

# Telecommunications systems for ITS solutions

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# 1 Introduction

As it was discussed in the previous chapter of this book processes in the ITS (Intelligent Transport Systems) architecture are defined as a chain of system components interconnected by the information links – see Figure 1.1. Each system component carries the implicit system function (like F1, F2, F3, G1, G2, G3, etc.). The terminator (e.g. driver, consignee, emergency vehicle) is often the initiator and also the terminator of the selected process.

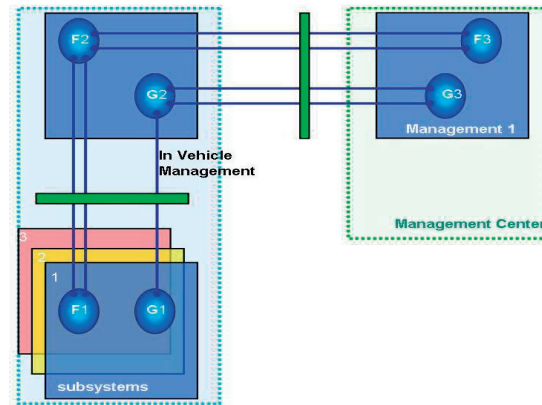


Figure 1.1 ITS architecture

The processes flows are mapped in physical subsystems or modules. The second process is defined by chaining the functions G1, G2 and G3 and the information flows between that functions which specify the communication links between the subsystems or modules. If time, performance or other constraints are assigned to different functions and information links, the result of the presented analysis is represented by table of system requirements (sometimes even contradictory) assigned to each physical subsystem (module) and physical communication link between the subsystems.

It is possible to consider a single universal subsystem fulfilling as precisely as possible system parameters. However, the creation of several subsystem classes according to a set of system parameters is feasible, as well as the creation of a modular subsystem where the addition of another module or module functionality upgraded entails the increase of system parameters, etc.

In case all the processes are already mapped by the physical subsystems or modules, the following results of process analysis can be achieved:

- functional specification assigned to each selected subsystem or module,
- interface specification,
- performance specifications of processes.

These decomposition results specify the requirement on each telematics service as well as allow to quantify the risks related to the requested parameters which are not reached.

The telematic subsystem performance quantification based on the performance indicators (see e.g. [2]) - is structured as follows:

- Availability – this service is available after the initiation process within the defined time interval on the certain probability level,
- Reliability – the defined level of service accessibility in the appointed time interval on the certain probability level,
- Accuracy – the maximal measuring error is less than the defined limit on the certain probability level,
- Continuity – time period, the service is not available in the defined time interval on the certain probability level,
- Integrity - if an accuracy exceeds the defined limit, the central system must be informed within the defined time interval, on the certain probability level,

- Security – the risk analysis and relevant classification must be done based on the detailed knowledge of the system environment and potential risks. The relevant solutions I usable if application of the additional security tools and redundant architecture.

The same approach can be applied on the other parts of ITS system as well as in case of different transport modes, e.g. road and railway transport. It is necessary to consider whether each transport mode has to have specific subsystem next to available ones or whether there is an opportunity for sharing such subsystems.

The telecommunication environment integrated in ITS solution significantly impacts ITS performance. The telecommunications solutions have the same modular principles, so that e.g. higher system parameters on the information transmission can be achieved by replacing/upgrading or adding the additional (SW/HW) modules and parameters of each module as well as of the whole telecommunication system can be described by a set of specific performance indicators set. However, for ITS system it is reasonable if both performance indicators systems are defined in a way. Under such condition the telecommunications subsystem impact on performance of the ITS system can be identified from identified values of telecommunications performance indicators.

## 2 Position of telecommunications solutions in ITS solution

Each telecommunications service is characterized by its performance quantified by set of performance indicators, and, such service is based on application of the telecommunication network. Telecommunications infrastructure, i.e. telecommunications network, can be shared by many users as well as services of different parameters. It is commonly accepted that telecommunication services are provided in limited number of categories or more frequently presented as classes. This approach is connected with general orientation of telecommunication business to commodity business where majority of provided services are limited number of off-shelf commodities with defined range of performance indicators values ready to be directly applied by clients individually or in combination of the services which are “wrapped in the package”.

It is not possible to expect public off-shelf telecommunications service providers selective approach in case of telecommunications services provided for ITS solution namely if ITS applications requirements are typically remarkably above requirements of majority of public telecommunications services users. It is explained by “core business” public providers concentration.

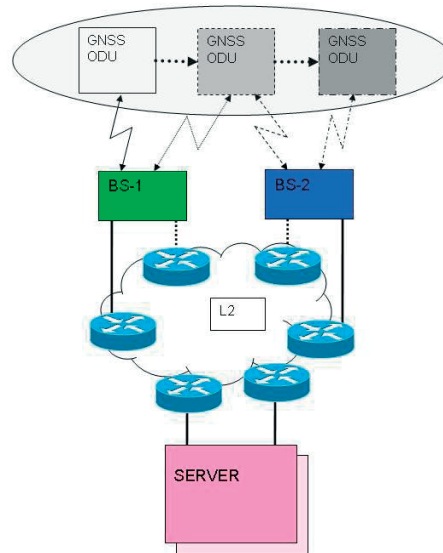
The specific products and approaches have been required in a case of ITS solutions. If such service quality and coverage is for any reason refused to be delivered by public providers there usually exists private solution with relevant parameters, however, the cost of service often represents the mail problem that is for such solution not acceptable. The limits calculated by financial models lead designers to apply mass market available telecommunications products if it is technically acceptable. If it is feasible the dynamical combination of more different services in the multipath regime (either public or privately designed and operated services) are acceptable to cover such specific requests. It can of course be done if such approach is justified by the financial models. Frequently it can be reached in a case of life-critical services.

**Tab. 2.1 Private vs. publically available telecommunications services**

	Private	Public
<b>Service quality management SLA</b>	Typically Available	Low (if any)
<b>Signal coverage</b>	Selectable/Cost dependent/Typically low	High
<b>Pricing/cost</b>	High (-er)	Low (-er)

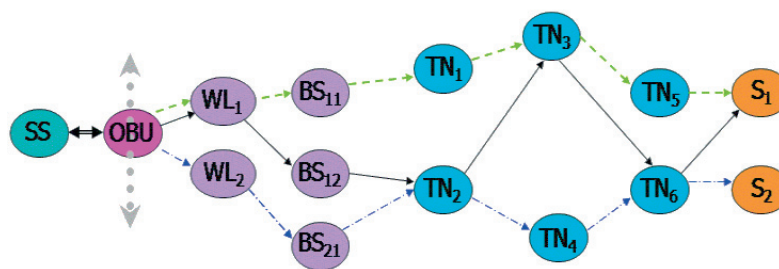
A typical telecommunication solution designed for ITS represents a chain – GPS/Galileo OBU (Out Board Unit) of GNSS (Global Navigation Satellite System), communication sub-chain of mobile part with system of BSi (Base Stations) network, terrestrial part equipped by routers/switches (L3 or L2 based solution) and a management central system (see Figure 2.1). Such a chain can be accepted as “typical” transport telematic architecture serving moving objects accessible by mobile access services. Base stations can be operated to the

same or more different providers (both private and public). The base stations may support different telecommunications technologies and relevant services from set of those which are supported by OBUs and installed in moving objects. The core issue of such approach lies on correct and the effective determination of the best possible wireless alternatives from the list of just available ones.



**Figure 2.1 Typical telecommunications network chain in ITS solution**

Figure 2.2 presents the architecture expressed in a form of telecommunications chain diagram. The introductory version of this architecture is connected with a pilot project of Airport Praha described later in this chapter, however, we later accepted it as the ITS telecommunications solutions general architecture. On Board Units (OBU), GNSS Sensing System (SS) and a set of Wireless Units (WL) are installed in the moving object. SS is now dependent exclusively on the GPS (Global Positioning System) with no SLA publically available services. However, it is expected that coming European Galileo GNSS as well as the 2nd generation of GPS will provide publically available paid services with guaranteed service quality. OBU represents not only control but also display and HMI (Human Machine Interface) and WL<sub>i</sub> represents i-th cellular technology of the wireless complex solution. The terrestrial communication part consists of set of mobile cellular Base Stations (BS<sub>ij</sub>) (j-th bases station of the i-th system) integrated by the terrestrial network based on L3/L2 switches/nodes (TN<sub>i</sub>) interconnected with Servers (S<sub>i</sub>). E2E (End to End) service is provided based on IP protocol, supported by L2 Ethernet protocol based switching.



Some alternative E2E communications paths: a)  $\cdots \cdots \cdots$  b)  $\longrightarrow$  c)  $\cdots \cdots \cdots$

**Figure 2.2 Telematics telecommunication scheme - chain diagram**

The typical general transport telematic access solution represents combination of wide range access alternatives. It is an area of spontaneously developed solutions. Generally accepted TCP/IP architecture offers the transparent interoperability, however, this approach needs to be combined with wide range of techniques and tools carefully installed and managed to obtain the requested service quality, reliability and security. Quite a few solutions in wireless area (Bluetooth, Zigbee, WAVE etc.) have not accepted TCP/IP architecture and such approach complicates the application of one OBU per vehicle due to problem of interoperability. The most

general concept presented by ISO (CALM) is ready to resolve such an issue, however, it is done on behalf of system complexity principal increase. This issue will be also discussed below. Correct and effective selection the best possible telecommunications solution takes in account the values of relevant performance indicators as well as the other appropriate parameters (like service cost). CALM ISO standards grouped in TC204, or WG15 by IEEE 802.21 standard represent the strongest solutions. However, the alternative approaches based e.g. exclusively on IP routing (categorized as “intelligent routing”) have been published, as well. One of such alternative approaches will be presented.

## 2.1 Telecommunications performance indicators

Definitions of the telecommunications performance indicators are outlined in the same way like are defined the telematics performance indicators. The main goal of such approach is to enable transformability of the telecommunications performance quantification into quantification of the whole telematics solution performance.

### 2.1.1 Service Activation Time (SAT)

SAT time needed for activation/modification of the network archived on certain probability level,

$$P(|a_i - a_{m,i}| \leq \varepsilon_1) \geq \gamma_1 \quad (1)$$

i.e., absolute value time difference of the  $i$ -th requested  $a_i$  and measured  $a_{m,i}$  system activation time is less than  $\varepsilon_1$  on the probability level  $\gamma_1$ .

### 2.1.2 Service availability

Service is available without any interruption in defined time interval:

$$P(|ca_t - ca_{m,t}| \leq \varepsilon_2) \geq \gamma_2, t \in \langle 0, T \rangle \quad (2)$$

i.e. absolute value of difference requested parameter  $ca_t$  and the measured (reached) one  $ca_{m,t}$  is less than  $\varepsilon_2$  on probability level  $\gamma_2$  within time interval  $\langle 0, T \rangle$ .

This parameter is typically replaced by known and frequently applied parameters MTBF a MTTR defined as follows

### 2.1.3 MTBF (Mean Time Between Failure)

MTBF is defined as time between two unexpected inoperable stages on certain probability level and

$$P(|f_i - f_{m,i}| \leq \varepsilon_3) \geq \gamma_3, \quad (3)$$

i.e.  $i$ -th absolute value of difference of requested  $f_i$  and archived  $f_{m,i}$  time between two failures is less than  $\varepsilon_3$  on probability level  $\gamma_3$ .

If time interval MTBF on the same probability level is representatively longer than time interval  $\langle 0, T \rangle$  of definition (2), it is possible to accept this parameter as insignificant.

### 2.1.4 MTTR (Mean Time to Restore):

MTTR is defined as time of service recovery from unexpected inoperable stage on certain probability level:

$$P(|rc_i - rc_{m,i}| \leq \varepsilon_4) \geq \gamma_4, \quad (3)$$

i.e. absolute value of difference of requested value  $rc_i$  and reached value  $rc_{m,i}$  of i-th recovery is less than  $\varepsilon_4$  on probability level  $\gamma_4$ .

This parameter is relevant in case if there is available automatic recovery functionality based on redundant network architecture. It is important to mention that namely in IP network processes this value varies in orders in dependence of recovery procedure PBB on L2 vs. MPLS/L3 based solutions.

### 2.1.5 Delay

Accumulative parameter defined as time frames which are delivered within a defined time period on a certain probability level:

$$P(|d_t - d_{t,m}| \leq \varepsilon_5) \geq \gamma_5, t \in \langle 0, T \rangle, \quad (4)$$

i.e. absolute value of difference requested maximum delay  $d_t$  and reached one  $d_{t,m}$  is less than  $\varepsilon_2$  on probability level  $\gamma_2$  within time interval  $\langle 0, T \rangle$ .

This parameter is effected by

- *interfaces rates*
- *trunks/links capacity,*
- *frame size, and*
- *load of all in line active nodes (e.g. CPU, memory).*

Some parameters like interface rates and packet size are static. Trunks/links and nodes load are dependent on other network users, they are predictable stochastically acting parameters. Probability of negative influence of any parameter overload in between others is namely in the hands of network manager and his responsible approach to forecast of each network bottle neck based on evaluation of all network flows. Correctly selected priority of each service selected for each application plays very important role, as well.

### 2.1.6 Packet/Frames Loss

Percentage of undelivered packets/frames within the defined time period on certain probability level:

$$P((pl_{t,d} / pl_t) \geq \varepsilon_6) \geq \gamma_6, t \in \langle 0, T \rangle \quad (5)$$

i.e. absolute value of ratio of delivered packets  $pl_{t,d}$  and sent packets  $pl_t$  is less than  $\varepsilon_6$  on probability level  $\gamma_6$  within time interval  $\langle 0, T \rangle$ .

This parameter has the same fatal consequences with network quality management and correctly selected priority of service. “Classical TCP/IP networks solution” is frequently replaced by UDP/IP with aim to reduce delay in the network with aim to postpone responsibility of evaluation and relevant action on identified missing packets on “more intelligent” application layer.

### 2.1.7 Security

$$P(|Wc_i - Wc_{m,i}| \leq \varepsilon_7) \geq \gamma_7 \quad (6)$$

i.e. absolute value of difference of requested value of risk situation and reached one  $Wc_i$  and the reached one  $Wc_i$  is less than  $\varepsilon_7$  on probability level  $\gamma_7$ .

Risk Analysis (RA) and classification must be done based on detailed knowledge of the system environment and potential risks. The risk of information integrity can be caused by attack on any part of the information transfer chain. Typically combination of more than one security tool – like authentication, coding or even tunneling on both  $L_2$  and application layer. Security and safety issues and methods represent complex issue and it is discussed in separate chapter.

### 2.1.8 Class of Service – (CoS)

CoS itself does not represent performance indicator. Each CoS represents some performance indicators values “tolerance range”. Such simplification approach has roots in effort of telecommunications services provided to offer maximum of services in off-shelf product regime. Such services are provided separately or in packages representing the set of applied services. These tolerance performance indicators values ranges are closely related with packet/frames/cells priorities and quality of service also determines the cost customer which is obliged to pay for selected CoS.

## 2.2 Communications design methodology

The impact of telecommunications services described by performance indicators values as defined in chapter 2.1 can be transformed into telematic performance indicators structure, and vice versa. Such transformation allows telematics system synthesis. Final additive impact of the vector of communications performance indicators  $\vec{tci}$  on the vector of telematic performance indicators  $\Delta \vec{tmi}$  can be expressed by equation (2.2.1), however, under condition that probability levels of all indicators are unified on the same level and all indicators are expressed exclusively by time or on time convertible value:

$$\Delta \vec{tmi} = TM \cdot \vec{tci}, \quad (7)$$

where  $TM$  is a transformation matrix. The identification of the  $TM$  represents an iterative process and it is handled in four steps. Identification process starts with matrix in the most general structure -  $TM_0$ . The transformation matrix takes into account all potential relations between telecommunications and the telematic indicators. The significance of any parameter of  $TM_0$  in context of the other processes is not evaluated in depth in this step. Matrix construction is logically dependent on the detailed communication solution configuration. Telematic performance indicators vector  $\vec{tmi}$  consists of:

- Accuracy  $p_i$ ,
- Availability  $t_{ds,i}$ ,
- Reliability  $t_{ma,i}$ ,
- Continuity  $t_i$ ,
- Integrity  $t_{tsna,i}$ .

Telecommunications performance indicator vector  $\vec{tci}$  is for described application:

- time to upload  $d_{u,i}$ ,
- time to download  $d_{d,i}$ ,
- handover within the same access technology  $rc_{hs,m,i}$ ,
- handover within different (CALM) media  $rc_{hd,m,i}$ ,
- feedback parameters settings period  $rc_{rp,m,i}$ ,
- MTTR of the terrestrial network service  $rc_{r,f,i}$ ,
- MTTR of the access mobile service  $rc_{r,m,i}$ ,
- time period fix service is not available (self-healing process not available or not successful)  $t_{na,f,i}$ ,
- time period mobile service is not available (self-healing process not successful)  $t_{na,m,i}$ ,
- time to accept OBU (On Board Unit) into relevant cell  $t_{oi,i}$ .

The general impact of the listed set of communications performance indicators on above defined set of telematic performance indicators is described for parameters:

*Accuracy* is expressed as distance vehicle reaches within the whole communication cycle

$$\Delta p_i = v_i * \left( d_{u,i} + rc_{hs,m,i} + rc_{hd,m,i} + rc_{rp,m,i} + \right. \\ \left. + rc_{r,f,i} + rc_{r,m,i} + d_{d,i} + t_{na,m,i} + t_{na,f,i} \right) \quad (8)$$

*Availability* represents the time required to accept wireless unit into a network plus the time to deliver information about successful acceptance

$$\Delta t_{ds,i} = t_{oi,i} + d_{d,i} + rc_{hs,m,i} + rc_{hd,m,i} + rc_{rp,m,i} + \\ + rc_{r,f,i} + rc_{r,m,i} + d_{u,i} + t_{na,f,i} + t_{na,m,i}, \quad (9)$$

*Reliability* means the time service is not available within defined period

$$\Delta t_{tna,i} = t_{na,f,i} + t_{na,m,i} + ns_{hsm,i} * rc_{hs,m,i} + ns_{hdm,i} * rc_{hd,m,i} + \\ + ns_{rpm,i} * rc_{rp,m,i} + ns_{sf,i} * rc_{r,f,i} + ns_{rm,i} * rc_{r,m,i}, \quad (10)$$

*Continuity* represents time period communications service is not available

$$\Delta t_i = rc_{hs,m,i} + rc_{hd,m,i} + rc_{rp,m,i} + rc_{r,f,i} + rc_{r,m,i} + t_{na,m,i}, \quad (12)$$

*Integrity* is expressed by the time needed to deliver information about the system failure

$$\Delta t_{tsna,i} = rc_{hs,m,i} + rc_{hd,m,i} + rc_{rp,m,i} + rc_{r,f,i} + rc_{r,m,i} + d_{d,i} + t_{na,f,i} + t_{na,m,i}. \quad (13)$$

*TM* structure for this set of parameters defined in Eq. (8) is as follows:

$$TM = \begin{bmatrix} k_{p,u,i} \cdot v_i, & k_{p,d,i} \cdot v_i, & k_{p,hs,m,i} \cdot v_i, & k_{p,hd,m,i} \cdot v_i, & k_{p,rp,m,i} \cdot v_i, & k_{p,r,f,i} \cdot v_i, & k_{p,r,m,i} \cdot v_i, & k_{p,na,f,i} \cdot v_i, & k_{p,na,m,i} \cdot v_i, & 0 \\ k_{d,u,i}, & k_{d,d,i}, & k_{d,hs,m,i}, & k_{d,hd,m,i}, & k_{d,rp,m,i}, & k_{d,r,f,i}, & k_{d,r,m,i}, & k_{d,na,f,i}, & k_{d,na,m,i}, & k_{d,oi,i} \\ k_{s,u,i}, & k_{s,d,i}, & k_{s,hs,m,i} \cdot ns_{hs,m,i}, & k_{s,hd,m,i} \cdot ns_{hd,m,i}, & k_{s,rp,m,i} \cdot ns_{rp,m,i}, & k_{s,r,f,i}, & k_{s,r,m,i}, & k_{s,na,f,i}, & k_{s,na,m,i}, & 0 \\ 0 & 0 & k_{k,hs,m,i}, & k_{k,hd,m,i}, & k_{k,rp,m,i}, & k_{k,r,f,i}, & k_{k,r,m,i}, & k_{k,na,f,i}, & k_{k,na,m,i}, & 0 \\ k_{i,u,i}, & k_{i,d,i}, & k_{i,hs,m,i}, & k_{i,hd,m,i}, & k_{i,rp,m,i}, & k_{i,r,f,i}, & k_{i,r,m,i}, & k_{i,na,f,i}, & k_{i,na,m,i}, & 0 \end{bmatrix} \quad (14)$$

where  $v_i$  is a vehicle velocity,  $ns_{hs/hd/rp,m,i}$  represents the number of phenomenon appearance (on appropriate probability level) in time interval  $\langle 0, T \rangle$ . Value of each parameter  $k_{xx,yy,m/f/-i}$  is in the end identified either as „0“ or „1“ in accordance to iterative process described below.

Each element of *TM* is consequently evaluated based on the detailed knowledge of the particular telematic and communications configuration and its appearance probability in context of whole system dynamics. This approach originated in physics so it represents the subsequent iterative process leading to the stage, where all the minor relations (indicators) are eliminated and the major coefficients of *TM* are identified under condition that all related telematic performance indicators are kept within given tolerance range. There are four steps of the process leading to the final stage:

- [I] *primary elimination* of communication parameter based on implementation of a relevant communication solution or setting (e.g. guaranteed homogenous radio signal coverage in defined area),
- [II] *primary disregarding* of communications indicator, if its weight can be justified as insignificant,
- [III] *identification* and exclusion of indicators with significantly *lower* level of their *appearance probability* (e.g. in case of coincidence of processes with unified probability level of their individual occurrence - the dominant one is appointed),
- [IV] *final iterative identification of dominant indicators* as the last step of the iterative process of the *TM* identification is based on the virtual communication solution parameters settings. Potential solution modification can however, lead the identification process back to step [I].

The presented method is designed as general as possible to cover the widest range of telematic solutions and it is applicable for CALM standards management criteria identification, as well.

## **2.3 Wireless telecommunications alternative used in ITS solutions**

The most of ITS are applied to serve moving objects, so the appropriate wireless access solution is needed.

### **2.3.1 Publically available services with national/global coverage (Europe based)**

#### *2.3.1.1 GSM (Global System for Mobile Communications networks) based basic non-packet services*

- DTMF (Dual-tone multi-Frequency) is applied on voice channel,
- CSD (Circuit Switched Data),
- HSCSD (High Speed CSD),
- SMS (Short Message Service),
- USSD (Unstructured Supplementary Service Data) is not usually available for public applications,
- UUS (User to User Signaling) served via GSM is not usually available for public applications,

#### *2.3.1.2 GSM packet based services*

- GPRS (General Packet Radio Service)
- EDGE (Enhanced Data rates for GSM Evolution) data services
- UMTS (Universal Mobile Tele-communications System) data services are partially available, namely in the cities, rarely or no available in the rural areas. UMTS potential to grow in coverage is limited and in future is expected to be likely replaced by LTE technology – typically accepted as beyond 3<sup>rd</sup> because of this technology architecture and system philosophy principally differs from those known from 2<sup>nd</sup> and 3<sup>rd</sup> generation.

### **2.3.2 Locally available mostly as no publically served services**

#### *2.3.2.1 WiMax – IEEE 802.16d and Mobile Wimax 802.16e*

WiMax has been understood as promising telecommunications system, which meets even very specific wireless access performance indicators requirements. However, its telecommunications market was significantly delayed by “blocked” certification process and WiMax has been mostly available only in static installations (i.e. 802.16d). Expected reasonable Mobile WiMax cost/value is said to be behind the key carrier class technologies vendors to make smooth path for LTE finalization and market implementation. Delay caused by long-lasting certification process had destructive impact on penetration of that promising technology. It is clear that newly coming LTE has remarkable similarity in representative percentage of its functionalities if compared with Mobile Wimax. It is, however, unclear if and when LTE appears as widely commercially successful alternative.

#### *2.3.2.2 WiFi IEEE 802.11 with wide range of Amendments*

WiFi 802.11 a/b/g/p represents service with the wide range of Amendments like quality of service “e”, cellular architecture r, MIMO “n” etc. All technologies will be in detail discussed in separate sub-chapters. Due to its approach to the market WiFi has been understood as “low end” technology with ability to provide only “low end” QoS, if any. However, portfolio of WiFi functionalities added by Amendments including such like QoS support, cellular coverage, high quality radio solution etc. can gain in future its market position specifically to cover not well enough served areas by public services where high quality of service is requested e.g. by the ITS solutions.

#### *2.3.2.3 DSRC*

- *DSRC 5.8*

Dedicated Short-Range Communications (DSRC) is designed to provide „asymmetrically initiated“ communication (with semi-passive on board transceiver) between a vehicle and infrastructure (e.g. ETC DSRC units located on highways or selected roads).

- *DSRC 5.9 - WAVE*

DSRC 5.9 - WAVE offers C2I (Car to Infrastructure) or C2C (Car to Car) communication served for transport telematics services offer in distances range up to 1 km.

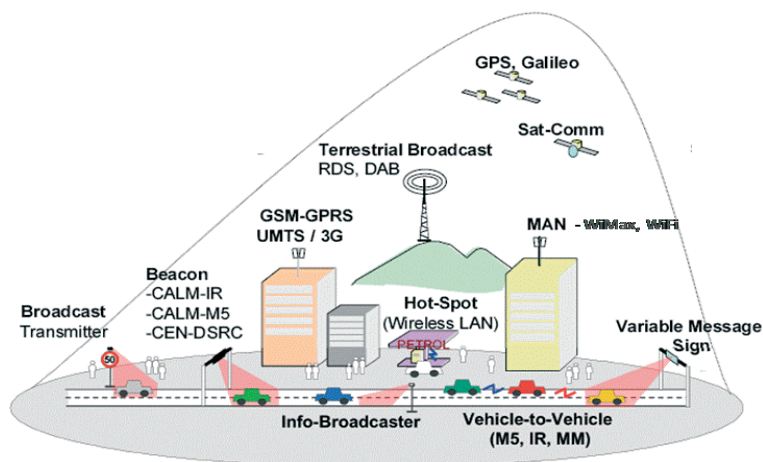
As it was described in chapter 2 the ITS solutions combine both mobile access and terrestrial backbone services. Final telecommunications performance indicators are influenced by performance of the terrestrial part of the overall telecommunications solution, as well. Terrestrial solutions are commercially available products and so the only basic principles description of core technologies and their system performance are described in Chapter 4. Even though MPLS and specifically MPLS over ATM solution of the IP networking are presented as the best backbone alternative, the cost of such approach as well as some performance parameters frequently cause ITS solutions migrate to the less expensive dynamically developing solutions, which can meet system performance requirements. Extended Ethernet based backbone solutions are getting priorities in remarkable number of backbone alternative. The background of the phenomena is discussed in Chapter 4, as well.

### 3 Telecommunications systems

A selection of relevant telecommunications solution represents the critical point of ITS solution design. Such a solution typically represents the combination of wireless and terrestrial solution. The performance of the whole telecommunication chain directly impacts the reached performance of the ITS solution.

The Ethernet family of standard was accepted first as protocol of a Local Area Network (LAN) L2. Mass production of Ethernet switches was caused by enormously growing Ethernet penetration into LAN solutions and it had strong impact on technology development as well as quick falling of market prices. The dynamically growing Ethernet services performance both in capacity as well as in variety of services on L2 caused its fast penetration leading to its L2 position dominance on the TCP/IP market. This “won battle”, however, was just the Ethernet first step in its positioning in WAN (Wide Area Network)/MAN (Metropolitan Area Network) and the definite turning point of Ethernet market position connected with fiber communication support implementation. It was started by the Fast Ethernet (100Mb/s) and the Ethernet is reaching today 100 Gb/s capacities with additional potential to grow.

The ITS requirements on size of the covered areas, the wireless mobile access as well as the wide range of required service performance represent the wide range of parameters and it varies from case to case. Figure 3.1 is a typical example of how the variety of different services are applied and each ITS application can represent the different system, i.e. performance indicators range. It is clear that kind of telecommunication service includes the combination of more services (e.g. terrestrial service approached by wireless mobile access). Frequently can be backbone (e.g. metropolitan MAN) solution shared by more access solutions with remarkably different requirements to be provided on the same network.



**Figure 3.1 Telecommunications Car2C and C2I in transport telematics**

Transport telematics systems apply the combination of different telecommunications systems, however, as telecommunications do not represent core business for ITS providers, it is expected that such services are

selected from portfolio of mostly publically available services - transport telematics companies do not have ambitions to become a telecommunications service provider. The private telecommunication solution is rarely provided by telematics provider and it is accepted just only if the required service in expected quality is not available in a certain area – often it is limited in size. This fact could be also a reason why only the very brief description of actually available telecommunications means are offered in this chapter with aim just to indicate for the user that the potential each discussed technology can reach in capacity, availability etc, i.e. in range of reachable performance indicators, and, only just a few technologies which are not dedicated for mass market are described more in detail.

It is also important to remind that the parameters reachable by technology solution do not have to match with the parameters which are offered by the providers. Namely the mobile wireless operators, but not only those, are concentrated on their core business with defined performance. ITS solutions requirements are typically above average requirements and it might be quite a frequent problem to find provider of services portfolio matching ITS services requirements.

### 3.1 Basic telecommunications solutions properties

Telecommunication solution represents application of one or more telecommunication services set up to meet requirements of the customer telecommunication task. Particular telecommunication services are provided based on telecommunication networks application. Correctly designed network architecture, appropriate implementation processes of the technical solution, flexible implementation of services and network management tools, centrally processed pro-active network monitoring, network and services day to day configuration and management are necessary conditions for offering specific portfolio of services as well as potential for services growth both in quantity and quality. Such system provides conditions for dynamic reactivity on demands both for communication system as a whole as well as for particular fragmental communication solutions which the system consists of. The structure of processes has classic pyramidal structure of the system management (Fig 5.1.1). The provider is capable, under certain technical conditions, to provide service of a defined quality class (Quality/Class of Service – QoS/CoS), i.e. to guarantee set of agreed service parameters (see Chapter 3.3) within an agreed interval. Agreed conditions are included in an agreement (contract) on provisioning of telecommunication services usually known as SLA (Service Level Agreement).

Cost of provided telecommunication solutions is an important measurable and it is usually limiting a faster development in applications areas. Safe and effective sharing of telecommunications networks capacity not only by more applications but also by more legal subjects represent one of the most effective approaches how to reduce the telecommunications services cost. Ability to guarantee agreed security level of each particular application represents the key issue of such approach (see Chapter 3.3).

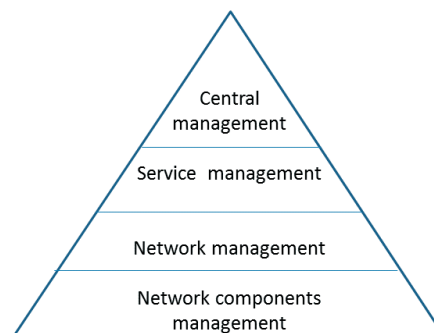


Figure 3.2 System of networks and services

### 3.2 Types of network according to topology

According to the nodes arrangement and their mutual connection network topology three principal network arrangements can be recognized:

- mesh – the mesh arrangement can be perceived e.g. in an analogy of sport networks, which does not mean unconditional connection of each node with each one but a mutual communication of all the nodes,
- ring – all nodes are connected into an endless ring,
- star – all terminating nodes are interconnected with central node enabling mutual interconnect of any node to any other one and it also (exclusively) interconnects this set of nodes with the other networks.

In most telecommunication networks there is a hierarchic structure which enables a combination of particular network typologies with different hierarchies considering the fact that a combination of different topologies can be combined even on the same hierarchy.

### 3.3 Types of network according to the hierarchy within a network:

Regarding the range of covering and relevant capacity on a relevant hierarchic level, the telecommunication networks are usually divided into several levels:

- Access network providing connection of a final customer
- Backbone networks on several levels - Metropolitan Access Network – MAN and
- National and international WAN (Wide Area Network) for a signal distribution in a hierarchic structure from one final customer to another one that does not have to be on the same hierarchy (for example connection of a final customer to a www server) indispensably.

The example of a typical typology of a network is in the following Fig. 5.3.1., in which application of various typologies of network on different and same levels of the hierarchy of a network can be seen.

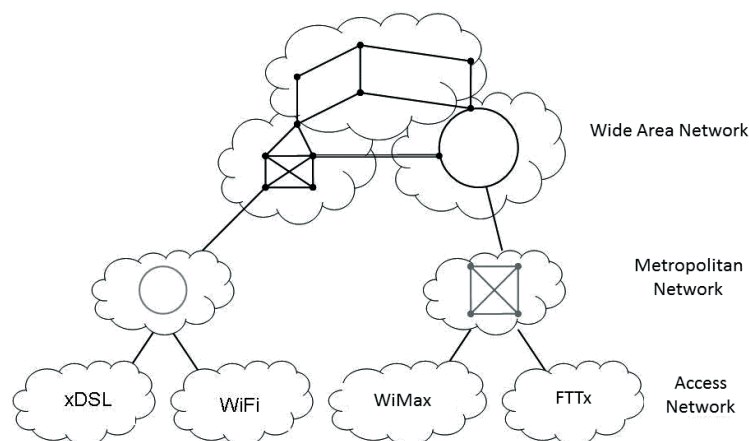


Figure 3.3 Demonstration example of hierarchic architecture of a telecommunication network

### 3.4 Types of networks according to the used physical layer:

#### 3.4.1 Metallic

Metallic/copper wired networks represent historic basis of a physical layer. They have gone through many stages of development and nowadays they are used only in two basic arrangements:

- UTP (Unshielded Twist Pair) and STP (Shielded Twist Pair) shielded either as a whole cable or as each pair (combination of both alternatives is an alternative as well); they are divided into categories (CAT 1 to CAT 7) and used from basic telephone services to full video or 10G Ethernet (mostly only in B2B – back to back application on a few meters distance). As an illustration of the development within this branch Table 3.1 introduces basic parameters of different TP categories,
- Coaxial cables – today spread particularly as access VHF lines of cable TVs with a possibility to use the transmission capacity for providing other services like access for the Internet services (step by step replaced even in this area by TP). The coaxial cables have, however, vanished from data networks. Originally the physical layer of the Ethernet networks was made of a coaxial cable. The same situation was in the backbone networks of the past years. Originally such long-distance lines including undersea cables were based on coaxial cables.

**Tab. 3.1 Twisted Pair (TP) categories**

<b>Category</b>	<b>Capacity</b>	<b>Application</b>
<b>CAT 1</b>	144kb/s, 24Mbps (2MHz)	<b>2B&amp;D (ISDN), xDSL 2+</b>
<b>CAT 2</b>	4 Mbps	<b>IBM Token Ring</b>
<b>CAT 3</b>	16Mbps	<b>10BASE-T Ethernet</b>
<b>CAT 4</b>	20Mbps	<b>e.g.16Mbps Token Ring (IBM)</b>
<b>CAT 5</b>	100Mbps, 1000Mbps (4TP)	<b>155 Mbps ATM – replaced by CAT 5E</b>
<b>CAT 5E</b>	1000Mbps	<b>155Mbps ATM, 1000Mb/s Ethernet</b>
<b>CAT 6</b>	10000Mb/s	<b>1000Mb/s - new installations</b>
<b>CAT 6E</b>	10Gb/s	<b>10GBASE-T Ethernet – short dist.</b>
<b>CAT 7</b>	<b>Banwidth to 700MHz</b>	<b>Full analog video</b>

### 3.4.2 Fiber

Fiber appeared firstly as a physical layer of the high-capacity transmission (Backbones), however, more and more it has been appearing as a physical environment of access networks (FTTH – Fiber To The Home ETA.), too. Its transmission capacity is significantly higher comparing metallic line (using DWDM up to orders of Tb/s). Remarkable advantage of fiber applications is no sensitivity on different and variable ground potential as well as on the lightening effects. Optical conductors are of two basic categories:

- Multi mode optical fiber – substantially cheaper alternative (fiber diameter 50 – 100 micrometers) with possibility to carry signal only within a short or medium distance (units of km at maximum),
- Single mode optical fiber – originally much more expensive alternative (diameter of the fiber up to 10 micrometers (8.3 – 10)) with a need of precise source of light of a narrow

### 3.4.3 Radio frequencies solutions

The physical layer of a radio solution is an electromagnetic field transmitted from a transmitter to a receiver. During the duplex communication each side has a transmitter and receiver and their work in the following modes:

- Simplex
- Half duplex
- Duplex

The electromagnetic field is divided into bands from 3 kHz to 300 GHz. The physical layer is created also on different wave lengths, for example IrDA on the wave length in the vicinity of a light spectrum (infrared) laser connection etc.

## 3.5 Circuit and packet oriented telecommunication systems

The telecommunication area is still even more and more only formally split into circuit and packet switched alternatives:

- Circuit switched communication systems - traditional telecommunication systems operating on a circuits principle that have, for example in voice systems, reached not only mass service coverage but also a high quality and reliability and
- Data packet based communication systems – data are organized in packets / frames / cells. Each such “unit” includes heading carrying originated and final destination address as minimum.

Historical development of this area is locked with national telecommunications monopolies and transparently defined standardization processes under the control of ITU-T (set up processes of the United

Nations Organization). On the contrary the TCP/IP communication “category” has been developing on and evolution principles with no central authority engaged in management of this process. Basic IP principles do not support any service quality management. For the applications of higher demands it is always necessary to apply a corresponding support

- on L2 layer of a network interface e.g. via Ethernet,
- on L3 layer for example by protocol of MPLS type
- on L4 layer for example by application TCP and UDP or
- on the application layer.

Their mutual combination is possible as well so that this communication system could meet the assignment and concrete demands on the basis of which a concrete telecommunication service is run.

### 3.6 TCP/IP architecture

#### 3.6.1 TCP – Transport Control Protocol

TCP and alternative UDP – User Datagram Protocol - have in their competence particularly (see e.g. [4]):

- transmission of data which are collected and distributed according to the type of application (via virtual ports).
- data construction and vice versa.
- management of service reliability – TCP only – next packet is not transmitted until the acceptance at final destination of a previously sent packet is confirmed to transmitting node. The simplification (for example comparing X.25) lies in the fact that only transmitting and receiving node are directly engaged in delivery management process. Positive aspects of such approach are simplicity and thus low costs of such solution. However, such approach does not resolve problem where it origins but “heals” the problem as a whole with a frequent impact on transmission speed which is not apparently a priority within this attitude and can be frequently principal issue.
- UDP does not support packet delivery management, which is the main difference comparing TCP.

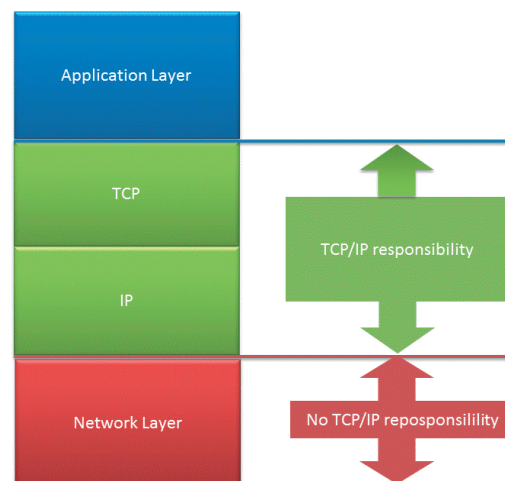


Figure 3.4 TCP/IP model

#### 3.6.2 IP – Internet Protocol

IP is responsible for routing of a packet dynamically routed from a router to another one. IP protocol (L3) is defined (see e.g. [5]) by these properties:

- not “reliable” in delivery protocol

- not circuit-switched” protocol
- routes packets that do not have any equivalent in a circuit switched systems,
- the only routing protocol of the TCP/IP model,
- signal transmission is transferred to responsibility of the Network Interface Layer (NIL) - link and physical layer of the RM OSI, i.e. L2 – the interface with this L2 is precisely defined and the application of a broad spectrum of NIL protocols is enabled, (ATM, Frame Relay, Ethernet etc.),
- IP support is implemented
  - in host computers
  - in routers
- nowadays version IPv4 with 32 bits IP addresses is dominantly used but already for quite a few years the gradual implementation of IPv6 version with 128 bits addresses has been carried out with remarkable extension of protocol properties improving between others principally its reliability and safety.

The properties of IP protocol, which can be summed up into the following items, are based on the introduced competences:

- it is universal,
- it offers unified transmission services,
- it creates universal inter-operability environment,
- it does not use specifics of physical transmission technologies – it only requires a “common minimum”,
- it tries to solve difference of users by creating unified environment for all the applications,
- it is focused on simplicity, efficiency and speed,
- it supports variable size of a packet – the size of a packet is set by a sender (application) – it can happen and it also often happens that – regarding a limited size of the second layer (L2) – the packet is fragmented into more frames/cells which is not however a fatal problem although it can impact the dynamics of transmission services
- it is not circuit oriented – it guarantees neither order nor time of delivery,
- it operates only in the “best effort” mode,
- it does not guarantee delivery of a packet
- it does not guarantee that the data content is not damaged or modified,
- it does not support any packet flow control,
- it “can” exclude a packet from transmission (throw away), if
  - wrong check total is identified,
  - the service life of a packet is exceeded,
  - node is overloaded unable to process packet,
- if a packet is excluded, node sends a report on it to the packet sender without its delivery guarantee,
- For information on non-standard situations needs the IP protocol contains ICMP which is its integral part and which has to be implemented compulsorily. The receiver of ICMP reports is the packet sender. ICMP packets delivery is not confirmed,
- The volume of a data part is variable with a maximum length of 64 Kbytes (65535 bytes)
- The minimum supported volume: 576 bytes, i.e. 512 bytes of data, the rest is running costs. At such volume of a packet fragmentation of a packet on L2 carried out by Ethernet protocol is not necessary – of course it is different in the case of ATM (see the following chapter).

### 3.6.3 IP addressing

In the most spread version IPv4 the address is a 32-bit binary number. A unified way of record is used: the content of each byte is expressed as a decimal number, particular parts are connected with dot, for example 193.84.57.34. IP V6 address is a 128-bit binary number. On the contrary to IPv4 it is written as a hexadecimal number, groups of four separated by colons. In a group of four naught is left out and if one or more groups of four are only made of naught, the presence of naught is identified by separating colon. The example of a V6 IP address: *4ef5:ffff:1::ba9/64* where the number after the slash sets the length of a so called prefix, that is number of bits of the address from the left that identifies the affiliation to the relevant network.

### 3.6.4 Routing

The routing is a choice of a direction in every router and a delivery of each packet on the basis of permanently revived routing information. The routing involves besides its own power function of delivering packets also preservation of routing information that is keeping routing tables, number of optimum paths which is a combinatory problem of searching for the shortest path in a theory of graphs. The result is “source materials for a choice of a direction”. Router itself keeps distribution of addresses into classes – FEC (Forwarding Equivalence Class) and each class uses the same path but only to the following router.

Each router tests approachability of its neighbors, i.e. the interface stage (including the link and the following node interface), it puts together a “link state packet” on approachability of neighbors – link state and its evaluation and it sends such information to all the nodes within a network.

Within IP routing each router analyses each IP header and according to its content it determines the next hop direction of a packet. This method of routing is demanding with respect to the need to process the content of each IP packet.

### 3.6.5 AS – Autonomous systems

Big volumes of transmitted routing information are solved through the division of the entire network in autonomous systems (AS). AS does not spread detailed routing information but it only provides information on approachability of particular networks which are within the relevant AS. Each autonomous system has a certain (small) number of input/output points for interconnection with other autonomous systems. Through these points the information on approachability (on its content) is exchanged and the mutual existence is tested, too.

The exchange of information on approachability, existence, establishment of mutual interconnection etc. must run among the autonomous systems. Appropriate protocols are necessary for this. Commonly applied BGP 4 supports general interconnection of autonomous systems and it enables to set various criteria when choosing among alternative directions. The AS administrator sets directions priorities, for example, according to the speed, capacity of links, reliability etc. Within Autonomous systems each AS can deal with routing as it suits itself. It can do its own routing policy and the way of update of routing information. There are available different alternative IGP (Interior Gateway Protocol) which can be used for updating routing information. It could be e.g. historical RIP (Routing Information Protocol) protocols, which operated according to the principle of a “vector distance” suitable for small networks, or e.g. well-known OSPF (Open Shortest Path First) IGP protocol. However, new much more effective and complex alternatives are available and applied, in present solutions.

OSPF is an open version of an older SPF (Shortest Path First) protocol. It was designed and its specifications are open to the public. Each node tests approachability of its neighbors (link state). Each node sends periodically link state packets to all the nodes or immediately when there is a data change. All nodes within the network have entire information on particular connections and they can compute optimum paths. Each node computes for itself, if any error occurs it only influences the relevant node. It enables to define various paths for various modes of operation and it supports a load balancing of a network. It also supports subsequent division of a network into smaller areas which are analogous to autonomous systems in the point that their typology does not exceed the relevant area. This way it decreases volumes of distributed updating information.

### 3.6.6 Summary of TCP/IP properties

The basic idea of TCP/IP lies in the fact that the transmission mechanisms should primarily transmit data and they should not be loaded with other functions. The transmitted data are not protected. On L3 data are neither encrypted nor protected. If any application needs some rate of security implementation on application layer or alternatively solution on L3 (MPLS) or on L2 (e.g. ATM, Ethernet etc.) must be provided.

The consequence of such approach is that transmission infrastructure is simpler, faster and less expensive comparing secured state. The security is to be solved on an application level particularly through authentication, encoding, tunneling, separating by security gates etc. MPLS represent alternative very complex approach and it can be implemented either on L3 or L2. Remarkable number of security tools is available with protocols applied on L2, as well. All these approaches to improve system parameters represent so additional cost, which can be higher than the basic network solution. However, it is installed only in case there is customer willing to pay for such quality of service.

For a transmission of control systems in real time it is necessary to transmit the data with a small and regular delay, i.e. with regular intervals between particular received packets which the TCP application cannot guarantee. The control of the Quality of Service (QoS) is basically the contrary of the maximum effort principle. The quality parameter can be improved by the following measures:

- Quantitative, i.e. on a basis of increasing available capacity. The principle of a maximum effort continues and the improvement is only statistical which means there is a lower probability of a decrease in requirements and it is applicable (with important technological and economic limits) at both transmission capacities and routing/switching capacities,
- Qualitative: implementation of the QoS support – an operation according to the principle of a maximum effort is replaced by different way of operation. The improvement is guaranteed but it is more expensive and more difficult. The following tools can be used:
- Using preferences – the packets/frames are assigned different preferences and they are treated differently, the transmissions of a higher preference get services of a higher quality in a form of allocation of sources at the expense of the transmissions of a lower preference. The solution examples are:
- Reservation – a property of TCP/IP which is not used very much – for the purpose of concrete services it is possible to reserve required sources and then to use them – it concerns reservation of transmission capacity, switching capacity etc. RSVP (Resource Reservation Protocol) and the following transport protocol RTP (Real Time Protocol that operates over UDP protocol is the example of the solution using the tools of L2 layer (Network Layer) as PBT, L3/L2 etc.
- MPLS (Multi Protocol Label Switching) carried out with the tools of the third or second layer (MPLS/L3, MPLS/ATM etc.),

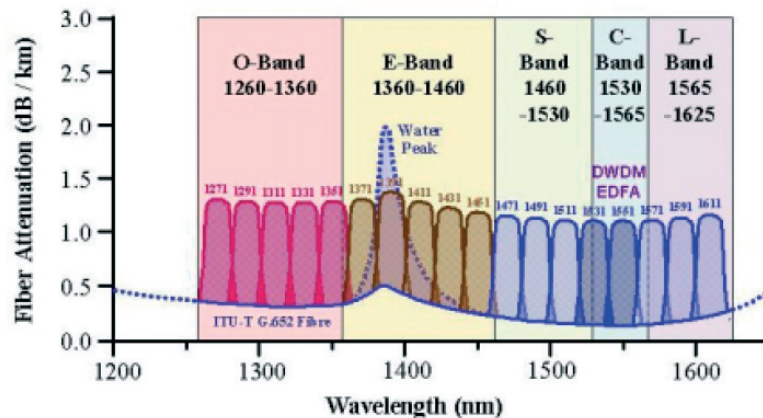
## **4 Backbone networks**

Backbone networks and appropriate backbone services are dedicated to cover wide areas. As already mentioned such services are usually provided by public provider and such services are typically designed for “average” customer requirements. ITS services requirements on services quality are frequently much above that average. We will provide reader with basic description of mostly installed and applied technologies. Some of these “historical” technologies provide remarkable selection of parameters hardly reachable if the up to date ones are applied, however, with no exception in remarkable value/cost ratio difference. These are namely SDH/SONET and ATM solutions developed with afford to be positioned as so called “carrier grade” technology (i.e. with redundant architecture and very attractive time to recover in case of any node/link failure - MTTR).

New backbone technologies lost their “carrier grade” character with consequences on their backbone systems performance. In ITS applications exists potential of fatal consequences if service performance (described by Service Level Agreement - SLA) does not meet required level. That is also reason why concentration of effective management of backbone services quality is getting remarkable position again – see e.g. PBT technology described in chapter 5.5.3.3.

### **4.1 Basic description of WDM**

Wavelength-division multiplexing WDM (Wavelength Division Multiplex) enables ability to share the capacity of fiber for more individual channels distinguished by the different wave lengths. Each WDM channel is totally independent of the other channels and such approach can be understood as the capacity or protocol-transparent multiplex. Total transmission capacity may reach with DWDM (Dense WDM) systems up to 10 Tb/s, i.e. 3 orders more than the transmission capacity SDH/SONET (10 GB/s for STM64) and 100 times more if compared with CWDM (Course WDM).

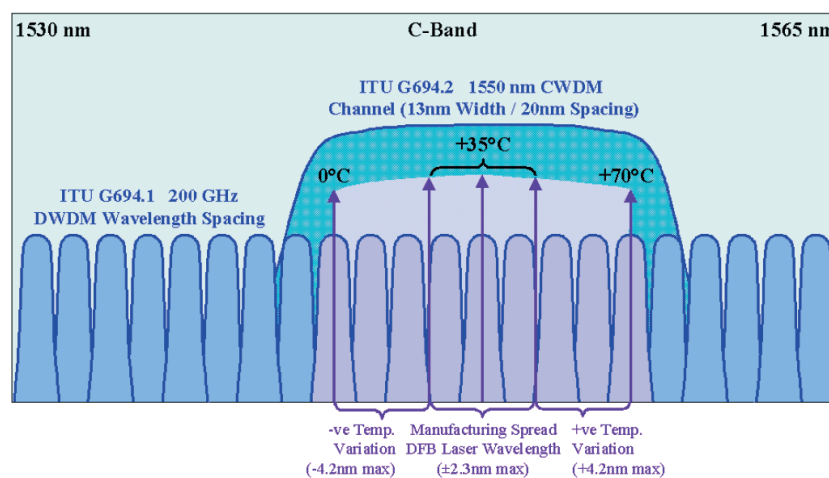


**Figure 4.1 ITU G.694.2 CWDM and DWDM (C-band)**

Basic DWDM channel bandwidth is 200GHz (1,6nm), however bandwidths of the only 50 or 100GHz are frequently used, as well. Total number of supported channel depends, of course, on each channel bandwidth. DWDM capacity can be also enlarged by extension of applied spectra to zones C in S and L. DWDM systems apply exclusively thermally stabilized LASER light signal source nowadays usually integrated on one silicone substrate. Its thermal stabilization ensures adequate thermal stability of the wavelength – see Figure 4.2. Receivers are based on different types of PIN broadband diodes and selectivity of the receiving party is given the characteristics of thermally stabilized optical filters.

CWDM channel bandwidth is 20nm in accordance to ITU standard G.694.2 – 1. Up to 16 channels are available. However C-band would be omitted for later DWDM application and in case of older fibers application remarkable part of E-band would be excluded, as well. We can anticipate that typically up to eight channels are useable with CWDM and this statement is proved by extended number of practical installations.

Figure 4.1 describes CWDM channels bandwidth distribution in accordance to standard ITU G.694.2. C-band is dedicated for DWDM applications, if it is relevant issue. Different requirements on the production tolerances and temperature stability of CWDM in zone C (1530 – 1565nm in accordance to G. 694.2) and DWDM requirements are shown on Figure 4.2. CWDM technology was designed for relatively cheap telecommunications broadband solutions with limited transmission capacity. DWDM has been accepted as robust high capacity carrier grade WDM alternative. This assessment corresponded in past with the ratio of the cost of LASER- transmitters and optical filters designed for CWDM and DWDM. Thanks to the progressive technological development cost of the CWDM and DWDM LASER transmitters, but also optical thermally stabilized filters, optical amplifiers and other system elements were principally reduced and trend of gradual price convergence of the various components is being identifiable.



**Figure 4.2 CWDM and 200GHz channel bandwidth of DWDM.**

Another reason of the original perception of DWDM vs. CWDM was based on application of originally very costly broad spectrum optical amplifiers exclusively applied in DWDM architecture. Application of optical

amplifiers eliminates need for repeated regeneration of each channel, i.e. conversion to the electrical signal and back to the optical one. Such approach is adopted in CWDM architecture and it copies in fact SDH/SONET regenerator/multiplex idea. Results of the financial modeling based on recent pricing (Y 2009) are available below and results anticipate very careful financial evaluation of both alternatives before the right one is chosen. On Figure 4.3 and Figure 4.4 are presented examples of CWDM DVDM configurations applied in mentioned financial model.

#### 4.1.1 DWDM

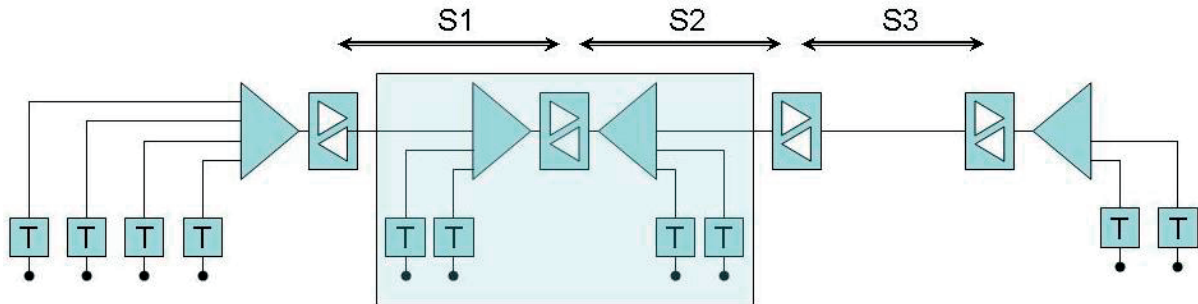


Figure 4.3 Tree segments of the DWDM chain with I/O possibility on each segment borders.

In DWDM architecture signal enters system via transponder T which ensures conversion of the input signal to optical signal of required by system wavelength. Subsequently, each channel is connected to the OADM (Optical Add/Drop Multiplexer) and output of the OADM is amplified by the OAMP (Optical Amplifier). After pass through selected number of segments channel is filtered on the relevant OADM and subsequently it is via transponder T converted to appropriate format to be connected to the end user. OADM units are implemented on boundary between segments where signals drop in/out are required.

Distance between two OAMPs depends on OADM capacity and its parameters, quality of fiber and parameters of OAMP and served distance can easily be more than 100km. Regeneration of signal is obligatory after degradation of optical signal below acceptable level which is not resolvable by the optical amplifiers. Number of “OAMP” hops before signal must be regenerated depends on parameters of all parts and number of simultaneous channels transferred via fiber.

#### 4.1.2 CWDM

In CWDM arrangement signal enters system via input transponder T to be transformed to corresponding wavelength and it is amplified on appropriate level. Then it is connected to the backbone fiber via OADM (Optical Add/Drop Multiplexer). At the end of fiber segment each channel is separated and it is either delivered to end use via transponder T or regenerated to appropriate parameters (optical/electrical/optical conversion) by transponder T and connected to further OADM. After n-segments processing channel is terminated and signal is subsequently dropped out by the OADM and connected via transponder T to the client in requested format. Interface unit located in connected Ethernet switch can act as transponder T if all required by CWDM system parameters are fulfilled. Such interface units are offered in wide price variability in dependence on vendor (quality/policy/competition). These units cover whole by ITU G 694.2 defined spectra range and required optical parameters.

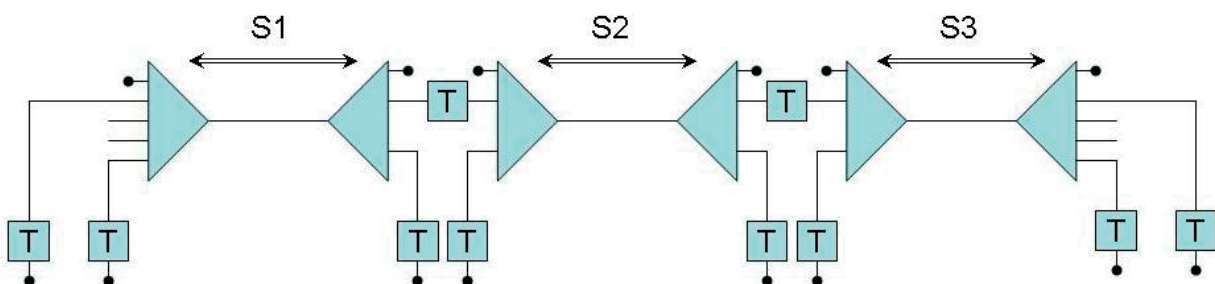


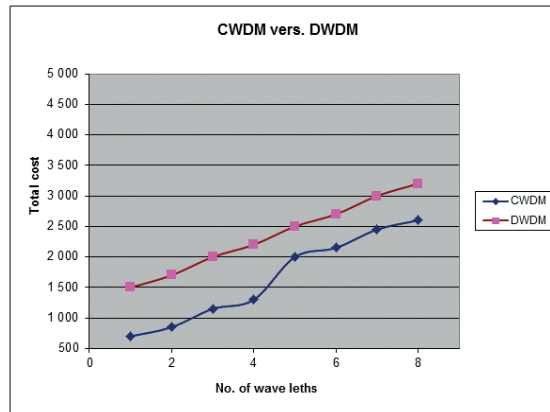
Figure 4.4 Tree segments of CWDM chain with I/O possibility on each segment borders

### 4.1.3 CWDM vs. DWDM

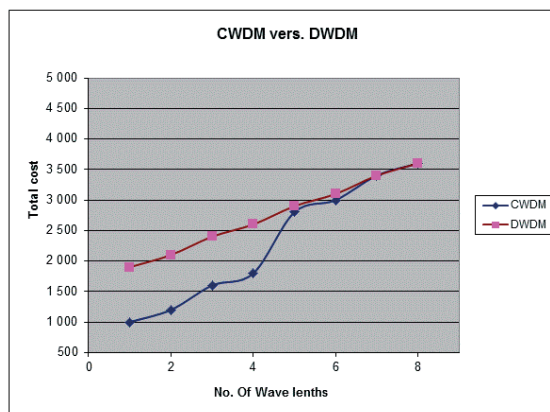
The analysis of the conditions of the WDM solution is derived from real prices around the year of 2008. The used values are always relative and can be only used for comparison purposes. The computation was carried out for two, three and four segments. The computation considers necessary number of transponders, optical add/drop OADM units or optical OAMP amplifiers if necessary. The results are displayed graphically in Figure 4.5 - Figure 4.7 for a two-, three- and four-segment solutions, considering that the investment costs are only of comparison value

**Tab. 4.1 CWDM a DWDM parameters selection**

	CWDM	DWDM
<b>Number of Channels</b>	Up to 18 (typically 8)	<b>256, ...</b>
<b>Spectra</b>	band O,E,S,C,L	<b>C, and add. L,S</b>
<b>Frequency range</b>	2500 GHz	<b>50 –200GHz</b>
<b>Fiber capacity</b>	40 Gb/s	<b>nTb/s</b>
<b>Distance</b>	hundreds of km	<b>Practically unlimited</b>
<b>OAMP</b>	-	<b>EDFA etc,</b>



**Figure 4.5 CAPEX - 2 segments of both CWDM and DWDM solution**



**Figure 4.6 CAPEX - 3 segments of both CWDM and DWDM solution**

The analysis considers eight wave lengths per one CWDM solution at maximum. This limit is derived from experience with physical properties of available fibers that were installed before, the effort to let C band free for possible DWDM application and economic evaluation of known CWDM solutions. As an important result we consider particularly the graphical expression of the concurrence of DWDM and CWDM costs in Figure 4.1, which proves the fact that under certain conditions CWDM may be more expensive than DWDM application within the framework of competent solution. Of course not all variables are included but in the submitted

analysis the used parameters represent parameters of dominant influence on studied connections. A higher flexibility of CWDM for connecting or disconnecting new customers on a route is a typical example of a group that does not have direct economic indexes though it can have an important impact on chosen solution.

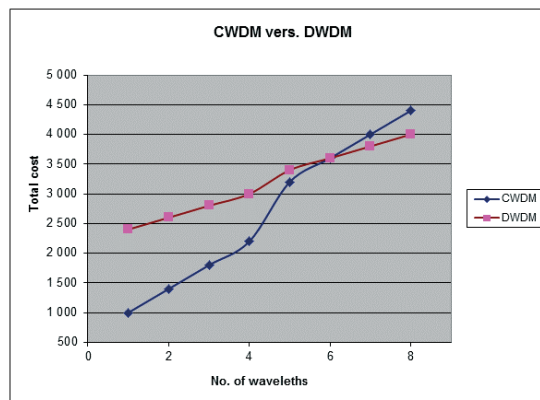


Figure 4.7 CAPEX - 4 segments of both CWDM and DWDM solution

#### 4.1.4 WDM - conclusions

At four-segment the CDMA solution may be less advantageous in purchase costs than the DWDM application. Such results are not for sure an extraordinary surprise but they may catch a user unaware if marketing analysis were not properly worked out or evaluated and there is an increase in capacity over five CWDM channels.

The presented results cannot be simply generalized and it is necessary to consider many other aspects as a higher flexibility and lower costs of CWDM when connecting customers on a route, an availability of transponders just in customers' units etc. The goal of this chapter was to show possible conditions in prices of WDM solutions that may bring unexpected, particularly economic, surprises.

The above mentioned properties of both systems lead to their mutual combination in many networks, where DWDM usually makes backbone for longer distances and "supplementary" CDWM may bring a solution, interesting from the economic standpoint, of an access of particular customers on a rout.

## 4.2 SDH/SONET

The first standard of synchronous optical networks was SONET (Synchronous Optical Network). This standard was developed and adopted particularly in North America. ITU standardized the modified system for European conditions under the name SDH (Synchronous Digital Hierarchy) by the standard ITU-T G.707, which was based on European definition of hierarchy PDH (interface E1 – E3) and from the level STM1 SDH is fully interoperable with the SONET standard. More details are available e.g. in [39].

SDH/SONET are systems only for application on an optical fibre with the transmission capacity limit of 10 Gb/s (STM64/OC192). The basis of systems is a very precious synchronization based on Celsius clock with a phase synchronization on the GPS system. The reference clock ( $10^{-14}$ ) has a regional range of effectiveness and the conditions in what extent it is possible to use one synchronicity source are set. The networks SONET/SDH can be unambiguously put into category of global telecommunication systems thanks to transparent standardization and mutual compatibility and it has been a backbone of global networks used for voice services up until now.

### 4.2.1 PDH

PDH (Plesidochronous Digital Hierarchy) is based on bit multiplex principle with additional support of limited difference of the reference oscillators of each node (nodes are not centrally synchronized like in SDH/SONET systems) by "stuffing" - see ITU-T G.745. Stuffing is based on redundant bits application which in accordance to situation are or are not used for information transfer. Number of stuffing bits represents the tolerance range of cooperating nodes reference oscillators frequency difference.

European/American PDH different multiplex levels bit rates are listed in Tab. 4.2. PDH interface is still alive for SDH interface which is still applied for remarkable number of highly demanding ITS applications not speaking about dominant users, i.e. both fixed and mobile digital voice services.

Tab. 4.2 SDH/SONET PDH interfaces

Technology	SDH				PDH			
Interface	E1	E2	E3	E4	DS-1/T1	DS-2/T2	DS-3/T3	DS-4
kbps	2,048	8,448	34,368	139,264	1,544	6,132	44,736	139,264

#### 4.2.2 SDH architecture

The multiplex SDH scheme is totally different from PDH. On the contrary to PDH the information is mapped into virtual blocs, the so called containers and this way it is possible to get information of a lower order from the information flow of a higher order without “demultiplexing” the flow in its full width because the transmitted information is unambiguously identified by its time position in the time scheme of the containers. The basic function structure is in Figure 4.8.

Basic signal structure mapping is provided by containers CXX into which are stored data from individual interfaces. Other levels are so called virtual containers VCX generated by adding the path header – POH (Path Overhead). TU (Tributary Unit) is generated by adding of the pointer PTR (Pointer) to the virtual container. PTR is mapping container time position. TUG (the Group) represents combination of VCX and administrative units (AUG) are subsequently added directly to the synchronous transport modules.

From the diagram Figure 4.8 it is clear that the STM1 can carry up to  $3 \times 7 \times 3 = 63$  E1 signal, or 3 E3/DS3 (T3). SDH can concatenate containers to increase transmission capacity (STM4, 16, 64), by simple n-times increasing bit rate of the STM1 frame.

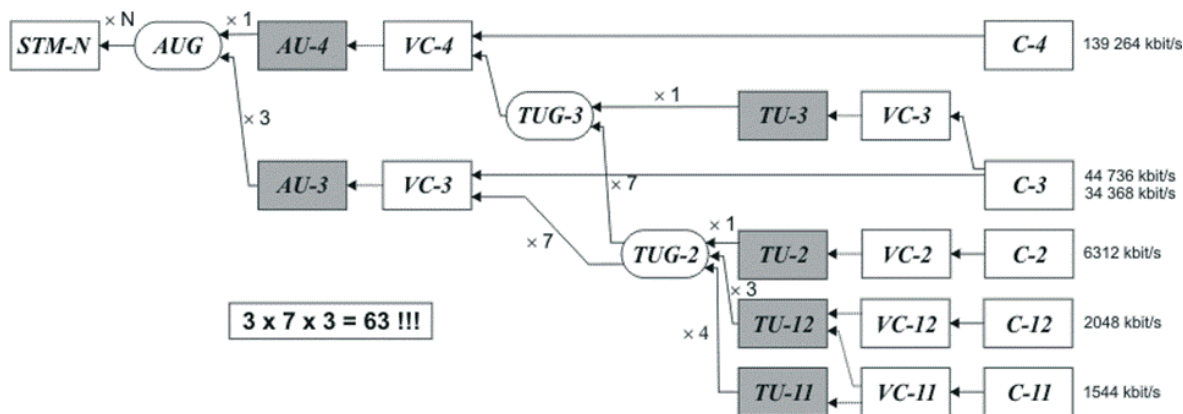


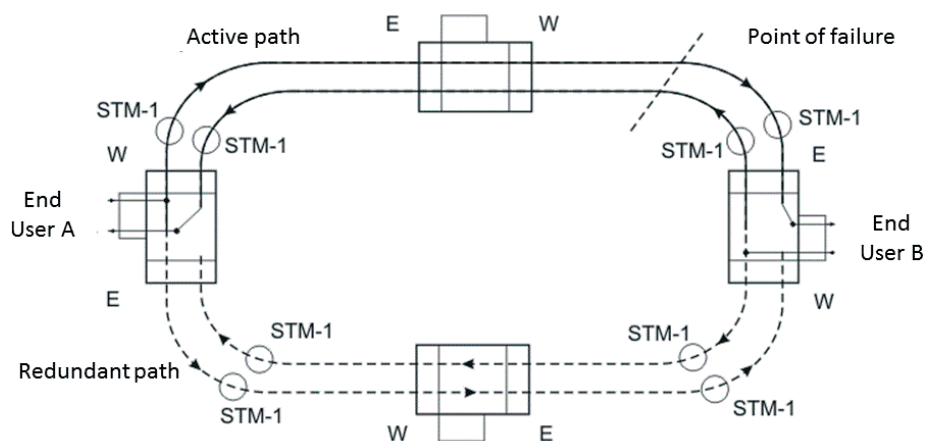
Figure 4.8 SDH system principles

SDH/SONET architecture is still far from data services packet arrangement, however, construction of TU and AUG indicates packet principles accepted in parallel – packet/frame heading caring information about data which are to be delivered to end user.

#### 4.2.3 SDH network architecture

The SDH network consists both of multiplexors and then regenerators which only contain necessary parts of multiplexor that ensure regeneration of signal on a route. The regenerators are used where the access to transmission capacity is not required. Most SDH producers enable to extend a used regenerator to a multiplexor in case of need by supplementing relevant modules.

On Figure 4.9 is displayed the typical SDH network ring topology arrangement. This arrangement offers 100% redundancy. SDH node A transmits signal to both directions and addressed node C in the case of any problem switches input from current direction to the alternative one and it is fully recovered in time period shorter than 50ms. SDH technology was for many years leader in time of recovery together with later coming ATM. Nowadays “Ethernet” based solutions like “HYPER ring” or PBT are comparable or even better in time to recover. Due to its high reliability and fast recovery times SDH has still been used in many sensitive industrial applications.



**Figure 4.9 SDH ring topology**

## 4.3 The ATM

The original idea of the ATM protocol authors was to improve the transmission in ISDN networks. So it is not surprise that ATM was originally known as B-ISDN (Broadband ISDN). However, ATM soon left its mother's area and it has been accepted by data networks family. ATM can be considered as network solution supporting both circuit-oriented solutions and packet data services. The system is not synchronized from view of individual cells (small frames) transmission. However, the transmission on bits level is on a synchronous principle with continuous flow of cells structures on the backbone, where the individual cells are either occupied or vacant.

### 4.3.1 ATM Architecture

ATM principle of universality (voice and data) is based on application of very small cells (53B frames with 48B data and 5B header). ATM cell size (and technology available in these times) led designers to hardware solutions of cells switching engines with a result of high switching capacity namely in the context of then available alternatives. The negative result of this approach is evident in high cost of these systems acting as killing factor of this very advantageous technology.

Data transmission follows the virtually created paths (VP). Each service is related to one virtual channel identified by a number – VCI (Virtual Channel Identifier). ATM network management is ready in the case of any problem on the virtual path to resolve the issue the reconfiguration/replacement issue of the problematic path. The same principle is applied in case of any local network overload.

### 4.3.2 Protocols of ATM Adaption Layer

- AAL1 – bit transparent service (circuit emulation) provided as CBR – Constant Bit Rate service),
- AAL2 – supports rtVBR – Variable Bit Rate for real time applications,
- AAL3/4/5 – supports nrtVBR – Variable Bit Rate and ABR – Available Bit Rate,
- LANE – LAN Ethernet emulation incl. support of broadcast and multicast,
- MPOA – Multiprotocol over ATM – supports wide range of protocols.

### 4.3.3 ATM performance

The great advantage of ATM is the ability to guarantee the various provided services parameters – QoS (Quality of Service). It is ready to support the wide range of real-time services ATM parameters are based on the size of the cell – the data capacity of 48B with a header just 5B. Overhead is kept on amazing the only 11%. The size of the cells is connected with high frequency of cells exchange with high probability that critical cell will be delivered in time.

Cell header contains between others also the information about cell priority (only one bit is standardized, however it was extended in some proprietary implementations to 3 bits – like with NORTEL) used

for determination of the order of the cells to be accepted for delivery. This is basis for ability to guarantee wide range of quality of services. Therefore ATM is able to guarantee even regular delivery of data, of course under the condition of fair and correct management of the network.

ATM protocol is significantly more flexible than SDH. Like SDH ATM provides the same reliability and security for sharing fiber capacity. Principal problem preventing faster ATM penetration was in its both CAPEX (Capital EXpenses as well as in OPEX (Operational EX) parameters. ATM can be considered as the reference telecommunications system solution between others in terms of offered services quality, reliability and safety, services management and rapid response on the failure or local overload in the network (MTTR) comparable with a solution based on SDH/SONET technology.

#### **4.4 ETHERNET – IEEE 802.3 and ISO 8802**

At the end 1970 decade was designed and implemented collision protocol to interconnect mini-computers with transmission rates of 2.9 Mb/s and the effective utilization on approx. 30% of the transmission capacity authors definitely did not anticipate that they are building the fundamentals for protocol of 21<sup>st</sup> century. For multiple collision access to bus method based on ALOHA principles was adopted. Now it is known as CSMA (Carrier Sense Multiple Access). Thanks to the activities of companies Digital Equipment, Intel and Xerox (the owner of the Ethernet® registered mark) and later IEEE Ethernet principles were accepted as a basis for standard IEEE 802.3. An access protocol has been extended by CD (Collision Detection) principles. So the full name of standard IEEE 802.3 is „Carrier Sense Multiple Access with Collision Detection Access Method and Physical Layer Specification“. Since the creation of the Standard it has been in permanent development and from the original default transmission speed 10 Mb/s 802 raised to 100Gb/s with the additional future expectations to grow).

IEEE 802.3 firstly won the battle to become the leader of LAN L2 TCP/IP networking. Implementation of the optical interface in combination with the non-collision switching principles (IEEE 802,1d integrated later in 802,1q) this has started its penetration into the other areas including global WANs. The basic concept for the standard remained unchanged, but the tools for the management of the quality of service, creating the VLAN tools etc. created to this standard possibility to significantly exceed the original boundaries of its original application. This was step by step processed without bombastic publicity, but its success generated an extraordinarily interesting economic consequences.

##### **4.4.1 IEEE 802.3 and IEEE 802.1q standards - alternative telecommunication networking**

The IEEE standards 802.3 and 802,1q and their corresponding implementation let "Ethernet" grow from its original LAN applications to the whole spectrum of networking – MAN and national/global WAN. The basic characteristics of this family of protocols are:

- 802.3x represents a wide range of standards in terms of transmission speed, support of different physical layers, interconnection capability in a real time (RT) with low delay. Its architecture enables interconnect of slow 10 MB/s PC client with fast 100 Gb/s port of the central servers without any additional network support,
- Although standard 802.3 is 100% compliant with CSMA/CD collision protocol 802.1q (former 802.1d) standard implemented no collision solution based on circuits switching where MAC address of the final destination port is used for a switching decision processes. This solution is termed as „Switched Ethernet“. This approach enables simultaneous application of network with no potential of mutual collision,
- If two switches interconnect line is overloaded, any new frame being above limited transfer capacity is lost. Frames heading optionally include priority level identifier (with extended heading only) applied by the switch to decide which frames with advantageous priority will be preferentially delivered and which of them shall be lost. Standard 802.1q includes between others also the 802.1p priority processing tools standard. It describes also a system of queuing. Each queue (FIFO) is filled with frames of the same priority and frames to be delivered are selected from different queues in accordance to the situation on relevant interface. Such system enables the simple and transparent implementation of priority management, which represents necessary condition for QoS management implementation to guarantee e.g. delivery time as well as delivery frames jitter,
- Standard 802 .1q (Virtual Bridged Local Area Networks) between others also defines extended header information (see Figure 4.10):

- TCI – type of a frame,
- P – priority level (0–7),
- C – Canonical ID – identification MAC address type,
- VID – VLAN ID – Virtual LAN identification,

P	DA	SA	TCI	P	C	VID	T/L	DATA	FCS
8B	6B	6B	2B	3b	1b	12b	2B	...	4B

TCI *Tag Control Info type of frame is indicated*

P *Priority 0 -7*

C *Canonical ID (MAC address type*

VID *VLAN ID*

**Figure 4.10 Extended Ethernet Header**

- A later versions of the standard IEEE 802.1q adopted former standard 802.1v (VLAN Classification, Protocol and Port) and 802.1s STP (Spanning Trees Protocol). These tools enable networking with ring topology construction. Their implementation enables conversion of multi-ring topology of tree topology as well as enables reconstruction of the network in a case of any line or node failures. STP was not well accepted due to serious problems with its stability and convergence times. 802.1q later adopted RTSP (Rapid Spanning-Tree Protocol – originally IEEE 802.1w) improving STP functionality and stability. This protocol relies on the active ongoing communication between active elements with principal improvement of recovery times,
- Protocols GARP (Generic Attribute Registration Protocol), GVRP (GARP VLAN Registration Protocol) and GMRP (GARP Multicast Registration Protocol) enable dynamic declaration and propagation of the optional attributes and their values. Protocol GARP is a tool for declaration and network distribution of attributes and their values. With the support of the protocols GVRP and GARP are dedicated to declare network terminal point membership to private network and via GMPR group membership of the terminal point is declared.

Thanks to the dominant position of „Ethernet“ on local LANs switches and the other network elements production reached mass volumes and very competitive conditions on market. Its impact on market has been similar to that what has happened in the area of PCs. Such situation has got significant impact on remarkable decrease of the final network elements prices. Low prices do not mean in such situation critical difference in system elements as well as whole network parameters quality. Just opposite situation is identified and „Ethernet“ based networks solutions are “in principles” getting closer and closer to the solution based on the ATM protocol.

#### **4.4.2 „Ethernet“ and its applicability in WAN**

Historical development of a performance of family of “Ethernet” protocols, i.e. the performance of the solutions based on standards IEEE 802.3 and 802.1q, shows that this concept step by step has created the condition for its penetration on WAN level. Reachable node to node distance range of more than 100 km predetermined this technology application range for both national as well as global networking.

“Ethernet” concept has been a subject of many controversies. Commercial success of already operated solution on all network levels (up to global WAN) significantly reduces strength of core arguments presented by its originally very strong opposition. It is still clear that this technology represents “less sophisticated” solution if compared with complex ones like MPLS or IPv6 based networks. Already available managerial tools usually sufficiently support both configuration and management of the network. Principal improvement of managerial tools cause an additional cost customer has to pay for. However, price/performance ratio still remains as the key advantageous value of this approach. Another remarkable step in „Ethernet“ area is represented by PBT (Provider Backbone Transport) solution (see 5.5.3.3), which is able to offer parameters comparable e.g. in network convergence with SDH/SONET or ATM systems.

### 4.4.3 Convergence times reduction

#### 4.4.3.1 STP/RSTP

WAN and MAN backbone solutions based on "Ethernet" accepted multi-ring topology. This is similar to an approach applied in SDH systems. Due to the appearance of broadcast and multicast frames, typical for "Ethernet", it is necessary to virtually open every ring and create tree topology arrangement of the network. For this purpose was developed and accepted STP (Spanning Tree Protocol) known as unstable and in convergence slow tool. The problems STP based systems stability and convergence times were relevantly resolved by implementation of RTSP (Rapid Spanning Tree Protocol) adopted in IEEE 802.1q standard. Convergence times achieved in networks equipped with approximately one hundred nodes sub-second values, of course under condition of correct network configuration.

Convergence time is definitely one of the critical parameters for ITS. If acceptable (sub-second recovery times) solution can be based on ring topology supported by RSTP. "HYPER" ring" (5.5.3.2) type of approach can offer substantial convergence acceleration.

IPv6 implementation is the other way how to improve IP networks performance e.g. in QoS management, VPN support etc. Expected IPv6 massive penetration, however, has not happened, and, nowadays most of installed solutions are implemented either on L2 with RSTP support or via MPLS either on L3 or L2 (ATM). High expectation are linked with below described (5.5.3.3) PBT technology installed on the „Ethernet“ networks. Such solution can also meet even highly demanding requirements. Nevertheless, we can identify that the specific requirement are frequently resolved by L3/SDH, L3/ATM and MPLS/ATM with remarkable impact the solution cost.

#### 4.4.3.2 HYPER-ring

Major acceleration of convergence was firstly introduced by company Hirschman as "HYPER" ring. This solution is based on the competence transfer to the individual rings local management units (switch) which are dynamically cooperating with their neighbors. Network convergence time based "HYPER" ring is comparable with convergence times of SDH/SONET. This solution represents the important result for relevant selection of solutions of "Ethernet" networks with critical requirements on convergence times, i.e. between others also some ITS applications (e.g. such solution was applied at Prague airport for time critical applications). The idea of original HYPER ring approach has been implemented in the various modifications by majority of world manufacturers of the "Ethernet" technologies.

HYPER-ring strategy is based on RM (Redundancy Manager) installed in one of nodes on each ring. RM node decides which node on ring disconnects the ring. Tree structure it locally created by mechanisms accepted from RTSP (Rapid Spanning Tree Protocol). The ring disconnect is valid for all data frames except of the Watchdog frames. These frames periodically test integrity of the ring and localize the potential failure. For minimizing of the data loss during failure localization and following network recovery in this time period delivered frames are stored in the stack to be transferred after recovery.

The described solution guarantees recovery time in tens of ms. Presented architecture and implemented SW does not guarantee the "only" recovery times, but provides also the effective administration and maintenance of the network.

#### 4.4.3.3 The PBT - Provider Backbone Transport

High expectation can be identified from the new technologies based on simple extension of native and widely spread Ethernet technology. This new approach named as Provider Backbone Transport (PBT) is based on PBB (Provider Backbone Bridging). Basis for PBB represents standard IEEE 802.1ah known as Mac-in-Mac tunneling. MAC tunneling principally improves limitations if compared with solution using Q in Q principles, i.e. solution standardized by IEEE 802.1ad. PBB approach reduces maintenance requirements and enhances security by clear separation of the customer and provider addressing space. PBB Ethernet extension does not require multiple complex control protocols and it does not require any extension of the existing hardware, as well.

PBT provides TDM-like connection management characteristics into the traditionally connectionless Ethernet area. PBT approach creates point-to-point Ethernet „tunnels“ based on specification of the paths related traffic data will be carried via Ethernet environments. Each path is equipped with the ability to reserve appropriate bandwidth so that PBT supports QoS (Quality of Service) metrics that guarantees SLA provisioning regime. Carrier Ethernet Operations, Administration, and Maintenance (OAM) standards (IEEE 802.1ag) can

be applied to provide fault notifications in milliseconds so that „carrier grade“ failover times can be achieved. PBT guarantees MTTR even much better than 50 ms. It means that PBT telecommunications system is comparable or better if compared with SONET/ SDH networks.

PBT Ethernet operates in a connection-oriented model. Conventional learning behavior of Ethernet switches is replaced by provisioning of the paths across the network based on VLAN ID as a virtual circuit identifier, similar to other connection-oriented technologies like Frame Relay and ATM or MPLS. For that reason flooding of broadcast, multicast and unknown unicast traffic are not applicable. Such approach enables representative network scaling to the sizes necessary in core networks. (R)STP is replaced by the management systems of the service provider similar to those available with MPLS or ATM models. Both connectionless (like Ethernet), or connection-oriented (like MPLS) alternatives can be provided.

These two standards offer a simple, cost effective “Ethernet” based evolution strategy that is easy to implement scale and manage. Additionally service and tunnel layer approach is similar to other technologies. If the tunnel or transport layer is “abstracted” from the service layer, PBT can be used to deliver not only Ethernet services, but MPLS based services or more generically in fact any type of traffic - voice, video or data can be delivered.

Recently several major advancements in Ethernet OAM (Operations, Administration, and Maintenance) have been recently archived. OAM has become an important differentiator between different Ethernet service providers - especially for customers expecting OAM functionality similar to that they obtain from TDM or circuit based networks. The following Ethernet OAM standards represent tools for operators to provide effective management of their Ethernet infrastructure:

- *Fault management and performance monitoring (ITU-T Y.1731)* - performance monitoring measurements like frame loss ratio, frame delay and frame delay variation to assist with SLA assurance and capacity planning. For fault management the standard defines continuity checks, loopbacks, link trace and alarm suppression (AIS, RDI) for effective fault detection, verification, isolation and notification in carrier networks,
- *Connectivity fault management (IEEE 802.1ag)* - Defines standardized continuity checks, loopbacks and link trace for fault management capabilities in enterprise and carrier networks. This standard also partitions the network into 8 hierarchical administrative domains,
- *Ethernet in the first mile (IEEE 802.3 ah)* - monitoring and troubleshooting Ethernet access links. It defines tools for discovery, remote failure indication remote and local loopbacks and status and it allows performance monitoring.

#### **4.4.4 Backbone solutions based on „Ethernet“**

The family of switched “Ethernet” solutions, i.e. based on standards IEEE 802.3x, 802.1d a 802.1q represent unique L2 communications system applicable and applied both for local as well as wide area alternatives. Due to reasonable performance/cost ratio, availability of service quality management and selection convergence treatment provided in different categories „Ethernet“ solutions have quickly penetrated in area of public services. However, also ITS specific requirements can be met and it is not surprise that such solutions were already implemented.

#### **4.5 IP Virtual Private Networks (VPN) solution**

Increasing needs of private solutions provided on public networks can be clearly identified. Particular IP VPN (Virtual Private Network) solutions that safely and reliably share communication network means by multiple users represent hot topic caused primarily by economic pressure.

Service users do not always make the rational decisions because they are not able to separate the important properties described by complex performance parameters from those of the second order. The second order parameters represent e.g. actual technology trends, similar applications implemented by either partners or competitors, incorrectly understood marketing presentations of producers/suppliers etc. This part might help decision makers to improve their decision abilities to obtain solution with relevant parameters and appropriate cost.

L2 VPN implementations are due to limited network tools usually significantly different if compared with traditional VPN/IPv6 or MPLS VPN models. L2 VPN is typically identified by list of MAC addresses accepted for each closed group – i.e. VLAN (Virtual LAN). The certain level of priority is assigned to such group of

MAC VLAN based on IEEE 802.11q standard tools. By assigning certain priority to each VLAN not only privacy but also QoS is assigned to each group. If a customer requires the combination of various QoS provider usually creates more VLANs with different priority level. Priority selection is locked with adequate VLAN selection based on connect to the relevant MAC address.

#### 4.5.1 MPLS IP VPN

IP VPN can be efficiently implemented in MPLS based networks. MPLS eliminates most of the peer-to-peer IP model problems. Although IP network uses OSPF, RIP, EIGRP or other protocols, MPLS VPN solution requires BGP protocol capabilities implemented in IBGP which includes specific features needed for VPN support. The input node of the MPLS network converts protocol used by customer to BGP. The IP address of the customer is extended by RD-Route Distinguisher. Parameter RT (Route Target) carries the information about the original user protocol BGP and it will be converted back at the output node. Routing is then managed by MPLS with BGP protocol support.

At the MPLS output node each packet is delivered in a standard form with the relevant IP address. MPLS VPN is able to ensure the required level of separation of individual subscribers, and, it allows resolving of any IP address conflicts. MPLS allows also the efficient management of network capacity exploration (traffic engineering), i.e. optimizing utilization of the widest possible nodes and trunks of the network. This MPLS features represent the major advantages of MPLS. The unique achieved parameters, however, caused significant increase of the solution complexity connected with higher cost of service. Another critical issue for ITS applications is represented by long convergence of MPLS VPN networks.

#### 4.5.2 MPLS IP VPN vs. L2 VPN

MPLS VPN is a comprehensive and very robust tool for the construction, maintenance and optimization of IP VPN solution. Its important limitation for ITS can be recognized in extremely high MTTR parameter. It must be also pointed out that such robust solution caused relatively high purchase, operation and maintenance cost.

Effective application of combination of protocols, IEEE 802.3 and 802.11q (RTSP is included, now) can provide much faster convergence times than it can be reached with MPLS/L3 solution. If RSTP protocol is applied network convergence with more than one hundred nodes **use to be** below 1s. Convergence can reach tens of ms e.g. in case of „Hyper ring“ and it can be even shorter in a case PBT approach is applied. The remarkable advantage of „Ethernet“ based solution is the total cost (purchasing, operational and maintenance).

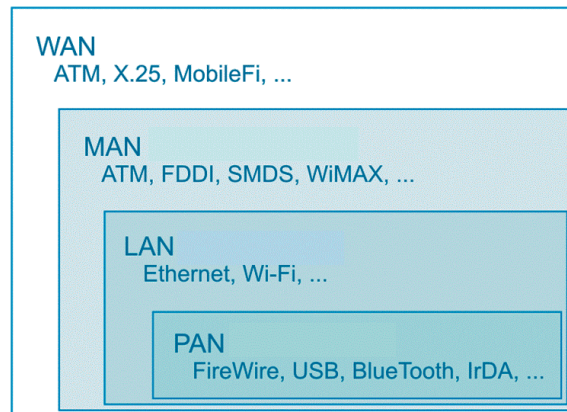
However, MPLS networks management tools and engineering support are remarkably above that what Ethernet based solutions can offer. Final decision so always remains on network designer who must balance cost and system parameter with required service performance parameters.

## 5 Access wireless mobile solutions

The access wireless telecommunications solutions are one of the key components of the telematic systems. These systems apply service of the second generation of GSM (Global System for Mobile communications networks) such as DTMF (Dual-Tone Multi Frequency), CSD (Circuit Switched Data), HSCSD (High Speed CSD), SMS (Short Message Service), USSD (Unstructured Supplementary Service Data), UUS (User to User Signaling). 2.5<sup>th</sup> generation of GSM offers packet data service GPRS (General Packet Radio Service) and EDGE (Enhanced Data rates for Gsm Evolution). The third generation of mobile networks UMTS (Universal Mobile Tele-communications System) has been growing, however, availability of UMTS services are usually concentrated to the cities. "Beyond the 3-rd generation." represented by the LTE (Long Term Evolution) technology is coming with user's high expectations not excluding the ITS solutions.

In ITS systems DSRC (Dedicated Short Range Communications) systems are used in 5.8GHz and 5.9GHz alternatives or IrDA (Infrared Digital Access) solutions. However, the standard group 802 of IEEE represents the most important part of ITS telecommunications solutions. The well-known group of standards legally incorrectly labeled as „Ethernet" (registered officially by company Xerox) is the only one representative of this 802 working group standards applicable in WMAN, WLAN and WPAN (Wireless MAN, LAN, PAN). These standards can cover wireless solutions in urban, local, and personal networks and they can be understood as mostly private alternatives to public mobile services:

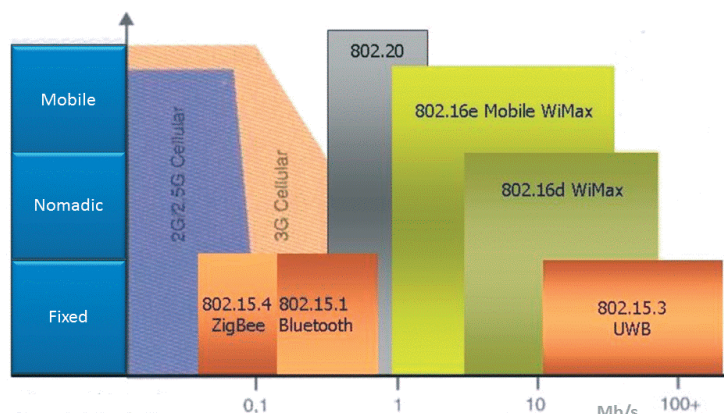
- IEEE 802.11 is on a way to gain important role in ITS solution due to QoS management option (Amendment 802.11e) and high velocity tolerance reached by Amendment 802.11p - accepted as MAC layer of the WAVE/DSRC 5.9,
- IEEE 802.16d is usually accepted as WMAN solution, and IEEE 802.16e represents the mobile solution with wide area coverage. The approach to WiMax technical solution is a very progressive and some its key principles will be most probably adopted in the other coming soon systems,
- IEEE 802.15.1 Bluetooth (P2P WPAN), 802.15.3 UWB (57-64 GHz bandwidth, up to 2Gb/s bit rate P2P WPAN,), 802.15.4 ZigBee (small areas (max 50m P2P) Low Rate “ad hock” networking).



**Figure 5.1 Hierarchy of different services coverage**

Figure 5.1 describes hierarchy of service area coverage (with the relevant technologies mentioned) and Figure 5.2 Data access technologies - mobility and bit rate capacity (Mb/s)

bandwidth and mobility positioning of the wide range of mobile access technologies. Such comparison can have the only informative value due to the fact that public massive solutions can be hardly precisely compared with the dedicated private alternatives. This comparison is provides exclusively from the user's point of view.



**Figure 5.2 Data access technologies - mobility and bit rate capacity (Mb/s)**

The applications of satellite communications systems are quite rare and they are mostly applied if any other services are not available. Most of the satellite projects with goal to offer mobile public service based on LEO (low orbit) and MEO (medium orbit) satellite system collapsed. On the contrary, the oldest system based on GEO (geo-stationary) satellites Inmarsat originally dedicated for maritime transport has remained as unique communication service used in some specific applications including air transport communication etc.

Most of above mentioned access wireless systems can be combined in multi-path communication solution in ITS frequently connected with family of standards CALM, where 2<sup>nd</sup> generation of handover was firstly explicitly defined.

## **5.1 Mobile telecommunications services**

The mobile services operators in Europe operating dominantly GSM based system play key role in wireless public telecommunications. Mobile services description is available in wide range of literature e.g. like [43] and there is no reason why to repeat all of that what was written. We will focus only on the key parameters of the individual services which have direct impact on ITS solutions. GSM providers offer a wide portfolio of services where voice still has kept its key position as the core mobile service. Public providers offer services with clear concentration to fulfill requirements primarily for voice services. The guarantee of the data solution quality is not mostly provided.

The alternative solution of different robustness are available, however, namely signal coverage is mostly limited just due to the fact, that services are not publically provided. On the other hand such services can be operated with guaranteed QoS. WiMax (IEEE 802.16), WiFi (IEEE 802.11 with QoS and mobility amendments) are typical representatives of such services and their very progressive technical principles will penetrate most probably not directly but via their “mutations” recognized transparently or in hidden form in coming solutions like LTE technology or DSRC 5.9/WAVE with high potential to be massively applied in future ITS applications.

### **5.1.1 DTMF (Dual Tone Multiple Frequency)**

Circuit setup belongs to basic functionalities of both digital and analog voice networks. In voice analog terminal DTMF modulation is used to deliver digital information describing selected final destination number (address) as well as the other user requirements on functions represented by other than number characters. DTMF system uses eight different frequencies signals transmitted always in pair defined sequences. Each pair represents one of sixteen different numbers, symbols and letters. The signal is transmitted 50ms and after each pair transmission must follow at least 50ms without a signal. DTMF capacity is 10character/s. In past DTMF was applied for more different applications. Due to unavailability of IBS (In-Band Software modem) standard DTMF was also applied in e-Call (automatic emergency call) pilot project in the Czech Republic to deliver information packet from terminal to e-call center via voice channel, i.e. as is required by e-call standard. The volume of delivered data was due to very low transfer speed shortened to the only 29 bytes. This pilot project was successfully finalized and in final solution DTMF is logically replaced by IBS.

### **5.1.2 CSD (Circuit Switched Data)**

CSD technology is based on voice channel utilization. Data are not delivered based on modem technology known from terrestrial voice networks, but via integration of the data bit flow into GSM data flow schema. Basic supported bit rate is 9.6 kb/s and if redundant channel coding is reduced capacity can reach 14.4 k/sec per time slot (TS).

### **5.1.3 HSCSD (High Speed Circuit Switched Data)**

HSCSD is so-called high-speed circuit switched data services based on the same principle like CSD. To increase the bit rate is based on multiple time slots application. In configuration 3 + 1 43.2Kbps download/14.4 kb/s upload bit rate is provided.

### **5.1.4 IBS (Inband Software modem)**

The European service e-call was standardized as a service operated exclusively via (virtual) voice circuit. IBS converts the digital data on audio tones to be transferred via GSM voice channel without any damage or content loss. However, frequencies and reached bit rates are close to DTMF, i.e. it is much slower than CSD or ordinary modems applied on metallic lines. Due to specificity of the GSM audio codec the particular audio tones are selected in a way that the compressing process can have minimal impact on transferred data. IBS modem SW based solution, which can be linked to the existing software of both terminals and network nodes (typically PBX). Programmable functional implementation is used in both the existing mobile network nodes and terminal equipment, as well as in the fixed network communication platform, i.e. typically PBX. The term PBX SW therefore represents the PBX software installed in powerful distribution nodes of the e-Call center.

The principles of the IBS modem are based on the properties of GSM voice codec equipped with adaptive filters which automatically set necessary bandwidth in accordance to voice signal properties. If codec identifies voice signal with remarkably limited frequency spectrum then GSM adaptive codec filter significantly reduces

channel bandwidth. Such situation can e.g. occur if longer sequence of the identical bits is transferred and consecutively bit value modification appears. IBS packet is protected against such sequence impact. To avoid the influence of adaptive filters in cellular codec packet header contains random or alternating bit sequence. The header is set to cover sequences which are similar or the same as those contained in the payload of the packet. Therefore IBS encoder analyzes the sequence of bits and IBS can subsequently place relevant subset or sequence of the bits in the header section. It is so guaranteed that the adaptive filters do not filter tones, which represent the transmitted digital data.

#### **5.1.5 SMS (Short Message Service)**

SMS is a service originally provided in European GSM networks later being spread globally. SMS is applicable for specific data transfer. Voice channels are not utilized, but transmission is based on utilization of signaling system network spare capacity. SMS message offers maximum of 160 characters for each message. Messages are centrally routed through the SMS Centre, which shall forward message to the end user if there is available transmission capacity signaling system and at the same time the customer terminal is available on the network. Therefore, service delivery is not guaranteed and it can be delivered with significant transport delay. Uncertain delivery represents critical parameter for many data applications. Benefit of this service can be, on the other hand, identified in possibility of automatic notification of message delivery as well as in its cost.

#### **5.1.6 GPRS (General Packet Radio Service)**

GPRS fundamentally differs from the (HS)CSD data service. It is a packet-based service. The bit rate of the service depends on the encoding scheme, which is in GSM automatically selected from the four available options according to the intensity of the electromagnetic field signal received from the transmitter. Data bandwidth also depends on the number of simultaneously used time slots (TS). For IP communications 160 kb/s can be theoretically reached, however, the realistic bit rate use to be 40 kb/s, if 5 of the 8 TSs are allowed to be used for GPRS service. GPRS is typically operated in the configuration 4 + 1 (4 for download and 1 for upload). The transmission is faced to relatively large delays reaching easily 500ms and even more.

#### **5.1.7 EDGE (Enhanced Data Rates for GSM Evolution)**

EDGE is based on the GPRS principles. The main difference is in the modulation upgrade and in potential number of TS (Time Slots) dedicated to the service. Instead of GMSK used with GPRS in case of EDGE phase-modulation 8-PSK is applied. In case of EDGE system management releases for EDGE service all 8 TSs, then maximal theoretical speed 473,6 Kbps can be reached. With 4 released TSs can be achieved the bit rate of 236,8 Kbps. The realistic bandwidth is around 200 kb/s for download and 100 kb/s for upload, if 3 + 2 TSs are released. An important difference compared to GPRS is that technology EDGE implementation is locked with all base stations radio upgrade as well as every used terminal must be EDGE compliant.

#### **5.1.8 CDMA (Code Division Multiple Access)**

CDMA 2000 Technology is applicable for relatively wide frequency bandwidth (1.25 MHz with CDMA 2000). Each traffic signal is encoded by unique “pseudo-random” digital code and it is so spread across the whole spectrum of the transmission channel. This widely known principle provides a single channel of communication for many users at the same time. 40% higher spectra utilization is identified, if it is compared with the traditional TDM/FDM (Time/Frequency Division Multiplex) systems like GSM: This advantage is paid by higher system complexity and more demanding procedures of the quality of service management.

CDMA 2000 technology includes following standards: CDMA 2000 1xRTT standards includes (1 x Radio Transmission Technology), CDMA 2000 1xEV-DO (1 x Evolution Data Optimized) and CDMA 2000 1xEV-DV (1 x Evolution Data/Voice).

The standard RTT (Radio Transmission Technology) is the oldest version and it offers the double the capacity for the transmission of voice and data offers a maximum speed of 614 kb/s. Standard EV-DO (Evolution Data/Voice) is focused purely on the data transfers, the voice service on this technology can be implemented only through a VoIP (Voice over IP). Data bit rate can reach 2.4Mbps. The realistic bit rate 500-700Kbps with approx. 200ms delay. The third standard EV-DV provides a maximum speed of 3.1 Mb/s for download and 1.8 Mb/s for upload. At the same time it supports the simultaneous use of both voice and data services. CDMA 450 is based on CDMA EV-DO and allows the use of this technology on the 450 MHz bandwidth of former analog NMT (Nordic Mobile Phone) provided e.g. in the Czech Republic.

### 5.1.9 UMTS (Universal Mobile Telecommunication System)

UMTS uses W-CDMA (Wideband Code Division Multiple Access) on 5MHz channels which are used in FDD (Frequency Division Duplex) regime. This solution, known as UMTS FDD, applies some provides (e.g. Telefonica) for provisioning of packet data services combined with the voice services based on switching circuits. Alternative solution is in TD-CDMA (Time Division CDMA) that uses TDD (Time Division Duplex). UMTS TDD is an alternative choice (e.g. T-Mobile) providing only the packet data services. Frequency spectrum of the UMTS in Europe consists of one two-zones (1920–1980MHz + 2110–2170MHz) and one unpaired (1910-1920MHz + 2010-2025 MHz). Three other bands are available soon, but the only one will be applicable in Europe (2500-2690MHz).

UMTS exists in several versions. The older versions support the relatively small bit rate – the first version of UMTS R99 (R3) in practice reached a speed of 120/50 kb/s, Today's in version of R5 are referred more frequently by one of the technologies that this standard includes - HSDPA (High Speed Downlink Packet Access). With this version it is possible to achieve speeds of up to 14.4Mbps. HSDPA currently exists in 4 versions. Framework of the R5 is an HSDPA Phase I in the versions of SoC I (speed up to 1.8Mb/s) or SoC II (speed up to 3.6 Mb/s) is available in Europe.

R6 was already standardized and HSDPA Phase II in versions of SoC I (speed up to 7.2 Mb/s) and SoC II (speed up to 14.4 Mbps) are available. Version R6 reduces delays of transmission to less than 50 ms (see ITS requirements!!). Speed increase is achieved by adjusting the radio interface by HSUPA (High Speed Uplink Packet Access).

Generally, UMTS implementation is behind expectations and services are focused only on major urban areas. Future attention is more and more concentrated towards the new technology LTE.

### 5.1.10 LTE

Mobile system LTE (Long Term Evolution) is considered to be the next step in mobile wireless communications and it was introduced in the framework of the third generation partnership project - the 3rd Generation Partnership Project (3GPP) Release 8. 3GPP agreement represents the ARIB, CCSA, ETSI, ATIS, TTA, and TTC, in which are involved more than 60 operators, producers, technology and research institutes.

3GPP defines IP "area" (flat) architecture, which is defined in the framework of the activities of the SAE (System Architecture Effort). LTE-SAE architecture and concept are designed with mass IP wireless market. LTE was built on the GSM/WCDMA solution with maximal simplification and discounting operation and implementation if compared with the previous "evolutionary" stage. Additionally the recently initiated cooperation between 3GPP and 3GPP2 (CDMA standardization group) with the aim optimizes the interconnectivity between CDMA and LTE-SAE. Effort is to ensure that operators of CDMA networks were able to economically develop their networks towards LTE-SAE. The standardization process of LTE was initiated within the 3GPP RAN, held in Toronto in November 2004. The objectives of the project can be summarized as follows:

- Reduction of prices per transmitted bit – more services at a lower price,
- Flexible use of existing and new frequency bands,
- Simplified system architecture and open interfaces,
- Acceptable energy consumption terminals.

#### 5.1.10.1 Architecture

LTE has got the flat architecture of SAE (System Architecture Evolution). This architecture was designed for optimizing the network properties of the solution, its prices and simplification were designed for the public service. The existing 3GPP (GSM and WCDMA/HSPA) and 3GPP2 (CDMA2000 1xRTT, EV-DO) systems are integrated through the standard interface in order to optimize mobility with LTE. The server HSS (Home Subscriber Server) is not connected on the core network via SS7 signaling, as it is the case of GSM, but via application of the protocol Diameter – AAA (Authentication, Authorization and Accounting), which is frequently used for access to the network IP or mobile IP solutions. HSS system consists entirely of "flat" IP solutions used in land division networks. Existing GSM, CDMA and WCDMA/HSPA systems are integrated using the standardized interface. LTE-SAE adopted quality management system in the form of quality classes (Class based QoS) is a welcomed feature of wide area covering public services for ITS applications.

#### 5.1.10.2 OFDM technology

For a downlink, i.e. communication from the base station to the terminals, LTE applies OFDM (Orthogonal Frequency-Division Multiplexing), which allows flexibility of available spectrum use and it allows high-capacity reasonably priced solution – OFDM is applied also by WiFi, WiMax, DVB, etc. OFDM uses a large number of narrow bands of with 15 kHz step and QPSK, 16QAM or alternatives to 64QAM amplitude modulation alternatives are used. The individual zones are organized in the source blocks, which are composed of 12 sub-zones, i.e. 180kHz and time T slot duration of the is 0.5ms. TTI (Transmission Time Interval) consists of two T slots. The user is allocated to certain number of the source blocks, which represent actually defined space in time-frequency grid of the two-dimensional structure. With increasing number of dedicated source blocks and number of levels of the amplitude modulation is increasing transmission capacity dedicated to a specific user. Mechanisms how to dedicate capacity to the individual users is based on the principles applied in HSPA.

For uplink, i.e. from the terminal to the base station, uses the LTE SC OFDM. This approach compensated the disadvantage of OFDM based on high coefficient of PAPR (Peak it Average Power Ratio). The high values of PAPR lead to relatively costly power amplifiers with the high requirements on linearity. These requirements cause increase of the terminals prices and high battery capacity requirements. SC-FDMA reduces the energy of PAPR value and so the limited energy requirements and linearity issue improve service coverage as well as handover processes can be smoother.

#### 5.1.10.3 „Advanced antennas“

The „Advanced multi-antenna“ has already been applied in eHSPA (evolved High Speed Packet Access) and it is used in LTE with a view to compliance with the requirements of mobile high-capacity solution with large permanent as well as extreme peak transmission capacity and large service area coverage. "Advanced multi-antenna" consists of multiple solutions, which are applied according to the set of scenarios. For example, high transmission capacity leads to multilayer antenna solution 2 x 2 or 4 x 4 MIMO (Multiple Input Multiple Output) where service coverage by formation of the beams is significantly extended.

LTE uses both FDD (Frequency-Division Duplexing) and TDD (Time-Division Duplexing). FDD can be considered as rather efficient approach, but there is at least a good opportunity to use TDD in unused secondary zones with an important economic impact. LTE is designed to support the bands from 5 MHz (or even less) up to 20 MHz in a wide range of frequencies and modes of both FDD and TDD, which offer a high flexibility for both existing operators and new provider. Not only the "classic" mobile terminal users form the customers target group, but also the laptops, gaming systems, applications such as video cameras, as well as fixed terminals (FWT – Fixed Wireless Terminals). The links between the capacity of wireless and terrestrial transmission solutions and potential difference of price/performance ratios could be interesting as well.

#### 5.1.10.4 LTE summary

LTE is well-designed solution for public services and it is mostly understood as the next generation of mobile networks. The category of 4th generation is, however, for basic principles discontinuity replaced by the “beyond 3rd generation“. LTE concept allows the users to combine services of existing providers with services of newly coming operators. LTE attacks the mass market with promises of high transmission capacity in both directions and very low latency. This technology offers flexible use of the bands in the range from approximately 5 MHz after 20 MHz in a combination of TDD and FDD. The LTE-SAE architecture reduces the nodes number and supports the high flexibility of the network configuration and high availability services. LTE-SAE transparently resolves the interoperability with GSM, WCDMA/HSPA, TD-SCDMA and CDMA. LTE offers QoS management tools which area is a very important parameter for telematic systems network industry with the requirement of mobility of telecommunications services.

Potential transmission capacity leads to enthusiasm that today mostly only fixed networks applications will be available to realize via LTE. But we would like to point it out as frequently unrealistic "optimism" generated namely by manufacturers of wireless technologies. We call for the realistic comparison with capacity of optical and metallic networks as well as the prices we are obliged to pay for these different services.

## 5.2 WiFi – IEEE 802.11

WiFi standard belongs to typical „surprises“ in the telecommunications evolution. WiFi was by market originally accepted as „low end“ technology in the same way like was understood Ethernet in comparison with its “older brothers”. Due to its massive application WiFi technology was produced as „mass“ product with remarkable influence of components pricing. Authors of this standard [47], however, step by step extended system parameters by, for telematics very important, Amendments like IEEE 802.11e/r/p principally improving WiFi systems performance possibilities. We will so only shortly describe basic parameters of core standards IEEE 802.11a/b/g and we will introduce also mentioned Amendments which significantly move potential of WiFi position on the “professional” market.

### 5.2.1 MAC (Media Access Control) layer of WiFi networks

IEEE 802.11 standard defines two methods of access to media – DCF (Distributed Coordination Function) and PCF (Point Coordination Function).

DCF (Distributed Coordination Function) and PCF (Point Coordination Function). The basis of all three standards (802.11a, b, g) consists of the DCF method, which is based on CSMA/CA (Carrier Sense Multiple Access) combined with CA (Collision Avoidance) method approach preventing collisions and resolving issue in a case collision appears. Optionally PCF based on RTS/CTS (Request To Send/Clear To Send) might be applied – the only one in time is confirmed by the central section (CTS) and thus is prevented from others in the broadcast. Then the process is confirmed by ACK frame to minimize the frames/packets loss.

The key weakness of DCF method if applied in telematic system represents the absence of QoS management. DCF cannot provide service in a way that selected client "gets right" of exclusive access to shared media. On the other hand there is no set time limit one client communication procedure must be finalized. This fact generated unequal conditions and it can cause degradation of service quality for majority of clients.

The second method can be used only if PCF infrastructure is implemented - this feature is optional and is quite rarely implemented. Transmitted beacon frames at fixed intervals (usually 0,1 s) specify parameters of the PCF. The time between two beacons is divided into two periods. CP (Contention Period) represents the DCF period. In the second period capacity is distributed to clients with appointed right to transmit. This time interval is referred as CFP (Contention Free Period). One client is active and the other clients are prohibited from attempting to broadcast. This is suitable for real-time applications. Unfortunately, PCF is not sufficiently supported and this solution has considerable limits e.g. due to absence of the appointed frames/packets priority. Most of PCF problems were later resolved by amendment IEEE 802.11e.

### 5.2.2 IEEE 802.11a

Standard 802.11a was approved already in 1999. It is designed for 5GHz unlicensed band, specifically the band 5,470-5,725 GHz, where 11 of 20MHz channels are available. Transmitted power is limited to 1W e.i.r.p. (equivalent isotropic transmitters power) if the device can automatically control power. In other cases power is limited to 500 mW e.i.r.p. Theoretical data speed is 54Mb/s, however, effective bandwidth depends on many parameters a particular radio routes. Typically it reaches maximally 30-36Mb/s. To achieve these speeds OFDM (Orthogonal Frequency Division Multiplex) is applied. OFDM is generally accepted as the best alternative when combined with BPSK, QPSK, 16-QAM, 64-QAM or even higher modulation. However, under specific conditions DSSS can be more effective (see 802.11b).

### 5.2.3 IEEE 802.11b

This standard is one of the most basic IEEE 802.11 standards family. WiFi 802.11b operates in the 2.4GHz band. 13 channel stepped by 5MHz are available in range from 2,412GHz to 2,472GHz in the CR. Unfortunately, the one channel ideal width is 20MHz and there is remarkable potential of channels overlapping with three channels exceptions. Radiated power is limited to 100mW e.i.r.p. The standard 802.11b bit rate is up to 11Mb/s if additional key encoding CCK (Complementary Code Keying) in DSSS (Direct Sequence Spread Spectrum) modulation on the physical layer is applied.

In case of deteriorated conditions data bandwidth can be reduced from 11 Mbps to 5.5Mbps, 2Mbps or even 1Mb/. Typical maximal bit rate is 5-6Mb/s.

#### 5.2.4 IEEE 802.11g

IEEE 802.11g standard was accepted in 2003. The main reason for the formation of this standard was insufficient provided bandwidth by standard IEEE 802.11b. IEEE 802.11g supports the frequencies range and the same channels split like 802.11b to support backward compatibility. The maximum reached bit rate is 54Mb/s. In order to achieve a higher speed, but also compatibility with 802.11b both DSSS (compatibility) and OFDM (higher bit rate) are supported at the same time. OFDM support 54, 48, 36, and 24Mb/s if 16-QAM is applied, 18 and 12Mb/s with QPSK, 9 and 6Mb/sec with BPSK modulation. Other rates are in accordance with 802.11b when DSSS is applied: 11, 5.5, 2 and 1Mbps. Reached bit rate depends on the parameters of radio routes and moving up to 30 Mb/s, but also whether DSSS or OFDM is applied. The system is forced to accept PCF mode, even it significantly increases the overhead. Non-collision approach, however, balances this disadvantage.

#### 5.2.5 IEEE 802.11e – Wireless QoS

IEEE 802.11e was accepted in September 2005. This amendment added QoS (Quality of Service) management and it corrected some problems in sub-layer MAC (Media Access Control) specifically in a support of all physical layers used in IEEE 802.11 networks. This amendment is a crucial tool for applications sensitive on delay, and minimal available bandwidth, etc., i.e. typically for ITS solution. Among the main features standard it includes QoS mapping, selects the path (terminal) to which data will be transferred using either “time slots” reservation or priority service. IEEE 802.11e improves MAC methods of DCF and a new hybrid coordination function of the PCF via HCF (Hybrid Coordination Function) is introduced.

HCF has two methods of access to the media:

- HCCA (HCF Controlled Channel Access) and
- EDCA (Enhanced DCF).

Both methods define the TC (Traffic Classes). Figure 5.3 shows the architecture of the MAC layer standard (see also [www.ieee802.org](http://www.ieee802.org)).

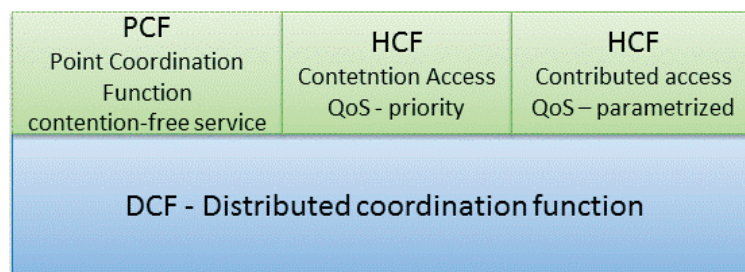


Figure 5.3 IEEE 802.11e MAC architecture

##### 5.2.5.1 EDCA (Enhanced DCF)

EDCA prioritize data with higher priority against data with a lower priority by generating shorter time to wait before it is sent. Each class of traffic has its opportunity to broadcast TXOP (Transmit Opportunity), it is bounded time interval each station can use to send maximum of available to transmission data. If packet does not fit to time interval (it is too long), it must be fragmented.

##### 5.2.5.2 HCCA (Hybrid coordination function Controlled Channel Access)

HCCA method is based on PCF (Coordination of Fiction) principles. However, HCCA is able to initiate CAP (Controlled Access Phase) at any time. It is initiated by the access point, whenever is requested to take or send the data. Ordinary during CP period EDCA is used. Another difference if compared to the original PCF is in availability of the traffic class (TC-Traffic Class) and traffic streams (TS – Traffic Streams). Queuing and final transmission can therefore be carried out according to the requirements of individual connections with various clients. In addition, clients provide information on the lengths of their queues for each traffic class, thus it is possible to achieve better results in the planning of the operation.

HCCA is generally considered the most advanced and most comprehensive coordination function. QoS can be configured and fulfilled very precisely. Each terminal has possibility to specify particular requirements on its

transmission (speed, delay, etc.). Such treatment allows the use of more sophisticated on-line applications such as real-time control, VoIP, video streaming, etc.

Improvements on MAC layer given by IEEE 802.11e can be listed as follows:

- *APSD* – Automatic Power Save Delivery – more effective transmission power management than in the original v IEEE 802.11.
- *BA* - Block Acknowledgments - whole TXOP can be acknowledged in one frame,
- *NoAck* - Not Acknowledged

**Tab. 5.1 802.11e/802.1d priority**

Priority	802.1d	802.1d	802.1e
<b>highest</b>	1	Background	<b>Background</b>
	2		
	0	Best Effort	<b>Best Effort</b>
	3	Excellent Effort	
<b>lowestest</b>	4	Controlled Loan	<b>Video</b>
	5	Video	
	6	Voice	<b>Voice</b>
	7	<b>Network Control</b>	

Service of sending frames with QoS can have one of two alternatives: QoSAck and QoSNoAck. Frames with QoSNoAck are not acknowledged with the result of short delay (see TCP/UDP).

A standard provides four categories of operation for the eight levels of priority (see IEEE 802.1d, [57], tab. 5.1) the basic method of EDCA adjusts waiting time between attempts to broadcast. AIFS (Arbitration Interframe Space) is a set of adjustment parameters. AIFS is extended with decreasing priority traffic. This means that the operation with high priority will be waiting less than traffic with low priority. Total wait time still added a random window potential collision (CW, Contention Window), its main purpose is prevention of a collisions of the packets in the same category.

The architecture of the MAC layer is shown on Fig. 5.3. It can identify compatibility with WiFi devices with no implemented 802.11e standard. These devices then communicate with each other, however, without any possibility to setup any priority for any service.

#### **5.2.6 IEEE 802.11i**

The IEEE 802.11i standard, also known as WPA2 was approved in June 2004. It modifies the security tools for the WLAN network. The use of effective methods of encryption and authentication algorithms this standard ensures better protection of networks and the transmitted data, in particular for professional applications. WPA2 uses the encryption AES (Advanced Encryption Standard), while the earlier WEP and WPA use the RC4 encryption. 802.11i architecture contains the following components:

- EAP (Extensible Authentication Protocol) and the authentication server,
- RSN (Robust Security Network) for the maintenance of record of the Association,
- AES based CCMP (Counter Mode with Cipher Block Chaining Message Authentication Code Protocol), which provides confidentiality authentication, integrity, and,
- TKIP (Temporal Key Integrity Protocol), which provides a combination of keys for the packets, checking the integrity of the message, and the mechanism of re-keying.

#### **5.2.7 IEEE 802.11n**

This standard 802.11n "Enhancements for Higher Effective Throughput" modifies the existing physical layer and MAC layer to achieve the throughput level on the MAC at least 100 Mb/s bandwidth (theoretically up to 600 Mb/s). It applies MIMO (Multiple Input, Multiple Output) approach, which is based on application of multiple antennas at both the receiver and the transmitter side. The original proposals were designed for 40 MHz bandwidth, however due to requirement of backward compatibility the bandwidth of 20 MHz was designed.

### 5.2.8 IEEE 802.11p

802.11p relates to IEEE activities and relevant standard IEEE 1609 known as WAVE (Wireless Access for the Vehicular Environment). It acts as MAC layer of the IEEE 1609 standard. This amendment represents also quite remarkable extension of 802.11 standards supporting mobile telematics applications. It includes communication between vehicles (C2C) and between vehicles and fixed infrastructure (C2I) with tolerance up to 240 km/h. Standard is designed for regulated band 5.9 GHz. Originally the authors wanted to create the basis for the CALM M5/late WAVE, and, promote the applications like ETC (Electronic Toll Collection), e-security services, e-call, cash transactions etc. Authors' vision is the uniform country-wide (global) network capable to communicate between the moving objects and fixed access points or between the mobile objects. Even though the basic goals were already reached, the problem of IEEE 802.11p which got in conflict with European system DSRC 5.8GHz and resolving of their mutual electromagnetic in-compatibility remains still as the hot topic for IEEE, ISO/CEN and ETCI.

### 5.2.9 IEEE 802.11r

The original handover principles described in IEEE 802.11f were too slow for real-time applications. Therefore IEEE 802.11r working group [48] has developed more reasonable solution which ensures a seamless transfer of communication between two base stations without noticeable transition effects (seamless handover) ensuring also continuous encryption based on 802.11i standard.

### 5.2.10 IEEE 802.11 - conclusions

IEEE 802.11 represents widely spread technology originally applied in public services as "low-end technology – see also Ethernet (IEEE 802.3, and 802.1q history). There was remarkable hesitation of the market to accept WiFi as professionally applicable technology. Amendment IEEE 802.11e introduces data packets/frames priority processing as an extension of the current WiFi technology. Amendment IEEE 802.11p means another breakpoint in a history of WiFi. IEEE standard 802.11 was via "partial" acceptance by WAVE as MAC layer de facto accepted as the professional partner and so accepted to family of ITS professional communications family.

## 5.3 WiMax

### 5.3.1 WiMax– IEEE 802.16d

This standard offers portfolio of wireless access services designed for professional applications (carrier class) in the robustness of solutions, flexible throughput and selectable levels of QoS. This standard has been described in detail in available literature [57] so that we will provide only basic characteristics of both fix and mobile version of WiMax. This standard offers Broadband Wireless Access (BWA) and it has been designed as the alternative to services fix technologies xDSL operated on metallic lines. It is suitable not only for data services, but also voice and video services. Properties of the system are given by its PHY (Physical Layer) and MAC (Media Access Control) as follows:

#### 5.3.1.1 PHY:

- Adaptive modulation (from 64-QAM to BPSK) and variable self-correcting coding provides flexible conditions for optimal usage of available spectra,
- OFDM 256 modulation effectively shares available frequency range for more clients,
- Both Time Division Duplex (TDD) as well as Frequency Division Duplex (FDD) are provided,
- Variably supported available frequency bandwidth supports conditions for optimization of its usage,
- Smart antennas optimize exploitation of available bandwidth – dynamic parameters of this solution can generate (in accordance to author's experiences) problem under dynamically changing conditions.

#### 5.3.1.2 MAC:

- 1 to 100 users can share one supported frequency bandwidth,
- System supports circuit oriented solutions based on QoS management with goal to support also voice/video on-line services,
- Security tools authentication and encryption (DES 3 only!!),

- Automatic control (minimization) of transmitted signal level,

#### 5.3.1.3 *Provided QoS levels – (ATM equivalent in brackets):*

- UGS – Unsolicited Grant Services – E1 circuits emulation for standard voice services (see CBR),
- rtPS (Real-Time Polling Services) – supports variable sized packet transferring with quality suitable for services provided in real time like MPEG or VoIP (see VBR),
- nrtPS (Non-Real-Time Polling Services) for variable length packet periodical transfer (see nrt VBR),
- BE (Best Effort services) for e-mail or web services (see ABR)

The Standard supports IP packet, Ethernet frames and ATM cells transfer. The Optimized support of basic protocols causes complexity of its own technology, but also service implementation and management. Despite the relative complexity of 802.16d WiMax is understood as prospective standard for stationary and nomadic telematic services. This standard does not provide any guarantee for mobile service, even though authors have positive experience with WiMax 802.16d reasonable movement tolerance up to 120km/h (see also e.g. experimental results of Prague Airport project in chapter 8).

### 5.3.2 **Mobile WiMax – IEEE 802.16e**

This standard was certified in the first half of 2008 [58]. It is based on standard IEEE 802.16d with an intention to resolve the mobility of wireless services. Mobility represents a substantial increase of system complexity if compared with stationary solution. Authors of Mobile WiMax standard accepted compromise in the tolerance of movement – the maximum speed is limited by "only" 150 km/h.

Radius of served area by systems based on IEEE 802.16d is up to 30 km, but typically in accordance to required traffic and effective usage of the bandwidth it will be usually much less. The same situation is with size of individual cell of the IEEE 802.16e based systems. Multiple usage of available bandwidth based on cell principles do not force provider to maximize coverage of each individual cell. Cell architecture support with handover principles as well a movement tolerance support generate significant additional overhead so that total communication capacity with the moving object (related to available bandwidth) is less than is the case of fixed stations. The system unlike 802.16d applies SOFDM (Scalable OFDM). This approach allows the standard behavior of the system in its entirety the bandwidth (from 1.25 to 14Gb/s). Mobile WiMax system accepted more powerful AES (Advanced Encryption System) replacing less effective older DES 3 algorithm applied in 802.16d.

Mobile WiMax 802.16e had no ambitions to be a direct competitor to the new generations of mobile systems. Its application fields were professional solutions with above standard service quality requirements. However, due to almost two years delay in certification procedure these ambitions will not be most probably fulfilled and Mobile WiMax future is uncertain. The very ambitious and sophisticated ideas of this standard will not be most probably materialized in wide implementation of Mobile WiMax networks, nevertheless, they certainly formed the basis of other high-performance mobile systems and both public and professional mobile services.

## 5.4 **PAN**

PAN (Personal Area Network) is designed for the mutual exchange of data between devices at a distance in units of meters. It is typically private solution for small groups of users. This type of solution was introduced by Bluetooth wireless technology used as wireless communication medium between mobile phones. Then the technology ZigBee (IEEE 802.15.4) and UWB (802.15.3) followed. Growth of WPAN (Wireless Personal Access network) solution is required by the development parameters of terminal equipment. Increasing the bandwidth for a short distance, as well as "ad hock" network architecture offers the new possibilities for applications in telematics, and therefore we will speak about each of these solutions.

These systems opened possibility to be applied for data collection as well as for wireless control of the equipment. Such technologies are used in households as the possibility for wireless various appliances control. Of course they are applied in various areas of industry, in the security engineering, medicine, telematics etc.

PAN can be classified according to the physical layer used on fix and wireless. Metallic PAN are represented e.g. by technology, USB (Universal Serial Bus) or FireWire (IEEE 1394 interface). The wireless networks are known as WPAN (Wireless PAN) and include standards like IEEE 802.15 or IrDA (Infrared). PAN systems do not frequently offer TCP/IP compatible architecture. These solutions have no ambition to

compete with WLAN in parameters like service coverage or robustness of construction, however, volume of transferred data can e.g. reach remarkable results (e.g. 2Gb/s with UWB). Relative simplicity of PAN solution offers substantially lower pricing on their implementation as well as operation exactly in context of PAN application. Each of the known standards has its specific parameters and must be fully respected when applied.

#### **5.4.1 Bluetooth – IEEE 802.15.1**

BlueTooth is designed for non-regulated band 2.4 GHz. To reduce the effect of other band users FHSS (Frequency Hopping Spread Spectrum) approach is applied. Version V1.1 uses 79 channels. Carrier wave is hopping from one channel to the other managed by the pseudo-random sequence generator. Frame hops rate is up to 1600 per second, i.e. each transfer would be carried out every 625µs. However, packets can last longer than critical 625 µs and such packets size reduces number of carrier frequency hops.

The data transmission is a regime of packet switching in asynchronous mode, the ACL (Asynchronous Global Link), where each packet may „last“ one, three or five time intervals. In each such case, the packet is transmitted on a single frequency, for the first time interval. The longest packet Qh5 contains 2871 bits and it transmits user data 2745. After sending this packet one basic sized packet must always be transferred back. Maximal transfer rate of version v1.1 during asymmetric mode 723,2 Kbps in one direction and 57,6 kb/s in the opposite direction.

Voice communication is carried out in synchronous mode the SCO (Synchronous Connection Oriented). Digitalized voice signal is split in time intervals and each interval is transferred one packet. In version v1.1 bit rate is 64 kb/s in both directions. Each packet begins with the access code of length 72 bits, which is unique for each autonomous site. Its task is to allow only authorized devices. The header is followed by 18 bits, however, its length increases to 54 protective encoding bits. The total length of the packet may be in the range of 126 to 2871 bits.

#### **5.4.2 UWB (Ultra-WideBand) – IEEE 802.15.3**

Working group 802.15.3 is split on 802.15.3a, 802.15.3b a 802.15.3c:

##### *5.4.2.1 802.15.3a*

The Working Group of IEEE 802.15.3a WPAN High Rate Alternative PHY Task Group addressed the use of the technology options for UWB wireless communication on short and medium distances. This group worked with two versions of the technology – DS-UWB and MB-UWB [51].

DS-UWB belongs to the Group IR-UWB (UWB Impulse Response). It is based on the principle of transferring one bit with one impulse. Extremely short pulse of 1ns duration is used. Modulation represents for logical zero time shift of half the duration of the pulse back compared with position of the reference pulse and logical one about half of the pulse duration forward. Due to the pulse modulation with pulse takes up the shape of the signal in the frequency bandwidth and several GHz. MB-UWB systems (MultiBand UWB) use of the above principles and OFDM (Orthogonal Frequency Division Multiplex). The total flow of data will then be divided into a number of incremental data flows the individual carrier. MB-UWB method places greater demands on hardware, but it is more robust to errors and interference against the IR-UWB. However, in 2006 activity of the working group 802.15.3a was suspended.

##### *5.4.2.2 802.15.3b*

802.15.3b address issue of MAC interoperability improvement and various small optimizations within the standard were introduced.

##### *5.4.2.3 802.15.3c WPAN Task Group: „Millimeter Wave Alternative PH“*

The Working Group 802.15.3 c was founded in 2005. This working group developed the solution for use of the unlicensed frequency 57-64 GHz. WPAN allows operation in accordance with all other microwave systems in a family of standards 802.15, and offers a high transmission speed (up to 2 Gb/s), thus meeting the requirements for applications such as high-speed Internet access, or wireless data bus for the replacement of cables between electronic devices.

Advantageous is that thanks to the use of shorter wavelengths, dimensions of the antenna is reduced and it can be incorporated into a single chip. The effectiveness of the integration of the antenna to the chip is still the subject of R&D experiments.

#### 5.4.3 ZigBee – IEEE 802.15.4

The latest member of IEEE 802.15 group - 802.15.4, represents MAC layer of the ZigBee architecture. It was accepted by IEEE in year 2006 [52].

Transport application field requires wireless self-configuring networks of mobile objects connected by wireless technologies known as MANET (Mobile Ad Hock Networks). In some applications like different sensors networking can represent mobility not as so important requirement, and, low power consumption, or driven power reduction etc. can be accepted as more important. Around 40 ad hock networking protocol are already described with still limited positive users response namely in a case of high movement dynamics is required.

The aim of the ZigBee Alliance activities was to develop the corresponding wireless communication environment for hierarchical network communications. Communication between the control unit or system of distributed control units and between the different sensors and actors is required. ZigBee Alliance focused its attention primarily on the creation of a set of standardized solutions tin open spectra 2.4 GHz and sub-GHz spectra. Sub-GHz spectra differ on each of the continents. Europe accepted 868 MHz, while North America and Australia 915 MHz.

ZigBee Alliance decided to support a standardized communication wireless access solutions for the widest possible range of users, i.e. from the private customer sector to industrial applications. Important criteria in the design represent requirement to support "remote" sensors/actors in a low consumption regime, dynamic networking the various active elements, and, reasonable price/performance ratio, which has been possible to reach the hierarchical distinctions of system parameters in the different categories of communications modes relevant to supported applications, level of sensors and actors etc.

ZigBee architecture consists of the following basic layers (i) Application Layer (APL), (ii) Network Layer (NWK) and (iii) Network Interface Layer (NIL). Functional decomposition of NIL on PHY and MAC sub-layers is compliant with family of standards IEEE 802. MAC layer is defined by standard 802.15.4, and PHY layer design is concentrated on low power consumption, low cost of elements as well as on low installation and operational expenses. NWK and APL layers are defined by ZigBee Alliance.

##### 5.4.3.1 MAC (Media Access Control) sub-layer

MAC sub-layer is designed to deal with the extensive number of network elements with relatively dense area coverage. To simplify the concept of communication system if compared e.g. with Bluetooth the only two modes only active and standby (sleep) mode are applied without any requirement on any application to choose from different degrees of reduced communication mode as it is usual with other systems.

##### 5.4.3.2 NWK

NWK layer offers dynamic "3-d" growth of the topology of network without any need of additional powerful transmitters. The other complex networks (expected range point-to point (P2P) is at least 50 m) are served relatively low value of the delay is reached. The network layer has implemented the dynamic routing table tool that is able to effectively adapt to network conditions changes caused by individual nodes or networks connection/disconnection.

##### 5.4.3.3 APL

ZigBee Application Layer APL consists of two sub-layers: APS (APlication Sub-layer) and ZDO (ZigBee Device Object). APS carries and manages tables, for Point to Point (P2P) interconnect and relevant mutual requests and data transfer. ZDO identifies hierarchical position of node in network manages whole system data networking.

##### 5.4.3.4 ZigBee properties

ZigBee is concentrated on:

- *Low power consumption* (battery life 6moths or more),

- *P2P minimal distance of 50m*, with potential to reach up to 500m in accordance to radio conditions,
- *Hand-shake protocol applied*,
- *Quality of Service (QoS) support* – selected time-slots are dedicated for guaranteed QoS, remaining time slots operated in slotted CSMA/CA (Carrier Sense Multiple Access Collision Avoidance) regime,
- *Extended number of nodes* – 64 bits address supports 264 nodes in max. 216 independent networks,
- *Simple set of codes (25% of codes used for Bluetooth systems)*,
- *Only 2 modes* – active and sleep,
- *Reasonable value/cost ratio*,
- *2 types of nodes* – FFD (Full Function Device) a RFD (Reduced FD),
- *Global applicability* (supports 2.4GHz. in sub-GHz band in Europe 868 MHz, in North America and Australia 915MHz).

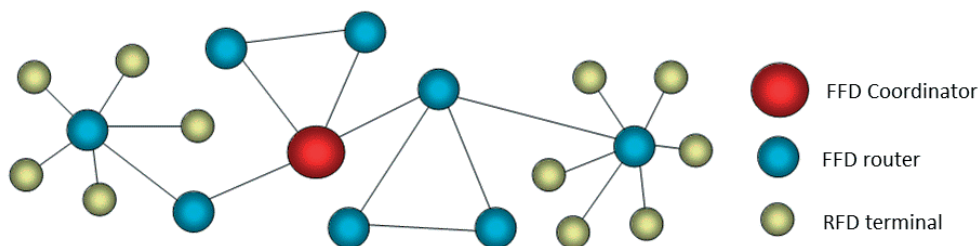
Data transfer regimes:

- *Periodically transferred data* - sleep mode is applied when transfer is not active,
- *Occasional data transfer* – triggered either by application (awaked from sleep mode) or by central unit via beacon delivery (cannot be switched to sleep mode)
- *Data service with defined QoS (Quality of Service)* - typically maximal time to deliver. It is based on dynamical time slot GTS (Guaranteed Time Slot).

ZigBee nodes are acting as:

*Network coordinator based on FFD* – setups network, transmits synchronization beacons, manages routers, collects information from routers, routes datagrams between nodes. It operates in receiver mode to act whenever required

- *Network router based FFD* - identifies available networks, supports data transfer under management of network coordinator,
- *Terminal device* - typically RFD – operates exclusively in star topology. Its functionality is limited and communicates exclusively with router node



**Figure 5.4 Example of ZigBee network topology**

#### 5.4.3.5 Security tools

Security tools do not only protect their own data frames, but the tools are applied to ensure the MAC control, signal and acknowledgment frames. MAC layer is used as the basis for the encryption algorithm AES (Advanced Encryption Standard) as well as many other security tools of ZigBee standard. These tools ensure required level of confidentiality and integrity of data as well as authorization of MAC frames. MAC layer does not control applied security processes. Process control is dedicated to NWK, which sets the security keys.

#### 5.4.3.6 ZigBee summary

ZigBee communication system has got significant potential in the professional communication applications where are offered different levels of security, guaranteed service quality and very low consumption of the lower layers nodes. Significant emphasis is concentrated on dynamic behavior of the network. Telematics definitely belongs between sectors with great potential to effectively use services parameters ZigBee communication solutions can be provided. ZigBee offers an optional selection of security tool which is not available in most alternative and even significantly more expensive technologies.

The main disadvantage of communication system ZigBee applied in telematics can be identified in low velocity of movement tolerance range required in ITS and limited distance individual nodes can communicate. However, distance limits are caused by low power consumption requirement due to fact that exclusively battery power supply can be used.

## 5.5 DSRC (Dedicated Short-Range Communications)

The goal of Dedicated Short-Range Communications (DSRC) goal is to enable providing „communications” between a vehicle and infrastructure. In accordance to ITU-R spectra dedication DSRC radio operates in radio frequencies range from 5,725 MHz to 5,875 MHz, i.e. Industrial, Scientific and Medical (ISM) band. However, in Europe frequency only range 5,795 – 5,815 GHz is accepted for all member countries and any part of the optional frequencies range 5,855 – 5,925GHz – see Figure 5.5. It may be used if released by national regulators. The communications based on DSRC standard provides communication between of Road Side Units (RSUs) and the On Board Units (OBUs) extended by US DSRC 5.9 WAVE standard also to pedestrian unit (PU) enabling communication between pedestrian and vehicle. European solution DSRC 5.8 selected asymmetrical model in which RSU represents an active side of communication and OBU is designed as semi passive unit, i.e. unit of low power consumption which does not require any battery energy for a signal transmission. For transmission OBU applies energy collected from received RSU signal via OBU antenna. Such a solution leads to the small compact units with long lasting battery life. Such an approach represents the remarkable advantage only in a case of exclusively microwave based system. With expected massive entry of units equipped with GNSS receiver and GSM mobile communication part such a low consumption is not the key issue – units use to be supplied from vehicle resources. DSRC 5.8 parameters, if compared with DSRC 5.9, are more and more frequently understood as a reason of future (mid or long term horizon) potential convergence to DSRC 5.9 more complex alternative.

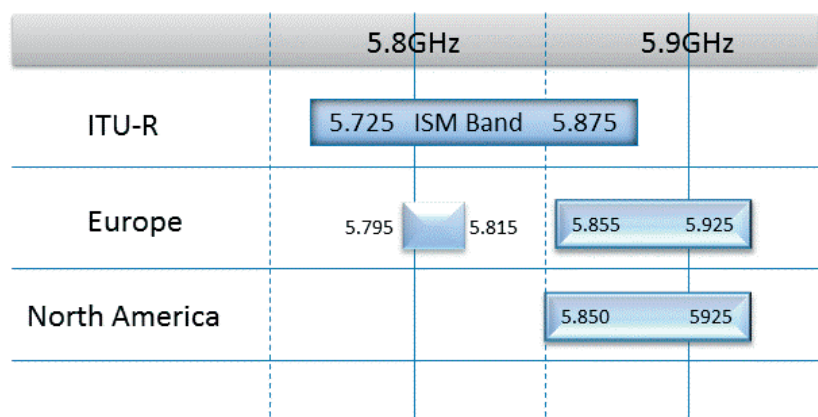


Figure 5.5 Dedicated for DSRC frequencies range in Europe and USA

### 5.5.1 DSRC 5.8

DSRC systems are applied in many European Union countries. Some reported interoperability problems between for signal transmission national alternatives must be solved before Pan European Electronic Toll Collection solution based on EU Directive No. 750/2009EU will be implemented to cover whole EU territory. This issue belongs to responsibilities of the EETC certifications laboratories. Some additional services using DSRC are expected to come, however, for signal transmission final acceptance of DSRC 5.9/Wave have generated a discussion about for signal transmission future co-existence of 5.9 and 5.8 DSRC systems and remarkable potential of their convergence to the IEEE alternative.

The DSRC architecture follows RM OSI model principles. Nevertheless, thanks to P2P (Point to Point) type of communication the only three layers were adopted.

Standard CEN 12253 specifies physical layer of the DSRC 5.8 GHz. The solution is connected with the short distances range between the Road Side Equipment (RSE) and On-Board Equipment (OBE). The standardized approach applies the cost-effective semi-passive transponder technology on the OBU side. It means that the energy of a signal received from the RSE is applied for uplink (from OBU to RSE) transmission. The signal (with new content) is transmitted with different frequency and modulation back to RSE. Such an approach does not require battery powered OBE transmission so that OBU size and power consumption limits

and the requirements on battery capacity lead to a low price and small-sized OBU unit. The default data rates represent 500 kbit/s for downlink and 250 kbit/s for uplink.

The Data Link Layer defined by CEN 12795 is split into Logical Link Control (LLC) and Medium Access Control (MAC) sub-layers. The LLC sub-layer has been adapted from the IEEE 8088.2 specifications (i.e. IEEE 802.2) applicable for both connectionless and connection-oriented services. The MAC sub-layer provides contention mechanisms to avoid and resolve, if not avoided, data collisions in multi-lane environments.

The Application Layer - CEN 12834 represents the service interface for the applications and resolves multiplexing and a common initialization mechanism for each communication process between the OBE and the RSE initiated always by RSE.

Application standards using described DSRC system have been developed by the application-oriented working groups within CEN TC 278, e.g. for Electronic Fee Collection (EFC) and Traveler and Traffic Information (TTI).

### **5.5.2 DSRC 5.9 - WAVE**

DSRC 5.9 known as WAVE (Wireless Access in Vehicular Environments) is dedicated for applications both C2C (Car to Car) as well as C2I (Car to Infrastructure) in the distance range up to 1000 m.

Group standards IEEE 1609 offers the widely acceptable telecommunication solution applicable in the automobile industry. WAVE provides basic communication model, optional standardized services and interfaces which together represent secure wireless communication vehicle-vehicle and vehicle-infrastructure. WAVE standards treat communication structure, security mechanisms, and high-speed (up to 27Mb/s) short-range wireless communication. The basic components are On Board Unit (OBU), Road-Side Unit (RSU) and the WAVE interface. WAVE solution provides the basis for a wide range of applications that are used in e-safety, ETC/EFC, enhanced navigation, traffic management as well as for many others.

Family of IEEE 1609 standards includes:

- IEEE 1609.0 - Architecture definition. It describes the architecture of the WAVE and services necessary for the multi-channel DSRC/WAVE device dedicated for communication in mobile automotive environment,
- IEEE 1609.1 - Resource Manager. It specifies the services and interfaces for application management WAVE resources. It describes the data service and the management of the services offered within the architecture of the WAVE. This part also defines the format of the messages and appropriate responses to them, the format for storing data, and more.
- IEEE 1609.2 - Security Services for Applications and Management Messages. It defines the secure message format and their processing. Further provides the conditions for the safe exchange of transferred messages, and, how these messages should be processed in accordance to their purpose and significance,
- IEEE 1609.3 - Networking Services. It defines the service network, the network and transport layer, including IP addressing and routing. It also defines the Wave Short Messages that provide some alternative to IPv6, which can be supported directly by application,
- IEEE 1609.4 - Multi-Channel Operations. It provides extensions to the Media Access Control (MAC) of the IEEE 802.11 MAC to support the WAVE. This standard defines the services and its security message format to enable support for crucial safe electronic payments and extensions to the IEEE 802.11 which are necessary to ensure the wireless communication in the environment of the vehicles.

IEEE 802 11p Amendment of the IEEE 802.11 standard meets requirement to support the distance range up to 1000 m, support for higher-speed vehicles (up to the relative speed of 240 km/h) and deals connected with specific transportations dynamic environment. Standard IEEE 802 11p is based on the standards IEEE 802.11a modulation and OFDM principle, it applies both CSMA/CA protocol and IEEE 802.11e message prioritization.

	No. Of layer	ISO/OSI net model	Data plane		Management plane	
IEEE 1609.1	7	Aplication	e.g. HTTP	WAVE applicatiom	WAVE station management Entity WSME	
IEEE 1609.2	4	Transport	TCP/UDP	WSMP		
	3	Network	IPv6			
IEEE 1609.3	2b	Data Link	802.2 LLC			
	2a		WAVE MAC			
IEEE 1609.4	1b	Physical	WAVE Physical Layer		PHY	
IEEE 802.11p	1a		WAVE Physical edium		management	

**Figure 5.6 WAVE/DSRC 5.9 architecture**

Communication can take place between:

- Infrastructure on the road and the mobile unit (I2C),
- two mobile units/ Cars (C2C) or
- Pedestrian and the mobile unit/Car (P2C).

## 5.6 Wireless access telecommunications solutions conclusions

Wireless access data solutions with guaranteed service quality have not been locally available. GSM operators offer data services as an option to voice, however, no service quality guarantee is provided. Originally expected breakpoint with UMTS entry did not come with expected coverage and nowadays operators are concentrated on LTE future with minor concentration on UMTS coverage grow.

The solutions based on WiMax (IEEE 802.16d and namely Mobile WiMax (IEEE 802.16e) were tested and promising results deduced remarkable expectations. However, Mobile WiMax 802.16e has not been offered massively to date and there is not expected massive entry of services based on these promising standards with clear ability to offer SLA on QoS.

On the other hand WiFi quietly entries ITS market and the first tangible result can be seen in acceptance of IEEE 802.11p in WAVE/DSRC architecture. Additionally Amendments to WiFi standards like 802.11e, 802.11n or 802.11r remarkably extend applicability of solutions based on IEEE 802.11 standard in ITS with reasonable decrease of the communication solution cost.

DSRC solution represents specific technologies dedicated for ITS applications. There are available two different systems (CEN and IEEE standards) and experts expect in mid or long term CEN DSRC migration to the IEEE solution.

The best approach to guaranteed coverage and limited QoS support for ITS applications is in the effective combination of different access solutions based on the second generation of handover described in Chapter 8.

## 6 Selected wireless telecommunications services performance

In the following paragraphs we will present the principles of the best available wireless communications solution selection based on evaluation of their performance indicators as well as other relevant parameters like service cost and corporate policy. The presented decisions processes are effectively useable to select the best possible alternative are effective if applied on IP layer (i.e. L3) of the TCP/IP communication model i.e. in contrary to the most of discussed alternatives implemented on L2. Before this solution is presented analysis of three most often applied wireless solutions of the key wireless technologies analysis are introduced and their principle performance indicators measured in our laboratories are provided.

### 6.1 Mobile services GSM

There is a view that GSM mobile network can provide fast and reliable data service with very reasonable signal coverage and very high level of availability. However, practical ITS implementation identified quite a remarkable problem with the performance of GSM data services applied within telematic applications.

Our study has been concentrated on identification of the critical internal performance indicators and analysis of their impact on the data services performance. All measurements were done exclusively within one GSM cell.

The principle change of service parameters caused either by too high number of customers or e.g. overloaded IP network caused e.g. by lack of routers CPU/memory capacity cannot be identified directly. Providers do not typically offer such information to their clients (even the top ones). The changes of the services performance indicators can be identified only indirectly by parameters like PLR (Packet Lost Ratio) or RTD (Round Trip Delay).

Data channel capacity  $CC$  can be calculated as  $CC = B \cdot \log_2(1 + C/I)[b/s]$ , where  $B$  is the bandwidth of the channel in Hz, parameter  $C/I$  is ratio  $C$  - total channel signal power in used bandwidth measured in mW and  $I$  - total noise power in applied bandwidth measured in mW.  $C/I$  typically used in GSM terminology is in Shannon – Hartley theorem known as  $S/N$ , i.e. the signal-to-noise ratio (SNR) or the carrier-to-noise ratio (CNR) of the communication signal and the Gaussian noise interference expressed as a linear power ratio. In GSM architecture bandwidth is constantly set to 200 kHz. Parameter  $C/I$  so represents the critical parameter which influences GSM data services technology performance expressed by performance indicators:

- Data channel capacity,
- Packets loss ratio (PLR) and
- PDD (Packet delivery delay) or RTD (Round Trip Delay).

Above mentioned parameters can be routinely identified on the IP layer using services of L3 (IP layer).

### 6.1.1 Methodology of experiment

Each of available 2.5<sup>th</sup> GSM generation data services i.e. CSD, HSCSD, GPRS and EDGE was individually studied. GSM laboratory was equipped by locally manageable fully calibrated base station with adjustable transmitter output power  $C$ . Additionally the signal noise with adjustable level  $I$  was generated by external calibrated noise generator. The power of the base station was set for each measurement period on defined level  $C$  and the power of noise  $I$  generated by additional noise generator was changed in defined limits step by step.

The service quality measurement is processed using L3 (IP) layer tools. Each individual measurement generated “ping -n 100 -l 10 ftp address”, where  $n$  represents the number of transmitted packets ( $n=100$ ) and  $l$  represents packet size ( $l=10B$ ). A small packet size was reasonable for identification of minimal time of service response.

The black line in graphs represents minimal RTD [ms], the dark grey stands for average RTD the light grey corresponds with average PLR [0% – 100%].

### 6.1.2 CSD measurement results

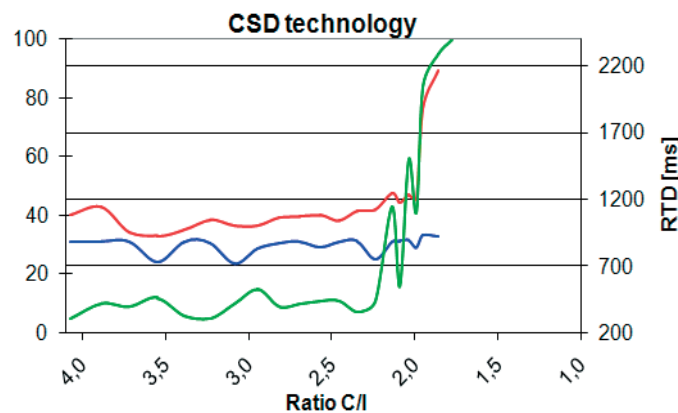


Figure 6.1 CSD technology

Fig. 2 shows that CDS technology offers data service with delay in the sub-second range. This service has relatively low sensitivity on signal to notice ratio. However, it must be stressed that CSD is circuit switched

technology with all well-known disadvantages of this approach in the field of data services.

### 6.1.3 HSCSD measurement results

Results obtained for CDS technology measurement displayed on Fig. 3 are valid for HSCDS technology, as well. The only difference is in a channel capacity due to fact that the increase of the capacity is exclusively reached by increasing of the applied time slots number.

### 6.1.4 GPRS measurement results

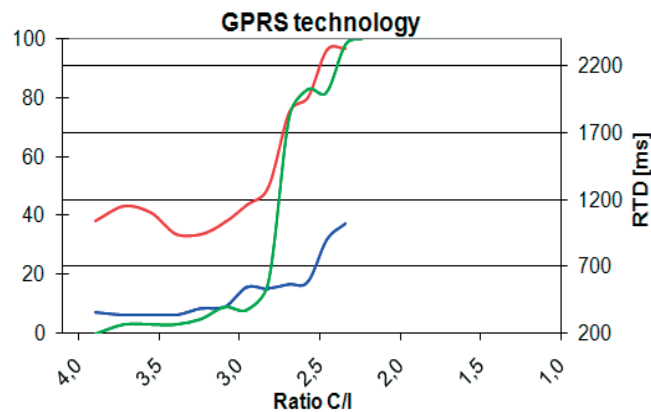


Figure 6.2 PLR and RTD - GPRS technology

Figure 6.2 shows that GPRS technology provides better delay than CDS (350 ms), however, only for small values of signal to noise ratio). With the increasing intensity of interference GPRS the service delay rapidly grows. The relatively high sensitivity packet loss on C/I can be identified, as well. For C/I above 2.69 packets loss is above 75%! The results identified mainly that GPRS technologies are applicable for “less demanding” applications where long delays and high potential of packet losses are not critical or it can be combined with alternative technology based on CALM principles.

### 6.1.5 EDGE measurement results

Figure 6.3 describes the fact that EDGE technology is better acceptable for telematics applications than GPRS, because of the remarkable improvement in both delay and packet loss. The minimal delay was in this case within interval from 258ms to 365ms and the high packet loss starts, when the value of C/I ratio is above 1.2. This technology so could appear even with more demanding telematic solutions if the service provider can guarantee appropriate priority of service provisioning.

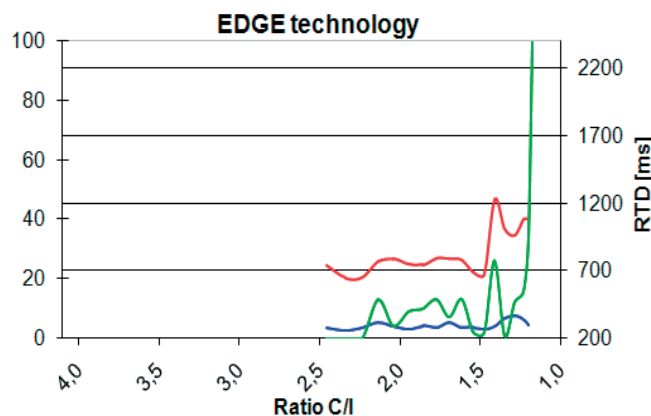


Figure 6.3 PLR and RTD - EDGE technology

### 6.1.6 GSM data services summary

GSM service providers have kept their focus on the core business - the mobile voice services and data services are provided as more or less complementary products with not guaranteed services quality. This disadvantage could be resolved by partial network capacity dedication (e.g. via virtual operator's services) to the "special" services portfolio with efficient the services quality management. However, the status of auspicious "virtual operators" does not have a good chance to be accepted due to strong "self-defense" afford of the powerful mobile operators.

Originally expected data services with global coverage based on the 3rd generation mobile data service (UMTS) don't have enough potency to reach all rural areas. Beyond the 3rd generation solutions (LTE) are very promising future solutions (expected latency approx. 10ms), however, such services cannot be expected sooner than in a few years.

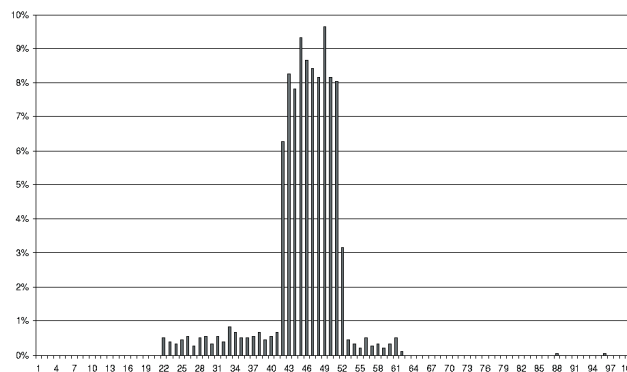
## 6.2 WiMax (IEEE 802.16d) measurement results

**Table 6.1 Results of the WiMax access study**

Site	Visibility	RTD [ms]	SNR [db]
1	LOS	45.6	33
2	LOS	47.1	32
3	NLOS	44.6	-26
4	NLOS	44.8	-27

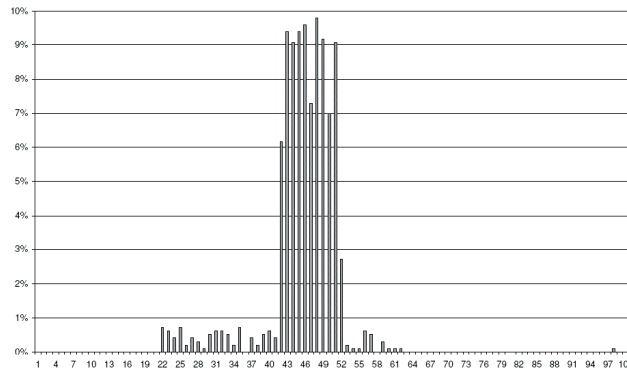
Remarkable potential can be recognized in a combination of all key GSM service providers as well as different alternative wireless access data services. Management can be based on the "CALM" or alternative principles with implemented effective classification and decision processes. Mostly alternative services are dedicated to fill the services gaps when/where globally available GSM wireless network cannot provide the service on required quality level.

The technology based on IEEE 802.16d/e standards known as (Mobile) WiMax represents one of the most promising substitutions. This technology (in version "d") was studied in detail in project called CAMNA. The research team had the unique opportunity to test WiMax technology in "real life" pilot application - see e.g. [10] or [36]. The basic results of WiMax measurement are in Table 3.2.1. Even though results are in different structure due to different method of WiMax technology performance identification the dynamic parameters can be quite easily compared.



**Figure 6.4 RTD spectra of LOS - SNR = +33db**

RTD presents "Round Trip Delay" in ms, SNR is "Signal to Noise Ratio" in dB, LOS represents "Line Of Sigh" and NLOS stands for "Non LOS". RTD results are displayed on Fig. 3.2.1 and 3.2.2. RTD is in average approx. 50ms, i.e. more than ten times faster than GPSR and EDGE.

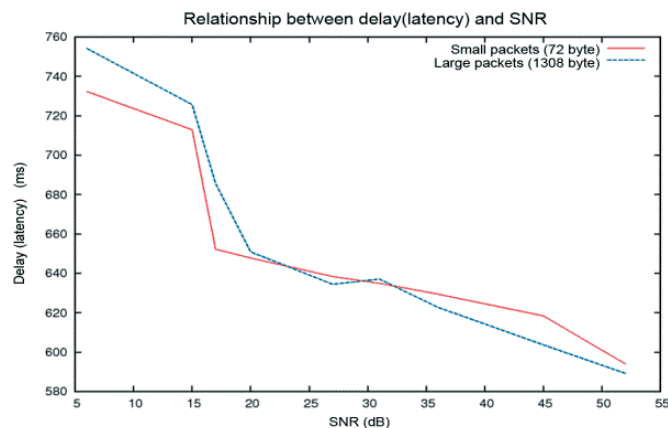


**Figure 6.5 RTD spectra of NLOS - SNR=-27db**

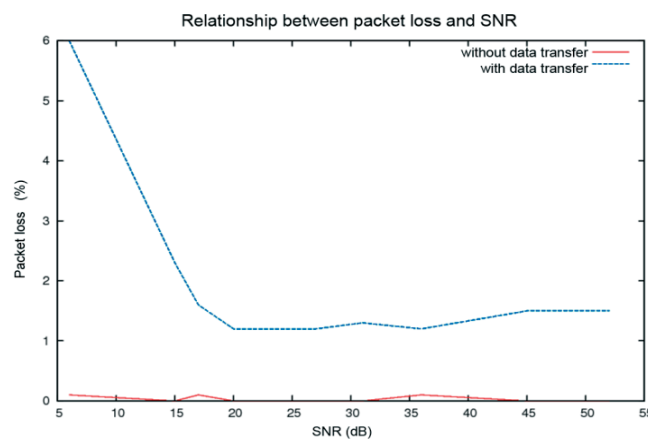
### 6.3 WiFi (IEEE 802.11) measurement results

Potential of WiFi technology based on IEEE 802.11 family of standards is in ITS dynamically growing. WiFi was primarily focused on the low end of the mass access Internet market and it has been operated mostly in the “non-licensed” frequencies bands. This standard, however, nowadays trends to the professional applications. New functionalities described in Amendments like 802.11e or 802.11p treat services quality management operated also in the licensed bands (e.g. 5.9GHz). Most of WiFi amendments have not been yet officially accepted and published, however, WiFi solutions will reach position of reasonably cheaper alternatives to more robust WiMax services.

The basic system parameters study was focused on latency and the packet lost ratio in dependency on SNR. Goal was to obtain representative set of data for this technology evaluation.



**Figure 6.6 WiFi – Packet delay vs. SNR.**



**Figure 6.7 Packet loss vs. SNR**

SNR values 20 dB represents the critical value. However, the both Delay and Packet loss are dependent on the other parameters - namely the network load. Collision access protocol has limited possibility to manage service quality. 30% traffic load can already cause latency up to several hundred of ms. Appendix IEEE 802.11e will, however, offer QoS (Quality of Service) management tools simultaneously with the much higher efficiency of available bandwidth usage.

## 7 Multi-path access solution structure

The multi-path concept provides the "most appropriate" telecommunications service based on continuous process of the suitable alternative selection with ambition to meet telematics application requirements. It is expected that seamless service for both vehicle with infrastructure (Car2X) and vehicle to vehicle (C2C) communication is provided. The selection of the best possible solution evaluated status of telecommunications performance indicators set (vector). The measured data can be combined with such parameters as the economic criteria, corporate policy etc.

The family of standards CALM (Communications Air-interface for Long and Medium-range) provided by ISO TC204 WG16, represents the comprehensive standardized solutions designed for transport telematics applications. The details of CALM architecture are described e.g. in [14] - [23]. CALM executive architecture design is in full compatibility with RM OSI / TCP/IP models are generating also the complex demand on some parts of modular solution.

IEEE 802.21 standard [32] is designed independent tool to optimize handover/handoff in heterogeneous IEEE 802 networks. It enables the selection of path in whole range of both wired and wireless IEEE 802 networks as well as some non 802 networks like 3G.

The author's idea was to build solution based on experience of IEEE 802 group as well as some other mobile groups (3G). Thanks to the detailed understanding of the both areas the authors designed effective handover processes for heterogeneous networks. The transformation of the available status information is positioned on the both link layers as well as of the other network layer to the upper layers with goal to provide effective handover represents the core of success. Standard 802.21 accepted continuity principles. This means that the same level service provisioning is maintained during the handover period. Such parameters cannot be reached if all the available parameters from all layers are not available both mobile nodes as well as all fix network management centers.

### 7.1 Basic ideas of the CALM approach

Standard ISO 21210 (CALM architecture see e.g. [16]) defines the CALM system architecture. Group of additional standards resolve the switching between the alternative paths of the access solutions. RM OSI based layer architecture is applied. It means that the vertical functional decomposition of the individual alternatives with vertical data (D)SAP ( (Data) Service Access Point) interfaces or groups of compatible alternatives sharing one (D)SAP are introduced. The system management copies the vertical decomposition, as well. However relevant modules communication is maintained exclusively in the horizontal layer structures via the management interface of the (M)SAP interface – see Figure 7.1. The management information exchange between layers is concentrated exclusively to the system management modules and no managerial information flows can be provided via DSAP interface. CALM family of standards exclusively uses on L3 IP in IPv6 with possibility to effectively trace each path of the access solutions: IPv6 offers application of its own private networks tools, and the quality of the service management. On the other hand the requirement of the IPv6 support represents one of the complications related to the smooth and the fast CALM system implementation due to fact that IPv6 support in the area terminal devices and nodes dedicated for local networks were slow and hardly practicable.

The technologies like Bluetooth, ZigBee have not accepted TCP/IP systems architecture so that compatibility is possible only on the MAC sub-layers of L2 (i.e. standardized by IEEE 802.15 series of standards). For such appearances CALM environment provides the construction of the alternative virtual TCP/IP solutions to support such technologies integration into the CALM architecture. It is therefore not surprising to identify many published results based on functional transformation of these non TCP/IP based architectures to the standard TCP/IP alternative. It has been interesting to track these trends.

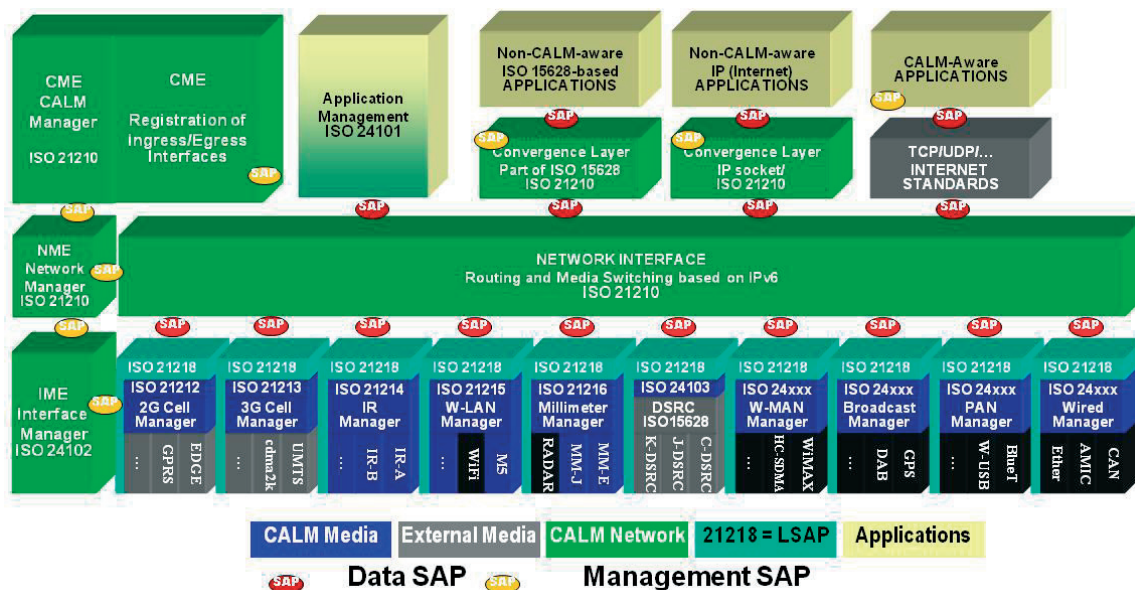


Figure 7.1 CALM detailed modular architecture - Geneva review

CALM solution for the systems with a different to TCP/IP architecture applies internal TCP/IP based simulation and both solutions are interfaced just on the border of the application layer. The future development will justify this approach or alternative TCP/IP based solution will be easily implemented. We understand that older graphical description (Geneva version) can very effectively describe all explained CALM ideas in simple drawings presented on Figure 7.1.

The complexity of the described modular architecture will require each model new design where the both “vertical” data interface defined by RM OSI or TCP/IP standard as well as the “horizontal” management interfaces are available. The process of the alternative wireless access solution substitution is understood as the second generation of the handover principle known in its first generation namely from the cellular mobile systems.

Communications CALM media are:

- Cellular systems including 2G and 2.5G GSM and UMTS,
- DSRC (5.8GHz) used worldwide for road tolling and access control,
- Millimeter wave technology (62-63GHz) used in conjunction with radar signal at similar frequencies,
- Satellite communications exclusively applied for emergency and “special applications”,
- Mobile Wireless Broadband (MWB) with the cells usually much larger than UMTS cells – today namely communications systems based on IEEE Std. 802.16e and coming IEEE Std. 802.20,
- IR (Infra Read) communications solutions,
- WiFi (IEEE 802.11 based) different alternatives - a, b, g, n,
- M5/WAVE based on standard IEEE 802.11p,
- IEEE 802.15.x based solutions: Bluetooth – 15.1, UWB (Ultra Wide Band) - 15.3, ZigBee - 15.4,
- W-USB (Wireless USB)
- ISO 15628 applications developed as application layer of European DSRC (5.8GHz). However CALM can support the only limited set of services,
- Other media to come.

Such extended list of technologies represents remarkable afford to develop and produce such extended number of modules and application of technology developed for the personal telecommunications systems and produced in mass production volumes might represent remarkable cost efficiency. The personal telecommunications systems are, however, based on simplified heterogeneous architecture with set of chips or modules designed to enable support of most today available technologies combination – like GSM voice and modem services, GPRS, EDGE, UMTS, WiFi, WiMax, DSRC 5.8, Bluetooth and also connectivity with the car internal communication bus CAN. The authors’ solution developed in a close cooperation with R&D partners represents example of suitable solution – see Picture 7.2. Such a solution, however, cannot fulfill the

requirements of CALM standards namely due to inconsistent system solution of the different chips or modules. Vendors apply cost effective „habitual“ alternative interfaces instead of one widely accepted universal bus. On the other hand such solution represents promising cost effective alternative, however, with no potential to apply described “CALM” ideas in all details. In a system with the described architecture the only limited number of the generally acceptable performance indicators is available. The second generation handover action can be typically activated by evaluation of the performance indicators set like Bit Error Rate (BER), Packets Lost Ratio (PLR) or packet Round Trip Delay (RTD) as well as other e.g. “radio” parameters with different level of influence on the final decision. Decision to switch to the alternative path is so issue with number of different frequently inconsistent input parameters. The number of inputs can be limited if significant parameters are identified and the other known parameters are identified as insignificant. Such afford to identify the key performance indicators has been basis for all the available telecommunications technologies used in the transport telematics.

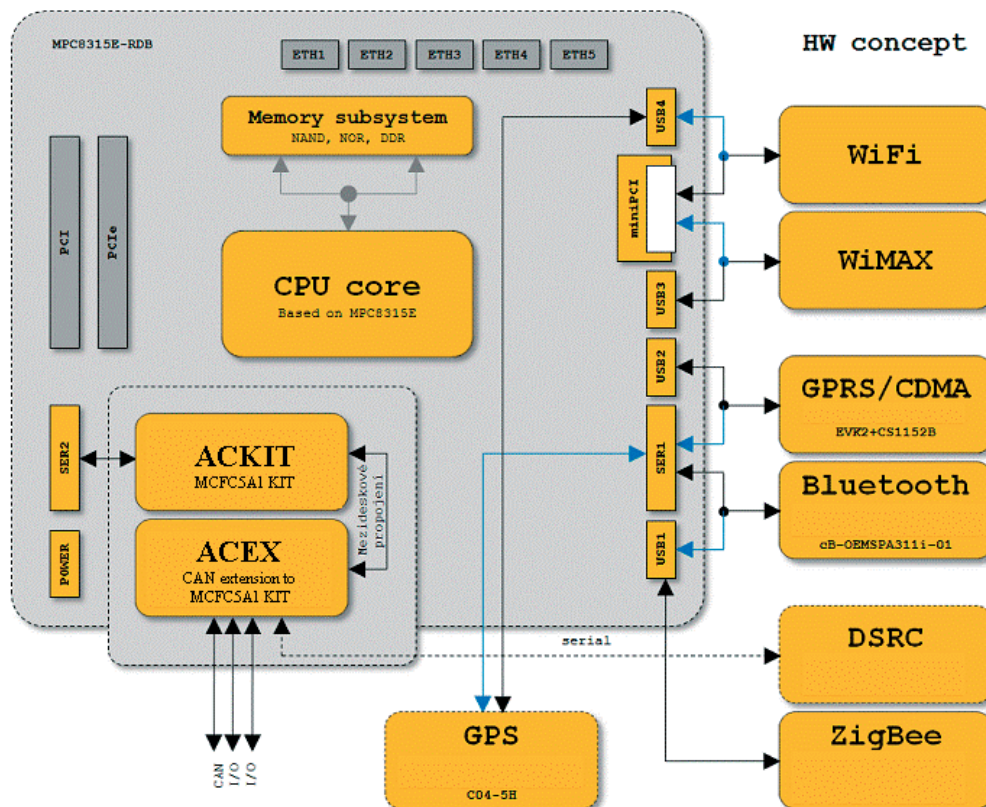


Figure 7.2 Frescale technology based alternative

Adaptive communications control system has following architecture:

- The 1-st layer – Cellular Layer (CL) - represents feed-back control processes of parameters like transmitted power, the type of applied modulation etc. Goal of processes on this layer is to keep the given set of managed parameters like e.g. Bit Error Rate (BER) or Round Trip Delay (RTD) within required limits.
- The 2-nd layer – the first generation of handover (1HL) represents the seamless switching process between the cells of the same mobile network. Such an approach is applied in the mobile systems like GSM, UMTS, Mobile WiMax or Mobile WiFi (802.11r). 1HL layer typically shares resources with CL layer (delivered usually as one system) so that there is no risk of contra-productively simultaneously operated processes on both layers - of course only - if it is correctly designed and operated. These solutions are, however, mostly designed as “close” ones, i.e. nothing like APIs are available.
- The 3-rd layer – the second generation of handover (2HL) is mostly dependent on the identification of the service performance indicators. It is for sure that the effective management on the 2HL layer can be much easier reached if 1HL and LC layers are opened for relevant information exchange with layer 2HL.

The critical issue can be identified in the potential simultaneous processing on the different layers of the

processes. Such activities can be contra-productive, and, all the potential decisions and actions should be well synchronized.

The family of standards ISO TC204, WG16.1 identified as “Communications Air-interface for Long and Medium range” (CALM) represents the concept of identification of the best available wireless access solution in given time and area. The process of the alternative wireless access solution substitution is understood as the second generation of the handover principle known in its first generation namely from the cellular mobile systems.

Each handover process is predestinated by the set of parameters range identified for the decision processes managed by control unit. Criteria for the “best possible” solution include indicators like BER (Bit Error Rate), packet RTD (Round Trip Delay), level of received radio signal, but also cost of provided service etc. Control system can take in account not only the absolute values of selected indicators, but also the specific parameters combinations trends.

Handover to alternative solution can be in principle evoked also by identification of more suitable alternative - e.g. by appearance of alternative service with more suitable cost conditions even though existing alternative has been technically sufficient and safe.

Communications CALM media are:

- Cellular systems including 2G and 2.5G GSM and UMTS,
- DSRC (5.8GHz) used worldwide for road tolling and access control,
- Millimeter wave technology (62-63GHz) used in conjunction with radar signal at similar frequencies,
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- IR (Infra Read) communications solutions,
- WiFi (IEEE 802.11 based) different alternatives “a, b, g, n, e, p, r”,
- M5/WAVE based on standard IEEE 802.11p,
- IEEE 802.15.x based solutions: Bluetooth – 15.1, UWB (Ultra Wide Band) - 15.3, ZigBee - 15.4,
- W-USB (Wireless USB)
- ISO 15628 applications developed as application layer of European DSRC (5.8GHz). However CALM can support the only limited set of services,
- Other media to come.

**Tab. 7.1 Examples of measured and control parameters on different layers**

	Measurable					Control tool				
	BER	Signal Intensity	S/N ratio	Packet delay	Packet loss	Transmitted power	Modulation	Channel coding	1st gen. handover	2nd. gen handover
<b>1<sup>st</sup> layer</b>	x	x	x			X	X	x		
<b>2<sup>nd</sup> layer</b>		x	x						x	
<b>3<sup>rd</sup> layer</b>				x	x					x

CALM standard resolves alternative access the path switching by vertical system de-composition to the individual subsystems for each communications access solution, however, management remains exclusively in the horizontal layers architecture. The relevant information needed for qualified decisions (incl. potentially of those from layers IHL and CL) are between layers shared exclusively via the control system structures. The details of CALM architecture are described e.g. in [15] - [23]. CALM applies exclusively still not widely spread IPv6 protocol which allows due to its extensive abilities to continuously remotely trace active applied alternative. Handover is in CALM accomplished exclusively on the L2 of the TCP(UDP)/IP model, i.e. out of TCP/IP competences. Handover competences given to this L2 is suitable way how non TCP/IP based system can be integrated into “CALM” architecture.

## 7.2 The IEEE 802.21 standard

The IEEE recently accepted standard 802.21 with the goal to support handover in the heterogeneous networks, i.e. *Media-Independent Handovers* (MIH). The standard enables mobile both users and operators to effectively use the advantage of overlapping and diverse access networks. It supports efficient networks discovering and executing heterogeneous handovers managed based on identified service capabilities and current link conditions.

IEEE 802.21-2008 offers the broad set of properties and tool that meet the requirements of effective heterogeneous handovers. It allows transparent service continuity during handovers by mechanisms of gathering and distributing of information from the various link types to a handover decision maker. The collected information includes consistent notifications about changes in the link conditions and available access networks. Scope of IEEE 802.21-2008 is restricted to access technology-independent handovers. Intra-technology handovers, handover policies, security mechanisms, media-specific link layer enhancements, Layer 3 (L3) and upper-layer enhancements are outside the scope of IEEE 802.21-2008.

### 7.2.1 The IEEE 802.21 Reference Model

IEEE 802.21 facilitates a variety of handover methods, including both *hard handovers* and *soft handovers*. A hard handover (break-before-make handover) typically suddenly switches between two access points or base stations or PoAs. Soft handovers initiate the new PoA connection process while traffic is still operated via serving PoA.

The main design elements of IEEE 802.21 can be classified into three categories: a framework for enabling transparent service continuity while handing over between heterogeneous access technologies; a set of handover-enabling functions; and a set of *Service Access Points* (SAPs).

#### 7.2.1.1 Transparent Service Continuity

IEEE 802.21 specifies a framework that enables transparent service continuity while a mobile node switches between heterogeneous technologies. The consequences of a particular handover need to be communicated and considered early in the process and clearly, before the handover execution. In soft handovers, it is crucial that service continuity, during and after the handover, is ensured without any user intervention. IEEE 802.21 specifies mechanisms to gather all necessary information required for an affiliation with a new access point before breaking up the currently used connection. In the case of hard handovers signaling preparation can initiate the connection context transfer from the serving PoA to the target PoA beforehand.

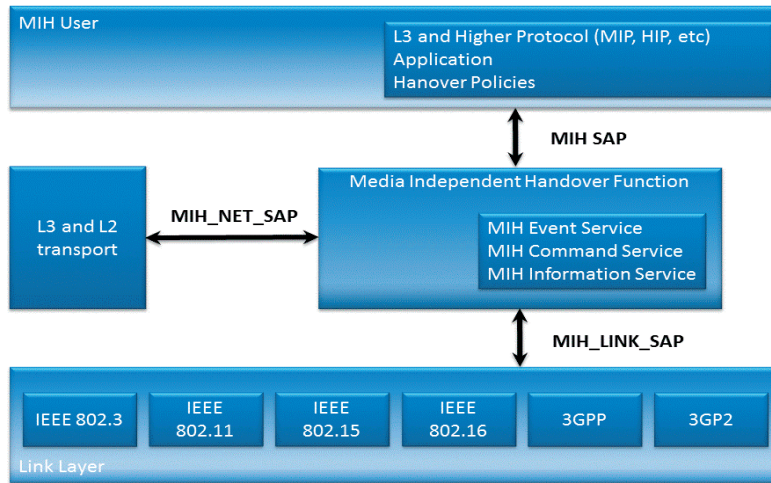
#### 7.2.1.2 Handover-Enabling Functions

IEEE 802.21 defines a set of handover-enabling functions specified with respect to existing network elements in the protocol stack, and introduces a new logical entity called *Media-Independent Handover Function* (MIHF). The MIHF logically resides between the link layer and the network layer. It provides alert services to entities residing at the network layer and above, called *MIH Users* (MIHUs), however, primary role of the MIHF is to assist in handovers and handover decision making by providing relevant information to management entities. MIHUs are projected to provide handover and link-selection decisions.

#### 7.2.1.3 Service Access Points

SAPs positioned between the MIHF and MIHUs (MIH\_SAP) provides MIHUs access to the following services MIHF:

- The *Media-Independent Event Service* (MIES) - event reporting about, for example, dynamic changes in link conditions, link status, and link quality. Events can be both local and remote. Remote events are obtained from a peer MIHF entity,
- The *Media-Independent Command Service* (MICS) supports MIHUs to manage and control parameters related to link behavior and handovers. Commands can be both local and remote. The information obtained with MICS is dynamic,
- The *Media-Independent Information Service* (MIIS) supports MIHUs to receive static information about the characteristics and services of the serving network and other available networks.



**Figure 7.3 IEEE 802.21 reference model**

Figure 7.3 describes IEEE 802.21 general reference model. There is needed definition of a technology-dependent interface of the corresponding media type - *MIH\_LINK\_SAP*. *MIH\_LINK\_SAP* enables MIHF to receive timely and consistent link information and control link operation during handovers. Currently supported link layers include wired and wireless media types from the IEEE family of standards (for example, 802.3, 802.11, 802.15, and 802.16), as well as those defined by the *Third-Generation Partnership Project* (3GPP) and *Third-Generation Partnership Project 2* (3GPP2). Besides these, IEEE 802.21 specifies a media-independent SAP (*MIH\_NET\_SAP*), which provides transport services for Layer 2 (L2) and Layer 3 (L3) MIH message exchange with remote MIHFs. Functions over the *LLC\_SAP* are not specified in IEEE 802.21.

### 7.2.2 Media-Independent Event Service

Events indicate or predict changes in the state and transmission behavior of physical, data link, and logical link layers. In general, events are triggers for initiating candidate network discovery and handover procedures. Depending on their origin in IEEE 802.21 events are categorized as either *Link Events* or *MIH Events*. The link events originate from the link layers, whereas MIH events originate from the MIHF and can be both remote and local. The local events propagate from lower layers to upper layers through the MIHF. Remote events occur at the protocol stack of another network entity and are transmitted from a peer MIHF to the local MIHF.

### 7.2.3 Media-Independent Command Service

The *Media-Independent Command Service* (MICS) enables higher layers to control the stream of events originating from lower layers, i.e. from MIHUs (MIH commands) or from the MIHF (Link commands) and the destination can be the MIHF or any lower layer, respectively. MIHUs can determine the status of different links in a uniform way, and control each interface accordingly. MICS defines the following set of commands:

- *MIH commands* typically used for network selection and handover management allowing the upper layers to initialize, prepare for, and execute handovers. MIH commands are also used to configure custom thresholds for link parameters.
- *Link commands* originate from the MIHF are sent to lower layers in order to control their operation. Link commands can be issued only locally. Nevertheless, Link commands can be executed on behalf of local MIHUs to process information received from a remote peer.

### 7.2.4 Media-Independent Information Service

The *Media-Independent Information Service* (MIIS) facilitates handovers through a unified set of tools that the MIHF can use to discover and obtain static (or rarely dynamic) information about networks. In other words, MIIS allows mobile nodes to check for available networks in range while using their currently active access network. MIIS information exchange occurs at the link layer (Layer 2) or network layer (Layer 3), so that all necessary information related to link layer or higher-layer services is collected before a mobile node authenticates with a new PoA. MIIS defines a set of *Information Elements* (IEs) in three groups:

- General Information and Access Network-Specific Information about neighboring networks
- PoA-Specific Information - reports location and addressing information, supported data rates, PHY and MAC layer types, and channel parameters that can optimize link layer connectivity. Some additional information from higher-layer services and individual capabilities of particular PoAs may be included as well.
- Other Information like network specific details.

The types of information handled by MIIS are exclusively related to handover decisions and conformance to the affiliation with the new PoA. The information relevant for assessing candidate networks by the handover machinery includes the connection establishment details, such as PoA address and location; which security mechanisms are supported in a given access network; and what QoS guarantees can be provided.

### 7.2.5 Service Management

IEEE 802.21 defines three service management functions:

- *MIH capability discover* uses the MIH protocol at Layer 2 or Layer 3, and media-specific Layer 2 broadcast messages are allowed; *MIH\_Capability\_Discover* broadcast messages and MIHF can also send *MIH\_Capability\_Discover* request messages using multicast or unicast to detect peer MIHFs in a solicited way
- *MIH registration* represents symmetric procedure by which two peer MIHFs authenticate and can then communicate with each other in a more trusted manner
- *MIH event subscription* enables MIHUs to subscribe to a particular set of events provided by MIES from the local or peer MIHF.

### 7.2.6 Media-Independent Handover Protocol

The *Media-Independent Handover Protocol* (MIHP) defines the rules and services for unified communication between the peer MIHFs. This protocol defines the message format, header, and encoding format and it is only used for communicating with peer MIHF entities. For internal communication no particular encoding is dictated.

### 7.2.7 MIH Communication Model

The MIHF communication model specifies different MIHF roles and their communication relationships and MIHF roles depend on their location in the network. For example, an MIHF on a mobile node can communicate directly with network-side entities called *MIH PoSs* using Layer 2 or Layer 3 communication. MIH PoSs may include the serving PoA or candidate PoAs. Network-side MIHFs can communicate with others at Layer 3 or above using the MIH protocol, introduced in the previous section.

### 7.2.8 Handover Execution

Actual handover execution is outside the scope of the standard. After the target is chosen the new connection is established while still routing traffic through the currently serving network. This process capitalizes described IEEE 802.21 services. The presented overview of the IEEE 802.21 Media-Independent Handover Services standard foresees that the standard IEEE802.21 adoption can principally improve network resource usage and permit multi-access devices to select the network access best suited for the communication needs based on information provided by described standard. The significant efforts in further amends and extends to 802.21-2008 to provide extended portfolio of services are expected. Work on defining a security-related extension to IEEE 802.21-2008. IEEE P802.21a WG has been already opened as well as IEEE P802.21b WG will take specific care of downlink-only handovers. Till now, however, it is not clear whether vendors will incorporate this standard in their future products and solutions.

## 7.3 Alternative approach based on the “intelligent routing”

Same authors propose an implementation of the decision processes in a TCP/IP environment. It represents routing process, which is used in the specific configuration and settings. The system tools that support handover action based on the results of identification processes have to be implemented. This approach minimizes the

need to the immediate investment in the development of the individual CALM type of modules/element with the specific requirements on their architecture including ability to be centrally managed. Proposed solution was originally designed as a "temporary" solution, but it will be applied until the availability of the CALM philosophy based systems are not resolved, implemented, produced and available on market in reasonable pricing. CALM approach to the multilayer management system resolves a problem of contra-productive processes done on different layers. The situation in an "intelligent routing" is different. The number of available parameters from different layers are very limited due to the fact, that, if commercially available modules or chip are applied such sub-systems usually do not offer open API (application Interface) where all available parameters could be available in a reasonable structure, i.e. equivalent of MSAP defined in CALM architecture and possibility to manage such parameters via open interface are not frequently available, as well. Communication of all critical internal parameters, as well as their management (minimum monitoring and control) in real time is needed, to avoid counterproductive simultaneous processes. In some implementations such functionalities are significantly minimized. Some solution like publically available services GPRS, EDGE, UMTS, etc., as well as for example Mobile WiMax (IEEE 802.16 e) operated by central service central management and only limited impact from terminal side can be expected.

It means that the concept of decision processes will not typically reach the full range of system information and possibility of intervention into centrally managed public systems system can be drastically limited. Additionally, it is necessary to take into account that there will be combined different telecommunications concepts "tuned" by interests of different services providers and from system obtained parameters might not have different the same representation. Above that all aspects available public services will have key position in application spectra due to their service provisioning coverage and decision systems have to combine limited available system parameters information (sometimes with very limited tools to influence service quality) with the quantification of indirect service quality tools to decide if the provided service is still appropriate or service quality is identified as critical to be changed to alternative one, of course – if it is available!

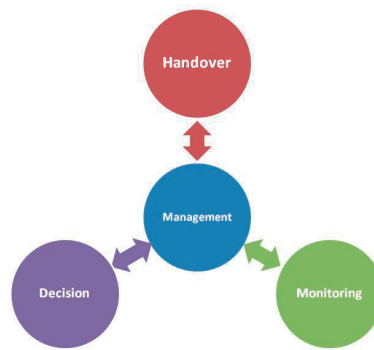
### **7.3.1 The DOTEK**

Decision processes representing basis for adaptability of communications wireless services have not been deeply enough resolved issue in CALM standards. We can identify recommendation based on PBM (Policy-based Management). This concept has been traditionally applied in the IP based networking and we can only state its remarkable success.

Above mentioned L3 routing based on "deterministic decisions" was applied in project of the communication module for transport telematics - DOTEK. "DOTEK approach" ensures the best wireless access service selection from the set of available wireless services and it is based on system parameters benchmarking derived from the telematic application requirement

The main objective of the DOTEK project is motivated by the "CALM principles". However, the main difference to CALM standards is implementation of routing principles on L3 layer replacing switching on L2 in case of CALM implementation. DOTEK project focuses mainly on the following areas:

- Analysis and selection of available wireless services applicable for different studied transport telematics services.
- Design of global and comprehensive management of these services including decision algorithm for selection of optimal data transfer technology.
- Provisioning of the continuous monitoring and evaluation of given services quality necessary for the correct decision to select appropriate service.
- Realization of the decision in order to ensure proper operation of telematics applications.



**Figure 7.4 General system architecture**

An important part of communication module is to monitor the current system parameters and communication technologies in order to assess their current situation and decide about their suitability for use according to the specific requirements of telematic applications. Telecommunication technologies are described by the system parameters like:

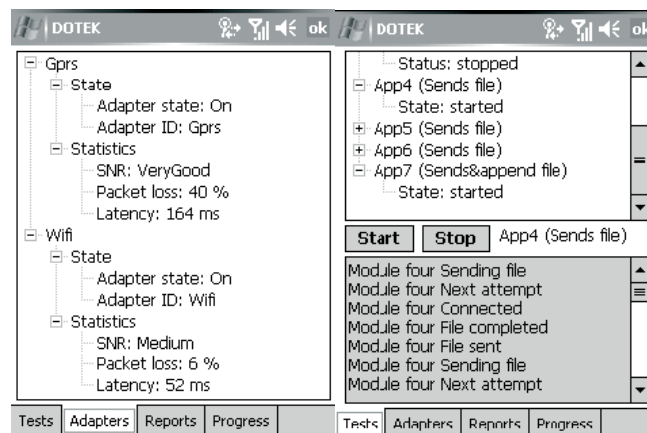
- availability,
- delays (latency),
- packet/frames loss,
- signal to noise ratio (SNR),
- received signal strength indication (RSSI),
- bit error ratio (BER),
- security level,
- etc.

For the final implementation in first working sample were chosen basic three monitored system parameters:

- signal to noise ratio,
- packet latency,
- packet loss.

For the further implementation it is possible and appropriate to include the other system parameters.

The implemented decision algorithm supports an appropriate access wireless service selection. It is based on the specific application requirements of the telecommunication service described by a set of parameters value range. The current status of available telecommunications technologies must be continuously available. The cost of each applied access wireless telecommunication service is taken into account, as well.



**Figure 7.5 Screenshots from testing application**

The decision to implement the described simplified approach was made based on evaluation of currently available research R&D manpower resources. The full adaptive version described below was out of implementation capacity and allocated resources, however, the only “extended PBM approach” was applied.

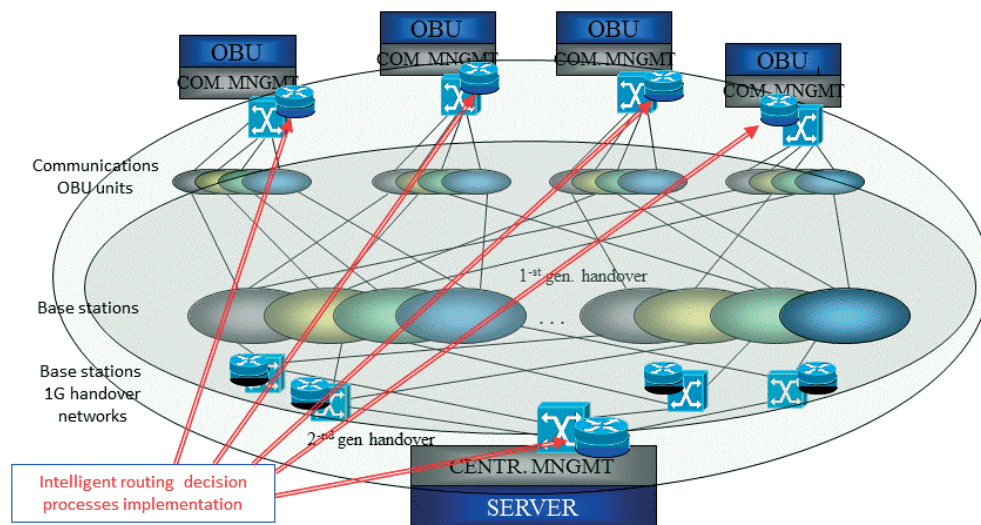
Presented system successfully passed the test scenarios to verify its basic functionalities, i.e. primarily time needed to analyze necessity to find relevant alternative path and to process complete second generation handover. Results are available in the table 2.

The DOTEK project was successfully developed and integrated in a universal vehicle OBU (On Board Unit) tested with four telematic applications – EFC (Electronic Fee (Toll) Collection), fleet management, e-Call and “intelligent” navigation. The solution is exclusively materialized in the SW package with no additional HW support required. This package has got modular structure and therefore it is practically technology independent and it can be integrated into existing systems to provide management of telecommunication technologies supported by these systems. The presented results prove that this system is fully functional and from presented handover times tests results it is clear that this approach can be applied in quite wide range of the telematics applications.

**Tab. 7.2 Results of test scenario focused on the time of handover**

Test No.	Handover time [ms]
1	208
2	137
3	41
4	108
5	362

### 7.3.2 Generalized DOTEK architecture



**Figure 7.6 Second generation of handover on L3.**

As a response on an urgent need of acceptable solution authors proposed alternative approach based on “intelligent” L3 routing operated in specific configuration and settings - see Fig. 3. This solution is understood as the interim alternative before full implementation of either CALM or IEEE802.21 is not implemented and available in reasonable pricing.

### 7.4 Adaptive decision processes

The decision processes are not discussed as frequently as it is with switching structures, even though implementation of the effective decision processes represents definitely the core of the issue. Due to the lack of communication abilities between managerial centers of the lower layers (usually closed system) and the 3-layer of the adaptive communication access system management decision of the top layer must be usually done based on insufficient or no information available from the lower layers. Mostly the only “macro” performance indicators of the whole communication chain are available. That is why authors concentrate afford on this area.

One of potential approaches to the decision processes is studied by authors and it is based on following principles:

- Measured parameters are processed by Kalman filter. Such process separates the reasonable part of present noise and also allows prediction of the individual parameters “near future” behavior.
- The set of measured parameters is extended by deterministic parameters like identification communicated with tall collection or economical parameter.
- The final set of data lines parameters vectors represent the basis for self-trained decision process. The best path selection algorithm processes available “cleaned” statistical data combined with the deterministic ones and decides based on “historical” training experience continuously improved by simultaneously processed self-training (see e.g. [66]). In the initial stage “minimal” high quality training data must be available for the first decisions.

Due to self-training the approach proposed solution does not strictly require 2HL layer communication with the other layers. Nevertheless, it would be more efficient solution if such communication between layers is at least partially available in future implementations.

Following paragraphs describe one of approaches to the decision processes, which are much less discussed than the switching approaches and their management. Proposed methodology is based on the following principles:

- Measured parameters are processed by Kalman filter. Such process separates reasonable part of noise and also allows prediction of the individual parameters near future behavior.
- Set of measured parameters extended by deterministic parameters like e.g. economical criteria is together available as vector  $x$ .
- Based on time lines of vector  $x$  it is feasible to classify the best possible technology selection. Classification algorithm is trained using time lines of training vectors  $x$  and relevant selected paths.

This solution does not necessarily require 2HL communication with the other layers, nevertheless, it will be much more efficient if such communication is at least partially possible in future implementations.

#### 7.4.1 Estimation and prediction of measured performance data vector $p(n)$

Let us define parameter vector  $p(n)$  in the time interval  $n$ . We will suppose that the dynamic of parameter  $p(n)$  evolves based on the following model (it is supposed that  $p(n-1)$  is known):

$$p(n) = A(n)p(n-1) + b(n) + q(n) \quad (15)$$

where  $A(n)$  is a transition matrix,  $b(n)$  is the deterministic vector of constant parameters and  $q(n)$  is the vector of Gaussian noise with the following property:

$$\begin{aligned} E[q(n)] &= 0, \\ \text{cov}[q(n), q(i)] &= 0 \text{ for } n \neq i \\ \text{cov}[q(n), q(i)] &= Q(i) \text{ for } n = i \end{aligned} \quad (16)$$

The equations (15) and (16) represent "evolution form of unknown parameters vector". In many cases we cannot measure the vector of an unknown parameter  $p(n)$  directly, however, we can measure another vector  $z(n)$  that depends on unknown parameters as follows:

$$z(n) = D(n)p(n) + r(n) + w(n) \quad (17)$$

where  $D(n)$  is a transition matrix,  $r(n)$  is a deterministic vector of constant parameters and  $w(n)$  is the vector of Gaussian noise with the following property:

$$\begin{aligned} E[w(n)] &= 0 \\ \text{cov}[w(n), w(i)] &= 0 \text{ for } n \neq i \\ \text{cov}[w(n), w(i)] &= W(i) \text{ for } n = i \end{aligned} \quad (18)$$

The equations (17) and (18) represent "evolution form of measurement vector". The algorithm for estimation of a vector  $\hat{p}(n)$  of unknown parameters together with its covariance matrix  $S(n)$  can be summarized:

$$\begin{aligned}\hat{\mathbf{p}}(n) &= \hat{\mathbf{p}}_e(n) + \mathbf{H}(n)(\mathbf{z}(n) - \mathbf{r}(n) - \mathbf{D}(n)\hat{\mathbf{p}}_e(n)) \\ \mathbf{S}(n) &= \mathbf{S}_e(n) - \mathbf{H}(n)\mathbf{D}(n)\mathbf{S}_e(n)\end{aligned}\quad (19)$$

where  $\hat{\mathbf{p}}_e(n)$  is an extrapolated estimate from the last step,  $\mathbf{S}_e(n)$  is a covariance matrix of extrapolation and  $\mathbf{H}(n)$  is Kalman gain. All the mentioned parameters are possible to be recursively computed from the last estimated parameters characterized by  $\hat{\mathbf{p}}(n-1), \mathbf{S}(n-1)$  according to the form:

$$\begin{aligned}\hat{\mathbf{p}}_e(n) &= \mathbf{A}(n)\hat{\mathbf{p}}(n-1) + \mathbf{b}(n) \\ \mathbf{S}_e(n) &= \mathbf{A}(n)\mathbf{S}(n-1)\mathbf{A}(n)^T + \mathbf{Q}(n) \\ \mathbf{H}(n) &= \mathbf{S}_e(n)\mathbf{D}(n)^T (\mathbf{D}(n)\mathbf{S}_e(n)\mathbf{D}(n)^T + \mathbf{W}(n))^{-1}\end{aligned}\quad (20)$$

Equations (20) and (21) are understood as "Kalman filtering algorithm". Now, we suppose the non-linear evolution of an unknown parameter vector (15) and a measurement vector (17) through known non-linear functions  $f(\cdot)$  and  $h(\cdot)$ :

$$\mathbf{p}(n) = f(\mathbf{p}(n-1)) + \mathbf{b}(n) + \mathbf{q}(n) \quad (21)$$

$$\mathbf{z}(n) = h(\mathbf{p}(n)) + \mathbf{r}(n) + \mathbf{w}(n) \quad (22)$$

The main idea is to linearize the equations (21) and (22) with the help of the first two components of Taylor series in extrapolated value  $\hat{\mathbf{p}}_e(n)$  (extended Kalman filtering):

$$f(\mathbf{p}(n-1)) \approx f(\hat{\mathbf{p}}_e(n)) + \frac{1}{2} \cdot \frac{\partial f(\mathbf{p})}{\partial \mathbf{p}} \bigg|_{\mathbf{p}=\hat{\mathbf{p}}_e(n)} \cdot (\mathbf{p}(n-1) - \hat{\mathbf{p}}_e(n)) \quad (23)$$

$$h(\mathbf{p}(n-1)) \approx h(\hat{\mathbf{p}}_e(n)) + \frac{1}{2} \cdot \frac{\partial h(\mathbf{p})}{\partial \mathbf{p}} \bigg|_{\mathbf{p}=\hat{\mathbf{p}}_e(n)} \cdot (\mathbf{p}(n-1) - \hat{\mathbf{p}}_e(n)) \quad (24)$$

Based on the equations (23) and (24) non-linear equations (21) and (22) are transformed into a linear form and Kalman filtering could be used. Kalman filtering can be started by the first measurement  $\mathbf{z}(1)$ . The initial parameters should be set up as:

$$\begin{aligned}\hat{\mathbf{p}}(1) &= \mathbf{H}(1)(\mathbf{z}(1) - \mathbf{r}(1)) \\ \mathbf{H}(1) &= (\mathbf{D}(1)^T \mathbf{W}(1)^{-1} \mathbf{D}(1))^{-1} \mathbf{D}(1)^T \mathbf{W}(1)^{-1} \\ \mathbf{S}(1) &= (\mathbf{D}(1)^T \mathbf{W}(1)^{-1} \mathbf{D}(1))^{-1}\end{aligned}\quad (25)$$

#### 7.4.2 Path selection as classification process

Let us introduce the vector  $\mathbf{x}$  as the vector carrying information about the values of performance parameters in sample time. The items of vector  $\mathbf{x}$  are either deterministic or random processes with help of Kalman filtering described above.

Let us define the classification problem as an allocation of the feature vector  $\mathbf{x} \in \mathbb{R}^D$  to one of the  $C$  mutually exclusive classes knowing that the class of  $\mathbf{x}$  takes the value in  $\langle \Omega = \{\omega_1, \dots, \omega_C\} \rangle$  with probabilities  $P(\omega_1), \dots, P(\omega_C)$ , respectively, and  $\mathbf{x}$  is a realization of a random vector characterized by a conditional probability density function  $p(\mathbf{x} | \omega)$ ,  $\omega \in \Omega$ . This allocation means the selection of best-fitted telecommunication technology based on knowledge of  $\mathbf{x}$  vector.

A non-parametric estimate of the  $\omega$ -th class conditional density provided by the kernel method is:

$$\hat{f}(\mathbf{x} | \omega) = \frac{1}{N_\omega \cdot h_\omega^D} \cdot \sum_{i=1}^{N_\omega} K\left(\frac{\mathbf{x} - \mathbf{x}_i^\omega}{h_\omega}\right), \quad (26)$$

where  $K(\cdot)$  is a kernel function,  $h_\omega$  is a smoothing parameter for  $\omega$ -th class,  $N_\omega$  stands for sample count in class  $\omega$  and  $\mathbf{x}_1^\omega, \dots, \mathbf{x}_{N_\omega}^\omega$  is the independent training data. The density estimate defined by (27) is also called the Parzen window density estimate with the window function  $K(\cdot)$ .

It is a well-known fact that the choice of a particular window function is not as important as the proper

selection of smoothing parameter. We use the Laplace kernel defined by the following Laplace density function:

$$f_L(x, \mu, \sigma) = \frac{1}{2 \cdot \sigma} \cdot \exp\left(-\frac{|x - \mu|}{\sigma}\right), \quad (27)$$

where  $x \in R, \mu \in R, \sigma \in (0, \infty)$ .

The product kernel is used with a vector of smoothing parameters  $\mathbf{h}_\omega = (h_{1\omega}, \dots, h_{D\omega})$  for each class  $\omega$ . The product kernel density estimate with Laplace kernel is then defined as

$$\hat{f}(\mathbf{x}|\omega) = \frac{1}{N_\omega} \sum_{i=1}^{N_\omega} \prod_{j=1}^D \frac{1}{2 \cdot h_{\omega,j}} \exp\left(-\frac{|x_j - x_{i,j}^\omega|}{h_{\omega,j}}\right). \quad (28)$$

Smoothing vectors  $\mathbf{h}_\omega$  are optimized by a pseudo-likelihood cross-validation method using the Expectation-Maximization (EM) algorithm [35].

To rank the features according to their discriminative power the standard between-to within-class variance ratio is employed. This method is based on the assumption that individual features have Gaussian distributions. The feature vector  $\mathbf{x} \in R^D$  takes value to one of  $C$  mutually exclusive classes  $\Omega = \{\omega_1, \dots, \omega_C\}$ . The probabilistic measure  $Q_{d,i,j}(d, \omega_i, \omega_j)$  of two classes separability for the feature  $d$  ( $d$ -th component of feature vector) is defined as

$$Q_{d,i,j}(d, \omega_i, \omega_j) = \frac{\eta \cdot (\sigma_i + \sigma_j)}{|\mu_i - \mu_j|}, \quad (29)$$

where  $\omega_i$  and  $\omega_j$  are classes and symbol  $\eta = 3.0$  denotes the real constant specifying the interval taken into account (probability that observation of normally distributed random variable falls in  $[\mu - 3.0 \cdot \sigma, \mu + 3.0 \cdot \sigma]$  is 0.998). The smaller the value of the measure  $Q_{i,j,d}$ , the better is separation of the inspected classes made by the feature  $d$ . For  $Q_{i,j,d} < 1$  both classes are completely separable. The measure is similar to the widely used Fisher criterion.

For multi-class problems, the two-class contributions are accumulated to get a  $C$ -class separability measure  $Q(d)$  for the feature  $d$ :

$$Q(d) = \sum_{i=1}^C \sum_{\substack{j=1 \\ i \neq j}}^C Q_{d,i,j}(d, i, j). \quad (30)$$

All the features in the training data are then sorted according to their  $Q(d)$  measures. The function  $Q(d)$  is similar to the significance of the  $d$ -th component of the measured feature vector. The subset of  $n$  first features is selected as an output of this individual feature selection method. The drawback of the method is the assumption of unimodality, in fact that just linear separability is taken into account. On the other hand, the individual feature selection method based on the between-to within-class variance ratio is very fast.

Presented classification approach is effectively applicable for relevant decision processes used to select the best possible alternative access from the set of available paths. The decision can provide evaluation of both random as well as deterministic processes and introduced approach enables continuous decision processes parameters training.

The presented method allows solutions implementations with limited information flows between telecommunications scheme layers. However, it is opened for any future changes of information resources. Such changes can lead to the principal decision processes parameters improvement. Due to this the self-training procedure of the new information resources integration is smooth and relatively simple.

It is important to stress that optimized number of the representative key performance indicators can lead to the significant reduction of required CPU capacity.

## 8 Conclusions of section Telecommunication systems for ITS solutions

Due to a regular complexity of telematic services and demand on size of the covered areas (wide area coverage, several classes of services with different system requirements) it is feasible to combine wireless access solution designed as seamless switched combination of more independent access solutions of the same or alternative technology with terrestrial backbone service to interconnect vehicles with central control system.

Terrestrial solutions are usually off-shelf available products. However, a designer must carefully calculate the impact of backbone performance on the whole telematics solution where dominantly the mobile access solution parameters are taken into account.

The public available GSM data services are designed to provide dominantly voice services. Selection of data services (2.5th generation) is provided with very limited or mostly no guaranteed performance parameters, i.e. as a complementary service in the best effort regime. Namely due to economical reason the 3rd generation (UMTS) mobile data services have not got potential to grow in appropriate way and service coverage is preferably concentrated on highly populated areas. Beyond 3rd generation solutions (namely LTE) are very promising future solutions, however, massive availability of these services cannot be expected sooner than in 2013 - 5 or even later.

The strong potential is so recognized in a combination of GSM data services with the alternative products provisioning namely if effective sharing of the GSM providers' infrastructure is reached. Alternative services are dedicated to fill the services gaps which cannot be provided by the core wireless network (continuously or in critical time periods, only). One of the most promising alternatives has been represented by technology based on standards IEEE 802.16d/e known as (Mobile) WiMax. Such access solution has been tested in areas, where served capacity and namely quality of GSM data services are recognized as not reasonable. WiFi services originally adopted as the low end of Internet access solutions dynamically grow in applicability in the "professional" solutions. Alternative services combination strategy with implementation of sophisticated decision processes (intelligent routing) can effectively extend potential of the widely spread GSM data services application substituted by the alternative solution where or when needed.

The decision processes representing the basis for adaptability of the communications wireless services are quite rarely resolved and published. Most of present implementations are based on Policy-based Management (PBM). Implemented "Extended PBM based" decision processes were presented as well as principle parameters describing system behavior acceptable for wide range of transport telematics applications.

Complex and flexible enough solution is, however, based on application of Bayes statistics and classification algorithms. A set of measured parameters can be so flexibly extended by deterministic parameters like economical parameter, corporate policy etc. Based on self-trained classification processes it is feasible to select the best possible alternative i.e. assigning data vector to one of set of classes. Classification algorithm is trained using time line of training data vectors extended by correct assignment to the relevant class, i.e. selected path.

The optimized number of the representative key performance indicators can, however, principally reduce requirement on CPU capacity. That is the reason why the detailed study of each applied telecommunications technology has been accomplished in our laboratory to identify specific representative key performance indicators for each technology potentially applied in the system.

The solution is materialized in the SW with limited or no requirements on available HW implementations. It is modularly structured with aim to be practically technology and operational system independent. It can be linked into existing systems structures with remarkable potential to provide effective management of the telecommunications alternatives being supported by the systems.

Described solution called DOTEK represent simplified solution successfully developed and integrated in the universal vehicle OBU. Implementation was tested with different telematic applications like EFC, fleet management, e-Call and "intelligent" navigation and reasonable parameters were reached. It was identified as applicable for wide range of telematic services. Additional improvement of the adaptive system behavior is expected in moment decision algorithms implementation based on described classification principles is completely finalized.

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