VILNIUS GEDIMINAS TECHNICAL UNIVERSITY

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INVESTIGATION OF EVALUATION AND IMPROVEMENT METHODS OF SERVICE QUALITY IN MOBILE NETWORKS

Doctor dissertation

Technological science, Electrical Engineering and Electronics – 01T

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MOBILIOJO TINKLO PASLAUGŲ KOKYBĖS VERTINIMO IR GERINIMO BŪDŲ TYRIMAS

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CONTENTS

1. INTRODUCTION

Relevance of the dissertation. GSM mobile networks are very widely used in the world and the number of GSM users exceeds 10^9. In the last decade, the GSM/GPRS network has been rapidly expanded in Lithuania as well. Coverage of the network has reached 99 % of territory of the country. Load of the network in largest cities reaches up to several hundred Erlangs for square kilometre. The voice traffic still generates the largest part of the network load, however the mobile data traffic also grows rapidly. Radio interface between the user of mobile services and the network is not stationary. The received quality of service (QoS) of every user in different places is different. Even in a single cell radio link characteristics differ in different parts of the cell. The signal level may easily vary about 10–20 dB just within several or tens of meters. Moreover, a mobile user experiences different link conditions while moving in the natural environment within or between cells. However, historically, characteristics of QoS of mobile services are measured and evaluated like the ones of the fixed network using statistical methods. Statistically calculated characteristics of QoS do not include information about the received quality of mobile telecommunication services of every individual user.

Conception of the received quality of services of individual users (individual QoS – iQoS) in the GSM/GPRS network has not been analysed before. QoS should be measured not only at the network level, as it is done today, but also individually in every user terminal. Possible ways of measurement and evaluation of the received voice and data QoS by an individual user need to be investigated. This would increase satisfaction of users, allow implementation of service level agreements for cellular network customers and introduce new value-added services of high benefit to users (such as quality dependent billing).

The objective and tasks of the dissertation. The objective of the dissertation is to investigate methods of measurement, evaluation and improvement of iQoS of data and voice users in mobile telecommunication networks.

The tasks for achieving this objective:

1. to create and present a conception of individual quality of services (iQoS) of the user in mobile telecommunication networks and offer a method of measurement of iQoS;
2. to develop a system adequate simulation model that will evaluate needs of users and predict the necessary network expansion and the QoS an individual user will receive;
3. for improvement of iQoS for mobile data users to investigate a possibility of transmitting data of users using parallel mobile data channels.

Scientific novelty of the dissertation. Quality of the received voice and data services is a widely analysed topic in the world, however QoS received by every particular user in the mobile network has not been analysed yet. Interface between the user and the network is not stationary, therefore statistically calculated QoS does not indicate the actual QoS the individual user receives. In the dissertation a new (not analysed before) conception of the received quality of services of individual users in the GSM/GPRS network is proposed.

Aggregation of parallel data channels is a quite old technology, however combination of parallel mobile data channels for connecting remote or moving LANs or WLANs has not been analysed yet. In the dissertation the existing technologies for combining data channels are analysed, modelling as well as an experimental investigation are done and suggestions for technology of combining parallel mobile data channels are formulated.

The main novelty elements in the dissertation are:

1. a new conception of individual user quality (iQoS) and measurement method of iQoS in the network;
2. measurement and modelling results of impact of radio channel quality on quality of voice and data services;
3. an improved model of common voice and data resources of the GSM/GPRS network and measurement results that confirm the model adequacy.
4. results of experiments of combining parallel mobile data channels using EIGRP and TEQL protocols.

In the dissertation the author defends:

1. The conception and measurement method of individual quality of services (iQoS) in mobile networks.
2. Measurement results of voice and data characteristics of QoS of a Lithuanian mobile network operator.
3. The updated model of management of common voice and data recourses of the GSM/GPRS network.
4. Experimental results that indicate a possibility to use a bundle of parallel mobile data channels for connection of remote or moving LANs/WLANs.

Approbation of the dissertation. The main results of the dissertation were presented during these reports in scientific conferences:

1. Batkauskas V. Evaluation of intensity of electromagnetic fields created by radio objects in close surroundings. In Lithuanian. Conference “Lietuva be mokslo – Lietuva be ateities”, Vilnius 2002.
2. Balkė R., Batkauskas V. Analysis of quality of service (QoS) of GPRS data channel. In Lithuanian. Conference “Lietuva be mokslo Lietuva be ateities”, Vilnius 2003.
3. Batkauskas V. Investigation of singularity of a GPRS data channel. In Lithuanian Conference “Elektronika 2003”, Vilnius 2003.
4. Dzindzelėta B., Batkauskas V. Software methods of increasing capacity of a GSM network. In Lithuanian. Conference “Elektronika 2003”, Vilnius 2003.
5. Chmylko V., Batkauskas V. Analysis of construction of a model of a GPRS data channel in a mobile telecommunication system. In Lithuanian. Conference “Lietuva be mokslo – Lietuva be ateities”, Vilnius 2004.
6. Balčiūnas D., Batkauskas V. Simulation of a common data and voice channel in a GSM/GPRS network. In Lithuanian. Conference “Lietuva be mokslo – Lietuva be ateities”, Vilnius 2004.
7. Batkauskas V., Balčiūnas D. Optimization of common voice and data channel resources in GSM/GPRS network. Conference “Elektronika 2004”, Vilnius 2004.
8. Batkauskas V., Kajackas A. Impact of electromagnetic interference on QoS in cellular networks. International Wrocław Symposium and Exhibition On Electromagnetic Compatibility EMC 2004, Wrocław 2004.
9. Batkauskas Vaid., Batkauskas Vygin. Investigation of control of remote microprocessor systems through telecommunication networks. In Lithuanian. Conference “Lietuva be mokslo – Lietuva be ateities”, Vilnius 2004.
10. Krisiūnienė A., Batkauskas V. Accounting and billing in GPRS data network. In Lithuanian. Conference “Elektronika 2005”, Vilnius 2005.
11. Kvederas A., Batkauskas V. Influence of radio channel quality on the mobile data transmission trough EDGE mobile data channel. In Lithuanian. Conference “Elektronika 2005”, Vilnius 2005.
12. Batkauskas V. Experimental investigation of third generation mobile network in Lithuania. In Lithuanian. Conference “Elektronika 2005”, Vilnius 2005.
13. Batkauskas V. Use of parallel GPRS/EDGE/UMTS mobile data channels for connecting of WLAN access points. In Lithuanian. Conference “Elektronika 2005”, Vilnius 2005.
14. Batkauskas V. Investigation of technology for connecting of WLAN hot spots using a pool of parallel mobile data channels. IADAT-tcn2005 International Conference on Telecommunications and Computer Networks. Portsmouth 2005.

Practical use of results of the dissertation. Results of the dissertation are used for development, standardisation as well as implementation of measurement and evaluation technology of iQoS in mobile telecommunication networks. These new ideas were presented in the IEEE Communication magazine in 2004. Later iQoS was analysed and quoted in five publications and in proceeding books of three conferences abroad as well as was included in recommended article lists of two universities. The updated simulation model was used for calculation of QoS received by users and forecasting of necessary GSM/GPRS network resources in the Omnitel mobile network. The proposed technology of combining parallel mobile data channels will be used for connection of moving WLANs to the internet in public transport.

Structure and scope of the dissertation. The dissertation includes 5 chapters. In the first introduction chapter relevance, the objective, the main tasks of the work as well as scientific novelty, practical use of dissertation results, defended statements and structure of the dissertation are presented. In the second chapter individual quality of services received by the user is investigated. The new conception of the received individual quality of services of users in the GSM network is proposed. It is shown, that QoS should be measured not only at the network level, as it is done today, but also individually in every user terminal. In the third chapter a GPRS/EDGE data transmission technology is analysed. A mathematical model of distribution of common voice and data recourses is made. During the modelling common channel utilization, the number of served users, user data throughput as well as blocking probabilities of voice and data users are analyzed. In the fourth chapter possible solutions of connecting remote or moving LANs using a bundle of parallel GPRS, EDGE and UMTS mobile data channels are investigated. Modelling and measurement results are presented there as well. All investigation results are summarised in the fifth conclusive chapter. At the end of the dissertation the list of references is presented. It includes 117 bibliographical sources and 11 publications of the dissertation author.

Dissertation includes 94 pages of text, 11 tables, 51 figures and 128 references.

Acknowledgement. I would like to thank my scientific supervisor Algimantas Kajackas for everyday help during all dissertation work. I would also like to thank my brother Vygintas and my parents for assistance during the work. Most of all I would like to thank my wife Vaida for her help, support and understanding during these hard working years.

I would also like to thank JSC “Omnitel“ for close cooperation and possibility to perform an experimental investigation using Omnitel the GSM/GPRS/EDGE/UMTS network.

2. QUALITY OF ACCESS AND RECEIVED SERVICES OF USERS IN THE GSM NETWORK

GSM mobile networks are very widely used in the world. Load of the GSM network in largest cities reaches up to several hundred Erlangs for square kilometre. Unfortunately, the main GSM system resources (possible number of usable frequencies) remain the same as in the beginning of evolution of the GSM network while number of users and network load is increasing. Therefore, for network operators and system manufacturers it becomes more and more difficult to guarantee high quality of mobile services.

Radio interface between the user of mobile services and the network is not stationary. However, historically, characteristics of QoS of mobile services are measured and evaluated like the ones of the fixed network using statistical methods. Statistically calculated characteristics of QoS do not include information about the received quality of mobile telecommunication services of every individual user.

In this chapter GSM standards and architecture of the GSM network are shortly reviewed, factors that influence network capacity are analysed and possible ways of increasing GSM network capacity using power control, discontinuous transmission, adaptive multi rate (AMR) half rate codecs and frequency hoping technologies are investigated. During the modelling GSM/DCS network capacity, using different frequency reuse schemes and different capacity increasing technologies in Vilnius city, is calculated. Statistics of dropped calls during 4 year period in the Omnitel network is presented. Interference impact on voice quality in the GSM network is thoroughly analysed. A new precise individual carrier over interference (C/I) measurement method is proposed. Measurements of dependencies of GSM network quality parameters on C/I are presented.

A new (not analysed before) conception of the received quality of services of individual users in the GSM network is proposed. It is shown, that QoS should be measured not only at the network level, as it is done today, but also individually in every user terminal. The offered concept of rating of individual QoS will allow a dynamic assessment, recording and subsequent analysis of the actual QoS, received by a particular mobile user (individual QoS). Realization of this novel service will be achieved by integrating a special QoS module (software agent) into user terminals for monitoring and recording of data of individual QoS.

These new ideas were published in the IEEE Communication magazine in 2004. Later iQoS was analysed and quoted in five publications and in proceeding books of three conferences abroad as well as was included in recommended article lists of two universities [33, 58, 59, 61, 66, 68, 88, 108, 110].

2.1. The Global System of Mobile Communications

The first widely used mobile cellular system was NMT (Nordic Mobile Telephone) created in 1981.

In 1982, CEPT (Conference of European Posts and Telecommunications) established a new investigation group GSM (Group Special Mobile) that was responsible for development of a mobile telecommunication system covering all Europe. The main requirements for the new system were good speech quality, high efficiency of frequency spectrum, compatibility with fixed ISDN networks, low operational cost, support of international communication (roaming), possibility to use small mobile phones and support of new future services.

In 1990, ETSI published the first GSM specification. The first commercial network began operating in 1991. In 1993, 36 GSM operators already operated in 22 countries.

In 1996, the second phase of the integrated GSM 900 /DCS 1800 system was established, the list of possible network services was expanded and HR as well as EFR voice codecs were introduced.

During the phase 2+ in 1997–1999 new data communication technologies HSCSD (High Speed Circuit Switched Data) and GPRS (General Packet Radio Service) were introduced. New VAS (value added services) protocols like CAMEL, SAT, etc. were introduced as well.

In 2001, a new mobile data technology EDGE (Enhanced Data for GSM Evolution) was introduced. The EDGE technology retains the same structure of time slots (TS) of GSM systems and uses 8-PSK modulation. It rapidly increases the maximum end-user data throughput (up to 59.2 kb/s per TS).

The GSM network coverage territory is divided into cells. Every cell is served by a dedicated base transceiver station (BTS). All mobile stations (MS) in a particular cell territory are connected to that cell. When a MS reaches the cell border it reconnects to another BTS serving a neighbouring cell.

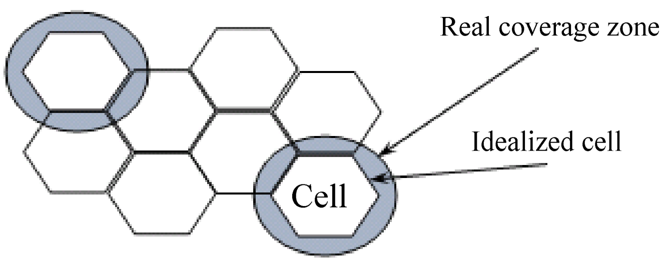

Fig. 2.1.1 An idealized coverage structure of a mobile GSM network

In a real network, every cell is divided into 3–4 sectors. It allows reaching better efficiency of frequency spectrum and higher maximum capacity of the system.

All architecture elements of the GSM network and interfaces among elements are standardised by ETSI [1–4]. Structure of the GSM network is presented in Fig. 1.1.2.

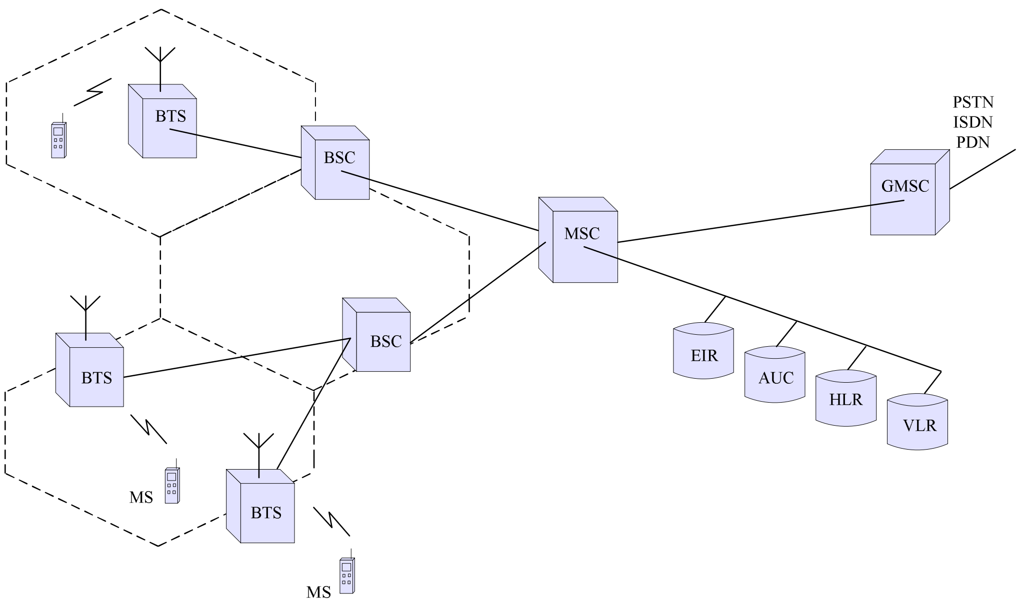

Fig. 2.1.2 Infrastructure of the GSM network

There: MS – a mobile station; SIM – a subscriber's identity module; BTS – a base transceiver station; BSC – a base station controller; MSC – a mobile switching centre; HLR – a home location register; VLR – a visited location register; AUC – an authentication centre; EIR – an equipment identity register.

2.2. Coverage and capacity of the GSM network

GSM networks cover all Europe and are the most popular mobile telecommunication networks in the world.

GSM network coverage is the territory where GSM network services of high quality could be provided. In Lithuania GSM network coverage has already reached 99 % of territory of the country.

High success of the GSM system is high challenge for creators of standards, system manufacturers and network operators. Mobile networks should transport high amount of information, therefore network capacity should be constantly planned and expanded very carefully.

In general, capacity of a GSM network could be measured as the amount of sent information from network square kilometre (Erl/km^2). If network capacity is evaluated using this method, it reflects the real situation of the network in biggest cities where lack of available frequency resources is the highest.

In this section the traditional GSM network coverage and capacity planning are described. Factors that influence network capacity are analyzed and possible ways of increasing GSM network capacity using power control, discontinuous transmission (DTX), AMR half rate codecs and synthesized frequency hoping technologies (SFH) are investigated as well. During the modelling GSM/DCS network capacity, using different frequency reuse schemes and different capacity increasing technologies in Vilnius city centre, is calculated. Statistics of dropped calls during four-year period in the Omnitel network is presented.

2.2.1. Planning of coverage and capacity of the GSM network

The GSM network is planned using rules provided in [1, 18] so that the received signal level and C/I in the covered territory would be higher than the minimum acceptable level. Network planning in rural areas is simple. However, in big cities where high system capacity is required network coverage depends on many factors. To find what cell coverage could be in Vilnius city, modelling of typical city centre environment was done. Results of this modelling are published in [A2].

Modelling was done using typical distances between BTS in Vilnius city centre (0.5; 1 and 2 km). 10 dB dispersion of the signal and 150 dB GSM channel path balance was used. Modelling results are shown in Fig. 2.2.1.

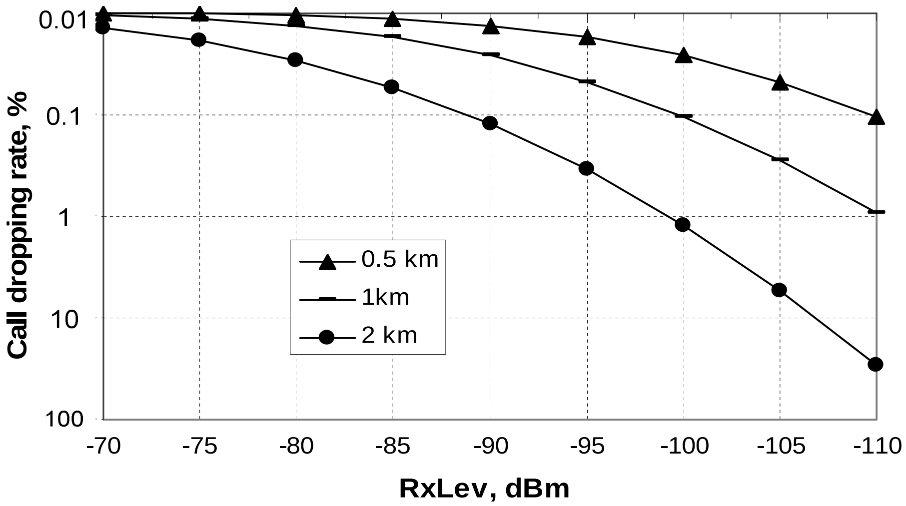

Fig. 2.2.1 Dependency of call dropping rate on the received signal level (RxLev)

From modelling results it could be noted that the marginal signal level (–104 dB) and the minimum acceptable signal quality in dense urban areas could be achieved only in 0.5–1.5 km distance from the base station. In reality, this distance additionally depends on the relief as well as upon the height and type of buildings.

The second modelling was done in order to determine the level of C/I when different frequency reuse schemes are used. Modelling results when frequencies are reused by 1, 3, 9, 12 and 21 sectors (1×1, 1×3, 3×3, 3×4 and 3×7 reuse schemes) are presented in Fig. 2.2.2.

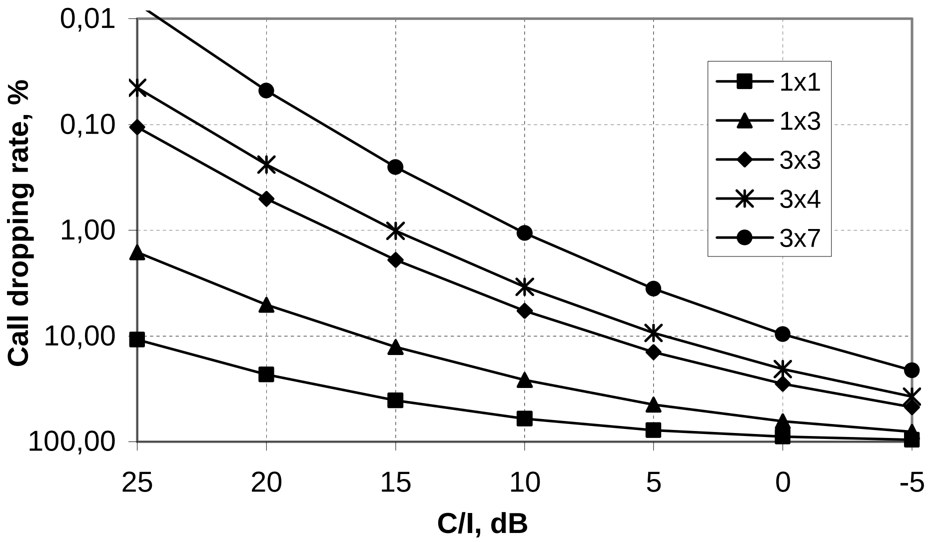

Fig. 2.2.2 Dependency of call dropping rate on C/I

From modelling results it could be noted that the minimum acceptable C/I level (9 dB) and good signal quality in dense urban areas could be achieved using 12 unique frequencies (3×4 reuse scheme). However, practically, good signal quality in 0,5–1,5 km distance could be achieved only using 21 unique frequencies (3×7 reuse scheme). This kind of a reuse scheme has a 4 dB higher C/I level. The 3×7 reuse scheme is needed, because distances between BTS are different and different BTS have different height and propagation conditions. These differences create signal dispersion that is higher than 10 dB.

2.2.2. Impact of Uplink/Downlink Power Control and Discontinuous Transmission on capacity of the GSM network

To minimise system interference and save batteries of mobile phones the transmitted power of BTS and MS is continuously monitored and accurately controlled. The transmitted power during an active GSM session is measured every 480 ms and regulated every 1–2 s. The power of BTS and MS is controlled so, that the signal level and quality would be in the defined range. It works like a typical system with negative feedback. Every moment power of the transmitter is set to the minimum value that could guaranty the minimum acceptable signal level and quality in the receiver.

Power of the transmitter is controlled basing on these two continuously measured system parameters [3, 4]:

- The received signal level (*RxLev*),
- The received signal quality (*RxQual*),

$$RxQual = F(C/I, BER). \tag{2.2.1}$$

Where: *C/I* – carrier to interference ratio, *BER* – bit error rate.

If radio channel quality:

$$Q_i = F(RxLev, RxQual), \tag{2.2.2}$$

is higher than the required quality Q^{req}:

$$Q_i > Q^{\mathrm{req}}, \tag{2.2.3}$$

then C/I and power of the transmitter should be reduced.

If Q_i is lower than the required quality Q^{req}:

$$Q_i < Q^{\mathrm{req}}, \tag{2.2.4}$$

then C/I and power of the transmitter should be increased.

In the GSM system power of the transmitter could be changed in 2 dB steps in range from 20 mW (13 dBm) up to 2 W. The power control algorithm of the transmitter is presented in Fig. 2.2.3 [4].

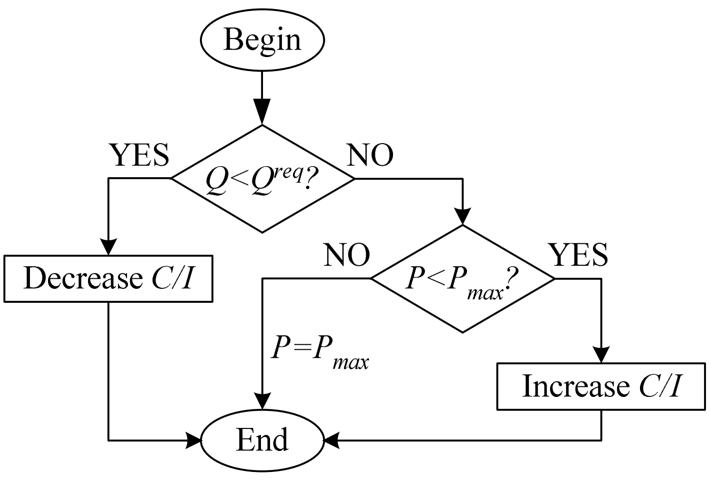

Fig. 2.2.3 The power control algorithm of the transmitter

It is possible to increase power of the transmitter (P_i) till the maximum allowed power of a particular cell or a mobile station transmitter $P_{\max}$ is reached:

$$P_i \le P_{\max}, \tag{2.2.5}$$

For optimal work of the power control algorithm, right system parameters that support the following criteria should be set:

1) maximally reduce the transmitted power in the whole GSM network;
2) to reach the maximum power control speed;
3) minimise possible system instability that emerges due to high system feedback or control speed;
4) to keep radio channel quality in an acceptable range.

Channel quality characteristics using different quality control algorithms [22, 111] are presented in Fig. 2.2.4. These characteristics are results of modelling using many phones taking into account possible signal dispersion as well as fluctuation in time and space.

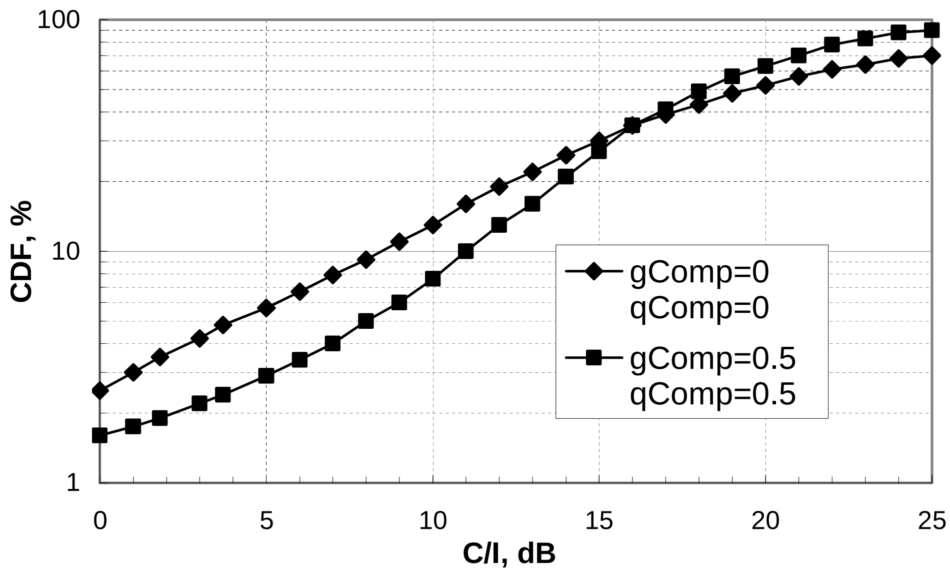

Fig. 2.2.4 Channel quality characteristics using different quality control algorithms

In Fig. 2.2.4 a cumulated distribution function (CDF) of C/I is shown for two cases:

1) power control is switched off (gComp=0 and qComp=0);
2) power control depends on the received signal level (gComp=0.5) and the received signal quality (qComp=0.5).

From Fig. 2.2.4 it could be noted that when power control is switched off about 10% of all calls in the system will be of low signal quality (C/I < 9 dB). When power control based on the received signal level and quality is switched on the percent of low quality calls decreases about two times.

The achieved quality increase could be changed into capacity increase. It could be noted that when power control is enabled the level of C/I increases by 4 dB (if CDF=10 %, C/I increases from 8 dB up to 12 dB). From Fig. 2.2.2 it could be noted that difference of 4 dB of C/I is between 3×7 and 3×4 frequency reuse schemes. If discontinuous transmission (DTX) is switched on it additionally increases C/I by 3 dB. So total C/I increase by 7 dB allows to use even a 3×3 frequency reuse scheme. It rapidly increases the possible maximum system capacity because the same capacity could be achieved using fewer frequency channels.

In cases, when during the power control process the maximum possible power of the transmitter is reached $P_i = P_{\max}$ and if there is no possibility to handover to a better cell, quality of GSM services could be bad. Variations of the received signal level when a MS moves in a low signal level area are presented in Fig. 2.2.5 [A7].

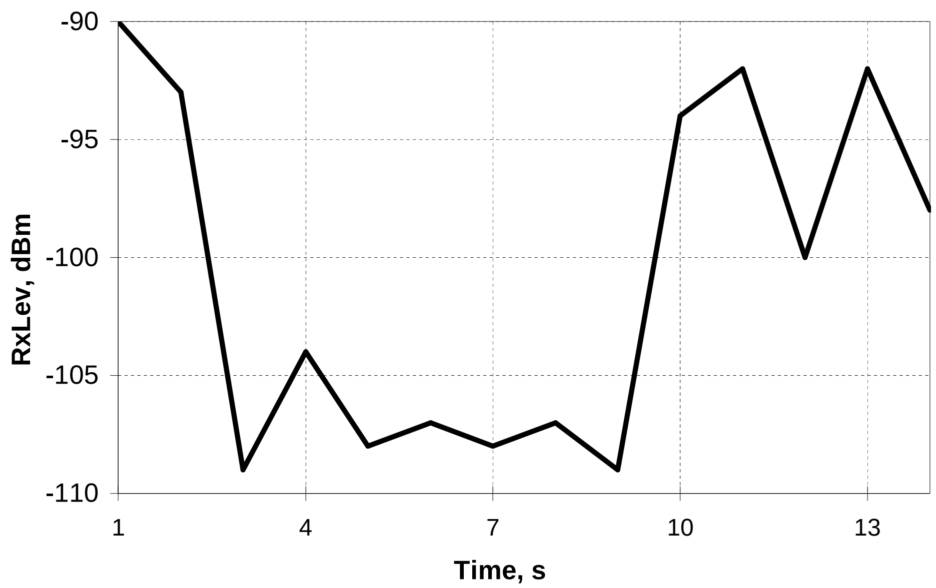

Fig. 2.2.5 Variations of the received signal level when a MS moves in a low signal level area

The real process of power control is difficult because control of power of a particular MS influences working environment of other MS (increase of power in one MS decreases C/I level of all other MS in a particular area).

Evaluation of end user services of the GSM network can not be done basing only on the received signal quality index RxQual (2.2.1) because signal quality and the received service (voice, SMS and etc.) quality are measured in different open systems interconnection (OSI) levels and do not have direct dependency.

2.2.3. Use of the AMR HR codec for expanding of capacity of the network

Use of the Half Rate codec could increase network capacity up to two times. However, a HR GSM vocoder performs reasonably well under optimal network conditions (high C/I). The performance of the Half Rate vocoder quickly deteriorates in low C/I conditions, especially when ambient background noise is present. As a result, Half Rate calls reduce call holding time and customer satisfaction making carriers reluctant to enable Half Rate in their networks.

The Adaptive Multi-Rate (AMR) speech codec has been introduced in GSM standards and is now available in many GSM network products and devices. The AMR codec has the potential to improve significantly call quality and network capacity.

The AMR speech codec includes a set of fixed rate codec modes for full rate (FR) and half rate (HR) operations. It has the ability to switch between different channel coding modes as a function of the propagation error conditions [6, 7]. The AMR HR for speech gives enhanced capacity over the air interface and speech quality using codec mode adaptation. This requires selecting the optimal channel (HR or FR) and codec mode (speech and channel bit rates) in order to deliver the best combination of speech quality and system capacity.

The AMR half-rate channel mode allows two AMR calls to be placed on a single air interface timeslot. Use of the AMR half-rate channel mode increases cell capacity without the need for extra radio hardware. The cost of the increase in cell capacity is lower quality of service provided by AMR half-rate calls. Given that, potentially, 16 AMR half-rate calls can be supported on an AMR half-rate carrier. Due to reduced bandwidth, an AMR half-rate call has, in general, lower perceived Quality of Service (QoS) than a full-rate call. When the received Bit Error Rate (BER) and the received signal quality (RxQual**)** indicate that, a half-rate channel is suffering interference and that the speech quality of the call is therefore impaired, an intra-cell handover back to full-rate is supported to maintain quality of service.

Additionally, some of GSM network manufactures offer proprietary voice quality improvement software. For example Ditech's Voice Quality Assurance™ (VQA™) software improves speech quality employing background noise reduction, speech enhancement, background noise compensation, acoustic echo control, voice level adjustment and optional hybrid echo cancellation. As a carrier deploys a new vocoder, such as an AMR, or modifies the mix of vocoders available, Ditech's VQA features continue to perform voice improvement in the network. Ditech's GSM Capacity Enhancement solution is compatible with all vocoder types – HR, AMR, FR, and EFR – providing carriers with flexibility while transitioning their networks [6, 7].

Use of the AMR half-rate channel mode and additional voice quality improvement software allows increasing network capacity up to 20 – 40 % without noticeable degradation of voice quality. Additional cell capacity depends on C/I conditions in every cell.

2.2.4. Frequency hopping technology for expanding of capacity of the network

To future enlargement of GSM network capacity dispersion of the signal and noise should be minimised. The most effective way to minimise dispersion is the frequency hoping technology. Using this technology, frequency of every cell transmitter is permanently changing 217 times per second. Deviation of level of the signal very strongly depends on the used frequency and signal averaging method. Using frequency hoping low signal level situation decreases very rapidly. It is equivalent to 3–5 dB density increase. Dispersion of the noise signal decreases in the similar way as well. Therefore using frequency hoping technology C/I increases and the system becomes more resistant to noise. Summarised modelling results [99] are presented in Fig. 2.2.6.

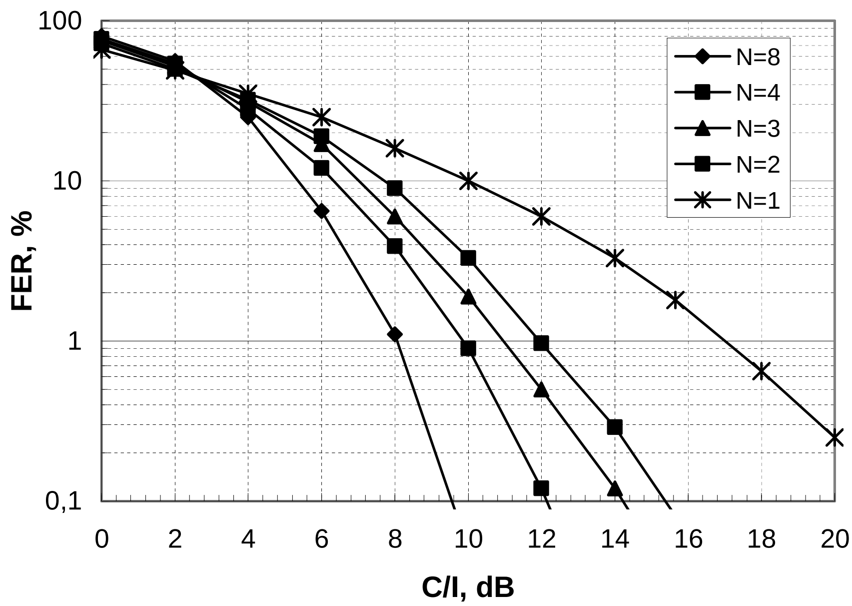

Fig. 2.2.6 Dependency of FER (Frame error rate) on C/I and on the number of unique frequencies in the cell

From modelling results it could be noted that using up to eight frequencies in the cell resistance of the system to noise could increase up to 9 dB. This resistance increase is even higher than resistance increase using power control and discontinuous transmission technologies. This increase in system resistance to noise allows to use more aggressive 1×3 and 1×1 frequency reuse schemes. Then same frequency pool is used in every base station (1×3) or even in every cell (1×1).

2.2.5. Calculation of the maximum capacity of the GSM network in Vilnius city

During the modelling the maximum capacity of the GSM/DCS network in Vilnius city centre is calculated and influence of high network load on C/I in the network is presented.

During the modelling capacity increasing technologies, described in sections 2.2.1–2.2.4, are taken in to account.

Theoretically, it is calculated [91] that using power control, discontinuous transmission and frequency hoping technologies frequency efficiency is limited to 40 % using the 1×3 frequency reuse scheme and to 20 % using the 1×3 frequency reuse scheme. If higher frequency efficiency is used, C/I in some places will drop below 9 dB and degradation of signal quality will be noticeable.

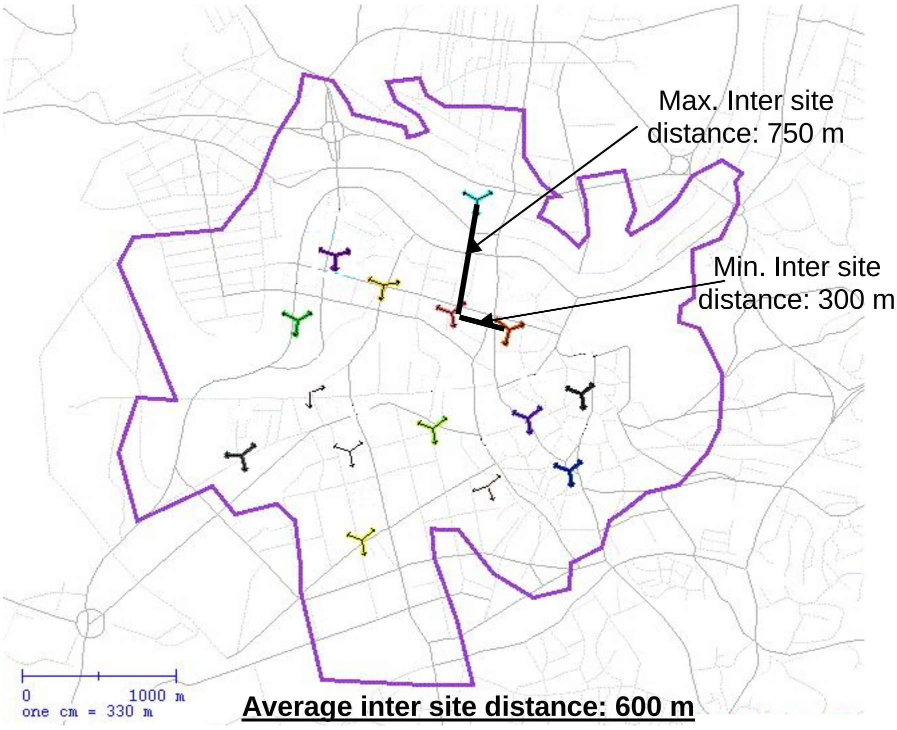

Fig. 2.2.7 GSM/DCS site locations in Vilnius centre

From Fig. 2.2.7 it could be noted that inter site distance in Vilnius city centre varies from 300 m to 750 m. However, in order to simplify calculation an average model inter site distance of 600 m is used. For modelling 40 GSM and 100 DCS frequencies are used. It is almost the maximum possible number of frequencies for one operator when three mobile network operators are in the country.

Summarised calculations of dependency of the maximum capacity of GSM and DCS systems on the frequency hoping reuse scheme and upon use of the AMR HR codec are presented in Table 2.2.1.

Table 2.2.1 Summary of calculations of the maximum capacity of GSM and DCS systems

1% blocking probability	GSM			DCS		
Reuse scheme of BCCH channels	7×3	7×3	7×3	7×3	7×3	7×3
Reuse scheme of TCH channels	7×3	1×3	1×1	7×3	1×3	1×1
Total number of BCCH channels	21	21	21	21	21	21
Total number of TCH channels	19	19	19	79	79	79
Number of BCCH channels in the cell	1	1	1	1	1	1
Number of TCH channels in the cell	0.9	6.3	19	3.8	26.3	79
Number of FR TS in the cell	12.5	26.3	36.4	26.1	86.3	128.4
Number of GPRS TS in the cell	4	8	8	8	16	16
Number of FR Voice TS in the cell	8.5	18.3	28.4	18.1	70.3	112.4
Maximum capacity of the cell, Erl	3.1	10.4	18.6	10.4	56.12	95.4
Maximum capacity of the system, Erl/km2	**32.9**	**110.4**	**197.5**	**110.4**	**595.8**	**1012.7**
Number of Voice TS in the cell using AMR HR	12.8	25.6	36.9	25.3	91.3	134.9
Maximum capacity of the system using AMR HR, Erl/km2	**62.6**	**173.0**	**270.7**	**173.0**	**802.5**	**1234.6**

Summarised modelling results are presented in Fig. 2.2.8 and Fig. 2.2.9.

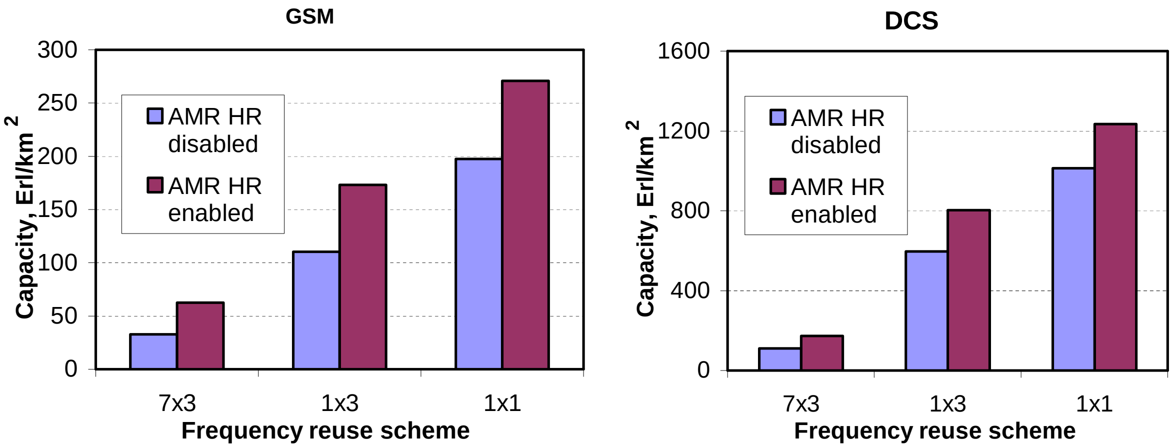

Fig. 2.2.8 Dependency of the maximum capacity of GSM and DCS systems on the frequency hoping reuse scheme and upon use of the AMR HR codec

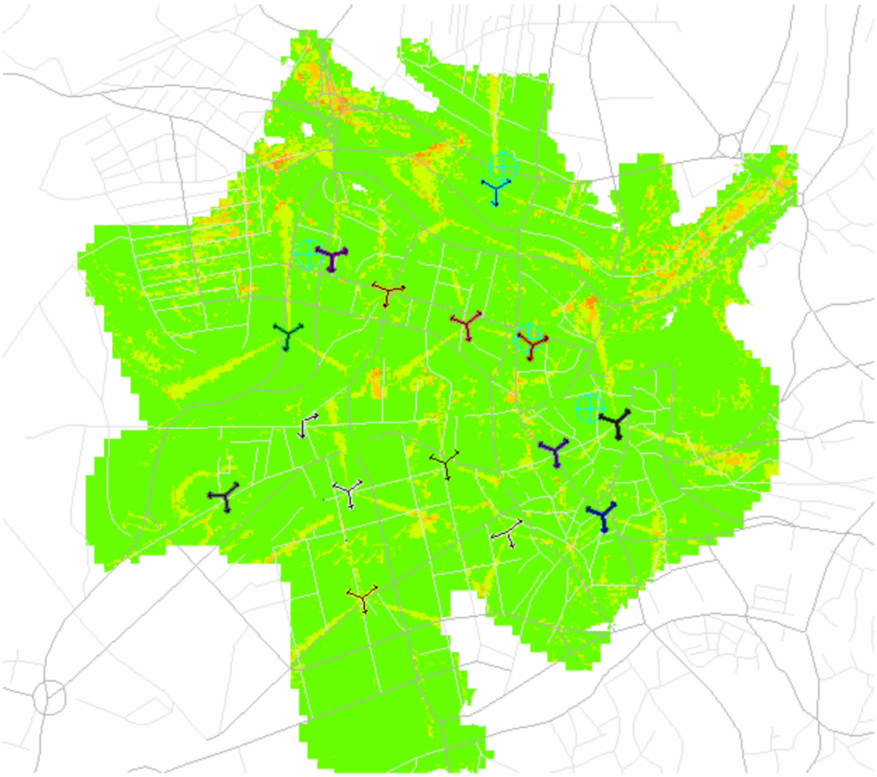

Fig. 2.2.9 Distribution map of C/I of the GSM network when the maximum GSM system capacity is reached (green – C/I > 10 dB, yellow – 10 dB > C/I > 8 dB, red – C/I < 8 dB)

Originally, the GSM network was developed to have higher cell radius and much lower system capacity. However, nowadays capacity requirements in big cities are very high. From modelling results it could be noted that maximum system capacity using frequency hoping could be increased up to 10 times. Use of AMR HR codecs additionally increases system capacity up to 30 %. If GSM and DCS cells are collocated the maximum GSM/DCS system capacity can reach 1500 Erl/km^2.

2.2.6. Results of measurements of dropped call rate in the Omnitel GSM network

During the last six years the number of GSM users and the required system capacity has been growing rapidly. During this time all described capacity increasing technologies were one by one implemented in the Omnitel mobile network. Efficiency of the described capacity increasing technologies could be evaluated observing variations of QoS parameters that occurred during this time. One of the main characteristics of QoS of the network in the Omnitel network is dropping rate of voice calls.

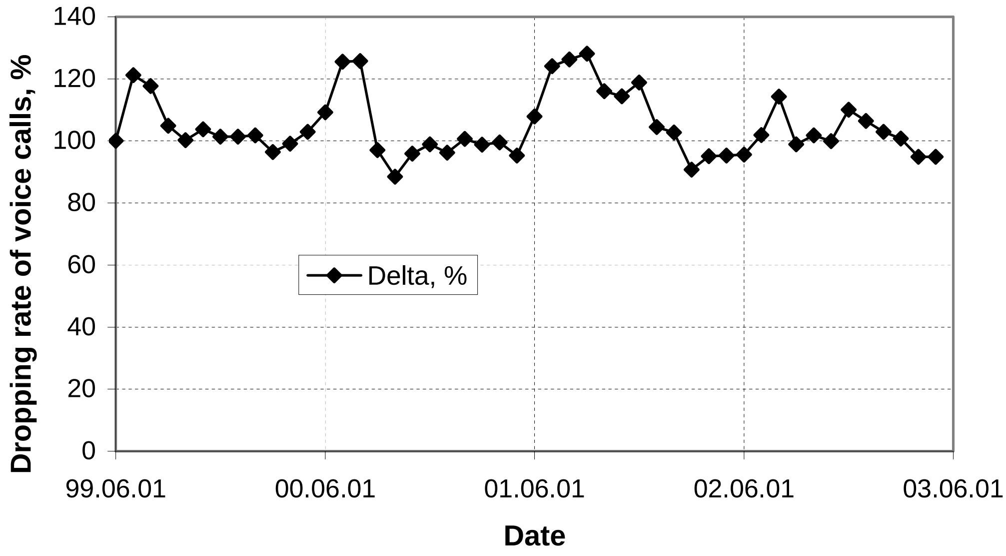

Fig. 2.2.10 Variation of dropping rate of voice calls in the Omnitel mobile network during the years

From Fig. 2.2.10 it could be noted that implementation of the described technologies has allowed to keep average call dropping rate the same during the years while number of users and capacity requirements has been growing rapidly.

2.3. Impact of electromagnetic interference on QoS in cellular networks

Influence of interference noises on GSM QoS was experimentally measured and results of measurements are provided. These results indicate that actually received QoS in different places and for different users is different. Degradation of QoS is noticeable in zones where carrier over interference ratio (C/I) reaches the lowest acceptable level. For determination of C/I situation in the network two different methods are used: simulation and measurement. These C/I definition methods are not precise enough to realize a dynamic and reliable QoS monitoring system. There is a need to find a new measuring method that would indicate C/I situation for each network user and at the same time monitor all network users. In order to achieve high accuracy of C/I measurement, it is proposed to carry out measurements involving all mobile stations (MS) in the network. The analysis of what kind of changes should be made in the equipment of the user and the network in order to support this type of C/I measurement is done. Having collected C/I measurement results from each network user and having correlated them with standard GSM measurements, it is possible to form a detailed and dynamically changing C/I matrix. This matrix would include information about all network users in their natural environment and C/I values would be measured quite precisely.

Use of a precise C/I matrix, especially in heavily loaded networks would make planning of the network and control of QoS easier.

It is well known that GSM cell serving area and capacity are interference limited. Nevertheless, GSM networks should satisfy requirements of quickly growing number of users, pass huge amounts of information at the same time guarantying high QoS.

While capacity of the GSM network grows rapidly, interference reaches a critical level. Even in a single cell area C/I can vary quite much. Therefore, increasing network capacity and at the same time guarantying high QoS, low C/I zones should be indicated very accurately and all possible means of reduction of interference should be applied.

2.3.1. Measurement of interference

There are two different types of interference in cellular systems: internal GSM system interference and external interference. External interference includes natural noises and interference created by other systems. Natural noises during development of the GSM system were evaluated and their influence on QoS is known. The main cause of interferences in a GSM network is interferences of the internal system. They are caused by interference of the same and adjacent frequency channels in the space. Interference appears between fixed BCCH as well as between frequency hoping (FH) channels.

From the practical research it is known that good quality of fixed BCCH channels could be guarantied when the 3×7 frequency reuse scheme is used. The main reason is that distance between sites, height and signal propagation environment are different for different sites. If a frequency reuse plan with fewer frequencies would be used, C/I in some zones would degrade quickly and drop call rate would increase [A2].

In TCH channels for increasing of capacity FH (frequency hoping), DTX (Discontinuous Transmission) and PC (Power Control) algorithms are used. These technologies allow usage of more aggressive frequency reuse schemes 1×3 and 1×1. Theoretically and practically it was determinated that SFH fractional load could be 40 % for the 1×3 reuse plan and 20 % for the 1×1 reuse plan [91]. If the calculated fractional load will be exceeded, the average C/I value will decrease quite quickly [A2].

Influence of interference noises on GSM and GPRS quality of services was experimentally measured using TEMS drive testing equipment [54] in urban, suburban and rural areas. Summarized results of dependence of Bit Error rate (BER) during a voice call on the reception level (Rx Level) and C/I are shown in Figures 2.3.1 and 2.3.2.

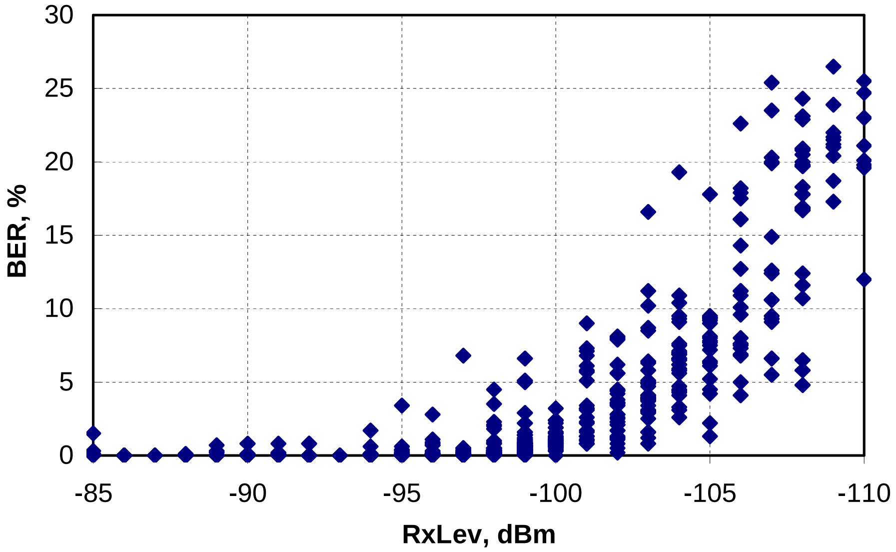

Fig. 2.3.1 BER as a function of the received signal level

As a part of the power control process, both, mobile user terminals and network base stations, measure the level of the received signal. From practical measurements of BER as a function of the received signal level, it could be noted that BER increases significantly when the Rx Level drops below –98 dBm (Fig. 2.3.1).

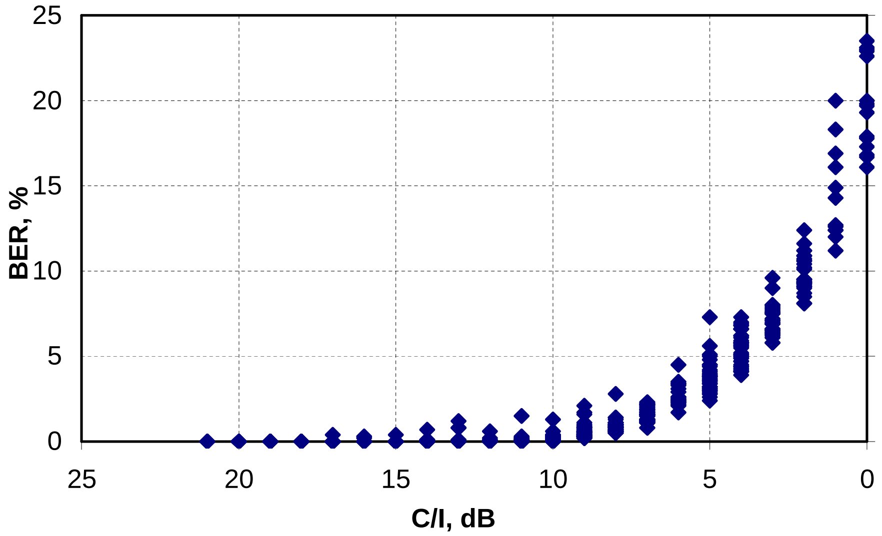

Fig. 2.3.2 BER as a function of C/I

From Fig. 2.3.2 it can be observed that BER increases when C/I drops below 7 dB.

In rural areas interference level is considerably lower than in urban areas. According to that, low C/I zones in rural areas are in places, where carrier signal strength is near the lowest sensitivity level of a mobile receiver. In cities low C/I zones could be found in places, where carrier signal strength is quite high. Therefore low C/I zones and low carrier signal strength zones in urban areas can be different.

The mean values of BER are used in receivers for evaluating the overall quality of radio link – a parameter denoted as RxQual. This parameter describes quality of the received signal in that particular channel, averaged over the reporting period of length equal to certain number of frames. When a user terminal moves in area with poor radio coverage, quality of the radio link is not optimal and RxQual constantly changes reflecting that.

2.3.2. A proposed method of C/I estimation

Rules of planning of the GSM network [1, 18] state that in cell serving area good voice quality will be guarantied when the lowest C/I value will be not less than 9 dB. The measurements show that C/I has high influence on GSM/GPRS QoS. Measurement results indicate that the actually received QoS in different places and for different users is different. For guarantying high QoS for all network users and trying to guaranty high QoS for each network user C/I situation in the network should be monitored and managed very carefully. Low C/I or potentially low C/I risk zones should be indicated very accurately.

Nowadays, for determination of C/I situation two methods are used: simulation and measurement. Simulation systems use digital maps, a detailed database of base transceiver stations (BTS) and mathematical models of electromagnetic field propagation. Measurements are made during drive tests using special measuring equipment.

Both C/I estimation methods are not enough precise. Accuracy of simulation is very influenced by inaccuracies of digital maps, heights and tilts of antennas as well as other kind of initial data. Big errors can occur because of low precision of field propagation models. Precision of drive testing measurements is high. However, these tests cover small territory and cannot be carried out very often.

It could be stated that existing C/I definition and measuring methods do not allow to estimate a dynamic C/I situation in every network place a mobile user can be.

In order to achieve high accuracy of C/I estimation measurements involving all mobile stations in the network should be done. Today measurement of C/I of an individual user during the call and transmission of received results to a BTS are not described in GSM or WCDMA standards. For GSM this parameter is measured only with specialized equipment like Ericsson TEMS [112]. TEMS equipment could be used as a prototype for C/I measurements in mobile phones.

C/I could be simply measured in dedicated mode and average C/I could be presented several times per second. In order to obtain correct C/I values possible use of power control and/or discontinuous transmission (DTX) should be taken into account. If frequency hopping is employed, the average C/I for each frequency should be measured.

The measurement range could be from –5 dB to +25 dB. C/I below –5 dB can be regarded as highly unlikely; in addition, if the number of hopping frequencies is low, C/I values below this limit would normally result in a dropped call.

If downlink DTX is used, the number of bursts transmitted from the base station to the mobile station may be lower than the maximum (depending on the speech activity level of the transmitting side). Measurements should be made only of bursts that are actually sent from the base station and not transmitted bursts should be disregarded.

The number of hopping frequencies determines the number of bursts used for C/I measurement of each frequency. For example, if four frequencies are used, 25 bursts (on average) per frequency are received in every 0.5 s interval. With more frequencies, there are fewer bursts for each frequency. This implies that accuracy of measurements is higher for small sets of hopping frequencies.

For example, if true C/I is within the range 0 to 15 dB and four frequencies are used for transmission, and there are no DTX interruptions, the measurement error is typically smaller than 1 dB. Accuracy of C/I measurements of non-hoping BCCH channels is higher.

If the number of frequencies per one carrier is higher than four and the DTX technology is used, then to obtain 1 dB accuracy, the averaging interval should be increased up to one or two seconds. A detailed explanation – in [112].

If special C/I measuring software (Ericsson TEMS could be used as a prototype) will be implemented into user terminals then QoS could be measured in every user terminal.

Details of C/I measuring, processing of received results and their transmitting to the network should be discussed separately. Presently it could be stated that QoS together with location-based service [55] creates a new functionality in the network management and control centre. Use of this functionality allows to create a detail C/I map (matrix) of the network. Then, during a mobile call, QoS expressed as $Q_k(Y, Z, t)$ would constantly change reflecting the actual QoS received by that particular user. Evaluation of the received quality may be done over a reasonably short time, e. g. several seconds. Since a person can tell a word in a one-second interval, impairment of quality over that period could be noticeable.

Nowadays Nokia, Motorola and other vendors offer call trace possibility of all network users. This system is made of data capturing units that are connected to every BSC, as well as the central server, that is responsible for collection of data from capturing units.

This equipment collects only standard service information from the GSM/GPRS network. This kind of information is used in an interface between a MS and a BTS (for example: the received signal level (RxLev) and the received signal quality (RxQual) in uplink and downlink, timing advance (TA), the cell number, handover information ant other). If a MS would be capable of sending results of C/I measurement to a BTS, then further collection of data and transfer of it to the central server could be realized through the existing system.

Having collected C/I measurements from each network user and having correlated them with standard GSM measurements (RxLev, TA, the cell number) and with position of the cell, it is possible to form a detailed and dynamically changing C/I matrix. This matrix would include information about all network users in their natural environment and C/I values would be measured quite precisely. Moreover, this matrix could be dynamically updated every 24 hours or every hour depending on the needs of operator and capabilities of data collecting hardware.

The described system could be used in order to evaluate not only the C/I situation of each user, but also to receive information about QoS of voice and data calls of every user. The proposed method could provide objective estimation of successful call setup and completion as well as instant monitoring of the received service quality.

2.4. Rating of individual QoS for voice services in cellular networks

Observation and management of Quality of Service (QoS) in cellular networks is a very important function. In this section it will be shown that it should be performed not only at the network level (as it is done today), but also individually in every user terminal. The existing centralized network QoS systems only collect data, that allows evaluation of average QoS at the network or cell level. However, they do not provide any objective data for estimating QoS that was actually served to individual mobile users. The offered concept of individual QoS rating will allow dynamic assessment, recording and subsequent analysis of the actual QoS, received by a particular mobile user (individual QoS). So the obtained individual QoS ratings could be then reported to the user directly (as a part of an intelligent user terminal concept) or otherwise become a part of service level transactions (billing, etc) between the user and the network operator [A3]. Realization of this novel value-added service can be achieved by integrating a special QoS module (software agent) into user terminals for monitoring and recording of individual QoS data.

2.4.1. Existing methods of evaluation of QoS received by the user

Quality of service (QoS) in telecommunication networks is an important topic that has been analyzed previously by many authors in various aspects. However, there is at least one aspect of QoS that seems to be undervalued. It is insufficiently represented in literature and is largely non-existent in telecommunication networks. This missing part is the estimation of QoS provided de facto to individual users (hereinafter referred to as individual QoS or iQoS).

It can be noted, that estimation of satisfaction of end users plays a key role in analysis and design of telecommunication systems. QoS perceived by end users is one of the main metrics for comparison of different system options and choice of alternative solutions for improvement of the system in the network design phase. The task is usually to achieve a satisfactory trade-off between the need to guarantee a certain degree of QoS and minimization of the cost of network design.

When dealing with cellular systems, system analysis and design are usually based on estimation of some critical parameters at the cell level. These parameters include, for example, the average number of busy channels and call blocking probability. The average number of busy channels is a measure of resource utilization and is typically an indication for the system operator about costs and efficiency of investments. Call blocking probability, instead, is a probability that a connection request cannot be satisfied due to the lack of available resources. Blocking probability at the cell level is thus used as an indication of dissatisfaction of the user. However, QoS perceived by the user in reality is much more multifaceted as it reflects probability that a whole call is completed successfully [A1].

In addition to the prior QoS modelling in the network design phase the operational QoS control is then performed in a live network via the QoS monitoring function in the Network Management and Control centre [A2]. Monitoring of QoS is needed in order to detect and locate any degradation of QoS performance below planned values. In addition, distribution of QoS across the whole network, instead of simply end-to-end QoS, needs to be monitored (distribution of QoS as experienced in real-time in different network segments is monitored) [6, 7].

Some novel concepts of QoS and underlying network architectures have been realized during design of 3G networks [5], providing means for distributing radio resources among different groups of users according to their individual preferences and QoS demands.

However, it may be observed that all these current efforts to model and monitor QoS in mobile networks address only the global layer (at the network, sometimes cell level) of management of QoS, do not providing any real means for monitoring of the actual QoS received by every user in the given network. Therefore, the QoS metric, as it is applied today, looses its meaning of ranking the actual service level, delivered to individual subscribers, rather being a generalized estimate of average QoS provided to all users. Moreover, while this traditional simplification worked reasonably well for POTS in fixed telephone networks, it may be potentially deceptive in the case of voice services provided in cellular mobile networks.

2.4.2. Arguments for adopting the new concept of Individual QoS

In its general meaning, QoS refers to ability of a telecommunication system to provide an appropriate transport service to deliver various types of communications traffic to different users at satisfactory measure. Sometimes it can be difficult to define the exact technical parameters required to ensure such delivery, especially because perception of "service quality" may differ from one user to another.

Thus it may be observed that whatever global QoS management concept is realized in the network, already by definition it could never produce results that would guarantee the same level of satisfaction to each and every user. This becomes even more complicated in the cellular network, where the interface between the network and users – realized via the radio connection – is not stationary. This non-stationary nature of connectivity in cellular networks is partly a result of circumstances that are common to any other kind of telecommunication network (bandwidth overloading, its randomness over the time). Connectivity is not stationary also due to inherent mobility features of cellular networks, i. e. the unexpected and ever changing physical location of mobile users.

Non-stationary nature of interface of radio network means that chances of successful communication (establishing and completing a call) will depend on the physical location of a user relative to the serving network node, which typically will be the closest base station. It is obvious that if a user terminal will be located within an optimal distance and at favourable radio visibility conditions, it would greatly increase chances of successful communication with high QoS. Contrary, being located near the edge of the coverage area (cell) makes communication more difficult and resource demanding.

Our analysis of traffic in one of the largest Lithuanian networks showed that the average rate of dropped calls in the whole network is reasonably small (1–2 %), that is less than the typical license-set limit of 5 %. However, a few users experienced much more frequent occurrences of dropped calls, some of them even having extreme cases of up to 30 % of dropped calls. This means that while the overall network-wide QoS remained at the sufficiently high level, some of the users may become victims of this averaging process and be charged by their network operator for the service of quality, that they actually did not receive.

It is clear that there might be different contributors having impact on iQoS, some of them radio interface related (like coverage issues), some of them depending on the user terminal itself (e. g. dropped calls due to weak battery unable to sustain the required call duration) or even user’s personal behaviour, etc. However, the most important factor is the impact of physical location of a user with regard to the serving (closest) network node and the resulting quality of an attainable radio communication link, that is not accounted in the current QoS management system.

Even in a single cell, radio link characteristics differ in different parts of the cell, more so within the whole cellular network. A mobile user experiences different link conditions while moving in the natural environment within or between the cells. The medium signal level (the long term characterization of radio link budget for a particular point) may easily vary by 10–20 dB just within very short distances of few or tens of meters.

All this supports a basic assumption of regular differences in link conditions that will be observed by different users within the same cell, especially if they are located at different spots for longer periods (housed, working). Moreover, this tendency and relationship between dominant location of users and the received iQoS does not appear sufficiently addressed by monitoring of QoS in current networks.

This calls for a need to establish some means for monitoring and recording of the actually experienced iQoS. This recording should be done for the whole duration of the call from the attempt to set up a call until the moment the call is terminated (naturally by the user or dropped call). The recorded data should be then collected and analyzed, first within a user terminal itself, then in some summarized form provided to the user and uploaded to the network management and control centre as appropriate.

This would allow, first of all, singling out those users, who suffer worse-than-average QoS and look for possible reasons of the poor service they received, and, secondly, it would give users an objective tool for gauging the grade of the received telecommunication services. The latter should not be underestimated. From user’s psychology point of view it may seem that if a user would be able to know “what QoS” he/she has received, this would already increase his/her overall satisfaction with that service because of feeling of “being in control”. This satisfaction of knowing what is happening and subsequent feeling of being “not cheated” may be further re-enforced by, e. g., making the measured iQoS one of parameters in the billing system. In that case users might be charged by operators not only by the amount (minutes/bytes) of received services, but also in accordance with the actual iQoS, i. e. charges could be made proportionate to the real user satisfaction.

Availability of objective information of iQoS would also allow introducing the practice of service level agreements between the network operator and users, that require individuating QoS to the service utilization of each single user [15]. This is an essential part of modern relationships between users and service providers, but up to now, it was not possible to realize this in mobile cellular networks due to the described uncertainty of the actual QoS provided to every individual user.

2.5. A proposed concept for rating of Individual QoS

At the initial stage, it is proposed to define the iQoS parameter and metrics for only one type of telecommunication service – voice, as it remains the most commonly used in cellular networks.

In general, the QoS parameter has many components defining the required quality for voice and data communication services, as e. g. given in numerous ITU-T recommendations. Many of those components apply equally to all users of a given network, while others are user-specific. This work further addresses only those components of QoS, that may differ between different users in a cellular network.

2.5.1. Call setup and dropping

It may be noted that the GSM MoU Association offers a voice communication scheme as an object for control of QoS [1]. To estimate impairments to iQoS, the following three events should be controlled and their appearance noted: access failure, setup failure and the dropped call.

Of these three, the two events, that happen after initiation of the call, are the most undesirable from the user perspective, namely the setup failure and the dropped call. This is because in addition to psychological discomfort, the user will have to pay for services he did not receive, if the error appeared already after the billing timer was triggered.

It may be noted that most of modern telecommunications networks recognize the event of dropped call and register it in the QoS system of the network. This information is then used for statistical analysis, in form of the averaged rate of dropped calls for overall number of calls. This simplification, as mentioned, works very well in fixed networks where all users have reasonably uniform conditions. However, in the case of uneven and non-stationary connectivity in a cellular network, such averaging loses its meaning.

It is therefore proposed that control of iQoS in cellular networks should first of all involve differentiation of those mobile users who have not experienced dropped calls (over the control period) and those who have had dropped call occurrences. The number of dropped calls may sort the latter group, so that users with higher number of dropped calls might be noticed and separately analyzed for reasons, etc.

For successfully completed voice calls, additional factors of iQoS might be analyzed basing on the speech transmission quality, as described in following sub-sections.

2.5.2. Radio link performance

All of modern 2G and 3G radio network interface standards foresee individual power control mechanisms [18], that are carried out by every user terminal and base stations for the whole duration of calls. Adaptive controls of transmitting power at the user terminal (mobile station) as well as at the base station are implemented in order to optimize the performance of the given communication link by properly balancing the radio link budget. Keeping power of the transmitter to the least sufficient levels allows conserving cell capacity (in CDMA systems) and equalizing the received signals, while also minimizing co-channel interference to other cells of the same system or to other systems. In more advanced systems, the power control process is not only based on monitoring the level of the received signal, but also linked to the communication quality, e. g. constant monitoring of BER at the receiver output.

However, while trying to keep the radio link budget and radio interface as a whole at its optimum, this process does not provide a complete remedy for ensuring high QoS for the carried communication services, because of its inherent finite latency, principal inability to cure ultimate faulty situations as e. g. the drop of signal strength below some specified limit, etc. Yet it appears that the derivatives of the power control mechanism may provide convenient means for estimating iQoS.

For example in the GSM system the mean values of BER are used in receivers for evaluating the overall quality of the radio link, a parameter denoted as RxQual. This parameter describes quality of the received signal on that particular channel, averaged over the reporting period of length equal to certain number of frames. When the user terminal moves in area with poor radio coverage, quality of radio link is not optimal and RxQual constantly changes reflecting that.

It would be very convenient to use the RxQual parameter for monitoring of iQoS, as it is already measured in all user terminals. However, it appears that as such it has no direct relationship with the actual speech transmission quality; therefore possibility to use it for estimation of iQoS would require applying certain adaptation and proposed means for doing this are explained below.

2.5.3. Estimating impairments to the voice quality

Voice quality can be measured by subjective methods such as the Mean Opinion Score (MOS) mandated in ITU-T Recommendation [62] or parametric estimation (objective) methods like PSQM [65] or PSQM+. The subjective methods are very time-consuming and expensive to use, while parametric estimation can be done quickly and inexpensively in the voice codec.

Recently, the ITU-T has created a so-called “E-Model” [64] for estimating the voice quality in packet-based transmission networks. For transmission planning purposes, the E-Model is a useful tool for estimating the combined effect of voice quality parameters (like speech level, attenuation distortion, transmission delay and echo path loss, etc.) on the overall voice quality perceived by the user.

Another advantage in favour of using the E-model as basis for evaluation of iQoS is the already existing results of the performed research, where acceptable values of rating (*R*) are derived, as shown in Table 2.5.1 [65].

Table 2.5.1 Proposed values for rating factor *R* and MOS

R value	R value	MOS	User satisfaction
100 - 90	Best	4.5 - 4.34	Very satisfied
90 - 80	High	4.34 - 4.03	Satisfied
80 - 70	Medium	4.03 - 3.6	Some dissatisfied
70 - 60	Low	3.6 - 3.1	Many dissatisfied
60 - 50	Poor	3.1 - 2.58	Nearly all dissatisfied

For evaluation of the rating *R* a suitable analogy with packet networks may be considered. Deterioration of radio link conditions in digital cellular networks causes loss of bit stream packets and fall-outs of noticeable intervals of voice conversation. The influence of the packet loss ratio on the voice quality impairment factor (I_e) in IP telephony is specified in the ITU-T Recommendation G.113 [62], more data is provided in [116]. The summarized data is presented below in Table 2.5.2.

Note that Table 2.5.2 shows data for specific packet rate per codec (20 ms for G-729 and GSM EFR codecs, 30 ms for G.723.1-A codec).

Table 2.5.2 Equipment impairment factor I_e as function of packet loss

Packet loss ratio, %	**0**	**1**	**2**	**3**	**4**	**5**
GSM EFR	5	16	21	26	30	33
G.729-A+VAD	11	15	19	23	26	28
G.723.1-A+VAD	15	19	24	27	32	

In accordance with observations in Table 2.5.2, the increase of packet loss from 0 % to 1 % produces an impairment factor ΔI_e =11 for the GSM EFR codec. This means that at packet loss ratio of 1 % the aforementioned typical mobile voice rating value would drop from $\boldsymbol{R}$=72/64 to $\boldsymbol{R}$=61/53. This corresponds to user satisfaction levels described as "Many dissatisfied" to "Nearly all dissatisfied". However, the values in Table 2.5.2 were derived for the case of IP telephony, so they should not be applied directly to the case of standard voice communication over a cellular mobile network. On the other hand, it should be also kept in mind that the data in Table 2.5.2 is based on the assumption of random distribution of lost packets. This is not always the case in cellular networks, where poor link conditions may cause prolonged periods of packet loss, that could not be repaired completely even by interleaving.

It should be also noted that evaluation of the equipment impairment factor as well as other elements of the rating factor $\boldsymbol{R}$ might not be objective. All these evaluations are based on empirically measured data and expert evaluations.

MOS can only be representative of test conditions in which they were recorded (speech material, processing, listening conditions, language and cultural background of listening subjects, etc.). Listening tests performed in different conditions than those used in the characterization phase could lead to a different set of MOS results. Regardless of that, the MOS method seems being very suitable and is used for experiments of evaluation of QoS when analyzing new coding or transmission methods and devices for resulting voice quality impairments [112].

For evaluation of voice impairment factors, the differential MOS $(\Delta\mathrm{MOS})$ methodology is proposed [A1], where the dynamically measured MOS is noted as:

$$\Delta\mathrm{MOS}(X) = \mathrm{MOS}(X) - \mathrm{MOS}(X_0)\,. \qquad (2.5.1)$$

Where: X – value of the voice impairment factor; X_0 – initial conditions.

During this study a comparative analysis was performed on the performance of different codecs (reported in [62, 65]) in the conditions of a mobile cellular network. The result of this analysis is presented in Fig. 2.5.1, that depicts the ΔMOS as a function of C/I. It may be shown that the curves are contained within the limits described by approximation curves:

$$\Delta\mathrm{MOS}_{\min}(z) = \mathrm{e}^{a/z_0} - \mathrm{e}^{a/z_0}, \quad a = 6.5;$$
$$\Delta\mathrm{MOS}_{\max}(z) = \mathrm{e}^{a/z_0} - \mathrm{e}^{a/z_0}, \quad a = 3.32. \qquad (2.5.2)$$

Where: z – is a C/I ratio; z_0 = 16 dB.

It may be also observed that voice quality deteriorates quickly when C/I becomes less than 7 dB.

Many results of analyzing the AMR codecs for the voice impairment due to the varying Frame Error Rate (FER) parameter are provided in [7]. Basing on this data, it may be concluded that the negative dependence of MOS evaluation on the FER parameter could be well represented by the following formula:

$$\Delta\mathrm{MOS(FER)} = 1 - \mathrm{e}^{\mathrm{b}\cdot\mathrm{FER}}. \qquad (2.5.3)$$

Where empirical parameter b = 0.09–0.11.

The results reported in [7] further lead to a conclusion that for the AMR codec the ΔMOS becomes approximately –2, after FER reaches 10 %.

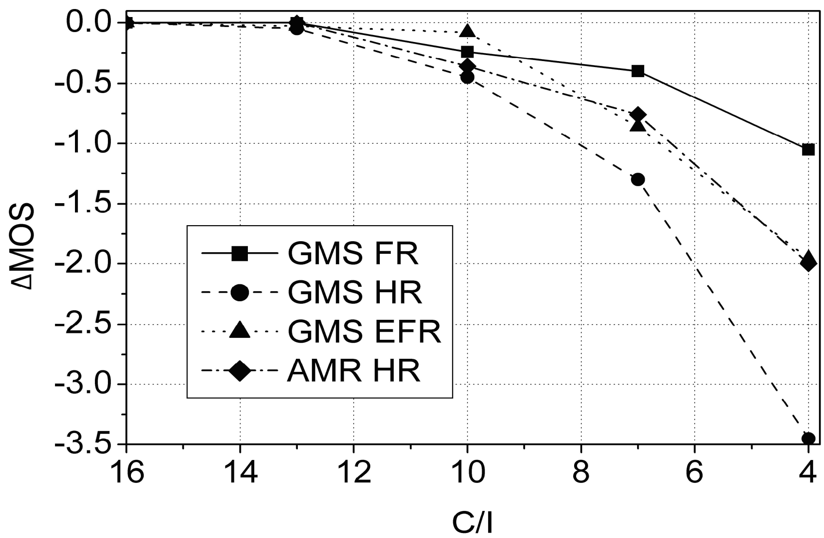

Fig. 2.5.1 Differential MOS as a function of radio link quality

Measurements indicate that in cases of poor quality (for $-110 < \text{RxLev} < -96$ dBm), the BER > 10 % was observed for 17.9 % of measurement points, and C/I < 6 dB was observed at 38 % of all points. The occurrence of both BER > 10 % and C/I < 6 dB was observed for 17 % measurement points. This confirms that in areas with poor radio coverage the obtainable voice quality will be significantly deteriorated.

2.5.4. Relating voice quality impairment factors to the measured radio channel parameters

In real life conditions it is impossible to measure directly either the Δ MOS or components of the rating R, such as the equipment impairment factor I_e. Therefore, any dynamic evaluation and recording of iQoS would require finding a connection between the voice quality and the parameters that might be or already are being measured in user terminals, like C/I, BER or RxQual parameters. It appears that it would be easier to link these parameters to either MOS (ΔMOS) or the equipment impairment factor I_e. As a result, an abstract function $Q_k(Y, Z, t)$ could be built, expressing iQoS received by the user k as a function of time t and radio link parameters (Y, Z).

Then, during a mobile call, iQoS expressed as $Q_k(Y, Z, t)$ would constantly change, reflecting the actual QoS received by that particular user. Evaluation of the received quality may be done over a reasonably short time, e. g. several seconds. As a person can tell a word in a one-second interval, impairment of quality could be noticeable.

The seconds seeing no voice quality impairments would get $\Delta Q_k(Y, Z, t)=0$. If corrupted frames would appear, the actual value of $\Delta Q_k(Y, Z, t)$ might be easily calculated like in [6]. This process would produce series of quality impairment estimates $\Delta Q_k(Y, Z, t_1)$, $\Delta Q_k(Y, Z, t_2)$, …, $\Delta Q_k(Y, Z, t_i)$.

The next step would be to provide a summary evaluation of iQoS over a given period (a month, a day, for every call). This may be done by various averaging techniques, possibly assigning different significance factors to various kinds or levels of recorded impairments, distinguishing between time-adjacent repetitive impairments and randomly spaced ones and so on. Work on this issue is still in progress.

2.6. Chapter conclusions

1. Usage of mobile services has been growing rapidly and therefore need arose to find new ways how to expand capacity of the GSM system at the same time guarantying high

QoS. For this reason power control, discontinuous transmission and frequency hoping ware implemented. System capacity could be additionally increased using AMR codecs. Using all these capacity increasing technologies the common GSM/DCS system capacity could exceed 1500 Erl/km^2.

2. Measurement results indicate that implementation of the described GSM capacity increasing technologies allows to keep average quality of GSM services at the same level. For example, average call dropping rate in the Omnitel network has remained the same during the years while the number of users and capacity requirements has been growing rapidly.

3. The updated analysis and measurement results indicate that a radio interface between the network and a particular user is a highly unstable phenomena. It affects quality of a communications channel on an instant basis. Therefore, it is impossible to guarantee priori a certain level of service quality to every user of the cellular network.

4. Traditional averaging methods for monitoring of QoS that are used in fixed telephony networks are not appropriate to introduce any level of certainty of the provided mobile telecommunication services to individual users.

5. For determination of C/I situation in the network measurements could be done involving all mobile stations in the network. Having collected C/I measurements from each network user and having correlated them with standard GSM measurements (RxLev, TA, a cell number) and with the cell position, it is possible to form a precise and dynamically changing C/I database.

6. An individual QoS – iQoS evaluation mechanism has been proposed. This evaluation mechanism would enable to provide objective information about the actually received iQoS of the user as well as of the network provider. This would increase certainty and satisfaction of users, allow implementation of service level agreements for cellular network customers and introduce new value-added services of high benefit to users (such as quality dependent billing).

3. DEVELOPING A MODEL OF THE GPRS/EDGE DATA CHANNEL FOR THE INDIVIDUAL USER

A GPRS/EDGE system was introduced using the present GSM infrastructure. The same radio resources are used for transmission of packet data and circuit switched voice. Management of common voice and data channel resources depends upon many factors, such as the number of data and voice users, usage behaviour, radio channel resources and quality as well as on internal management rules. Quality of data services received by an individual user is very unstable and depends on many factors.

In reality quality of services received by an individual user could be objectively evaluated using characteristics of a mobile data transfer channel such as data throughput, packet delay, packet blocking rate and etc. Verification of QoS received by an individual user, taking into account that the number of mobile data users in one network exceeds a million, is very complex and expensive. It is done very seldom and only in a small part of the network.

Development and use of the system adequate simulation models allow evaluation of varying needs of users and calculation of optimal system parameters before introducing any changes into a live commercial GSM/GPRS/EDGE network. It also allows predicting what QoS and with what probability an individual user will receive.

In this chapter the GPRS/EDGE data transmission technology is analysed. Measurement results of QoS characteristics of GPRS and EDGE data channels are presented. A mathematical model of common voice and data resources based on the system analysis and measurement results is made. During the modelling the common channel utilization, the number of served users, user data throughput as well as blocking probabilities of voice and data users are analyzed. To confirm modelling results and the model adequacy measurements of characteristics of a real system in a commercial GSM/GPRS/EDGE network are done.

In this chapter the created model that evaluates QoS received by an individual user is used for producing the data transmission technology using several parallel mobile data channels.

3.1. Analysis of the GPRS/EDGE mobile data network

In 1994 the General Packet Radio Service (GPRS) system was introduced using the present GSM infrastructure [36–44]. GPRS in GSM provides packet-switching logical channels (Packet Data Channel, PDCH) for data applications. At the air-interface radio resources are assigned to the mobile station only temporally and on a per-packet basis. The

PDCH's basic transmission unit is a *radio block* that in case of one PDCH requires four time slots in four consecutive Time Division Multiple Access (TDMA) frames.

The length of a time slot is 4.615 ms and every 13th burst is not used for transmission. Thus, the mean transmission time of a radio block of 456 bits sums up to 20 ms. Four different Coding Schemes (CS) are defined providing data rates from 9.05 kbit/s to 21.4 kbit/s per PDCH [41]. Since in GPRS access of all eight slots of a TDMA frame is forecasted, data rates up to 160 kbit/s can be achieved. For a single mobile station it is multi slot capability that defines how many consecutive slots within the TDMA-frame can be used.

In this way the packet switching technology was introduced in mobile networks. It allows using network resources optimally for data transmission at the same time serving high number of data subscribers. It also allows data users to be always on line.

After several years the EDGE (Enhanced Data rates for GSM Evolution) technology was introduced. The EDGE technology reuses the same frequency and time division access as in GSM/GPRS. However, a new 8PSK modulation, 9 coding schemes and advanced error control algorithms were introduced. It increased the average and maximum data throughput up to 2–3 times if comparing with GPRS. For the first time EDGE was commercially launched in the USA by the Singular Wireless operator in Indianapolis. Now EDGE operates in more than 80 mobile networks in the world. In Lithuania two mobile network operators support EDGE – JSC "Omnitel" and JSC "Bitė Lietuva".

In the GSM/GPRS/EDGE system the same radio resources are used for transmission of packet data and circuit switched voice. Management of common voice and data channel resources depends upon many factors such as the number of data and voice users, usage behaviour, radio channel resources and internal management rules.

3.1.1. Infrastructure of the GPRS/EDGE mobile data network

The GPRS/EDGE network infrastructure is build on the GSM network infrastructure (section 2.1) adding several functional components. One of the main new components is SGSN (serving GPRS support node). SGSN handles all control signalling for establishment, maintenance and release of packet data end user services. SGSN provides functionality for forwarding and handling of user information (IP packets), coming from/to the radio network and GGSN for packet switched services. The second core node is GGSN (gateway GPRS support node). GGSN supports control signalling to external IP networks for authentication and allocation of IP address. It communicates with one or several SGSNs. GGSN provides functions for forwarding and handling of user information (IP packets) to and from external networks (Internet/intranets).

During implementation of the GPRS technology in a BSS and other network elements only software upgrade is required. However, implementation of the EDGE technology requires hardware replacement of radio transceivers in order to support not only GMSK but 8PSK modulation as well.

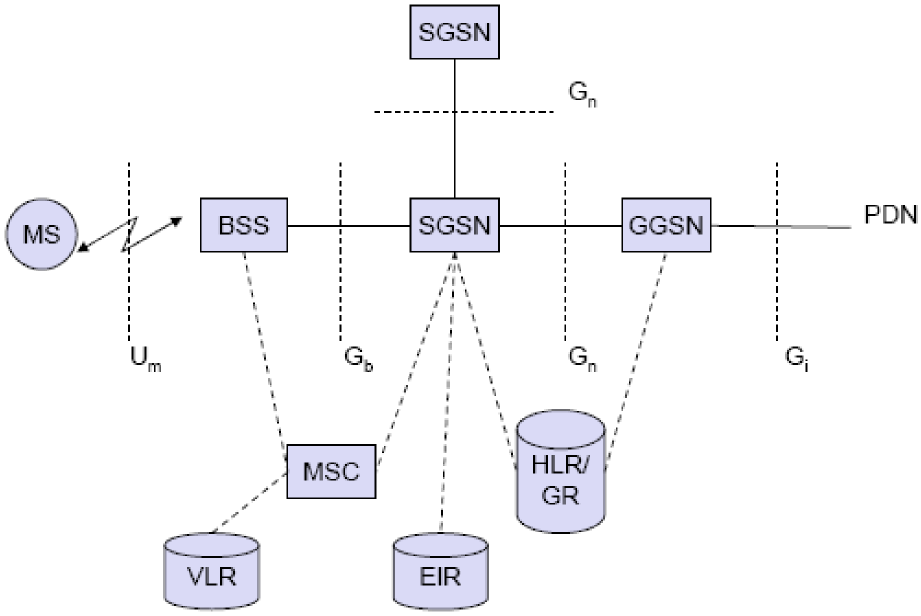

Fig. 3.1.1 Infrastructure of the GPRS/EDGE network

Interfaces among network service providers, GGSN and SGSNs (interfaces G_i and G_n) are based on standard solutions of IP networks like leased lines, local Ethernet connections and etc. Interconnection of SGSN with a BSS (G_b interface) uses the Frame relay (FR) data transmission technology [8, 12, 41, 46, 71].

The bottleneck of the mobile data transmission system is a radio interface (U_m) set between a MS (Mobile Station) and a BSS (Base Station Subsystem). This interface has the highest impact on QoS of the mobile data channel, therefore this interface will be deeply analysed [9, 16, 47, 51, 100, 105].

3.1.2. Transmission of data packets in the GPRS/EDGE network

FDMA (Frequency Division Multiple Access) and TDMA (Time Division Multiple Access) radio channels are used to create an Um connection between a MS and a BSS.

According to [21, 28, 40, 42] the layer 2 of GPRS is subdivided into Radio Link Control (RLC) and Medium Access Control (MAC). The RLC layer deals with radio block reassembly and ARQ error retransmission. The MAC layer is responsible for resolving access attempts of the radio block among many mobile users. Before a MS sends one or more than one block on uplink in the first place it should send a request to the BS (Base Station) via PRACH. Then the BS responds to the access grant of the MS indicates that the corresponding MS can send its block(s) on specific position(s) within a 52-multiframe. The BS sends the access grant using the PAGCH.

In the uplink one PRACH is mapped into one radio block within a 52-multiframe. This is determined by the Uplink Status Flag (USF=FREE) that is broadcasted continuously on each downlink radio block. These uplink radio blocks with corresponding downlink radio blocks marked as non-free USF are reserved for data blocks.

An LLC frame is then segmented into RLC data blocks that are formatted into the physical layer. Each block comprises four normal bursts in consecutive TDMA frames. Figure 3.1.2 summarizes the data flow in GPRS/EDGE.

All MS that use data services in a particular cell share the available PDCHs. The average data transmission speed for one individual user depends on the number of simultaneous active users, available radio channel resources and radio channel quality therefore iQoS perceived by the user changes very dynamically.

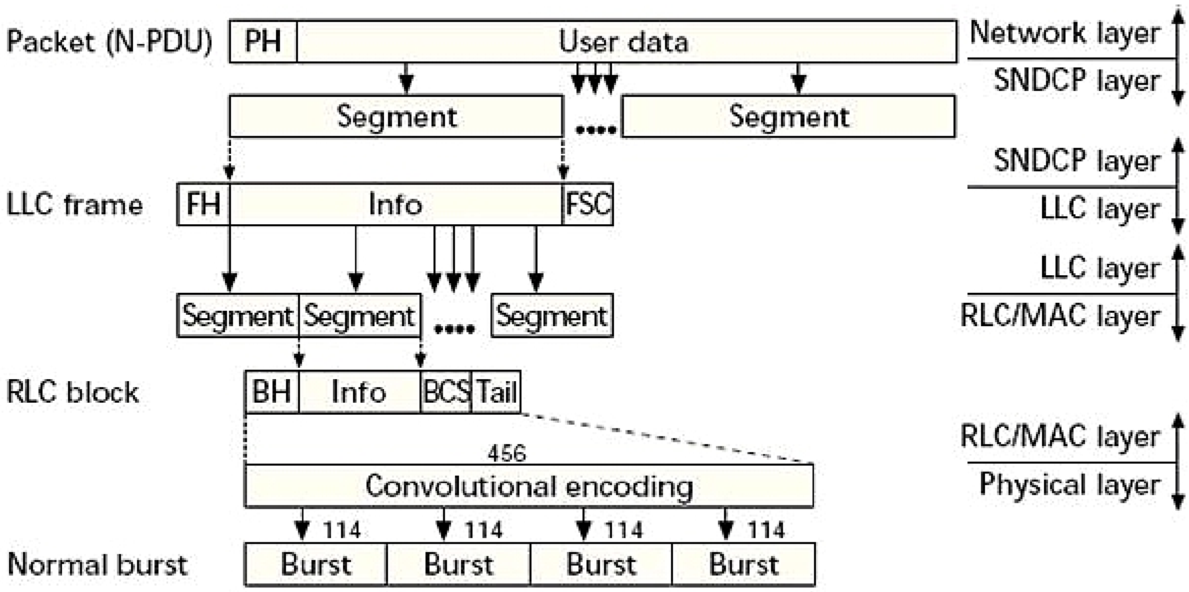

Fig. 3.1.2 Flow of user data packets in the GPRS/EDGE system [16].

PH – packet header, FH – frame header, BH – block header,

FCS – frame check sequence, BCS – block check sequence

Structure and building of all data packets in details are described in ETSI standards [42].

3.1.3. Coding of GPRS and EDGE radio data packets

For data transmission in the GPRS network up to 8 time slots (TS) on each carrier could be used. For coding of data packets one of four coding schemes (CS) could be used (Table 3.1.1) [43].

Table 3.1.1 Data rates using different coding schemes and different number of time slots

Channel coding scheme	CS1	CS2	CS3	CS4
1 time slot	9.05 kb/s	13.4 kb/s	15.6 kb/s	21.4 kb/s
4 time slots	36.8 kb/s	54.2 kb/s	63 kb/s	86.2 kb/s
8 time slots	72.04 kb/s	107.2 kb/s	124.8 kb/s	171.2 kb/s

To every RLC block a RLC/MAC header and a block check sequence are added. For CS1, CS2 and CS3 the radio block is convolutionally coded with code_rate = ½ and then punctured to the desired code rate. CS4 does not contain error correction.

The service data rate is calculated tacking into account 240 ms duration of a GPRS super frame. The super frame consists of 52 GSM frames including 48 traffic frames and 4 control frames. The 48 traffic frames carry 12 RLC blocks. Therefore, the data rate is:

$$\frac{12 \cdot B_{\mathrm{CS}}}{0.4 \cdot S} \cdot 8 = V_{\mathrm{CS}} . \tag{3.1.1}$$

There: B_{CS} – the number of user data bytes per a RLC block (depends on used CS), S – duration of the GPRS super frame (240 ms), V_{CS} – user data throughput in kilo bits per second

Considering the four coding schemes, CS1 has the lowest coding rate. It is also the most robust coding scheme. Therefore, CS1 is used for all control messages. CS2 and CS3 have lower code rate and more information bits. CS4 is the most data efficient coding scheme and is the most vulnerable to channel impairment. The Base Station Subsystem can choose any of the coding schemes to optimize throughput and delay performance [44, 50, 60, 89].

The CS selection algorithm is described in section 3.1.5. The main reason of CS selection is to protect user information from errors and distortion. In worse radio environment the protection level of user data should be increased and lower CS should be used. However lower CS could transmit less user information.

EDGE (Enhanced Data rates for GSM Evolution) is a radio interface technology that has been created for better utilisation of the available radio resource and for increasing of the maximum end user data rate. The main EDGE differences comparing it to GPRS are a new 8PSK modulation, advanced coding of data packets and error correction algorithms [15, 35].

EDGE increases speed of end user data transmission up to 2–3 times. However, fluctuations of data transmission speed during the data session are also increased. EDGE uses nine coding schemes. The first four CS correspond to GPRS CS and use the GMSK modulation. However data rates of 1–4 CS a little bit differ from GPRS CS because of different headers of packets. Other 5–9 CS use the 8PSK modulation and have higher data rates.

If a corrupted data packet is received it could be retransmitted using a lower CS. It dynamically adjusts a coding algorithm to the quality of radio environment. In GPRS it can not be done and the corrupted packet will be retransmitted using the same CS.

In GPRS/EDGE the bit rate seen by the user strongly depends on the user location within the cell. Users in favourable positions get higher throughput than the indicated mean bit rate. Others get less, i.e. the average bit rate cannot be guaranteed. Dependency of average user data throughput on distance to the serving BS using EDGE on the BCCH carrier is presented in Fig. 3.1.3 [20].

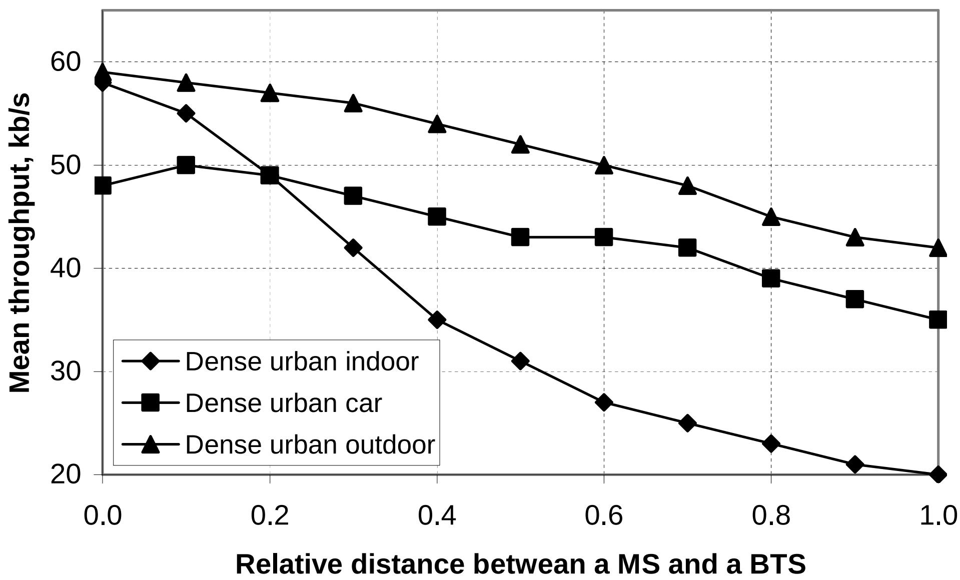

Fig. 3.1.3 Dependency of EDGE user data throughput on distance between a mobile and a base station in dense urban environment

From Fig. 3.1.3 it could be noted that in dense urban environment good EDGE rates could be guaranteed only in case when the site radius is 400–500 metres. If the site radius is higher EDGE users, located near the cell border, do not have throughput advantage in comparison with GPRS users.

It could be also noted that for an individual EDGE user one of the main QoS characteristics, data throughput, strongly depends on the user location in the cell and QoS for different users is different.

3.1.4. Multislot classes of a MS

GPRS and EDGE mobile stations use different modulation and different packet coding algorithms. Additionally, all mobile stations are divided into multislot classes and can allocate different number of radio interface TS. The multislot class of the mobile station describes:

1. the maximum number of TS for DL transmission of data packets in one TDMA frame (*Rx*);
2. the maximum number of TS for UL transmission of data packets in one TDMA frame (*Tx*);
3. the total maximum number of TS (*S*) that could be allocated in one TDMA frame by one mobile station (*Tx* + *Rx*):

$$2 \leq S \leq (Rx + Tx). \tag{3.1.2}$$

All GPRS and EDGE mobile stations are divided into 18 multislot classes [37, 38]. Nowadays the most popular multislot classes are 10 and 12. Mobile stations with higher multislot classes are very difficult to manufacture – if a multislot class is higher than 12, mobile stations should transmit and receive the radio signal at the same time. It is practically impossible to implement UL and DL duplex communication in small mobile stations.

3.1.5. Dependency of usage of a coding scheme on quality of radio environment

Different radio interface coding schemes (CS) are used to minimise data corruption during transmission of packets and maximise data transmission throughput. Low CS have good error protection but low data throughput (high CS – vice versa).

A CS of the DL transmitted RLC block depends on the parameter *S* [42]:

$$S = a \cdot B + b \cdot P + c \cdot L + d \cdot D. \tag{3.1.3}$$

Here: B – Block Error Rate (BLER) of DL transmitted RLC blocks; P – coefficient that depends upon acknowledgement of RLC packets; L – coefficient that depends on moving window stalls; D – coefficient that depends on CS changing history; a, b, c and d weigh coefficients.

Table 3.1.2 DL CS dependency on the parameter S

S	0	1	2	...	15	16	...	30	31	...	45	46	...	75	76	...	100
Existing CS	Future TS																
CS1	2	2	2	1	1	1	1	1	1	1	1	1	1	1	1	1	1
CS2	3	3	3	2	2	2	2	2	1	1	1	1	1	1	1	1	1
CS3	4	4	4	3	3	2	2	2	2	2	2	1	1	1	1	1	1
CS4	4	4	4	4	4	4	4	4	3	3	3	2	2	2	1	1	1

The coefficient P will be increased by a determined value, if RLC DL packet acknowledgement is missing. If the second packet is acknowledged, P will be decreased by another determined value.

The coefficient L will be increased by a determined value, if the RLC/MAC protocol moving window stops. L linearly increases until the RLC/MAC protocol window starts moving. When the window moves, L linearly decreases until it reaches 0.

The coefficient D equals 0 until the CS index decreases. In this case D is increased to the fixed value X and starts slowly decreasing with the speed $v = \frac{D}{T}$. If $D \neq 0$ and the CS index is decreased for the second time D is again increased to the fixed value X and decreasing speed drops twice $\frac{v}{2}$. In this way the system is protected from quick increase of the CS index, if previously CS was often decreased.

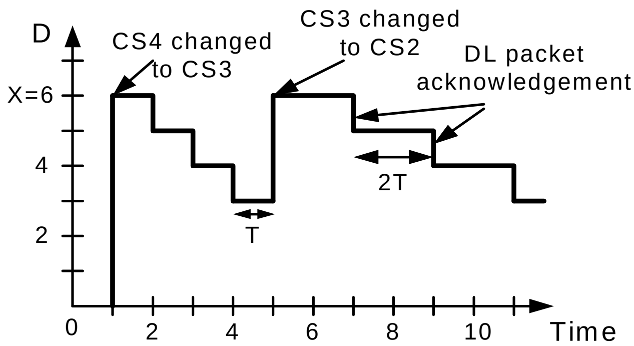

Fig. 3.1.4 Dependency of the coefficient D on history of CS changing

A CS of the UL transmitted RLC block depends on C/I. The C/I value is measured in the BTS receiver and is continuously reported to the MS. Dependency of the used CS on the C/I level is presented in Fig. 3.1.5.

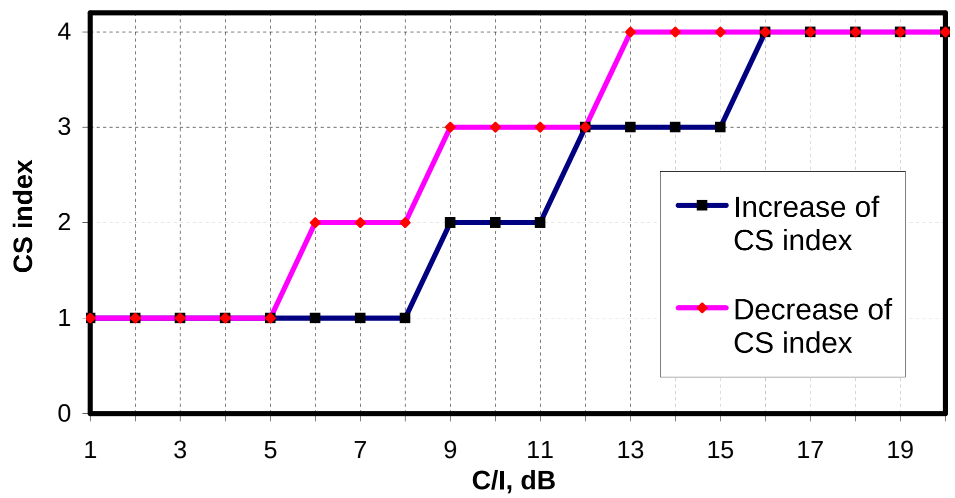

Fig. 3.1.5 Dependency of CS, used for uplink RLC block transmission, on the C/I level

In low C/I zones the first CS is used for user data transmission. If the C/I level reaches 8 dB the C/I index is increased and the second CS is used. If C/I is higher than 15 dB the fourth CS is used. If C/I decreases to 12 dB the CS index is switched from 4 to 3 for higher user data protection. To start using the fourth CS the C/I level should increase up to 16 dB. Hysteresis of 2–3 dB is used to prevent changing of UL CS too often.

Depending on the used physical GPRS channel (BCCH or frequency hoping TCH) measurement and averaging of C/I in a BTS receiver takes some time (up to 1 second). If the C/I level for a particular user will change rapidly, a MS will not be able to adopt the CS index very fast. During the time when an inappropriate CS is used packet distortion will appear. Damaged packets will be retransmitted using the RLC error control and retransmission algorithm, however individual QoS of this user will be lower.

3.2. Experimental measurement results

Experimental measurements are one of the methods to evaluate precisely QoS received by an individual user. During the investigation dependency of QoS parameters on conditions of radio environment was measured.

Measurements of GPRS data throughput, packet delay, the used CS and the allocated number of TS during the data session was measured in five places of Vilnius city. In every place using the FTP protocol 100 kb file was downlink transmitted 25 times. For evaluation of reduction of QoS when a MS reselects from one cell to another, measurements in moving environment (average car speed about 50 km/h) were performed.

For measurement results in every test place the average M and the standard deviation D were calculated.

$$M = \frac{1}{N} \cdot \sum_{i=1}^{N} x_i \ ; \qquad (3.2.1)$$

$$D = \sqrt{\frac{1}{N} \cdot \sum_{i=1}^{N} \left(x_i - M \right)^2} \ . \qquad (3.2.2)$$

Summarised measurement results are presented in Tables 3.2.1–3.2.2 and Figures 3.2.1–3.2.8.

Table 3.2.1 Measurement results of quality parameters of the GPRS radio channel

Test place	Average data throughput, kb/s	Standart deviation, kb/s	Maximum throughput, kb/s	Minimum throughput, kb/s	Average number of used TS	Average used CS	Delay (MS-GGSN), ms
Saulėtekis	66,8	2,6	70,4	48,6	4	3,6	435
Antakalnis 1	45,4	5,4	58,9	29,0	3,85	2,1	486
Antakalnis 2	63,0	6,4	76,4	30,0	3,9	3,5	442
City centre	69,4	2,8	76,0	53,0	4	3,7	429
Naujamiestis	64,6	4,3	69,8	43,8	4	3,4	436
Average	61,8	4,3	70,3	40,9	3,95	3,3	446

Table 3.2.2 Comparison of packet delay of GPRS, GSM-DATA and PSTN data networks

		RTT of 500 bytes packet, ms				RTT of 10 bytes packet, ms			
		min	average	max	Deviaton	min	average	max	Deviaton
GPRS	LLC unack.	1392	1606	3224	211	540	674	1603	66
	LLC ack.	1412	1634	3532	190	561	730	5118	125
GSM DATA	RLP transp.	1592	1645	3515	45	751	808	1081	32
	RLP nontransp.	1572	1649	4323	103	731	839	4532	120
PSTN		160	211	1983	37	120	160	2704	30

From Table 3.2.2 it could be noted that the average data packet round trip time RTT (MS – GGSN – MS) strongly depends on the packet size. RTT of a 500 B packet is about two times longer than RTT of a 10 B packet. It could be also noted that RTT in mobile data networks is from 4 to 7 times longer than RTT in fixed data networks like PSTN. It is so because GPRS was introduced using the existing GSM convolution coding and deinterleaving. For mobile data transmission control of errors and RLC packet retransmission mechanisms were additionally used.

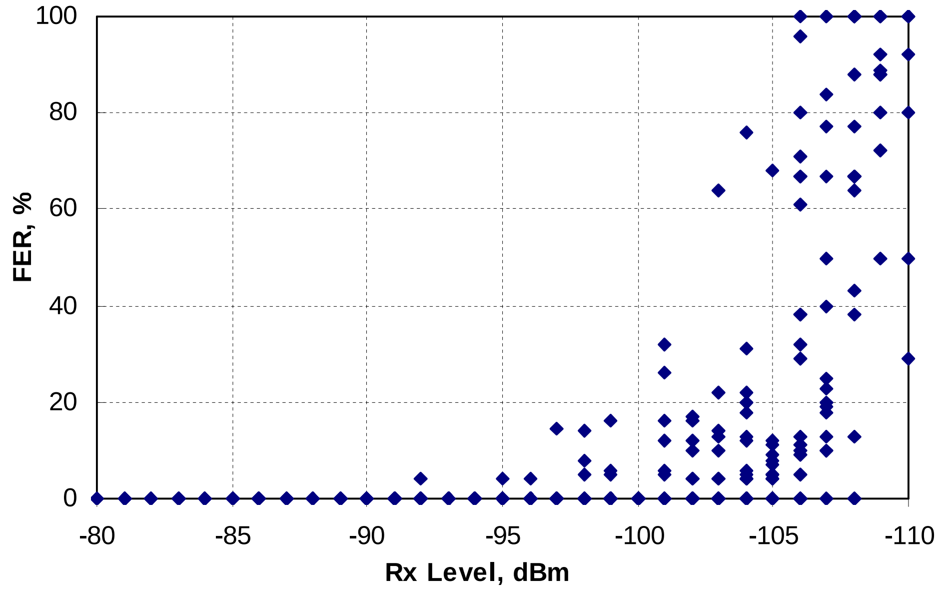

Fig. 3.2.1 Dependency of frame error rate (FER) of GPRS packets on the received signal level

From Fig. 3.2.1 it could be noted that frame error rate of packets is within an acceptable level till the signal reception level reaches –95…–100 dBm. If the signal level becomes lower than –100 dBm the frame error rate of packets increases significantly and data transmission becomes impossible.

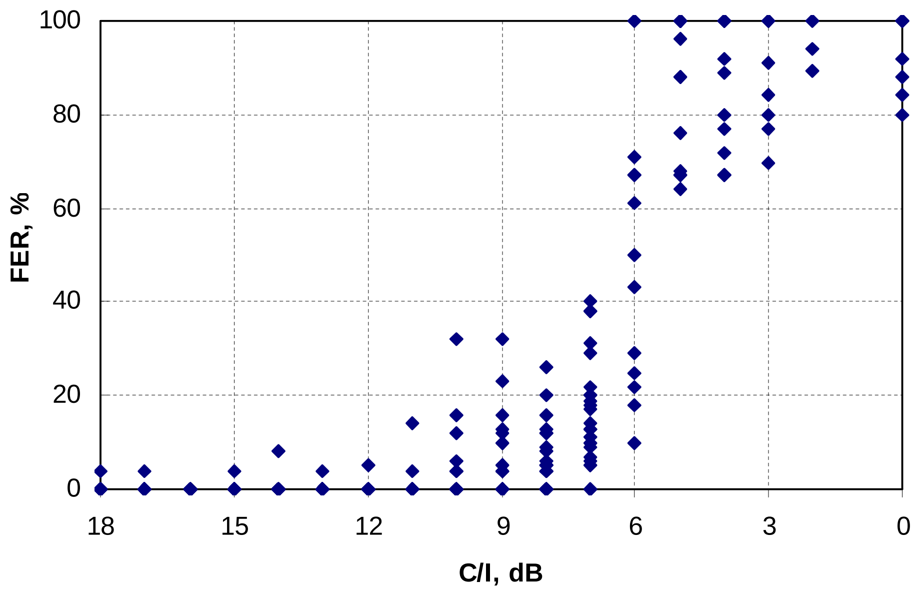

Fig. 3.2.2 Dependency of frame error rate (FER) of GPRS packets on C/I

From Fig. 3.2.2 it could be noted that frame error rate of packets increases when C/I is lower than 12 dB and increases significantly when C/I level drops lower than 6 dB.

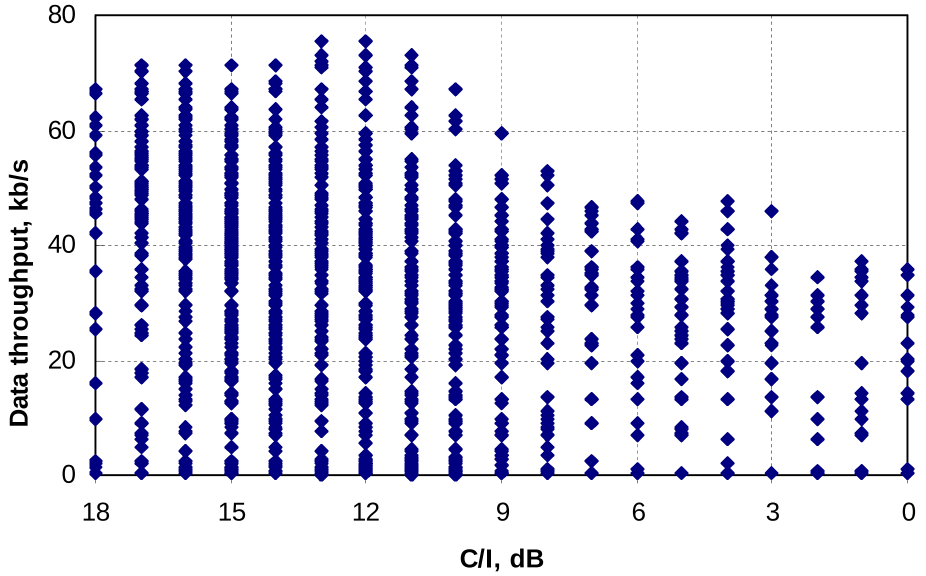

Fig. 3.2.3 Dependency of individual data throughput received by the user on C/I

From Fig. 3.2.3 it could be noted that the maximum data throughput starts decreasing when C/I is lower than 9 dB.

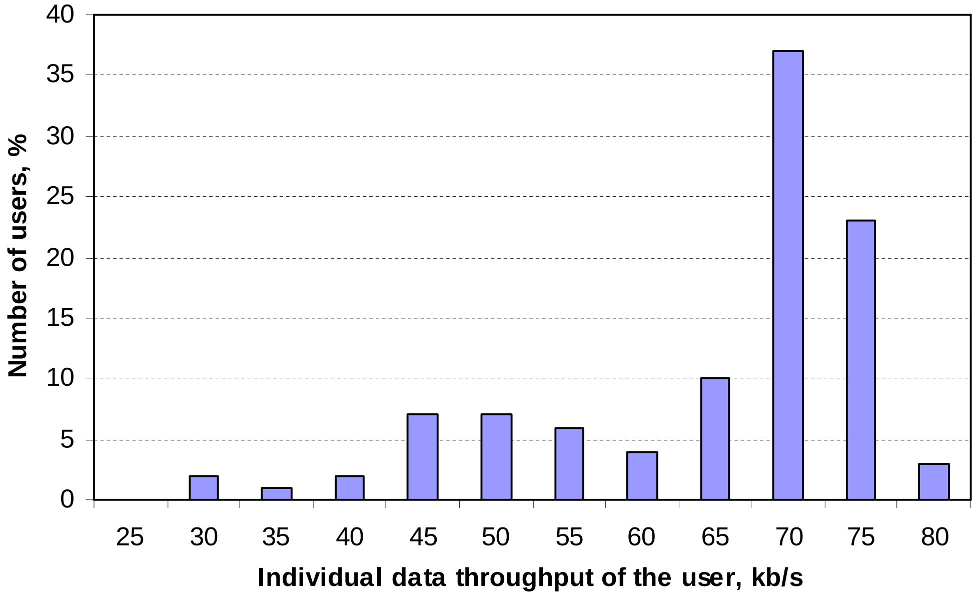

Fig. 3.2.4 Distribution of the received data throughput

Having analysed measurement results it could be stated that, using a MS of the 10th multislot class, the average GPRS MS DL data throughput was about 62 kb/s and the average GPRS packet delay between a MS and a GGSN was 446 ms. During most of the measurements four TS were assigned and the CS3–CS4 coding schemes were used. Frame error rate and data throughput were in an acceptable level while the received signal level was higher than –100 dBm and C/I was higher than 9 dB.

It could be also noted that QoS is not the same for all users in the cell coverage area. If distance to the BS increases data throughput received by the user decreases.

When the C/I level for stationary mobile data users is lower than 9 dB, users can improve the received QoS by themselves. It could be done by installing an omni directional antenna with the gain *G* oriented to the BTS. An external antenna increases the received signal level and C/I, therefore data throughput received by the user will increase and delay variation of packets will decrease. The required antenna gain depends upon user location.

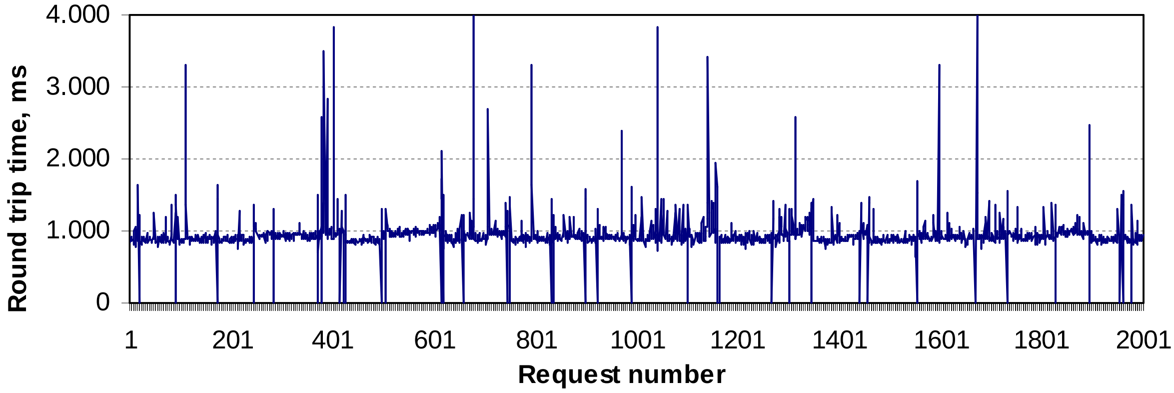

Fig. 3.2.5 Round trip time (MS-GGSN-MS) of GPRS data packets when a MS is moving in Vilnius city (average car speed is about 50 km/h)

From Fig. 3.2.5 it could be noted that when a MS moves and cell reselection occurs, higher packet delay (up to 4 s) and packet drops appear. Applications that use the mobile data network should tolerate this and use internal high level packet correction and retransmission algorithms.

For a detail analysis of CS usage in the GPRS/EDGE network additional measurements were done using measurement software of a TEMS investigator and Nokia 6230 (12 multislot class GPRS/EDGE mobile phone). Tests were done in stationary environment automatically generating HTTP as well as FTP UL and DL data sessions. The measurement sample was four hundred thousand readings.

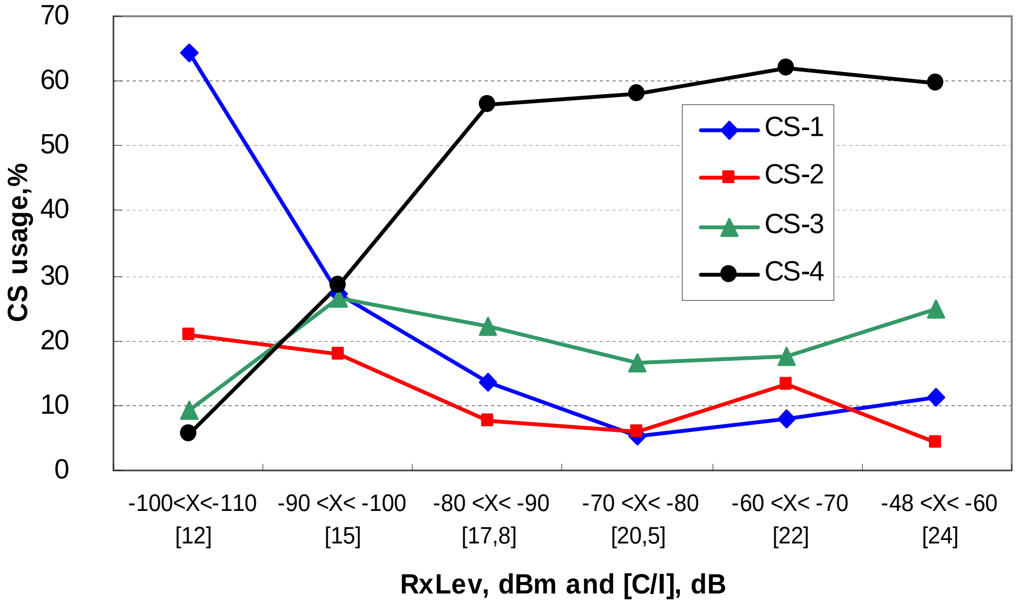

Fig. 3.2.6 Dependency of GPRS coding schemes on quality of the received radio signal (RxLev and C/I)

From measurement results it could be noted that, when RxLev is – 80...–90 dBm and C/I is 18 dB and more, a MS uses the fourth CS practically all the time. It means that there is a possibility to use more aggressive CS, however GPRS does not support them.

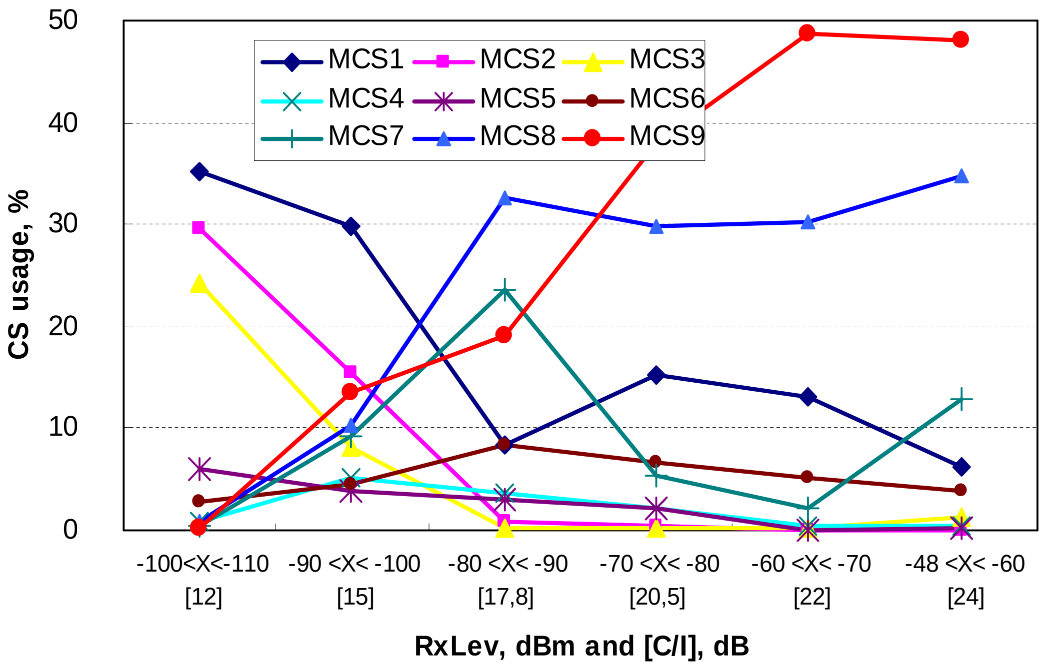

Fig. 3.2.7 Dependency of EDGE coding schemes on quality of the received radio signal (RxLev and C/I)

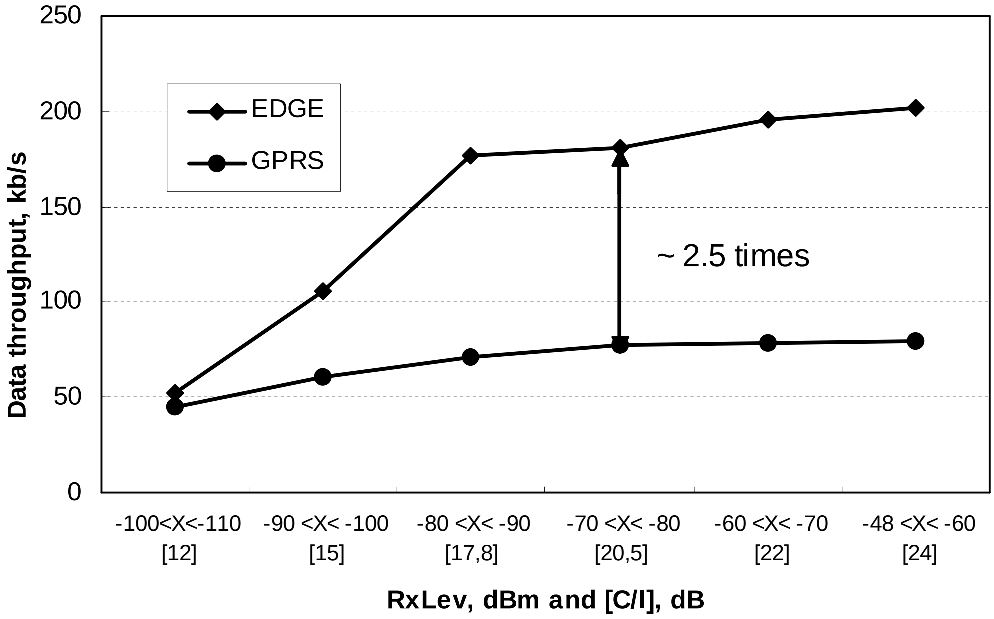

Fig. 3.2.8 Dependency of GPRS and EDGE data throughput on quality of the received radio signal (RxLev and C/I)

From measurement results it could be noted that individual data throughput of the user depends a lot on used CS and the number of assigned TS.

After implementation of the EDGE technology into the GSM/GPRS network the existing radio environment allows to use 8-PSK modulation and higher EDGE CS. It increases individual data throughput of the user 2.5 times in average (up to 150–200 kb/s). Only in places where the received signal level drops to –100…–110 dBm the 8-PSK modulation is not used and data throughput decreases rapidly.

In general data throughput V_n of a GPRS/EDGE user is a function of several variables:

$$V_n = F(CS_i, C_{\mathrm{R+SW}}, n_{\mathrm{CS}}, n_{\mathrm{D}}). \qquad (3.2.3)$$

There:

- CS_i coding scheme used for data transmission. The CS number i depends on quality of radio environment RxLev, C/I and etc.;
- $C_{\mathrm{R+SW}}$ the number of TS assigned to a particular user;
- n_{CS} the number of voice users (they could use SW channels and decrease the number of TS used for data transmission in the cell;

- n_D the number of data users who will share common available data recourses in the cell.

The scope of carried out measurements is not big, however measuring takes quite a long period of time. Evaluation of QoS in the whole network territory requires a lot of measurements that should be periodically performed in the whole network territory.

Comprehensive evaluation of QoS characteristics of the GPRS/EDGE network could be done using system adequate simulation models.

3.3. Resources of a GSM/GPRS network radio connection of an individual user

In this section the available GSM/GPRS/EDGE network common radio resources are presented and dependency of resources of an individual data user upon the number of voice and data users is evaluated.

Resources of the GSM/GPRS/EDGE system are presented in Fig. 3.3.1.

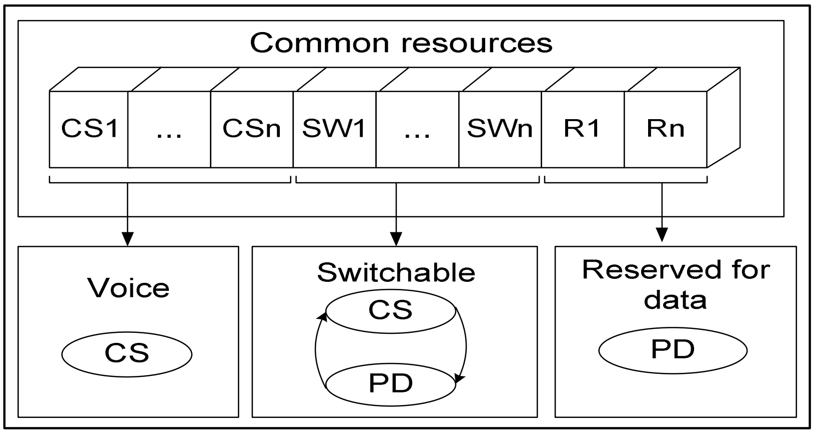

Fig. 3.3.1 Resources of a GSM/GPRS/EDGE channel.

CS (Circuit Switched) TS are dedicated for voice transmission. PD (packet data) channels are reserved only for packet data transmission (GPRS or EDGE). SW (Switchable) TS are common channels that could be switched from PD channels to voice channels depending on the number of data and voice users [14, 50, 69, 74, 76, 80, 95, 101].

In mobile GSM/GPRS/EDGE networks load peaks of voice calls and data sessions accur not at the same time. Use of SW TS allows flexible control of radio channel resources assigning them depending on voice and data load. In normal conditions SW TS are assigned for PD transmission and the system can handle high data traffic. However voice users in SW TS have higher priority than data users. Therefore, when all voice resources are used, voice users can preempt SW TS. Then SW TS will be transformed to CS TS for voice communication. It expands the number of voice channels. When the voice load reduces SW TS are transformed back to PD channels.

Use of SW TS introduces dependency of data transmission resources (the number of PD channels) upon voice load.

For example, the cell has C_{CS} - the number of voice TS, C_{R} - the number of TS reserved for data transmission and C_{SW} - the number of SW TS. All these TS could be used by all users in this cell serving area. It can be considered that the number of simultaneous voice users in the cell is n_{CS} and the number of simultaneous data users is n_{D}. Then the number of TS assigned for one individual data user, if $n_{\mathrm{CS}} < C_{\mathrm{CS}}$, will be:

$$k = \frac{C_{\mathrm{R}} + C_{\mathrm{SW}}}{n_{\mathrm{D}}}. \tag{3.3.1}$$

However, if the number of simultaneous voice users increases up to $n_{CS} > C_{CS}$, the average number of data transmission TS for an individual data user will be:

$$k = \frac{C_{\mathrm{R}}}{n_{\mathrm{D}}}. \tag{3.3.2}$$

Rules of channel switching, used by majority of network equipment manufacturers, are presented in Fig. 3.3.2 [A9].

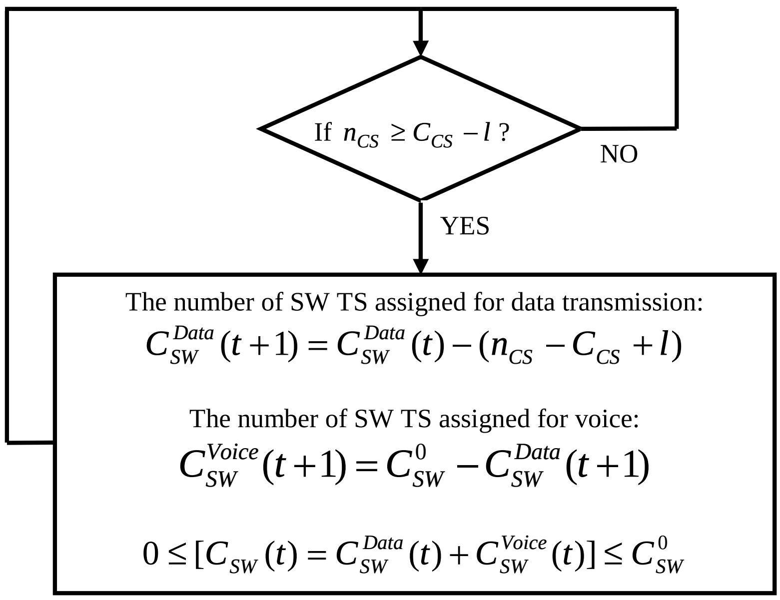

Fig. 3.3.2 The logic of SW channel assignment for data or voice transmission

In Fig. 3.3.2 C_{SW}^{0} is the configured number of SW TS in the cell; l is a SW TS transformation index. The transformation index indicates how many SW TS should be pre-empted and transformed to voice TS before the arrival of a new voice call when all TS, dedicated to voice, are already used.

Dependency of the number of data channels (GPRS/EDGE data network recourses) upon the number of voice users is an exceptionally unique feature of the data network. Dependency of data resources on number of voice users should be taken in to account during development of the model.

3.4. A model of distribution of common resources among voice and data users

In this section the model of common GSM/GPRS resources is presented. The model connects two systems: a CS (Circuit Switched) voice – GSM system and a packet date transfer – GPRS system. Peculiarities of interaction of GSM and GPRS users are investigated. During the modelling common channel utilization, the number of served users, individual user data throughput as well as blocking probabilities of voice and data users are analyzed. Suggestions for optimization of common data and voice channel resources are formulated.

Processes, that take place while dividing common data and voice resources, were thoroughly analyzed on purpose to create a mathematical model of a common GSM/GPRS channel. Background for the modelling is taken from [117] and [A4, A9, A11]. However, the model presented in [67, 117] was expanded and developed by adding SW TS transformation hysteresis, possibility of one TS to allocate up to 4 MS and possibility to use up to 8 TS for data transmission per carrier.

System capacity is the total number of allocated TS: $C_{\Sigma}=C_R+C_{CS}+C_{SW}$. Arrivals of CS voice call requests and GPRS Temporary block flows (TBF) form a Poisson process with a mean rate of λ_b and λ_d. Service time of voice calls and TBF sessions is assumed to be exponentially distributed with a mean of m_b and m_d.

Any state of the system may be described using probability of three variables:

$$P_{i,j,k} = F(i,j,k). \tag{3.4.1}$$

Where:

i – the number of TS, engaged with voice;

j – the number of TS, engaged with data;

k – the number of TBF data sessions.

One TS may include up to 4 TBF. A new voice call will be blocked, if (at its arrival) there are no available voice channels. In this case, the amount of voice channels in the system will be: $C_B = C_{CS} + C_{SW}$. A data session (TBF) will be blocked, when, creating a new TBF, all data time slots are engaged already: $C_D = C_R + C_{SW}$ and every data TS will include maximum 4 TBF sessions. Range of $P_{i,j,k}$ probability states is:

$$\begin{aligned}
&S = F(i,j,k),\\
&0 \le i + j \le C_{\Sigma}\ ,\ 0 \le i \le\ C_{CS} + C_{SW},\\
&0 \le j \le C_R + C_{SW},\ 0 \le k \le (j \cdot 4),\\
&C_{\Sigma}\ = C_R + C_{CS} + C_{SW}.
\end{aligned} \tag{3.4.2}$$

The diagram of the range of possible $P_{i,j,k}$ probability states is shown in Fig. 3.4.1.

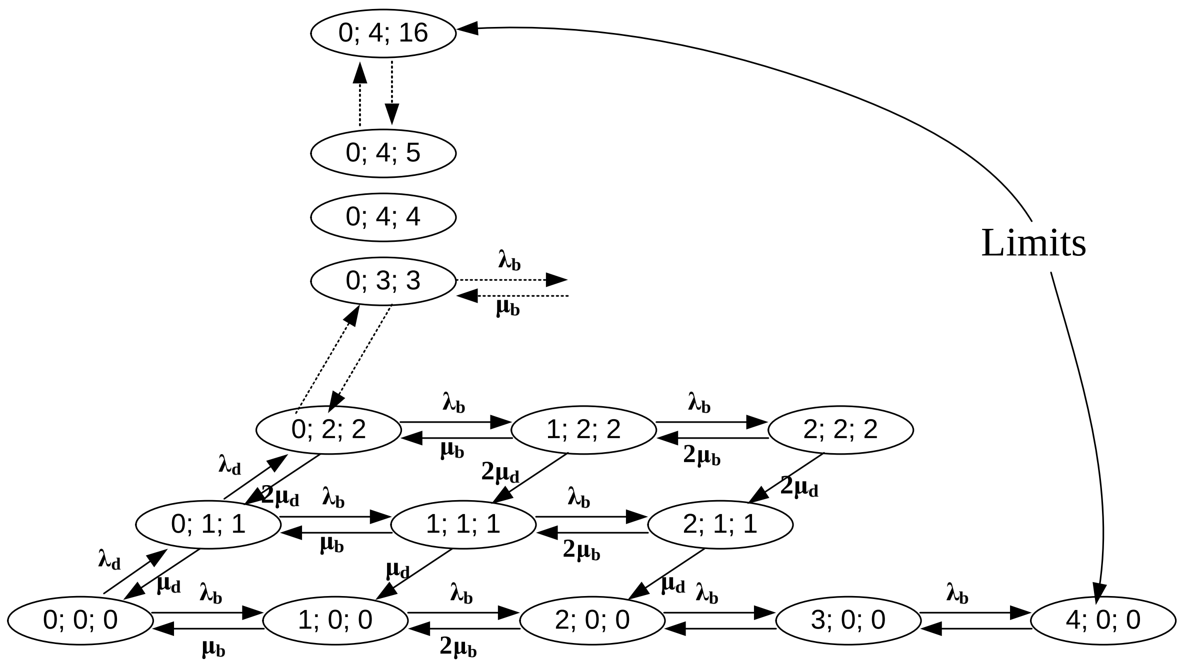

Fig. 3.4.1 The range of possible $P_{i,j,k}$ probability states

One mobile phone can maximally use up to 4 data channels (TS) at the same time.

The possibility of call blocking is equal to multiplication of probability, that all $i = C_{CS} + C_{SW}$ voice channels are engaged, by probability of arrival of a new voice call:

$$P_{CS} = P_{i=C_{CS}+C_{SW}} \cdot P_{CS\_new}\ . \tag{3.4.3}$$

Probability, that all voice channels are engaged, equals the sum of all probabilities of conditions of the system, when $i = C_{CS} + C_{SW}$:

$$P_{i=C_{CS}+C_{SW}} = \sum_{i=C_{CS}+C_{SW}} P(i,j,k), \tag{3.4.4}$$

$$P_{CS\_new} = l_{b} \cdot j\,(i+1, j, k). \tag{3.4.5}$$

Where: $j\,(i+1, j, k)$ is a function that equals one, when *i* variable increases by one (*j* and *k* do not change). It equals zero in all other cases.

The probability of blocking of a data session is equal to multiplication of probability, that every data TS includes 4 TBF, by probability of arrival of a new data or voice session:

$$P_D = P_{k=C_{4D}} \cdot P_{TBF\_new}. \tag{3.4.6}$$

Where:

$$P_{TBF\_new} = \lambda_{d} \cdot j\,(i,j,k+1), \tag{3.4.7}$$

$$P_{k=4C_D} = \sum_{i=4C_D} P(i,j,k), \tag{3.4.8}$$

Where:

$$C_D = C_R + C_{SW} - n(i, C_{CS}, C_{SW}). \tag{3.4.9}$$

The amount of SW TS reserved for data transmission depends on the amount of calls *i*, that take place at a particular moment.

$$n(i, C_{CS}, C_{SW}) = \begin{bmatrix} 0 & ,\text{if } i < C_{CS} \\ 1 & ,\text{if } i = C_{CS} \\ 2 & ,\text{if } i = C_{CS}+1 \\ C_{SW} & ,\text{if } i = C_{CS}+C_{SW} \end{bmatrix}. \tag{3.4.10}$$

If the maximum possible amount of voice calls $i = C_{\text{CS}} + C_{\text{SW}}$ is reached, only reserved channels C_{R} will be left for data transmission.

The probability of blocking of TBF sessions depends not only on the amount of the number of data users but also upon the number of voice sessions. Channel usage, that is estimated dividing useful work time of the transmitter by all work time, not estimating transformation time of SW channels, is:

$$U = \frac{\sum_{(i,j,k)\in S} (i+j)\cdot P_{i,j,k}}{C_{\Sigma}} . \quad (3.4.11)$$

Time of transformation of a SW channel (t_{TR}) includes time, needed to preempt the channel t_{h} , and time, needed to assign the channel for a new user t_{p} :

$$t_{\text{TR}} = t_{\text{h}} + t_{\text{p}} . \quad (3.4.12)$$

Time of transformation of a SW channel is an internal system parameter and can not be modified.

3.4.1. A model of delay of mobile data packets

In order to compute packet transmission delay of the *i*-th packet of size D_i, sent from *A* to *B* at time *t*, we segment the total delay into several components starting from channel assignment till final acknowledgement [96–98, 103, 109, 114]:

$$T_{\Sigma}(D,i) = T_{\text{ass}}(i) + T_{\text{tx}}(D) + T_{\text{error}}(i) + T_{\text{ACK}}(i) . \quad (3.4.13)$$

Where: D is the data packet size; T_{ass} is the time of the channel assignment procedure; T_{tx} is the data packet transmission time based on the packet size and channel bandwidth; T_{error} is the handling time of error control including retransmission; T_{ACK} is the processing and transmission time of the final acknowledgement message (for UDP protocol $T_{\text{ACK}} = 0$).

The time of channel assignment is measured between the moment when the first byte of the packet is ready for transmission $t_0(i)$ and the moment when the first byte of the packet is transmitted $t_S(i)$:

$$T_{ass}(i) = t_S(i) - t_0(i)\,. \tag{3.4.14}$$

If the bandwidth of data transmission channel b_{AB} between points A and B during some period of time is constant, then packet delay could be calculated like this:

$$T_{tx}(D) = \frac{D_i}{b_{AB}}. \tag{3.4.15}$$

Average data throughput of the channel could be calculated this way:

$$V(i,t) = \frac{\sum\limits_{i=1...N} \frac{D_i}{T_{\Sigma i}}}{N}. \tag{3.4.16}$$

Where N is the selected number of packets for average channel throughput calculations.

3.4.2. Creating a model in Matlab - Simulink

The analysis is done of the case, when common recourses include one TS reserved for data (*R*), three TS reserved for voice (CS) and four common SW TS. In SW TS voice transmission takes priority over data transmission. Therefore, when a voice request arrives, a SW channel will be transformed into a CS under the rules, described in [A9]. The general structure of the model is presented in Fig 3.4.2. The algorithm of control of data resources is shown in Fig. 3.4.3.

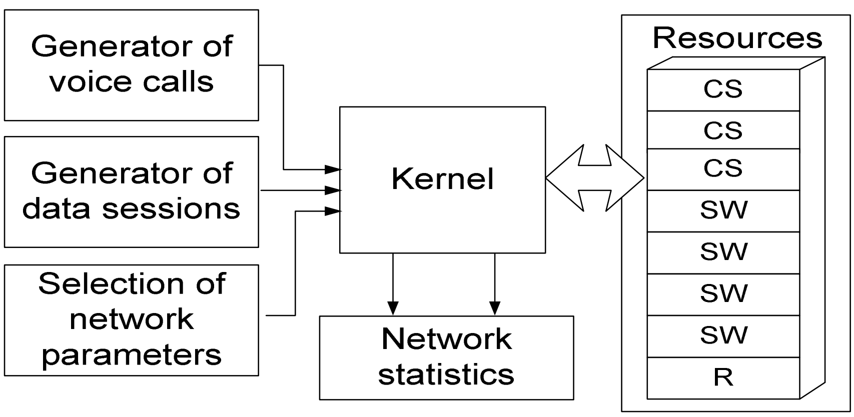

Fig. 3.4.2 General structure of the model

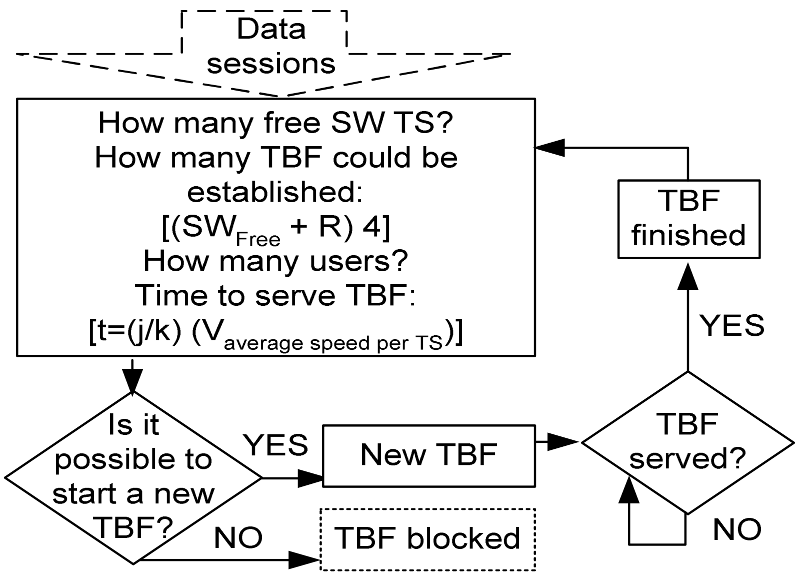

Fig. 3.4.3 The scheme of control of data transmission resources

Controlled parameters are: the transformation index, time of arriving and serving of voice requests and data packets. Time of transformation of a SW channel (t_{TR}) is also estimated while modelling.

The amount of voice requests, their serving time and the transformation parameter directly determine the amount of free SW channels. The less SW channels are assigned to data transmission, the less data users are served (data transition speed is reduced accordingly).

The algorithm of data transmission is much more complicated than that of voice transmission. When a new data request arrives the speed of data transmission and the time of data transmission are estimated. During the modelling they are continuously tuned depending on the data load, the number of data users and the amount of free SW channels.

3.4.3. Estimation of model input data

A Motorola mobile phone with an expanded menu, that allows observing the amount of transmitted data and used time slots, was chosen for the research. The mobile phone was connected to a personal computer. The “DuMeter” program was used to measure the average and maximum speed of data transmission, the amount of incoming and outgoing data and time required for transmission of data packets. Measurements were done in natural environment of the GPRS/GSM network.

Table 3.4.1 Measurement of data speed and transmitted data

	WAP	WWW	FTP/POP
The average speed of data transmission in 1 TS, kb/s	14.7	15.6	19.2
The average amount of transmitted data per one TBF, kb	24	1156	2453
Usage %	54	27	19

After evaluating the usage of every kind of service using the weight coefficient, the average amount of transmitted data is estimated:

$$N_{\Sigma} = N_1 \cdot k_1 + N_2 \cdot k_2 + N_3 \cdot k_3 . \qquad (3.4.17)$$

The estimated average amount of transmitted data $N_{\Sigma} = 791\,\mathrm{k}b$ and the average speed of data transmitting 15.8 kb/s per one TS are used to set the initial input parameters of the model.

3.5. Modelling results

The initial input parameters of the model are measured in a real GSM/GPRS network using TEMS drive testing tools and network statistics.

During the modelling the maximum voice call blocking rate could be up to 1.5 % and the maximum allowed blocking rate of data packets could be up to 10–20 %. Data interchange is controlled by a PCU (Packet Control Unit) [38] and dropped data packets are automatically retransmitted. Therefore, the system can accept packet blocking rate up to ten times higher than voice blocking rate. The initial parameters of the model are presented in Table 3.5.1.

Table 3.5.1 The initial parameters of the model

Parameter	**Value**
The average time for serving a voice request, s	40
The average time for transforming a SW TS, s	2.7
The average speed of data transmission per 1 TS, kb/s	15.8
The average amount of transferred data per a TBF, kb	791
Time of modelling, s	3600
Value of the transformation index ***l***	1

Dependency of the blocking rate and data transmission speed on the load of voice calls is presented in Fig. 3.5.1.

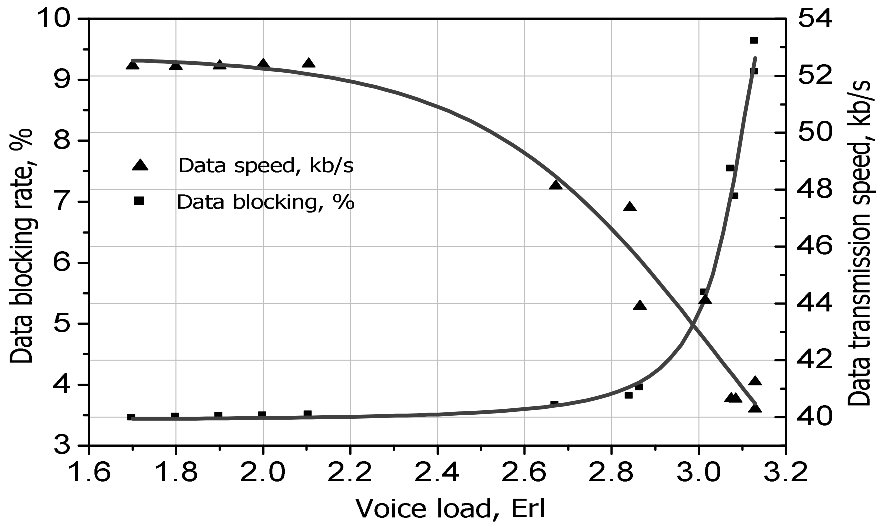

Fig. 3.5.1 Dependency of the blocking rate and data transmission speed on the load of voice calls

From Fig. 3.5.1 it could be noted that the average data transmission speed and probability of blocking of data packets in the GSM/GPRS system depends not only on the data load, but upon the voice load as well. The reason for this is that the growing load of voice requests reduces the amount of SW channels, assigned to data transmission. When the voice load grows, it could be noted that only reduction of data transmission speed appears. However, when the voice load becomes very high (more than 3 Erl), blocking probability of data packets increases very fast. Dependency of utilization of the GSM/GPRS system on the common load of equivalent voice and data requests is presented in Fig. 3.5.2.

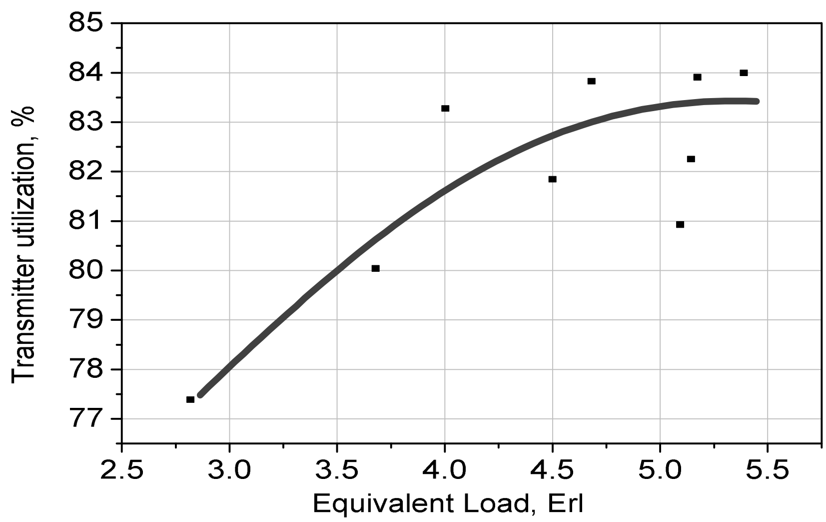

Fig. 3.5.2 Dependency of GSM/GPRS transmitter utilization on the common equivalent load of voice and data

It could be noted that the maximum utilization of transmitters reaches 83 %. During the modelling voice blocking probability is not allowed to reach 1.5 % and blocking probability of data requests is not allowed to reach 10 %.

One of the purposes of the modelling is to find an optimal index of transformation under certain determined conditions. When an optimal index is chosen, possibility of blocking a voice request can not exceed 1.5 % and possibility of blocking data requests must not exceed 15 %. The allowable voice blocking probability is up to 10 times lower than blocking probability of data. Voice blocking probability, that is higher than 1.5 %, can cause discomfort and dissatisfaction of mobile users. It could form a negative opinion about the mobile network operator.

Blocking probability of data packets could be higher. From measurement results it was noted that 10 % packet blocking probability does not considerably reduce the perceived service quality of the GPRS end user. In that case only short speed reduction occurs.

From Fig. 3.5.3 it could be noted that, when the transformation index equals 0, voice blocking is 1.6 %, whereas when the index is from 1 to 5, blocking does not reach 0.7 %. Blocking probability of data packets is within acceptable limits only when the transformation index is 0 and 1. If the transformation index increases, blocking probability of data packets increases rapidly as well and exceeds the acceptable level. When the transformation index is increased, the number of data channels and the average speed of data transition decreases.

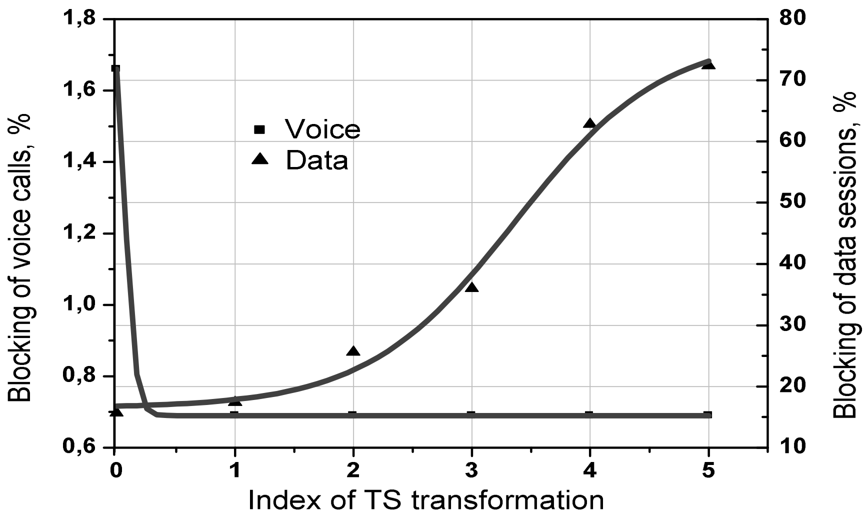

Fig. 3.5.3 Dependency of blocking probability of data sessions and voice calls on the TS transformation index

Utilization of a common GSM/GPRS transmitter with the transformation index equal to 0 and 1 is approximately 87 %. If the transformation index further increases, utilization significantly decreases, because more and more resources become unused.

When data and voice load peeks occur at the same time, usually the capacity of a base station is expanded and high blocking of data packet requests is avoided. During the research this critical situation is modelled in order to investigate system processes when the maximum load is reached.

After evaluation of measured system parameters and their influence on the perceived service quality of the end user, it is recommended to set the transformation index equal to 1.

The investigation of the most critical situation was done (when both data and voice request loads are the highest simultaneously). The voice load is approximately 2.5 Erl and the equivalent data load is approximately 2–4 Erl, depending on the amount of SW channels.

Dependency of blocking probability of data requests and the average speed dedicated for one user on the number of SW channels is presented in Fig. 3.5.4.

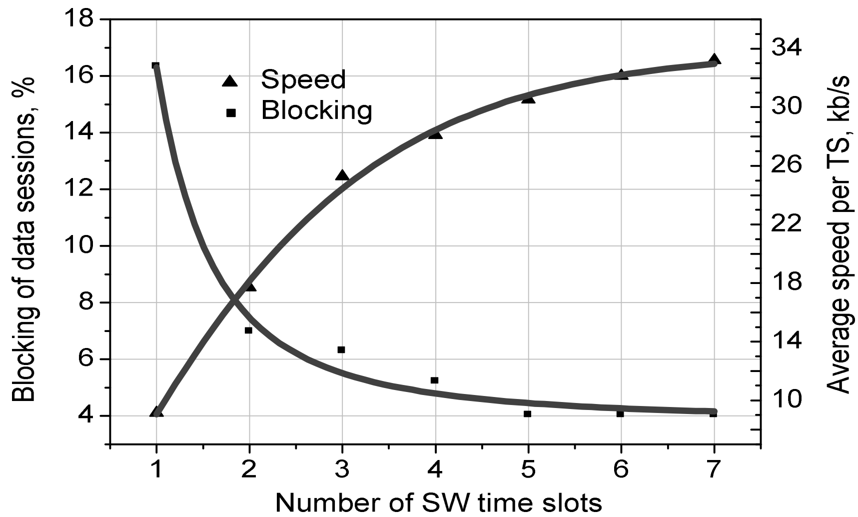

Fig. 3.5.4 Dependency of blocking probability of data requests and the average speed dedicated for one user on the number of SW channels

From Fig. 3.5.4 it could be noted that user data speed considerably increases when the amount of SW channels is increased. Blocking probability of data packets decreases quickly when the second and the third SW TS are added. So minimally 3–4 SW channels are needed for normal operation. However, optimal system operation will be reached only using 6–7 SW channels.

The average data transfer speed and data blocking probability in the GSM/GPRS system are determined not only by data, but also by the voice load. When the voice load increases significantly, data blocking increases as well. In that case data transfer speed decreases.

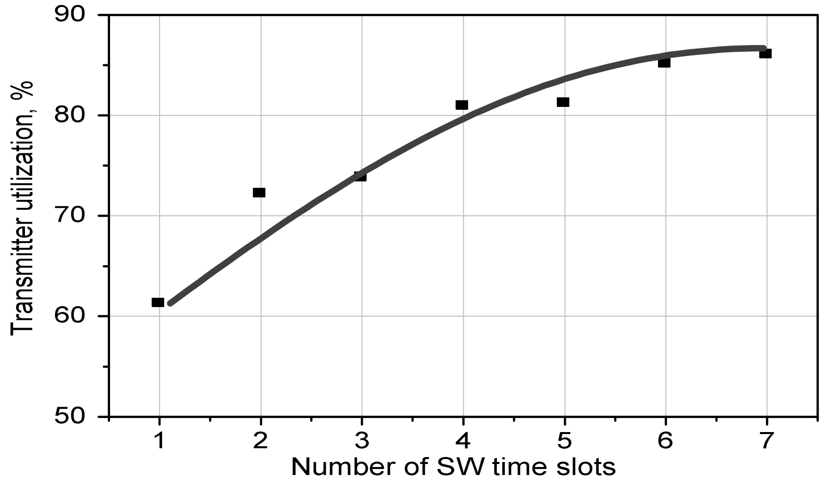

Fig. 3.5.5 Dependency of transmitter utilization on the number of SW channels (modelling results)

From Fig. 3.5.5 it could be noted that transmitter utilization depends a lot on the number of SW channels and varies from 60 to 87 %. This utilization increase is conditioned by fast reuse of SW channels for data transition after the end of a voice call.

3.6. Comparison of modelling results with the results of the research of a real GSM/GPRS system

An experimental research was done in a test base station with this configuration: R=1 TS; SW= 1 TS … 7 TS; CS= 7 TS … 1 TS. In total: R + SW + CS = 8 TS.

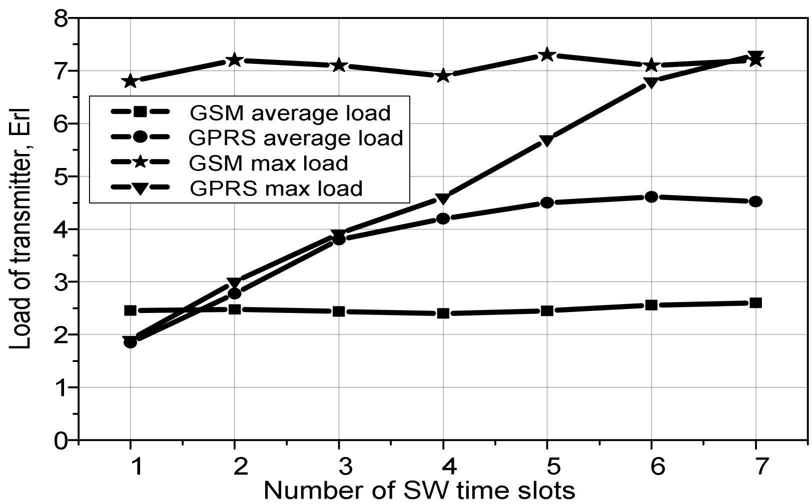

Fig. 3.6.1 The measured average and maximum voice and data load dependency on the number of SW channels

The most critical case, when both data and voice request loads are the highest at the same time, was investigated. During the experiment SW TS were changed from 1 to 7. The average and maximum voice and data loads were measured as well.

From results of experiments it could be noted that the average load was around 2.48 Erl. The equivalent data load, when the number of SW TS was increasing, was growing from 1.85 to 4.61 Erl. Furthermore, the maximum possible peek data load was increasing as well.

While the data load was increasing voice blocking probability was constant; data blocking was decreasing and the average data transmitting speed was growing.

The measured transmitter utilization dependency on the number of SW channels is shown in Fig. 3.6.2.

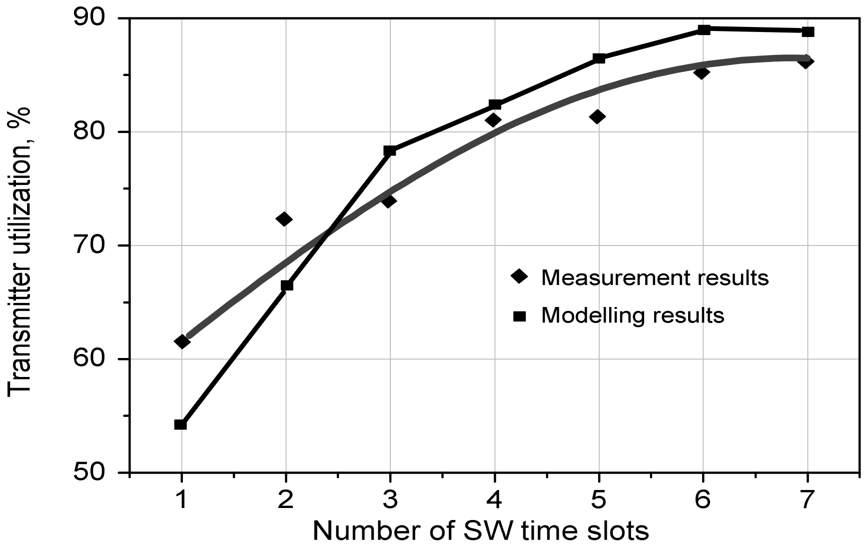

Fig. 3.6.2 Dependency of transmitter utilization on the number of SW channels. Results of real system measurement and modelling

Measurement results show, that a bigger amount of SW channels allows to increase data throughput for the end user and to increase utilization of common GSM/GPRS resources. This is possible, because channels, not used for voice transfer, are dynamically switched to data transfer. When a new voice call arrives for data assigned channels are dynamically pre-empted and without blocking the voice call are switched to voice transfer. In this case (for data transfer) not only for data dedicated resources, but also all other transmitter resources (not used for voice transfer) are used.

3.7. Chapter conclusions

1. A GPRS/EDGE mobile data transmission channel for an individual (not statistical) user is very different from Frame-relay, Ethernet or other data transmission channels. Throughput and other quality characteristics depend not only on specific characteristics of the radio channel (RxLev, C/I, etc.) but also upon network coverage, user location, characteristics of the user terminal as well as the number of data and voice users. An access channel of an individual user in the mobile data network should be considered as a channel with stochastically varying bandwidth.

2. Radio channel coding schemes (CS) and the CS selection algorithm have high impact on quality of the mobile data channel as well. When a MS moves away from a BS the C/I index decreases (the used CS index will also decrease). For that reason data throughput will decrease and users, located on the border of the cell coverage area, will receive services of lower quality.

3. Measurement results indicate that the average DL data throughput achieved by an individual user in the GPRS network was 62 kb/s, in the EDGE network – 158 kb/s and the average data packet delay was 446 ms. Frame error rate and data throughput are in an acceptable level while the signal level is higher than –100 dBm and C/I is higher than 9 dB.

4. Drive testing results indicate, that when a MS moves and cell reselection occurs, higher packet delay (up to 4 s) and packet drops appear. Applications, that use the mobile data network connection, should tolerate this and use internal high level packet correction and retransmission algorithms.

5. The existing radio environment allows use of the 8-PSK modulation and higher EDGE CS. It increases individual data throughput 2.5 times in average (up to 150–200 kb/s). Only in places where the received signal level drops below –100 dBm the 8-PSK modulation is not used and data throughput decreases rapidly.

6. Taking into account stationary mobile data users, QoS they receive could be improved by installing omni directional antennas with the gain *G* oriented to the BS. The required antenna gain depends upon user location.

7. Use of system adequate models decreases expenditures for experimental measurements and reduces investigation time. The modelling allows quick evaluation of varying needs of mobile data users and calculation of optimal network parameters before introducing any changes into the life network.

8. The model of common GSM/GPRS resources is presented. During the modelling common channel utilization, the number of served users, individual user data throughput as well as blocking probabilities of voice and data users are analyzed.

9. Measurement results indicate that load peeks of voice calls and data sessions do not occur at the same time. Therefore bigger amount of switchable channels in cells allows to increase data throughput for the end user and to increase utilization of common GSM/GPRS resources. This is possible, because channels, not used for voice transfer, are dynamically switched to data transfer.

10. If data and voice load peeks occur at the same time, usually the capacity of a base station is expanded and high blocking of data packet requests is avoided. During the research this critical situation is modelled in order to investigate system processes when the maximum load is reached.

11. After evaluation of measurement and modelling results it is recommended to set the transformation index (transformation hysteresis) equal to 1. This prevents switching of SW TS from voice to data and vice versa too often and reduces waste of cell recourses due to constant switching.

4. INVESTIGATION OF PARALLEL MOBILE DATA CHANNELS

Since quantity of e-services is expanding and the number of information sources and their users is growing importance of data transmission as well as Internet access is also increasing. More and more people use Internet access outside their offices while travelling or in special conference places. Nowadays, when wideband core data networks are used, the last mile access of the user is the “bottleneck” in the path between the end user and the information source. For example the “bottleneck” is a GPRS, EDGE or UMTS mobile data access channel that is analysed in this chapter.

In places where more data users are located one of the most popular access technologies is WLAN, because it is a fast and reliable access technology and WiFi end user equipment is implemented in most laptops. From the network side, for moving hot spots or for hot spots in areas where there is no possibility to install fixed radio links satellite links could be used. However, GPRS, EDGE and UMTS mobile data networks in rural areas are very low loaded and the GSM network is high loaded only in short peak hours. The GPRS data network covers all territory of the country. The EDGE technology has been already implemented in main towns of Lithuania and is being implemented in the countryside and near main roads very fast. Therefore, mobile data channels could also be used very effectively as a cheap and reliable backbone for moving or temporally installed WLAN hot spots [A6]. For example, parallel mobile data channels could be used for connection of local networks of village schools to the Internet. This technology could also be used for connection of remote automation systems that require high connection reliability [A10]. High reliability is obtained combining several parallel mobile data channels of one or several BTS or even several mobile network providers.

An access channel of an individual user in a mobile data network should be considered as a channel with stochastically varying bandwidth. Obviously using one of these channels it is not possible to ensure any QoS. Only combination of several independent channels with stochastically varying bandwidth could increase QoS of the channel. Independent channels could be realized connecting different MS to different BTS of one or several mobile network providers.

In this chapter possible solutions of connecting WLAN hot spots using a bundle of parallel GPRS, EDGE and UMTS mobile data channels are investigated. In order to ensure sufficient data transfer rate and high connection reliability combination of 2–6 mobile data channels from different base stations is analyzed. The main purpose of this investigation is to confirm a hypothesis that combination of several mobile data channels could improve quality of mobile data services and make it comparable to QoS in fixed data networks.

Today combination of mobile data channels or in other words inverse multiplexing is a poorly investigated field.

In this chapter a solution for connection of mobile data channels and load balancing, that was done at OSI – Network level using different load balancing protocols, is proposed. Continuous connectivity of user applications during a hot spot handover between technologies was realized using VPN tunnels for every data modem and supplying one gateway IP address for WLAN hot spot users [13]. One of the key features of the proposed solution is free selection of OSI – physical layer channels. GPRS, EDGE, UMTS and even WLAN or satellite channels could be combined there.

4.1. Combination of mobile data channels

Bandwidth of the access network of the user should be increased because of the expansion of the internet network, growing amount of IP data traffic and new data services. Quite a long time ago for improvement of the existing transmission systems bandwidth combination of several channels was used. It was done keeping the same network infrastructure and using the existing hardware.

Many authors have previously analysed transmission of a single user data stream through multiple channels in various aspects. In some cases synonyms of this research are: Inverse multiplexing, Multihoming, Bandwidth aggregation, Multilink, Data stripping, etc. [23, 25, 34, 53, 57, 67, 72, 73, 84, 106].

Multiplexing is a very well understood communication technique for transmitting multiple streams through a single interface. The basic concept of multiplexing is to divide (multiplex) and transmit a data stream via several (independent) channels in time (time – division multiplexing (TDM)), in frequency (frequency – division multiplexing (FDM); a typical example is orthogonal FDM (OFDM)), or in space (space – division multiplexing (SDM)).

Demultiplexing is the technique to recover a stream from the aggregated flow. Multiplexing and demultiplexing are used, for example, to send multiple TCP flows through a single IP interface [25–27]. The port number contained in the TCP header is used for demultiplexing the aggregated TCP streams, in order to deliver data to the right application.

Inverse multiplexing is the aggregation of multiple independent information channels across the network to create a single higher-rate information channel. Inverse multiplexing is used to transmit a single stream through multiple interfaces.

For example, ATM uses inverse multiplexing [29, 34, 107] to aggregate fractional interfaces into a single interface with higher bandwidth. This type of bandwidth aggregation can also be found in the Internet's link layer [30, 78]. The current techniques limit aggregation to links of the same technology between the same endpoints.

Striping is another name for the technique of aggregating multiple resources in order to obtain higher performance [53]. Striping was originally developed for use within disk subsystems, but has also become helpful in network subsystems. Data striping across multiple, parallel communication channels is a conventional communication technique used to improve system performance or reliability in relatively statically configured disk storage systems [17, 106] and fixed, wired LAN – WAN interconnection systems [34]. For any transport level striping mechanism to be effective, an essential characteristic is that congestion control must be performed independently for each network path.

Multihoming is defined simply as a host having more than one external link, either to a single ISP, or to different providers. In other words, a host is multihomed if it can be addressed by multiple IP addresses [10], as is the case when the host has multiple network interfaces. Multihoming has a multitude of benefits to offer, be it for the individual consumer, or for corporate entities. Multihoming could be used to increase fault tolerance, resistance to failures and total data throughput. However multihoming is basically provided to promote fault tolerance. If the primary address does not respond, the second address may be tried.

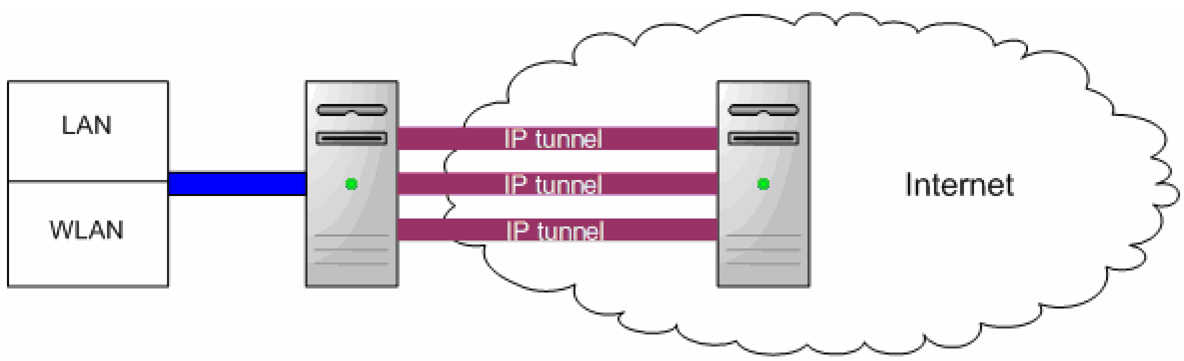

Fig. 4.1.1 Combining parallel IP data channels

In summary parallel channels could be combined in physical, channel or network OSI layers [23, 56, 84, 86, 87, 97]. Network fragment infrastructure with combined parallel IP data channels is presented in Fig. 4.1.1.

Combining channels in GSM standard networks

In the first GSM standards mobile data transmission was forecasted. For data transmission of every user one Circuit Switched Data (CSD) TS could be used. The maximum data throughput was 9.6 kb/s. This data throughput was limited by one GSM TS bandwidth.

Such kind of data transmission throughput does not fulfil user requirements and slows down expansion of mobile data service market. During development of GSM standards the TS combining technology was applied and higher throughput technologies like HSCSD and GPRS were implemented. Using these technologies several TS could be combined in order to create a higher throughput data channel for an individual mobile user.

In HSCSD up to 4 Circuit Switched channels could be combined [77], meanwhile in GPRS up to 8 Packet Switched channels could be combined for one mobile data user. Using these technologies the maximum data throughput for one mobile data user was increased up to about 80 kb/s.

Logically future development of combining of mobile data channels should be based on combination of channels of several frequencies. This is possible only using several transmitter/receiver modules (several MS). Frequency combining technology is not described in GSM standards and is not used.

Because mobile data networks are IP based networks, for combination of several MS IP channels we could use all network OSI layer IP protocols and standard IP network technologies. Several possible ways of combining of mobile data channels are analysed:

1. Parallel channels are created using recourses of one BTS. It is effective only when the BTS has several times more recourses than one MS could handle. Otherwise shearing of small BTS recourses among several combined MS decreases total data throughput of the combined channel.

2. Parallel channels are created using recourses of several BTS. Mobile data channels from several BTS of one or several mobile network providers could be combined. It could be expected that this method will create a combined mobile data channel with better QoS characteristics [6].

It could be noted that combining technology of several BTS channels is basically similar to nowadays widely analysed Spatial multiplexing technology, adopted for the higher OSI layer (IP networks).

Spatial multiplexing is a technique [63] that increases bit rate by using multiple antennas at both ends of the wireless link. This increase comes at no extra bandwidth or power consumption. The basic idea is that the use of multiple antennas in both the transmitter and the receiver opens up multiple parallel spatial data pipes within the same bandwidth and allows linear (in the number of antennas) capacity increase. Example of the typical architecture of spatial multiplexing system is shown in Fig 4.1.2.

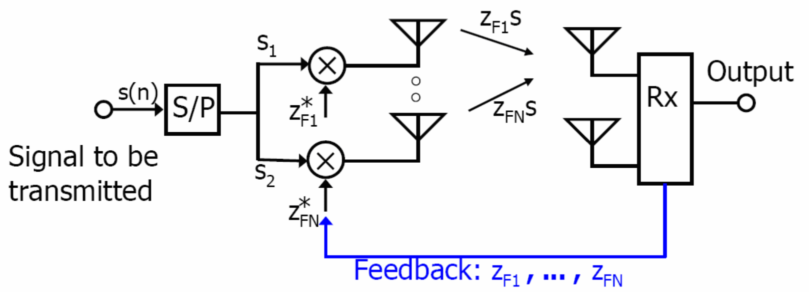

Fig. 4.1.2 Different data bits are transmitted via several independent (spatial) channels

It could be noted that combining technology of varying bandwidth of GPRS, EDGE or UMTS mobile data channels has not been investigated yet. In order to accumulate experience in this field, experiments of combining mobile data channels were done.

4.2. Experimental investigation of possibility to combine mobile data channels

In order to have better QoS characteristics of the total channel it is possible to bundle from 2 to 6 mobile data channels. One mobile data channel is realized using one mobile data modem. Using omni directional high gain antennas oriented to different sides, mobile data modems could be connected from 1 to 3 cells of base stations of the same mobile operator. If there is a need to combine more mobile data channels several mobile operators could be used. Different operators have totally independent networks, so the number of reachable different cells and reliability of bundled channels will increase.

It could be suggested to do aggregation of channels in OSI – Network layer using two software routers inserted into the network near the wired-wireless borders. The routers represent gateways between fixed and mobile data networks and between mobile and local data networks (Fig. 4.2.1). This system could be deployed by a network access provider, a wireless telecommunication service provider or a content distribution network operator.

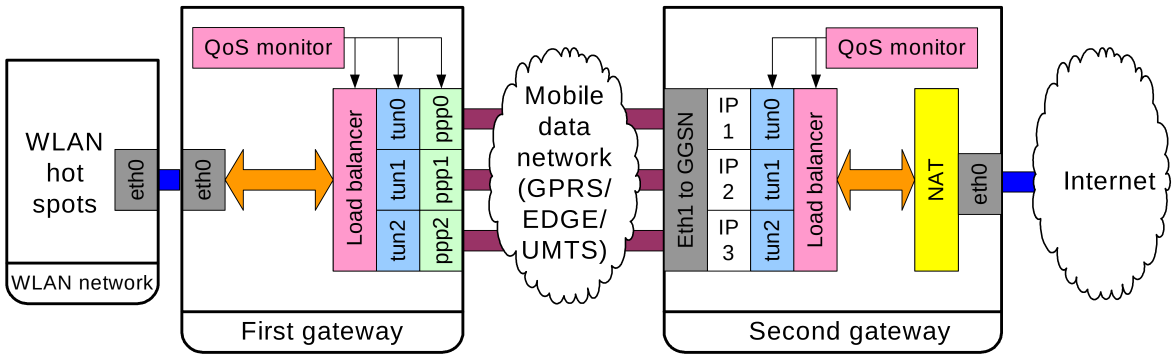

Fig. 4.2.1 LAN/WLAN connected to the Internet using a pool of parallel mobile data channels

The main gateway functions are:

1. Establishment of several connections to the mobile data network.
2. Establishment of several independent IP tunnels between the first and the second gateways via mobile data network.
3. Monitoring of quality characteristics of each tunnel.
4. Recovery of dropped mobile data connections and tunnels.
5. Real-time dynamic load balancing of user data packets among several parallel channels depending on QoS characteristic of each channel.
6. Adopting TCP flow of the fixed network to the mobile network environment. This could be done using a network accelerator [45, 90]. It breaks down TCP links, creates a reliable UDP connection inside the mobile network (between the first and the second gateways) and performs user traffic compression.

For every modem in the first server and for every interface in the second server fixed IP addresses should be assigned. During start up individual tunnels will be created for every modem. Depending on the used tunnelling technology data throughput of every data channel drops from 5 to 8 percents. Depending on the load of every channel user data packets should be dynamically load balanced for all active tunnels. Retransmitted packets should be sent with high priority.

The QoS monitoring system should monitor status of every channel evaluating confirmation messages of sent packets. When there is no user data traffic, periodical “keep alive” messages should be used. In case of tunnel failure, the tunnel should be removed from the bundle and the channel restoration algorithm should be applied. After successful reestablishment of the channel it should be returned to the pool.

4.2.1. Investigation of existing load balancing technologies of IP channels

Load balancing technologies based on the load balancing method are divided into two groups:

1. **Per-Packet load balancing.** The router periodically distributes all user packets for all parallel channels one by one. Using this method data throughput could be increased for every individual user. However, there is a risk that the packet will reach the destination address disordered.

2. **Per-Destination load balancing.** The router distributes packets taking into account their destination address. Using this method a packet to the same destination address can be sent only through one data channel. This load balancing method is effective only if there are many users going to different destinations.

Routing information protocol (RIP)

The RIP protocol supports traffic load balancing only among channels with the same QoS characteristics. For example, if there are two 512 kb/s channels every other packet will be send to a different channel. If one of the channels fails, then all packets will be send to another channel. This protocol cannot be used for load balancing among channels with different and dynamically changing characteristics, because the load balancing algorithm can not calculate how many packets were send to every channel. It can simply send equal number of packets to each channel.

Interior gateway routing protocol (IGRP)

IGRP is a routing protocol, developed to guaranty continuous connectivity among stand-alone systems. The protocol calculates the distance vector of each possible route. The distance vector includes route throughput, delay, reliability and load. Using this protocol load balancing of the packets could be realized among channels with different QoS characteristics. The number of packets sent to different channels depends on the distance vector. For a faster channel more packets will be sent, for a slower one – less packets. However, for load balancing to become effective QoS characteristics of channels should be carefully set by the system administrator. Therefore, using this protocol it is impossible to load balance packets among the channels with dynamically changing QoS characteristics.

The enhanced interior gateway routing protocol EIGRP

The EIGRP protocol has been developed by Cisco Systems and it is proprietary. This protocol could automatically evaluate changes of the network topology and QoS of the links. It periodically updates the routing table that includes:

1. The lowest throughput to the destination address;
2. Channel delay;
3. Channel reliability;
4. Channel load;

and etc.

This protocol supports the Cisco Express Forwarding (CEF) algorithm. Using the per packet load balancing CEF algorithm, the packets could be sent to the lowest loaded channel at a particular moment. It is done basing on the information, stored in the routing table. EIGRP can load balance user data packets maximum among 4 parallel channels.

Trivial Link Equalizer (TEQL)

It is a software routing protocol, developed for the Linux operating system. This protocol can load balance user data packets for many parallel channels. However, an equal number of packets is allocated for all channels. Therefore, only channels with equal QoS characteristics can be combined.

Open Shortest Path First (OSPF)

Only per-destination load balancing could be realized using the OSPF protocol. It is effective only when many users are connected and create many flows. If there are only several users, the maximum utilisation of the channels will not be reached and users will get services of lower quality. To achieve the maximum quality of a combined channel for an individual user per-packet load balancing should be realized.

4.2.2. Use of the EIGRP protocol for load balancing of data channels

In order to evaluate a possibility to use the EIGRP protocol for load balancing of user data packets among several mobile data channels, the modelling using the Opnet modelling SW and real experiments using Cisco routers were carried out.

The opnet modelling SW was chosen as the best modelling SW for the EIGRP protocol simulation. During investigation, other modelling tools like Omnet+, NS2, SSF Net and J-Sim INET were tested as well.

During the modelling common data traffic was set to 200 kb/s and the load was balanced among three 128 kb/s channels using load balancing per packet. After 2100 seconds the first channel was disconnected, therefore, total data throughput was load balanced between the second and the third channels. After 3600 seconds from the start of the modelling the second channel was disconnected as well, therefore capacity of the third channel became too low to handle all traffic and packet drops appeared (Fig. 4.2.3).

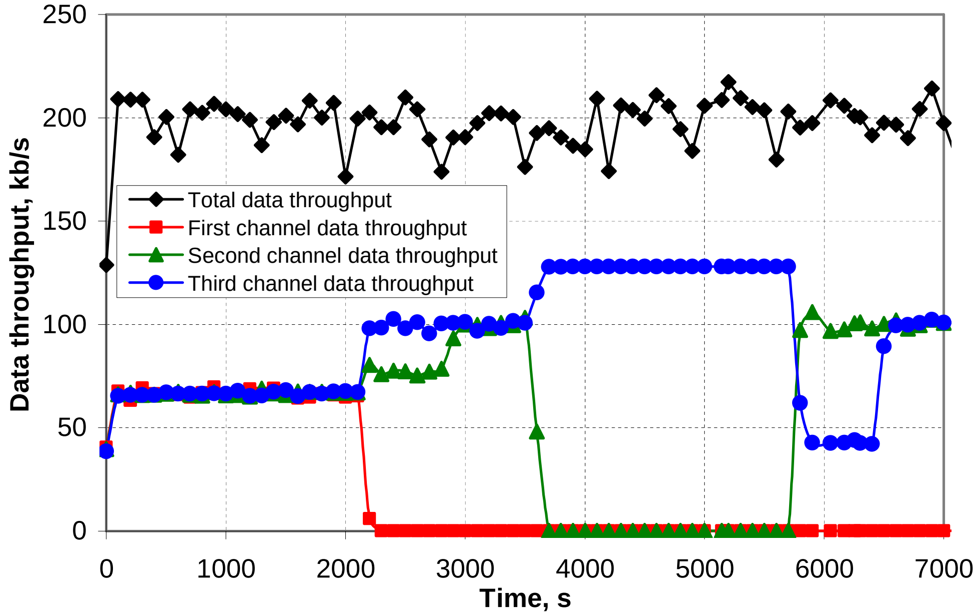

Fig. 4.2.2 Load of different channels during the modelling of load balancing using the EIGRP protocol in the Opnet modelling environment

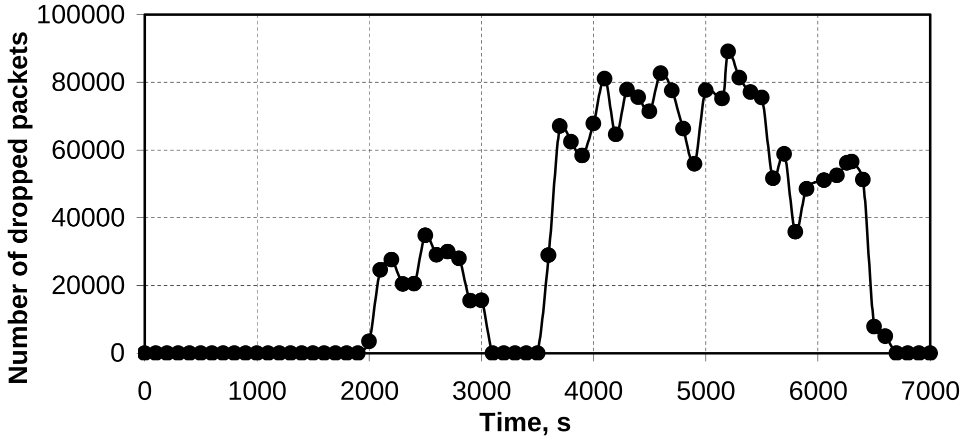

Fig. 4.2.3 The number of dropped packets of the total data flow during the modelling

Using the Opnet modelling SW it was possible to test load balancing only among channels with equal bandwidth. It means that the feature of the EIGRP protocol that allows load balancing among channels with different bandwidth is not realized in the Opnet modelling tools. Furthermore, it was not possible to realize load balancing among channels with different and stochastically varying bandwidth using any of the analysed standard modelling tools.

Therefore, experimental tests of the EIGRP protocol using HW Cisco routers were done. Network architecture presented in Fig. 4.2.1 was realized using a Cisco 1751 router, three Siemens MC35 mobile data modems and three omni directional antennas. In this way a remote LAN was connected to the Internet through three parallel mobile data channels. During the investigation user traffic was created using several personal computers that generated HTTP and FTP sessions and that were connected to the remote LAN.

For measurement of performance of each channel the Paessler Router Traffic Grapher (PRTG) SW was used. Data throughputs of these channels were measured:

1. Combined channel – FastEthernet0/0 interface (Fe0/0);
2. First parallel channel – Serial0/1 interface (Se0/1);
3. Second parallel channel – Serial1/0 interface (Se1/0);
4. Third parallel channel – Serial1/1 interface (Se1/1);

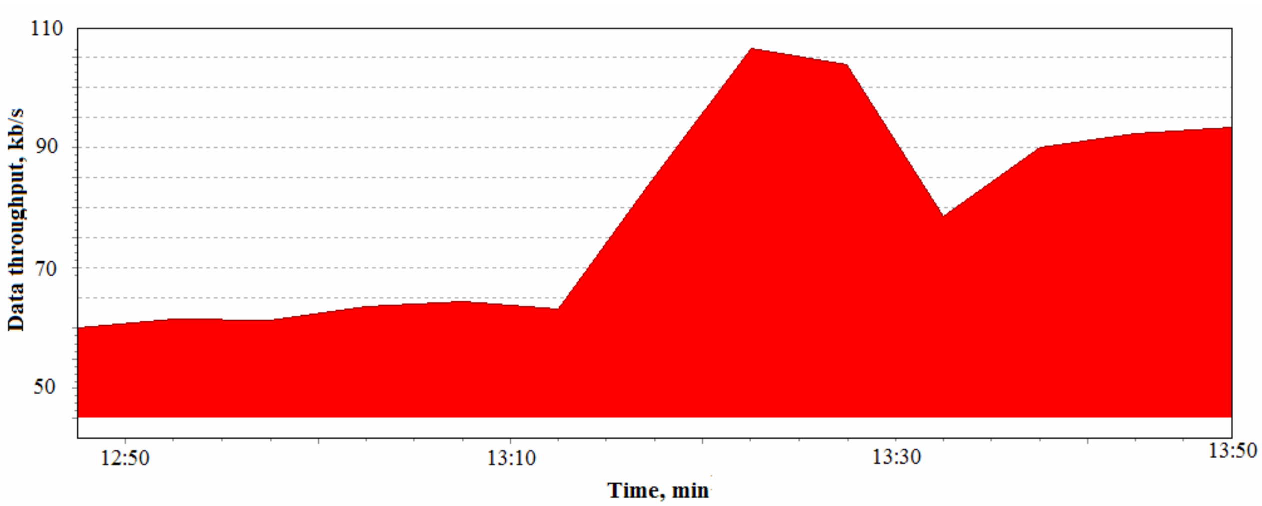

Fig. 4.2.4 Combined average data throughput (Fe0/0 interface). An averaging interval is 5 minutes

However, during analysis of not average but instantaneous data throughput it could be noted, that combined data throughput is not stable and fluctuates in big range.

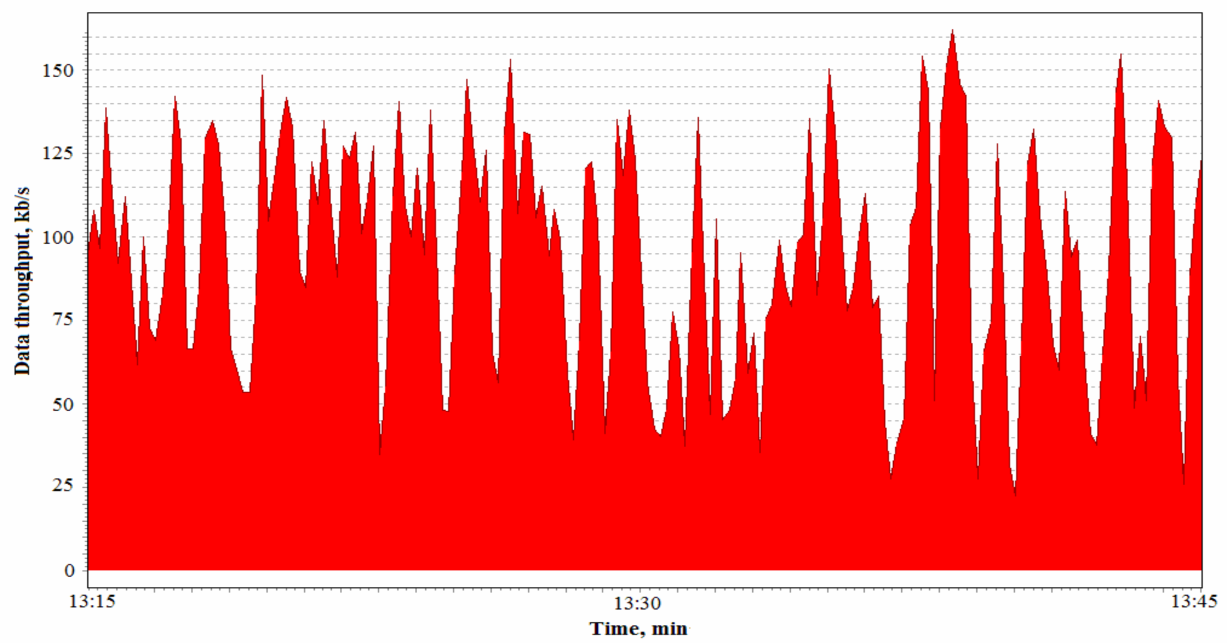

Fig 4.2.5 Combined instantaneous data throughput (Fe0/0 interface).

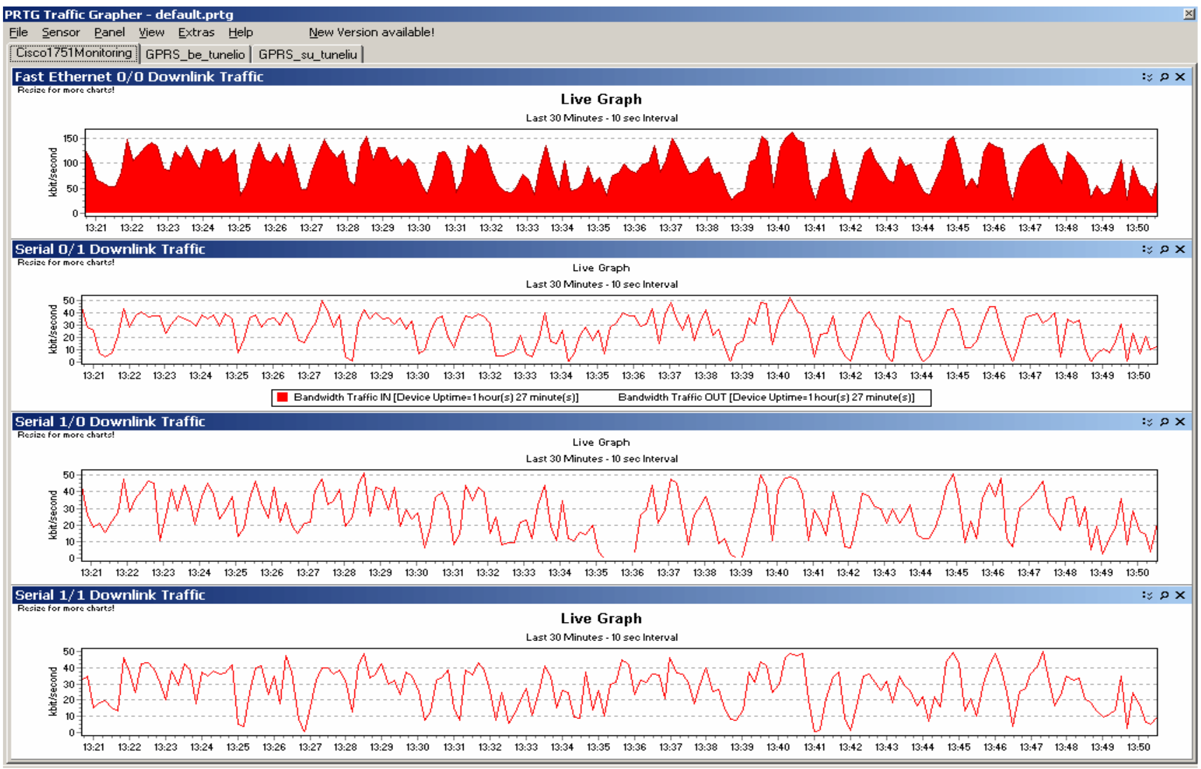

Fig 4.2.6 Combined data throughput and data throughputs of all three parallel channels

From Fig. 4.2.4–4.2.6 it could be noted that the combined average data throughput had reached up to 100 kb/s. The average data throughput of every channel was about 30 kb/s. From measurement results it could be stated that load balancing and combining of mobile data channels could be done using the EIGRP protocol, implemented in HW Cisco routers.

However, measurement results indicate that data throughput of every mobile data channel and total data throughput of a combined data channel is unstable. If only one MC35 modem would be directly connected to a personal computer, data throughput gaps would disappear and data throughput would become stable. Throughput instability using Cisco routers appears because the EIGRP protocol has been developed not for very dynamic and low bandwidth mobile data channels but for more stable and broadband data channels. The internal QoS monitor of the EIGRP protocol evaluates mobile channels and indicates that QoS of these channels is insufficient. Then it periodically applies a restoration algorithm of the channel that redistributes the load among other channels and sometimes even restarts of the mobile connection. It happens so because during development of the EIGRP protocol a possibility to use low and varying bandwidth of GPRS/EDGE mobile data channels has not been taken into account.

4.2.3. Use of the TEQL protocol for load balancing of data channels

For the second experimental investigation a Linux software router with the Trivial Link Equalizer was chosen. Tests were carried out in the Omnitel mobile network using two test Linux servers (gateways), 3 high gain omni directional antennas, three – GPRS, three – EDGE and two UMTS mobile data modems in three different ways. The user traffic was generated using IPERF SW. The network configuration is presented in Fig. 4.2.7. It could be noted that for combination of parallel channels 2***n*** external IP addresses are needed.

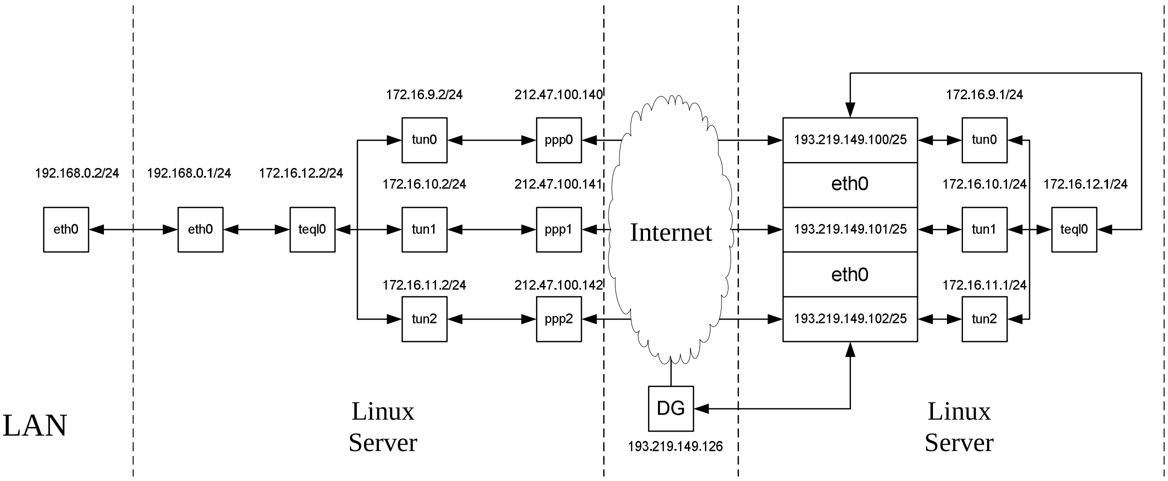

Fig. 4.2.7 Network configuration during combination of three mobile data channels

Queuing of packets, load balancing and tunnelling decrease the total data throughput about 10 % in average.

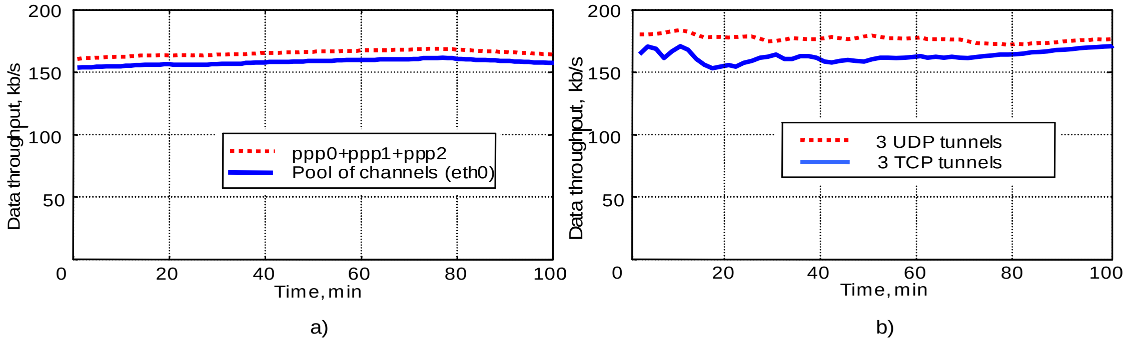

Fig. 4.2.8 Data throughput in the first server on Ethernet interface (eth0) and summarised data throughput of 3 GPRS modems (a) and data throughput on eth0 using two different (TCP and UDP) transport layer protocols for tunnelling through the GPRS network (b)

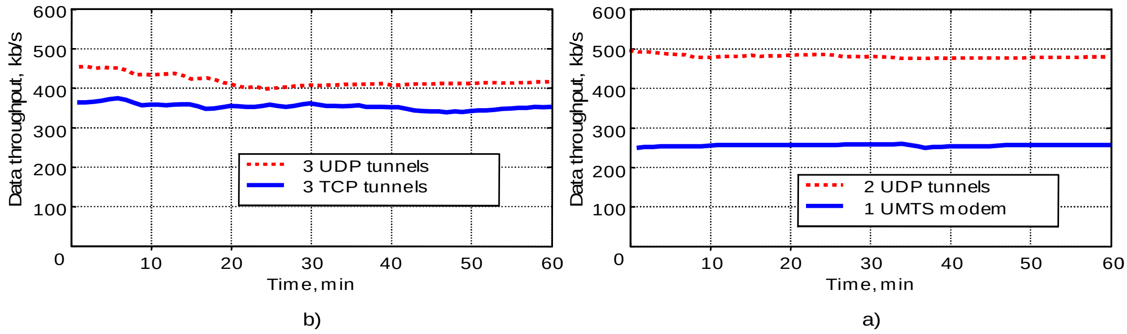

Fig. 4.2.9 Data throughput on eth0 using two different (TCP and UDP) transport layer protocols for tunnelling through the EDGE network (a) and data throughput measured in the first server on eth0 with one and then with two UMTS modems (b)

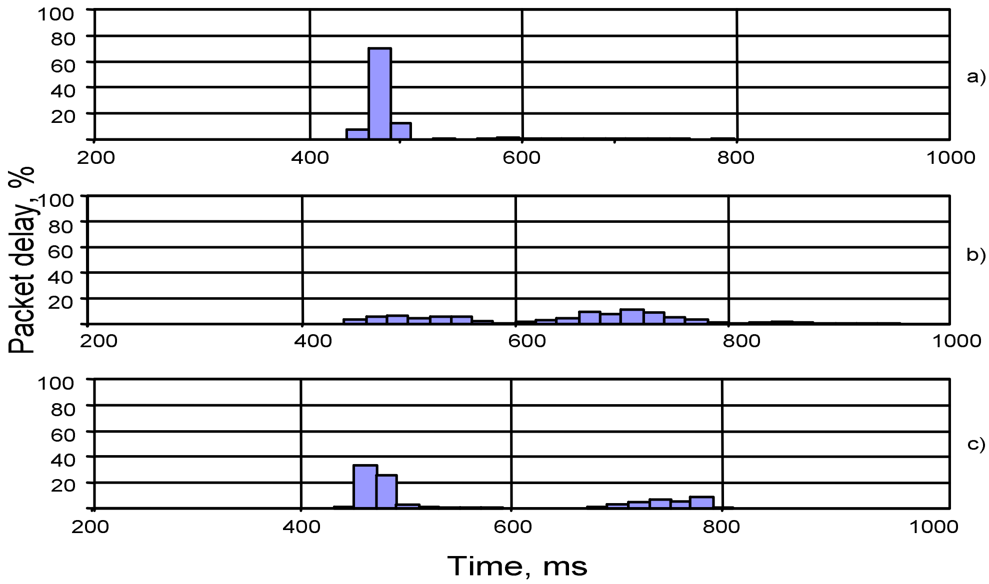

Fig. 4.2.10 Distribution of packet delay (round trip time) in the GPRS/EDGE network: a) one modem; b) pool of 3 parallel channels with the TCP protocol for transport layer tunnelling; c) pool of 3 parallel channels with the UDP protocol for transport layer tunnelling

Usage of TCP as a transport layer protocol decreases throughput of the total system from 15 % to 25 %. Using TCP packet delay variation is about 2.5 times higher. Use of a reliable UDP (rUDP) protocol with a simple packet retransmission algorithm resolves part of the QoS degradation problem. The TCP traffic loads the system not optimally and the end user receives services of lower quality. To resolve this problem an additional network accelerator of client-server architecture could be used [45]. It breaks down TCP links, creates a reliable UDP connection inside the mobile network (between the first and the second server) and performs user traffic compression.

4.2.4. Possibility to use the existing protocols for load balancing of data channels

Cisco EIGRP and CEF protocols, that were analysed during the experimental investigation, have been developed for load balancing of the traffic among channels with slowly fluctuating QoS parameters. These protocols cannot be used for load balancing among mobile data channels with stochastically varying bandwidth. Measurement of channel bandwidth and system adaptation is too slow and cannot dynamically evaluate the real situation in the channel. When channel bandwidth decreases very fast (normal situation in the GPRS/EDGE network) the internal QoS monitor recognises it as channel disconnection and restarts the connection.

The analysed TEQL protocol divides the common load for all parallel channels equally. These kinds of load balancing works well only if all channels have equal bandwidth and packet latency. However, when GPRS channels of different bandwidth are combined, common data channel throughput V_{Σ} could be calculated like this:

$$V_{\Sigma} = N \cdot V_{\min}^{n} . \tag{4.2.1}$$

Here N – the number of combined parallel channels, $V_{\min}^{n}$ – throughput of a data channel with the lowest bandwidth.

Using this protocol only a data channel with the lowest bandwidth is maximally loaded. All other channels are not utilised effectively.

During the investigation a protocol, suitable for dynamic load balancing per packet of traffic among several mobile data channels (taking into account bandwidth of channels and priority of packets), was not found. To fully utilise channels with stochastically varying bandwidth it is necessary to develop a protocol, supporting fast adaptation of QoS parameters to the channels and fast switching of packets.

4.3. Investigation of possibility to access several base stations

During planning of the GSM network some territory of the network is assigned for a particular base transceiver station (BTS). The number of areas where several BTS are serving is minimised. Therefore, connection of several cells of the same or different BTS without additional radio equipment is practically impossible. However, probability of connecting several BTS increases when high gain omni directional antennas oriented to different directions are used. In this case use of an additional antenna is equivalent to increase of cell serving area to a particular direction. A possible scheme of connecting a pool of mobile stations (PMS) to several cells of different BTS is presented in Fig. 4.3.1.

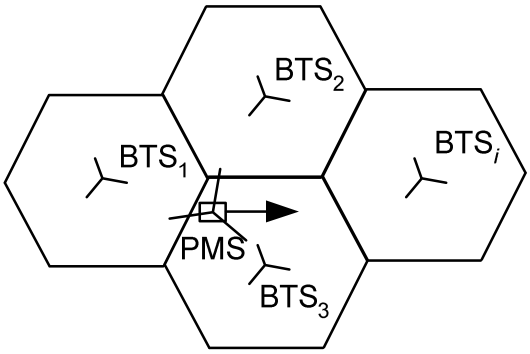

Fig 4.3.1 Pool of mobile stations (PMS) could be connected to several BTS using high gain omni directional antennas, connected to every MS and oriented to different directions

Requirements of QoS of every link are assumed to be fulfilled if C/I at the user terminal *i* (for all terminals $i \in R$) exceeds minimal C/I requirements g_0:

$$\mathrm{C/I}(i,j) = \frac{G_{i,j} \cdot P_i}{\sum\limits_{n \neq i} G_{n,j} \cdot P_n + N_i} \geq g_0, i \in R. \qquad (4.3.1)$$

Here P_i is to the user *i* transmitted power of the BTS; $G_{i,j}$ is the channel gain from the base station *j* to the terminal *i* including all channel elements; *n* is the number of all other user terminals; N_i is the noise power received at the receiver *i*.

If power of the BTS transmitted signal is *P*s then signal power of this BTS received by the user is P_{r} :

$$P_{\mathrm{r}} = P_{\mathrm{s}} - L(d, f, ...) \,. \tag{4.3.2}$$

Here $L(d, f, ...)$ is signal loss, d – distance between a MS and a BTS, f – transmitted signal frequency.

If the minimal acceptable power in the receiver is $P_{r\,\mathrm{min}}$ then the maximum distance between a BTS and a MS is d_{max} .

$$P_{\mathrm{r\,min}} = P_{\mathrm{s}} - L(d_{\mathrm{max}}, f, ...) \,. \tag{4.3.3}$$

If we connect a MS using an omni directional antenna with the gain G_A then

$$P_{\mathrm{A\,r}} = P_{\mathrm{s}} + G_{\mathrm{A}} - L(d_{\mathrm{A}}, f, ...) \,, \tag{4.3.4}$$

and

$$P_{\mathrm{A\,r\,min}} = P_{\mathrm{s}} + G_{\mathrm{A}} - L(d_{\mathrm{A\,max}}, f, ...) \,. \tag{4.3.5}$$

In (4.3.3) and (4.3.5) $P_{\mathrm{r\,min}}$ is the same variable, therefore both formulas could be equated.

$$G_{\mathrm{A}} - L(d_{\mathrm{A\,max}}, f, ...) = -L(d_{\mathrm{max}}, f, ...) \,. \tag{4.3.6}$$

Then

$$L(d_{\mathrm{A\,max}}, f, ...) = G_{\mathrm{A}} + L(d_{\mathrm{max}}, f, ...) \,, \tag{4.3.7}$$

and

$$\Delta d = d_{\mathrm{A\,max}} - d_{\mathrm{max}} \,. \tag{4.3.8}$$

The calculated increase in distance between a BTS and a MS Δd appears due to use of an external antenna with the gain G_{A} . In order to calculate the real increase in distance between a BTS and a MS a radio signal propagation model should be chosen.

For preliminary calculation a widely used ITU Hata radio signal propagation model was chosen. It is a simple model that is suitable for preliminary theoretical calculation. For future modelling of a real system PLANET modelling SW with a proprietary radio propagation model, digital maps and a precise BTS data base will be used.

The Hata radio signal propagation model is a simple formula, representing measurements of Okumura in the form: $L = A + B \log_{10}(d)$, where A and B are functions of carrier frequency and antenna heights. **d** is the distance between the transmitter and the receiver. The basic formula for medium path loss in urban areas was adopted by the CCIR in the form:

$$L_{\mathrm{U}} = 69.55 + 26.16 \cdot \log_{10} f - 13.82 \cdot \log_{10} h_{\mathrm{BTS}} - a(h_{\mathrm{MS}}) + (44.9 - 6.55 \log_{10} h_{\mathrm{BTS}}) \log_{10} d \,. \quad (4.3.9)$$

Here ***f*** is the frequency (in MHz) in the range of 450 to 1000 MHz, **d** is the distance (in km) between 1 and 20 kilometres, and h_{BTS} (in m) is the effective antenna height of the base station in the range of 30 to 200 meters. The height correction factor $a(h_{\mathrm{MS}})$ of a mobile station is applied for heights of mobile antennas h_{MS} (1 to 10 meters) for urban areas of small and medium cities:

$$a(h_{\mathrm{MS}}) = (1.1 \cdot \log 10 f - 0.7) h_{\mathrm{MS}} - 1.56 \log 10 f + 0.8. \quad (4.3.10)$$

In the Hata propagation model dependency of radio signal loss upon the distance between a BTS and a MS is described as $(44.9 - 6.55 \log_{10} h_{\mathrm{BTS}}) \log_{10} d$.

Therefore, increase in distance using an omni directional antenna with the gain G_A (using the Hata propagation model) could be calculated like this:

$$(44.9 - 6.55 \log_{10} h_{\mathrm{BTS}}) \log_{10} \cdot d_{\mathrm{A\,max}} = (44.9 - 6.55 \log_{10} h_{\mathrm{BTS}}) \log_{10} \cdot d_{\max} + G_{\mathrm{A}} . \quad (4.3.11)$$

Then

$$\log_{10} \cdot d_{\mathrm{A\,max}} - \log_{10} \cdot d_{\max} = \frac{G_{\mathrm{A}}}{(44.9 - 6.55 \log 10 h_{\mathrm{BTS}})}, \quad (4.3.12)$$

and

$$\log_{10} \left(\frac{d_{\mathrm{A\,max}}}{d_{\max}} \right) = \frac{G_{\mathrm{A}}}{(44.9 - 6.55 \log_{10} h_{\mathrm{BTS}})} \,. \quad (4.3.13)$$

From the formula (4.3.13) it could be noted that increase in distance between a BTS and a MS depends on the antenna gain G_{A} and effective antenna height of the base station h_{BTS}. Similar logarithmic dependencies could be found using other radio signal propagation models as well. It is so because of logarithmic nature of dependency of signal propagation distance upon signal loss in the channel.

Calculation results using the formula (4.3.13) as well as typical antenna heights of a BTS and typical gain of external omni directional MS antennas are presented in Table 4.3.1.

Table 4.3.1 Theoretical increase in distance between a BTS and a MS connecting the MS with external omni directional antennas

h_{BTS} , m	G_{A} , dB	$d_{\mathrm{A\,max}}/d_{\mathrm{max}}$
20	12	2.137
30	12	2.190
20	18	3.124
30	18	3.244

From calculation results it could be noted that increase of the possible maximum distance between a BTS and MS mainly depends on the external antenna gain. Theoretically, using an additional MS antenna with the gain of 12–18 dB, the maximum distance to the BTS could be increased up to 2–3 times. However, in real environment the maximum distance increase will be lower because of imprecise orientation of the omni directional antenna to the BTS.

Taking into account increase of coverage of cells, a pool of mobile stations could be connected up to 3 cells from one mobile network provider. Combining more data channels cells of different mobile network providers could be used.

Locations of BTS, load of cells and reliability are independent of networks of different providers, therefore combination of mobile data channels of several mobile network providers allows to increase bandwidth and reliability of the aggregated channel.

4.4. Influence of combining technology of data channels on QoS of an individual user

Combination of parallel mobile data channels will influence common data channel characteristics [73] of an individual user or a user group connected through a combined channel. The effected QoS characteristics will be:

1. Data throughput,
2. Packet delay and packet delay variation,
3. Channel reliability.

Data throughput

Mobile data channels have data throughput that is up to 10 times lower than that of a standard backbone technology used for WLAN hot spots [A5, A8, 11, 92, 104, 115]. QoS class in the GPRS/EDGE data network is still “best effort” and real data throughput depends on available recourses. However, the mobile data network in rural areas is loaded very low and experimental results show, that in most cases the achieved data throughput is near the maximum possible for terminals. Combination of mobile channels could increase the total achievable data throughput up to 3–5 times.

Table 4.5.1 Maximum data throughput that could be theoretically achieved by the user in up link (UL) and down link (DL) direction

	DL	UL
GPRS	85	43
EDGE	240	120
UMTS	384	384

Packet delay and packet delay variation

Packet round trip time is normally about 10 times higher in GPRS/EDGE. Nevertheless, during cell reselections several packets could be delayed up to 4–5 seconds or even dropped. Then, using bandwidth aggregation per packet, packet delay variation will be increased due to load of different channels. Additionally, some of packets could be delivered out of order. In this environment a standard TCP protocol works ineffectively. In that case there is a need for use of additional system components such as proxy flow aggregation [26] or server-client network accelerators [45]. They eliminate slow start, window stalls and etc. or even convert user side TCP traffic to a reliable UDP in the mobile data channel.

Channel reliability

Reliability of a mobile data channel is quite low. Depending on the mobile terminal, radio environment, time between reselections, the number of reselections and etc. disconnection from the network could occur in one or several hours. Use of several parallel data channels rapidly increases reliability of services for the end user. However, a special link monitoring and re-establishing system should be realized for every data channel.

Usually mobile data channels drop due to disconnection of the modem from the network. Therefore individual monitoring of every mobile data channel should be implemented. Reliability P of a bundled data channel that is made from k individual channels (if reliability of every channel p_i is mostly equal) could be calculated like this:

$$P = 1 - \prod_{i=1}^{k} (1 - p_i). \tag{4.5.1}$$

Reliability of a bundled channel increases additionally if mobile data channels of several mobile operators or different technologies (GPRS/EDGE/UMTS, WLAN/WiMax, satellite) are used. At present GPRS/EDGE networks have only “best effort” QoS, however in rural areas load of data networks is very low. If 3 mobile channels are combined, reliability of the bundled channel will be sufficient for connection of remote local networks and moving or temporal hot spots.

QoS characteristics of a bundled channel depend a lot upon the protocol used in the tunnel transport layer. A TCP protocol guaranties delivery of packets, however it is not adapted to the mobile environment. Such features as slow start, window stalls and etc. degrade performance of the data channel. A UDP protocol in this case is much more effective, though additional packet numbering and retransmission algorithms should be implemented.

4.5. Chapter conclusions

1. Experimental modelling results indicate that user terminals can be connected to the global network using several transceivers – several parallel mobile data channels. This increases bandwidth and reliability of the received individual user connection. However, combination of mobile data channels of different frequencies is not standardized.

2. Parallel GPRS, EDGE and UMTS mobile data channels could be effectively used as a quite cheap and reliable backbone for remotely installed local networks and moving or temporally installed WLAN hot spots. It is especially effective financially when the same operator provides WLAN and mobile data services.

3. From investigation results it could be noted, that, in order to guarantee sufficient data transfer rate and high connection reliability, combination of 2–6 mobile data channels from different base stations of one or several mobile network provides could be done.

Furthermore, theoretical calculations indicate that using high gain omni directional antennas a pool of mobile stations could be connected to several independent cells.

4. Bundling of channels increases the total data throughput, however queuing of packets, load balancing and tunnelling decrease the total data throughput about 10 % in average. Quality of every bundled channel should be monitored separately. Then, in the case of connection failure, every channel could be reset individually not influencing connectivity of end users. This rapidly increases reliability of data services the end user receives.

5. Measurement results indicate that data throughput of a combined data channel is unstable. Throughput instability using Cisco routers appears because the EIGRP protocol has been developed for stable and broadband data channels. It has not been adopted to be used for low and varying bandwidth of GPRS/EDGE mobile data channels. The internal monitor of QoS indicates that QoS of channels is low and periodically resets them.

6. During the investigation a protocol, suitable for dynamic per packet load balancing of traffic among several mobile data channels (taking in to account bandwidth of channels and priority of packets) has not been found. To fully utilise channels with stochastically varying bandwidth a protocol, supporting fast adaptation of QoS parameters to the channels and fast packets switching, should be developed.

5. GENERAL CONCLUSIONS

1. Measurement results in the GSM system indicate that the actually received QoS in different places and for different users is different. Statistical methods for monitoring of QoS, that are used in fixed telephony networks, are not appropriate to introduce any level of certainty of the provided mobile telecommunication services to individual users. For evaluation of individual QoS - iQoS in the network it is proposed to use special software modules installed in mobile phones. This evaluation mechanism would enable to provide objective information about the actually received iQoS to the user as well as to the network provider. This would increase certainty and satisfaction of users, allow implementation of service level agreements for cellular network customers and introduce new value-added services of high benefit to users (such as quality dependent billing).

2. From measurement and modelling results it could be noted that a GPRS/EDGE mobile data transmission channel for an individual (not statistical) user is very different from Frame-relay, Ethernet or other data transmission channels. Throughput and other quality characteristics depend not only on specific characteristics of the radio channel (RxLev, C/I end etc.) but also upon network coverage, user location, characteristics of the user terminal as well as the number of data and voice users. An access channel of an individual user in the mobile data network should be considered as a channel with stochastically varying bandwidth.

3. Parallel GPRS, EDGE and UMTS mobile data channels could be effectively used as a quite cheap and reliable backbone for remotely installed local networks and moving or temporally installed WLAN hot spots. It is especially effective financially when the same operator provides WLAN and mobile data services. From investigation results it could be noted, that, in order to guarantee sufficient data transfer rate and high connection reliability, combination of 2 - 6 mobile data channels from different base stations of one or several mobile network providers could be done. Quality of every bundled channel should be monitored separately.

4. A protocol, suitable for dynamic per packet load balancing of traffic among several mobile data channels (taking into account bandwidth of channels and priority of packets) does not exist now. To fully utilise a bundle of parallel channels with stochastically varying bandwidth a protocol, supporting fast adaptation of QoS parameters to the channels and fast packet switching, should be developed.

ABBREVIATIONS

AMR	Adaptive Multi Rate
ATM	Asynchronous Transmission Mode
AUC	Authentication Centre
BCCH	Broadcast Channel
BER	Bit Error Rate
BLER	Block Error Rate
BSC	Base Station Controller
BSS	Base Stations Subsystem
BTS	Base Transceiver Station
C/I	Carrier over Interference ratio
CDF	Cumulated Distribution Function
CEPT	Conference of European Posts and Telecommunications
CS	Coding Scheme
CSD	Circuit Switched Data
DCS	Digital Cellular System
DTX	Discontinuous Transmission
EDGE	Enhanced Data for GSM Evolution)
EFR	Enhanced Full Rate
EIGRP	Enhanced Interior Gateway Routing Protocol
EIR	Equipment Identity Register
ETSI	European Telecommunication Standards Institute
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FER	Frame Error Rate
FH	Frequency Hoping
FR	Full Rate
FR	Frame Relay
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GSM	Group Special Mobile
HLR	Home Location Register
HR	Half Rate

HSCSD	High Spead Circuit Switched Data
HTTP	Hypertext Transfer Protocol
IEEE	Institute of Electrical and Electronics Engineers
IGRP	Interior Gateway Routing Protocol
iQoS	individual Quality of Services
ISDN	Integrated Services Digital Network
ITU-T	Telecommunication Standardization Sector of the International Telecommunications Union
LAN	Local Area Network
MAC	Medium Access Control
MOS	Mean Opinion Score
MS	Mobile Station
MSC	Mobile Switching Centre
NMT	Nordic Mobile Telephone
OSI	Open Source Initiative
OSPF	Open Shortest Path First
PC	Power Control
PD	Packet Data
PDCH	Packet Data Channel
PMS	Pool of Mobile Stations
PSTN	Public Switched Telecommunication Network
QoS	Quality of Services
RIP	Routing Information Protocol
RLC	Radio Link Control
rUDP	Reliable UDP
RxLev	Received Signal Level
RxQual	Received Signal Quality
SDM	Space Division Multiplexing
SFH	Synthesised Frequency Hoping
SGSN	Serving GPRS Support Node
SIM	Subscriber's Identity Module
SW	Switchable
TA	Timing Advance
TBF	Temporary block flows
TCH	Traffic Channel

TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TEQL	Trivial Link Equalizer
TS	Time Slot
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunication System
VAS	Value Added Services
VLR	Visited Location Register
WLAN	Wireless Local Area Network

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