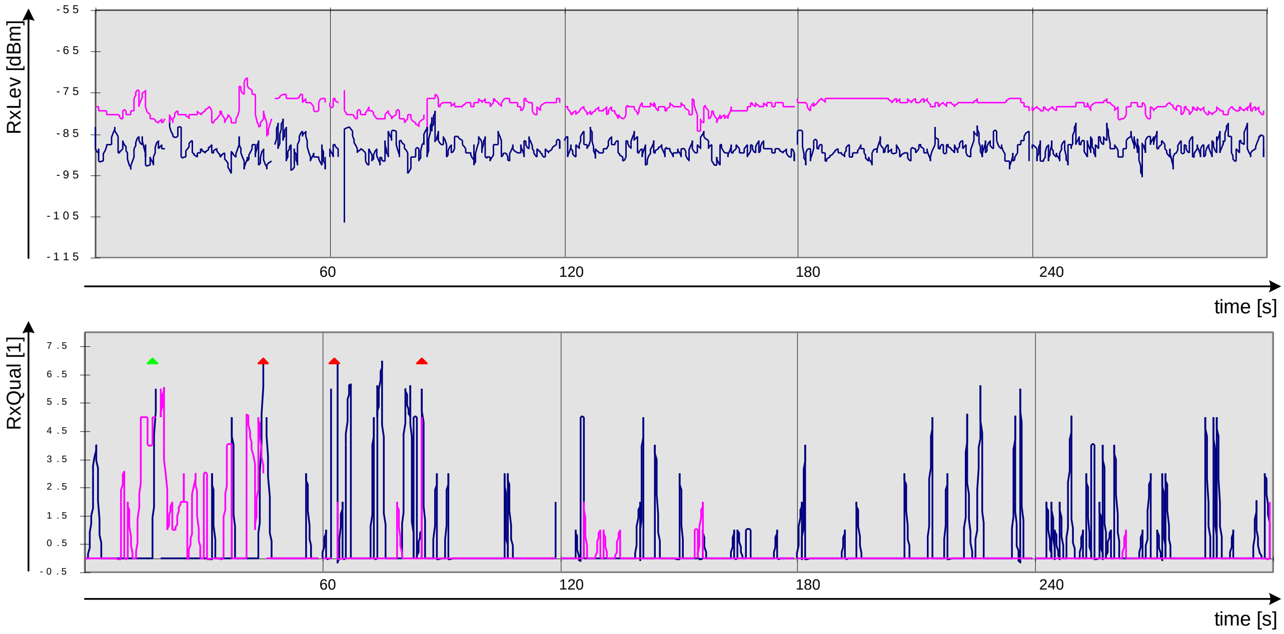


Figure 6-18 RxLev and RxQual during the period of the delay measurements

The statistics of RxQUAL show a more degraded environment than in the other presented measurements. The number of re-transmissions is also more important, highlighting one of the first observation made during the analysis (see Chapter 7.6.1). Nevertheless, the nature of the radio environment of Zurich is particularly limited by the interferences (which is more difficult to decode correctly for the GSM than a network limited by noise).

7. Behaviour of the transport protocol

The investigations related to the behaviour of the transport protocol have been based on the aspects relative to the reliability. Time-outs and sliding windows characteristics were particularly studied in order to understand the potential of the transport protocol.

1 Assumption PPP & IP

For the PPP and the IP protocols assumptions have been made. The focus of the study relies only on transport layer.

.1 PPP

To know the behaviour of the PPP protocol the RFC 1661 has been studied. The protocol has no timer which is important for our investigations.

The assumption for the PPP is that it has only influence on the throughput because the header size is 8 bytes.

.2 IP

The assumption for the IP is that it has only influence on the throughput (min. header size 20 bytes for IP version 4).

2 TCP

The investigations of the behaviour are based on the effect of delay and its variation, both influencing the sliding window and the time-outs. TCP has a minimal header of 20 bytes.

.1 Theoretical background

One of the most important and complex ideas in TCP is embedded in the way it handles time-out and retransmissions. Like other reliable protocols, TCP expects the destination to send acknowledgements whenever it successfully receives new octets from data stream. Every time it sends a segment, TCP starts a timer and waits for an acknowledgement. If the timer expires before data in the segment has been acknowledged, TCP assumes that the segment was lost or corrupted and retransmits it.

The TCP algorithm for retransmissions differs from the algorithm used in many network protocols because the TCP protocol is intended to use in an Internet environment. In such on environment it is impossible to know how quickly acknowledgements will return to the source.

To find an algorithm for retransmissions two aspects have to be taken into consideration. On one hand the *variance in delay* and on the other hand the *congestion*.

.1 Retransmission algorithm

New implementations of TCP can adapt to a wider range of variation in delay and yield substantially higher throughput. The approximations require the following computation which are used to fix the retransmissions (Go back N error control).

$$RTT_{i+1} = (1-\delta) * RTT_i + \delta * delay_i$$

This iterative formula calculates a round-trip estimation (Round Trip Time RTT). Not only the current delay is considered in the calculation but the past too. The *delay* is the measured RTT.

$$RTT\_DEV_{i+1} = \rho * \left| delay_i - RTT_i \right| + (1-\rho) * RTT\_DEV_i$$

RTT_DEV (Round Trip Time DEViation) is the estimated mean deviation between the delay and the calculated round-trip time. The previous RTT_DEV value is considered in the calculation too.

$$Timeout_{i+1} = RTT_{i+1} + \eta * RTT\_DEV_{i+1}$$

δ is a fraction between 0 and 1 that controls how quickly the new sample affects the weighted average, ρ is a fraction between 0 and 1 that controls how quickly the new sample affects the mean deviation, and η is a factor that controls how much the deviation affects the time-out. To make the computation efficient, TCP chooses δ and ρ to each be an inverse of a power of 2, scales the computation by 2^n for an appropriate n, and uses integer arithmetic. Research suggests values of δ=0.125, ρ=0.25, and n=3 will work well. The value for η is 4.

.2 Congestion Control

TCP must react to *congestion* in the Internet. Congestion is a condition of severe delay caused by an overload of datagrams at one or more switching points. When congestion occurs, delays increase and the router begins to enqueue datagrams until it can route them. In the worst case, the total number of datagrams arriving at the congested router grows until the router reaches capacity and starts to drop datagrams.

Endpoints do not usually know the details of where congestion has occurred or why. To them, congestion simply means increased delay. Unfortunately, most transport protocols use time-out and retransmissions, so they respond to increased delay by retransmitting datagrams. Retransmissions aggravate congestion instead of alleviating it. If unchecked, the increased traffic will produce increased delay, leading to increased traffic, and so on, until the network becomes useless. The condition is known as *congestion collapse.*

To avoid congestion collapse, TCP must reduce transmission rates when congestion occurs. Routers watch queue lengths and use techniques like ICMP source quench to inform hosts that congestion has occurred, but transport protocols like TCP can help avoid congestion by reducing transmission rates automatically whenever delays occur.

To avoid congestion, the TCP standard now recommends using two techniques: *slow-start* and *multiplicative decrease.* In the steady state on a non-congested connection, the congestion window is the same size as the receiver's window. Reducing the congestion window reduces the traffic TCP will inject into the connection. To estimate congestion window size, TCP assumes that most datagram loss comes from congestion and uses the following strategy:

Multiplicative Decrease Congestion Avoidance: Upon loss of a segment, reduce the congestion window by half (down to a minimum of at least one segment). For those segments that remain in the allowed window, backoff the retransmissions timer exponentially.

Because TCP reduces the congestion window by half for every loss, it decreases the window exponentially if loss continues. In other words, if congestion is likely, TCP reduces the volume of traffic exponentially and the rate of retransmissions exponentially. If loss continues, TCP eventually limits transmissions to a single datagram and continues to double time-out values before retransmitting. To solve these problems, TCP uses a technique called *slow-start:*

Slow-Start (Additive) Recovery: Whenever starting traffic on a new connection or increasing traffic after a period of congestion, start the congestion window at the size of a single segment and increase the congestion window by one segment each time an acknowledgement arrives.

The term *slow-start* may be a misnomer because under ideal conditions, the start is not very slow. TCP initialises the congestion window to 1, sends an initial segment, and waits. When the acknowledgement arrives, in increases the congestion window to 2, sends two segments, and waits. When the two acknowledgements arrive they each increase the congestion window by 1, so TCP can send 4 segments and so on.

To avoid increasing the window size too quickly and causing additional congestion, TCP adds one additional restriction. Once the congestion window reaches one half of its original size before congestion, TCP enters a *congestion avoidance* phase and slows down the rate of increment. During congestion avoidance, it increases the congestion window by 1 only if all segments in the window have been acknowledged. The following script and example illustrate the congestion control mechanisms. *TimeOut* becomes true if a timer expires and packets have to be retransmitted, *ssthresh* is the size of the threshold and *cwnd* is the current window size.

Calculation:

ssthresh = congestion avoidance threshold
cwnd = size of the current window

```
If (TimeOut) Then
      ssthresh = cwnd / 2
      cwnd = 1
Else
      If (cwnd < ssthresh) Then
            cwnd = 2 * cwnd
      Else
            cwnd = cwnd + 1
      End If
End If
```

Example:

1. The sender starts to send packets
2. A packet is loss and a time-out happens
3. The congestion avoidance threshold is set to half window size
4. The window size increases normally until to the threshold borderline
5. The window size increases only by 1

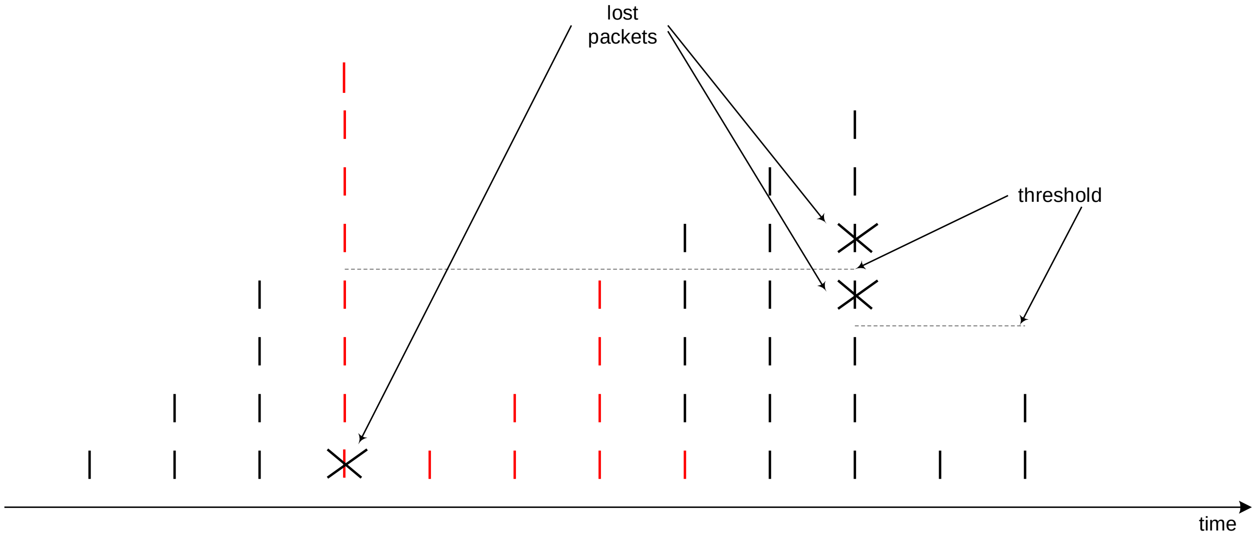


Figure 7-1 Behaviour of the Transport Protocol on a time-out

The sender sends every segment in a window one after the other without waiting for an acknowledgement. If the acknowledgement of the first segment in a window does not arrive (segment lost), all segments have to be retransmitted and it can take along time before new segments can be transmitted (as shown in Figure 7-1). If two or more time-outs happens in a short interval the threshold falls down to a minimum and it takes a long time until the window size arrives at the same level as before.

.3 *Flow control*

The flow control in our measurements has no influence because of data are sent very slowly and in this case we can assume that the recipient has infinite capacity.

.2 Behaviour of short TCP packets

If short TCP segments (for example 48 bits of data) are sent the PPP packets will be dispersed only over a few RLP blocks (in this case 3 RLP packets), knowing that such an RLP block contains 192 bits (24 bytes) of data. The round trip delay on the TCP layer is subdivided on one hand into the time to submit the RLP blocks in both directions (20 ms for each block) and on the other hand into the delay of the RLP blocks themselves (both directions). Only the delay of the latest arrived RLP block is important, because only the maximum total delay indicates when the TCP segment will be considered as received and acknowledged. Figure 7-2 illustrates the composition of the delay. The delay of transmissions of an RLP block corresponds to the one measured in the Swisscom GSM network (see chapter 7.6.1, 'General Comments').

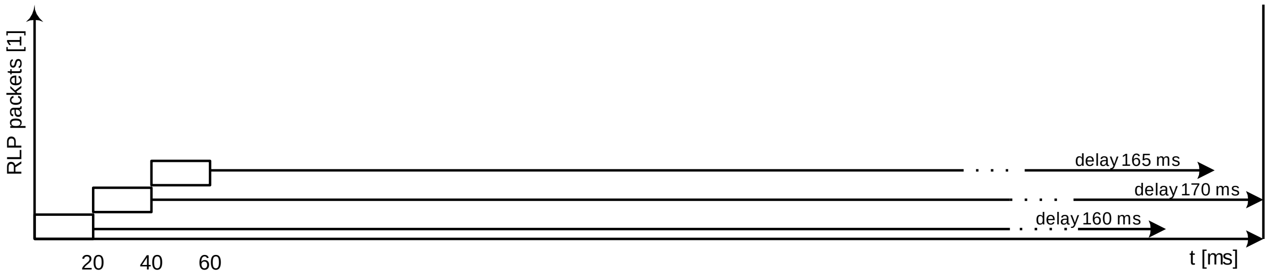


Figure 7-2 Composition of the delay while sending short TCP packets

The figure above shows that the variation of the delay has a big influence on the whole transmission time (especially if an RLP block has been one or more times retransmitted). Figure 7-3 illustrates the behaviour of the timeout computed by TCP assuming that all retransmissions are performed on the same link (worst cases).

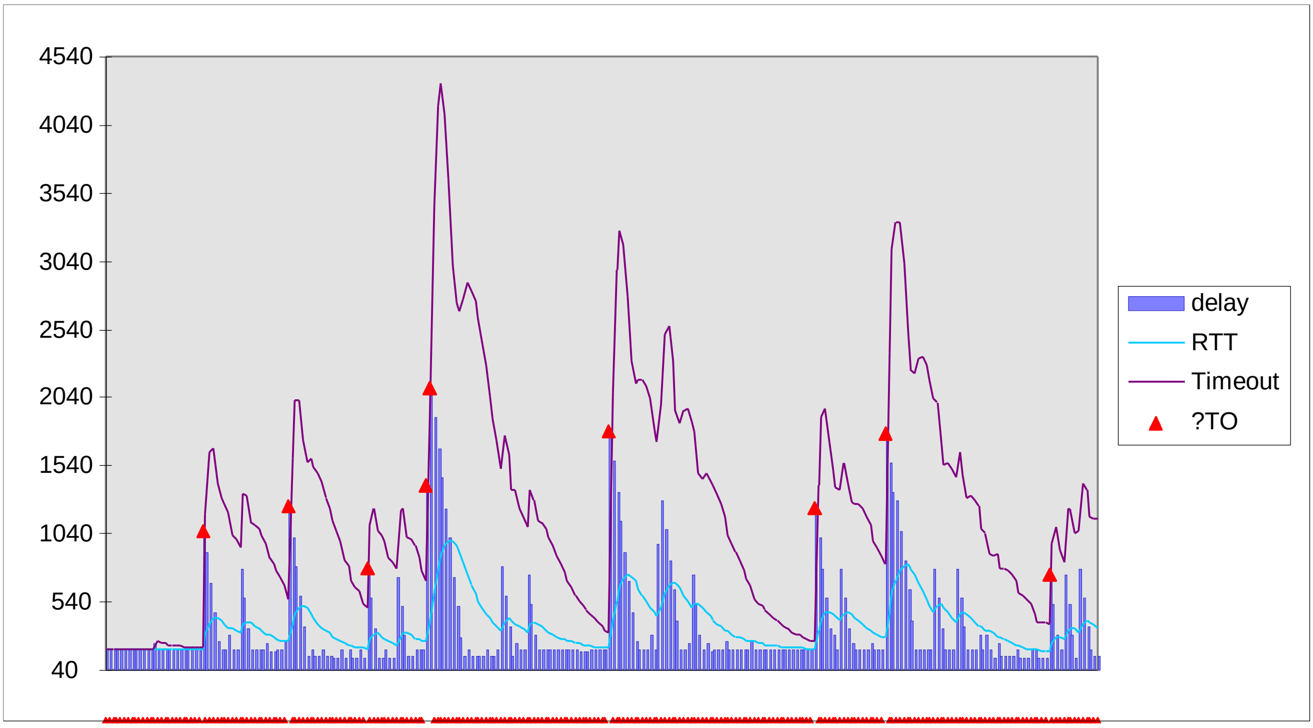


Figure 7-3 Behaviour of TCP while sending short packets

Figure 7-3 shows the foreseeable behaviour of the TCP Layer regarding the retransmission features. The considerations are based on the assumption that if a segment is lost the whole window has to be retransmitted. The blue graph shows the delay and its variation during a period of one minute of the measurements in Zürich. The RTT curve shows (using the formulas described in the theoretical part: δ=0.125; ρ=0.25; n=3; η=4 and the congestion control mechanism) the estimated delay for the next time stamp. The Timeout (also calculated with the formulas above) shows the maximum allowed delay. If the delay exceeds the Timeout a time-out occurs i.e. a packet is assumed as lost and is retransmitted. The red triangles shows the moment of a time-out. The algorithm for calculating Timeout tries to reduce the time-outs to a minimum. This has the effect that the Timeout curve increases very much in case of a time-out. Due to that some following big delays will not produce a time-out. This implies that if 2 (or more) retransmissions are enough closed each other, only the first one will induce a retransmission at the TCP layer. Such a situation can be found in bad GSM environments as shown in the precedent chapters.

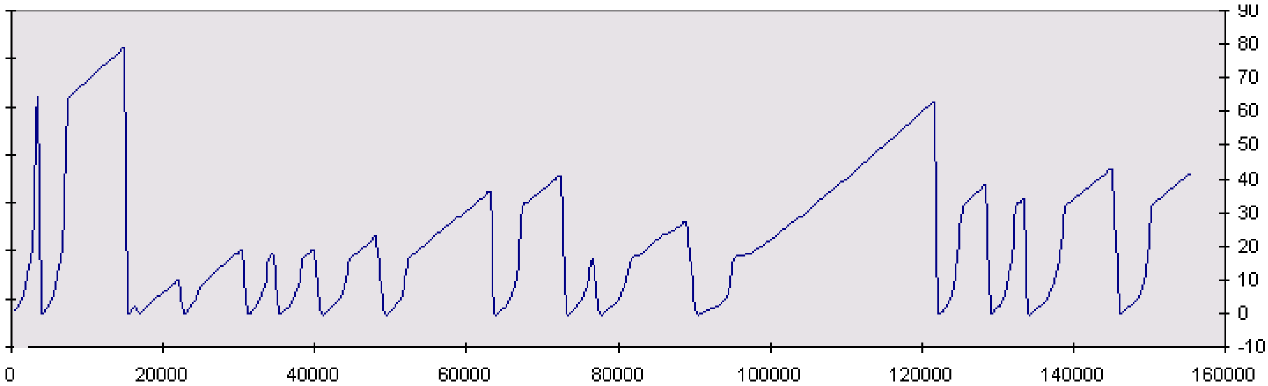


Figure 7-4 Behaviour of the TCP congestion window size while sending short packets

Figure 7-4 shows the behaviour of the sliding window during the same time period. The calculations are based on the formulas described in the theoretical part (*ssthresh* = 65535; *cwnd* = 1). The window increases very fast at the beginning because the threshold is set to 65535. After each time-out the algorithm allows a fast re-increasing until the threshold-line (indicating the congestion avoidance zone). Above this line the window is incremented only linearly. If two time-outs succeed another the window size rises very slowly and is often reduced to a minimum as shown in the figure above.

In conclusion if an application sends short segments as described above the application may show some degradation in the delivered QoS. For an application such as Telnet the GSM user will no be fully satisfied. An application like Telnet will have troubles with delay variations as shown in Figure 7-3. With this frequency of retransmissions the delay perceived by the user will not be regular. One the other hand the sliding window is no problem for Telnet because a constant throughput is not important for a good functioning.

If an application send short segments and need a constant throughput it will have troubles with the sliding window which changes its size very often and implies a non constant throughput.

The congestion control mechanisms of TCP are not well adapted to the kind of delays observed in the GSM environment. The particular nature of the delay statistics does not correspond to the one expected on a typical LAN. In very good GSM conditions (less than 2% of retransmissions), each retransmission of an RLP block will cause a time-out in the TCP layer. A retransmission of an RLP block is perceived as a congestion by TCP. The timer is accordingly adapted and the segment is sent again. Both mechanisms for providing data reliability (RLP and TCP over GSM) work not correctly together. In such a configuration, the TCP protocol is not suitable over the GSM link. The reliability is already ensured by the GSM layer 2. Using unreliable protocols for the transport of data over the GSM link solves the mentioned problem. UDP may be considered as a solution. The current approach proposed by the research community leads to replace the transport layer of the conventional applications by another mobile specific transport layer, which takes into account of the characteristics of the underlying protocols. Different proposals are in discussion, in particular the WTP (Wireless Transport Protocol) supported by the standardisation bodies.

.3 Behaviour of long TCP packets

If an application uses long TCP segments (e.g. 1000 bytes) the behaviour will be different. The PPP packet has to be segmented into different RLP blocks as shown in Figure 7-5. One RLP block will be sent every 20 ms.

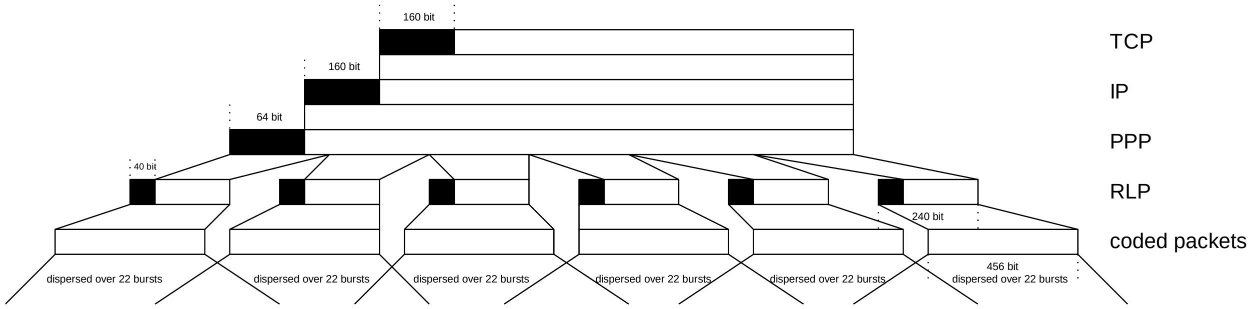


Figure 7-5 Segmenting of the PPP Packets

The PPP packets will be dispersed over a lot of RLP blocks. As described with short TCP segments the round trip delay on the TCP layer is subdivided on one hand into the time to submit the RLP blocks (20 ms for each block) and on the other hand into the transmission delay of the RLP blocks itself on layer 2. Only the total delay of the latest arrived block is relevant for TCP. Figure 7-6 illustrates the composition of the delay when long TCP segments are used. If for example 40 RLP blocks have to be used the RLP block "construction" time is about 800 ms.

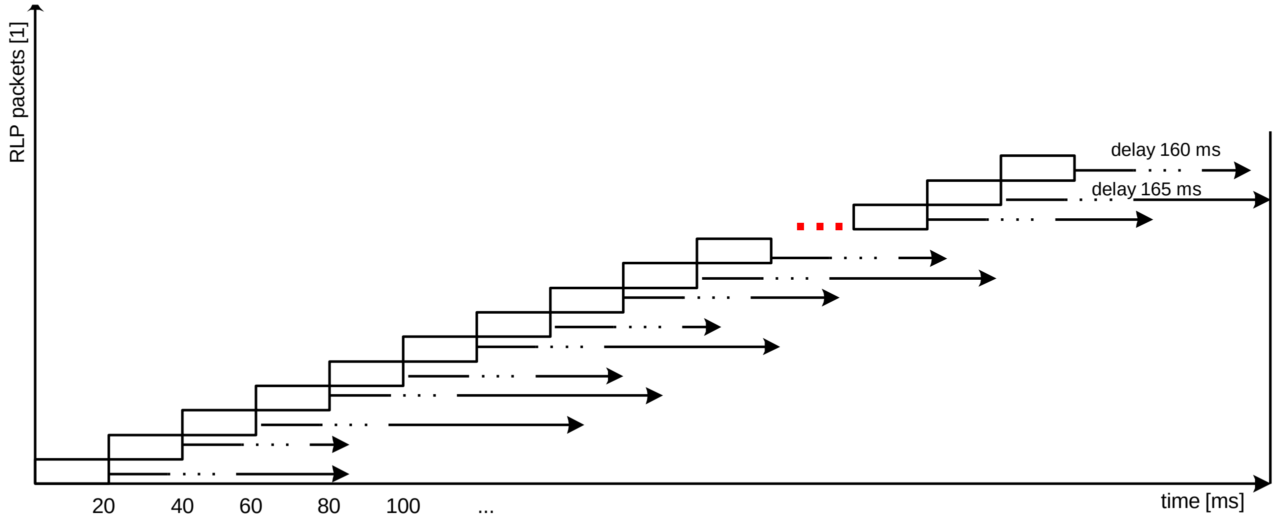
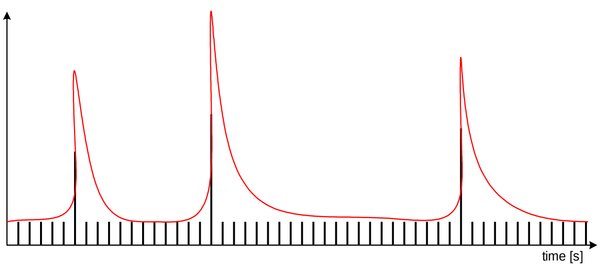


Figure 7-6 Composition of delay while sending long TCP packets

The sender (application) will try to send one segment after the other. Because of the limited throughput only one segment after the other can be sent. The result is that the window size will always remain at 1. The TCP will wait for the acknowledgement before sending a new packet.

The behaviour of the Timeout curve is different if long TCP segments instead of short TCP segments are sent. With short segments the transmission delay is closed to the one observed at the RLP level.



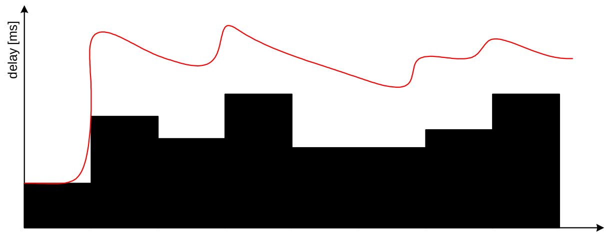


Figure 7-7 Behaviour of the Timeout with short and with long TCP segments

As shown in the figure above with short TCP segments there are a lot of segments having more or less the same transmission delay (in good GSM conditions, it may correspond to 2% of retransmissions). If a segment with a larger delay arrives a time-out occurs and the Timeout curve increases immediately. Because of the immense number of segments with constant transmission time the Timeout curve is adapted to the transmission delay again. This implies that at each RLP retransmission (in good radio channel environment), a TCP retransmission will also occur.

In the case of sending long TCP segments the transmission times have another statistical distribution. The delays of the individual RLP blocks has not the same influence compared to short segments. The transmission time for long TCP segments shows a more fluctuating behaviour. The number of variations is higher than for small segments, but the amplitudes of the variations are smaller. In other word, the delay does not show a "stepwise" characteristic (corresponding to the RLP block retransmissions). Due to the small but permanent fluctuations of the delay, the TCP protocol determines in average a higher timer value, because the deviations are taken into account in the computation of the time-out. For this reason, the probability of having retransmissions at the TCP layer is much smaller with long TCP segments than with small TCP segments. Unfortunately, the required time to construct and transmit the RLP blocks is important and can be a potential problem with application requiring short absolute delay (telephony, teleconferencing …).

Applications using long messages as described above and not sensitive to the absolute delay (like time critical applications) will work well. The throughput variation will be reduced to a minimum and the delay will be kept regular. Real-time applications e.g. will not work correctly because they require short absolute delay.

8. Measurement in the GSM Network (higher Layers)

1 Principle

The goal was to carry out measurements by GSM data with using the TCP / IP protocol stack. These measurement results should be used to verify the estimations about the behaviour of the transport protocol.

2 Definitions

The definitions of the measurements is documented in *Definition of Measurements in phase 3, version 1.0* (Appendix 3).

3 Measurement results / Verification of the estimations (see appendix 4)

Telnet: As described in *Definitions of Measurements in phase 3* there was no possibility to measure the transmission time of the TCP segments sent by Telnet.

To verify the estimations made in the previous chapter, a PPP connection to the IDA platform has been set up. After the login on the IDA platform a Telnet session to our Linux host has been set up. While typing different characters the following observations can be made:

1. If any characters on the computer keyboard are typed it takes a long time (more than 1 second) until the echo from the Linux host is returned.

2. The delaying is not regularly.

This verification showed exactly the estimated behaviour. The functioning of the Telnet application is not problematic but it is very uphill to use such an application over the GSM network. Someone using often the Telnet application becomes accustomed and can directly correlate the typed chars with the echo and retransmission time. If this is not the case commands often have to be retyped and the functioning of the application is not satisfactory.

FTP: To verify the estimations in the previous chapter a c++ programmed FTP tool has been used.

First a PPP connection to the IDA platform has been set up. The FTP tool allowed to make a FTP connection to our Linux host on the IDA platform. To measure the transmission time a file was downloaded from the Linux host. The TCP segments had a fixed size of 1460 bytes.

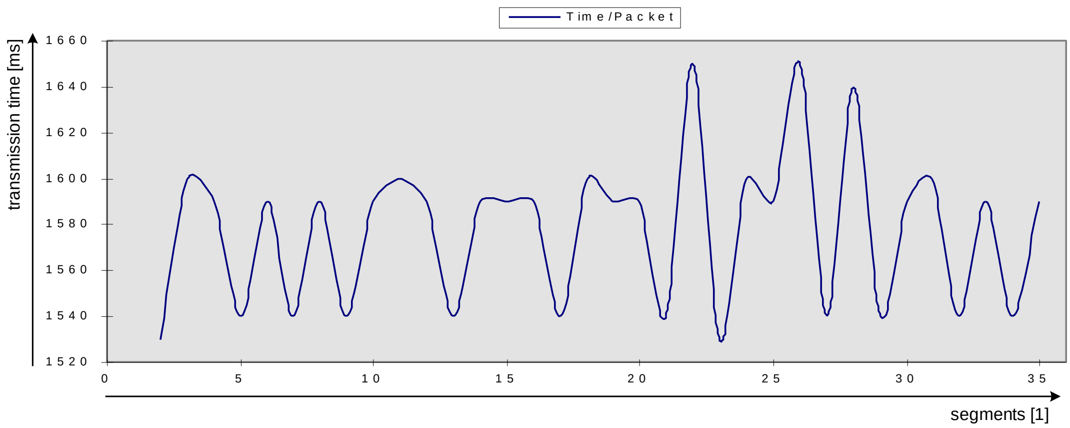


Figure 8-1 Variation of the transmission time using an FTP tool

The figure above shows the variation of the transmission time. It varies between 1530 ms and 1650 ms (see appendix 4). The maximal variation is about 120 ms. Correlated with the mean transmission time (1578 ms) the maximal deviation is 7.6 percent which is a very good result.

Because the absolute delay is not critical for the application, it works very satisfactory. This confirms our theoretical estimations. Such a stable transmission time illustrates clearly that no retransmissions are performed on the TCP layer.

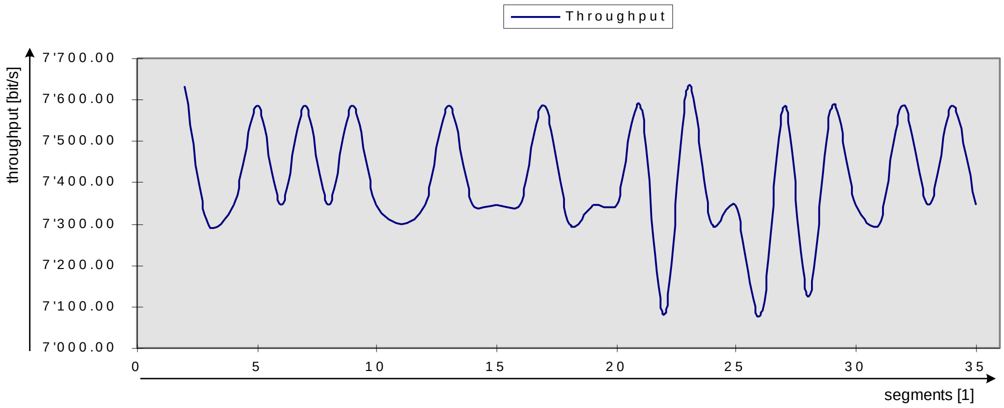


Figure 8-2 Variation of the throughput using an FTP tool

The figure above shows the throughput variation from one to next received FTP segment. The mean throughput is at 7404 bits/s. The maximal throughput variation is 325 bit/s or 4.4 percent which is also a very good and stable result (see appendix 4). The verification shows that it is true that the transmission delays on the RLP layer have no big influence on the throughput while sending long TCP segments.

9. Conclusion

The theoretical considerations related to the behaviour of TCP and its verifications have shown that certain applications will suffer from a degradation in the delivered QoS when used over a GSM network. On one hand applications using long TCP segments and not critical on absolute delay can be used without any problems. One the other hand real-time applications (having strict requirements regarding the absolute transmission delay) will not work correctly.

The introduction of HSCSD will not solve the problems experienced by the time critical applications. HSCSD will increase the bandwidth, but will have no influence on the transmission time. The absolute delay can not be reduce by HSCSD. It may be also expected that e.g. Voice over IP or Videoconference application will not work correctly. The bandwidth increase allows a higher throughput and can be very useful for Internet applications (such as FTP or WWW).

The coming GPRS will enable to work directly with IP (or X.25 packets). Unfortunately, the lowest layer of the GSM protocol stack remains unchanged (except of the coding schemes). The problem posed by the absolute transmission delay remains open. The current approach supported by different manufacturers and research institutes leads to replace the transport layer by another mobile specific transport layer (as middleware solution). Different proposals are in discussion, in particular the WTP (Wireless Transport Protocol) supported by the standardisation bodies. The goal is to use the different features already present in the underlying layers (reliability, synchronisation, …) in order to optimise the throughput and the delay of transmission.

I can say that this diploma work was very interesting. I didn't know the GSM technology before and I had the chance during these 9 weeks to profit from many GSM specialists by Swisscom. It was very interesting to define and especially to perform measurements in the real Swisscom GSM network. A lot of different problems had to be solved in this diploma work and I am happy that I had a grateful support from many people by Swisscom. I look forward that I can use these knowledge in my first job.

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