Mobile Networks

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Explanatory notes

	Definition
	Interesting
	Note
	Example
	Summary
+	Advantage
–	Disadvantage

ANNOTATION

The module contains information about basic principles used in different generations of mobile networks. Further there are described characteristics of different technologies implemented in the mobile networks (GSM, CSD, HSCSD, GPRS, EDGE, UMTS, LTE-(A), HSDPA, HSUPA, etc.).

OBJECTIVES

Explanation within the module depends on the second generation mobile systems GSM, i.e. digital cellular systems. The student is clearly acquainted with the principles of mobile networks, specifically with the functional organization of cellular networks and principle of operation of their individual parts.

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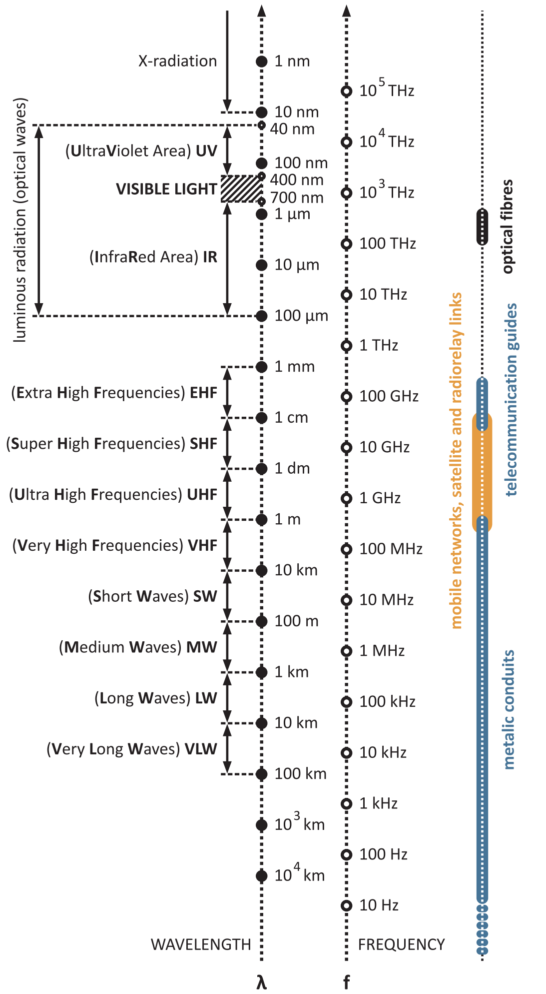
1 Introduction

1.1 Radio Transmission Paths

Electromagnetic waves, which can be effectively propagated on specific frequencies through free space, are used for implementation of radio transmission. A suitable group of electromagnetic waves is called as radio waves. Radio waves can be divided on the basis of their usage into these several categories (see the following list and figure):

- Radio Band (carriers from hundreds of kHz up to tens of MHz) – **LW** (*Long Waves*), **MW** (*Medium Waves*), **SW** (*Short Waves*) and **VHF** (*Very High Frequencies*)
- TV Band (carriers from tens up to hundreds of MHz)
- Mobile Networks and Microwave Band (carriers in order of ones of GHz)
- Satellite Links, Radio Relay Links and Broadband Wireless Access Networks (carriers up to tens of GHz)



Frequency bands utilized by radio systems

Radio systems may either complement the fixed or wired access resources, especially wherever it is convenient, or you can use them to create a distinct mobile and fixed network providing a wide portfolio of services.

The most economical benefit of radio systems is especially achieved in "hard to reach" areas and locations with sparse population where it is not cost-effective or suitable to build a cable network. The wireless network offers high flexibility, which can be temporarily used in case when the customer requires the connectivity to the network infrastructure immediately (for example concerts, meetings, etc.).

1.2 Basic Classification of Radio Resources

The radio resources can be distributed by a number of different sights. The radio resources can be classified for example according to:

- frequency band – narrowband radio systems and wideband radio systems
- direction of transmission – unidirectional radio systems and bidirectional radio systems
- configuration of radio systems – point-to-point and point-to-multipoint
- mobility of subscriber – fixed wireless local loop or mobile terminal
- used transmission resources – terrestrial links or satellite links
- licensed bands vs. unlicensed bands (quite essential for planning and creating of wireless networks)

According to portfolio of provided services, the radio resources and networks can be classified to voice, data, etc.

Other ways of the radio resources classification are for example according to:

- frequency band
- used modulation and coding scheme
- method used for sharing of the bandwidth (**FDM** (*Frequency Division Multiplex*) and **TDM** (*Time Division Multiplex*))
- access to shared resources (**FDMA** (*Frequency Division Multiple Access*), **TDMA** (*Time Division Multiple Access*), **CDMA** (*Code Division Multiple Access*) and **OFDMA** (*Orthogonal Frequency Division Multiple Access*)

Finally, we can also split the radio systems into public and private, etc.

1.3 Overview of technologies

For completeness and integrity of our initial review, it would be certainly more appropriated to introduce a comprehensive analysis of wireless technologies that are closely related to the topic presented in this module.

The first category of radio communication technologies can include analogue and digital cordless systems that complement, and in some cases completely replace, traditional telephones. This category undoubtedly includes **CT** (*Cordless Telephone*) system in versions CT0, CT1, CT2 and **DECT** (*Digital European Cordless Telephone*) system. The purpose of the above mentioned systems is to replace fixed wired subscriber line and enable the mobility within a limited area to the users. It is, therefore, in principle the most typical narrowband access system for the implementation of fixed wireless telephone network subscriber denoted as **WLL** (*Wireless Local Loop*).

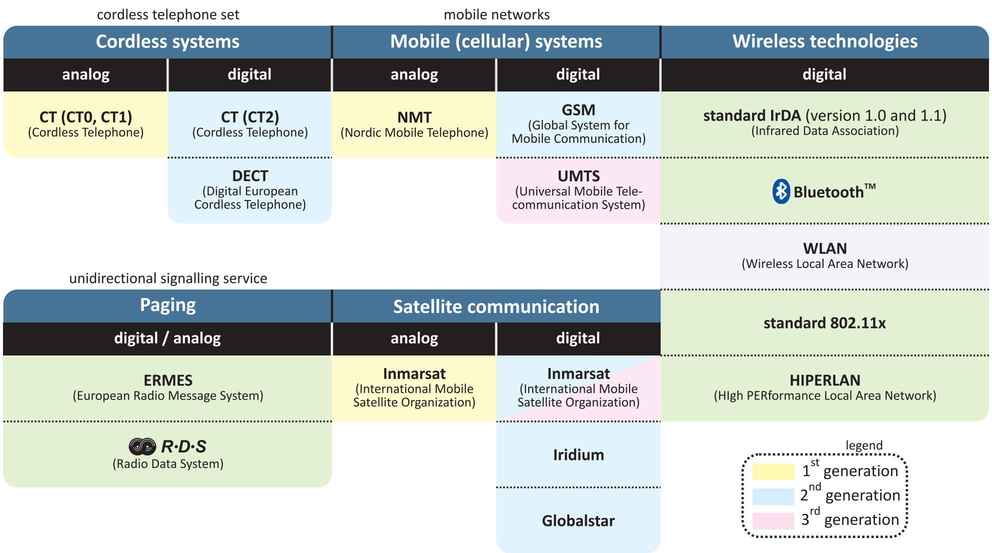
The second category is represented by both analogue and digital cellular (further in the module referred to simply as cellular) systems forming the infrastructure of distinctive mobile radio networks. As an example, the 1st generation systems (that are analogue) are denoted as **NMT** (*Nordic Mobile Telephone*). These were followed by systems of the 2nd generation (digital) like **GSM** (*Global System for Mobile Communication*), afterwards by systems of the 3rd generation known as **UMTS** (*Universal Mobile Telecommunication System*), and finally, by the systems of the 4th generation such as **LTE-A** (*Long Term Evolution-Advance*). These technologies will be described in detail in the following chapters.

As the third category, we can integrate wireless technology used to meet the requirements and needs of participants in the personal, local, metropolitan and wide networks by means of **PAN** (*Personal Area Network*), **LAN** (*Local Area Network*), **MAN** (*Metropolitan Area Network*) and **WAN** (*Wide Area Network*). These include standard **IrDA** (*Infrared Data Association*) in versions 1.0 and 1.1, then currently a dynamically developing Bluetooth ™ technology. To complete the survey we have to also indicate the **WLAN** (*Wireless Local Area Network*) technology based on 802.11x standard and **HIPERLAN** (*High Performance Local Area Network*), which is a European alternative for IEEE 802.11.

The fourth category can be combined with technologies that provide so-called paging. The term represents one-way radio systems providing radio contact services. The radio paging systems include European **ERMES** (*European Radio Message System*) or **RDS** (*Radio Data System*) global system.

The last and the fifth, but in its own way a little specific category of radio systems, are satellite communications. It is not an access system in the strict sense because its coverage includes a substantial portion of the earth's surface and next to intercontinental connections are of particular importance for marine and aviation coverage and inaccessible, sparsely populated areas. As a representative of this category can be mentioned analogue and digital systems such is **INMARSAT** (*International Mobile Satellite Organization*), **Iridium** or **Globalstar**.

The satellite positioning systems are a complementary category in our overview. This includes, in particular, the American **GPS** (*Global Positioning System*) system, the emerging European system named as **Galileo** and the Russian system known **GLONASS**. All these systems are abbreviated as **GNSS** (*Global Navigation Satellite System*). Categorized list of all technologies used for mobile communication is presented in Figure 2.



Overview of technologies for mobile communication

Further we will describe the digital cellular systems in details in the following chapters.

2 Mobile Telecommunication Networks

2.1 Introduction

Besides fixed wired networks, the mobile networks provide wireless connection and data transmissions. In addition to radio parts, the mobile network also includes the whole core network and fixed infrastructure to enable delivery of all kinds of services. With every new introduced generation, their capabilities are still growing and thus meet increasing requirements. Today, mobile networks utilizing digital concepts are the only alternative for the near and distant future.

GSM is a digital mobile system representing the second generation of mobile systems and it can be characterized as a digital cellular mobile radio telephone system.

The upcoming third generation of mobile systems is known as UMTS. These digital systems operate in the 2 GHz bands and integrate different wireless access technologies currently in a massive infrastructure capable to offer a wide range of multimedia services with guaranteed quality.

The fourth generation is denoted as LTE-A. It focuses on meeting further increase in demands of users to transmission rates and low delay for various set of services.

Generation of mobile systems

Generation	Name/Abbreviation	Characteristics
1^{st} generation (1980 to 1995)	**NMT** (*Nordic Mobile Telephone*); FIN, S, N, DK **AMPS** (*Advanced Mobile Telephone System*); USA **TACS** (*Total Access Communication System*); UK, IRL RADIOCOM 2000; FR	Analogue systems National systems VOICE

2^{nd} generation (since 1992)	GSM (Global System for Mobile Communication) DAMPS (Digital AMPS), resp. IS136; USA PCS 1900 (Personal Communication System); USA PDC (Personal Digital Communication) GPRS (General Packet Radio Service); denoted as 2.5 generation EDGE (Enhanced Data rates for Global Evolution); denoted as 2.75 generation	Digital systems VOICE + DATA
3^{rd} generation (since 2004)	CDMA 2000 (1×EV-DO, 1×EV-DV) **UMTS** (*Universal Mobile Telecommunication System*) **HSPA** (*High Speed Packet Access*), HSPA+; denoted as 3.5 generation **LTE** (*Long Term Evolution*); denoted as 3.9 generation	MULTIMEDIA
4^{th} generation (not available)	**LTE-A** (*Long Term Evolution-Advanced*)	MULTIMEDIA

2.2 Cellular Mobile Telephone Networks

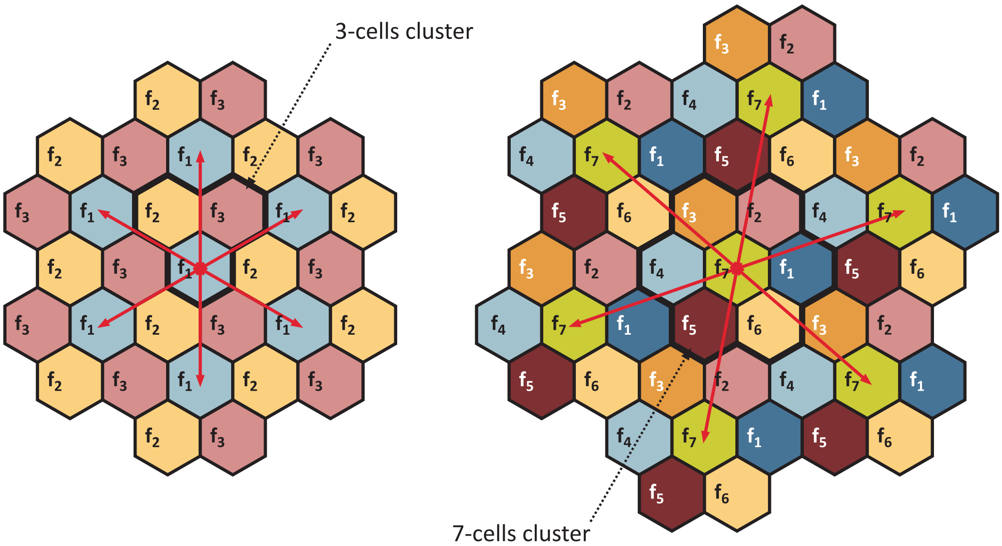
Mobile phones connection can be enabled by radio telecommunication resources and their operation usually follows the operation of fixed telephone networks. The final assembly consists of:

- framework of fixed **BS** (*Base Station*)
- **MS** terminals (*Mobile Station*)

One of the most fundamental principles applied in modern mobile telecommunication systems is based on dividing of the operated territory into partial elemental areas called as cells, which are always served by a particular base station.

The size of cells used in various mobile systems depends primarily on the type and purpose of the mobile system and can be classified as follows:

- femtocell (flats or offices) - intended for cover areas with low quality of signal from other cells, usually supposed to be indoor with radius of several meters.
- picocell (office and residential environment) - signal range is from tens up to few tens of meters.
- microcell (urban areas with dense housing) - focused mainly to the slower moving participants (for example to a car in city traffic or pedestrians), coverage within a single cell is up to few hundreds of meters.
- macrocells (large and sparsely populated areas) - primarily oriented for the high speed moving participants (for example, vehicles on the roads), diameter of the macrocells is maximally up to few kilometres.
- satellite cell (area accessible by telecommunications satellite) - allows the connection in locations inaccessible for the previous cell types, signal range is dependent on the position of satellites, respectively on their orbit and the parameters of the transmitting and receiving device.



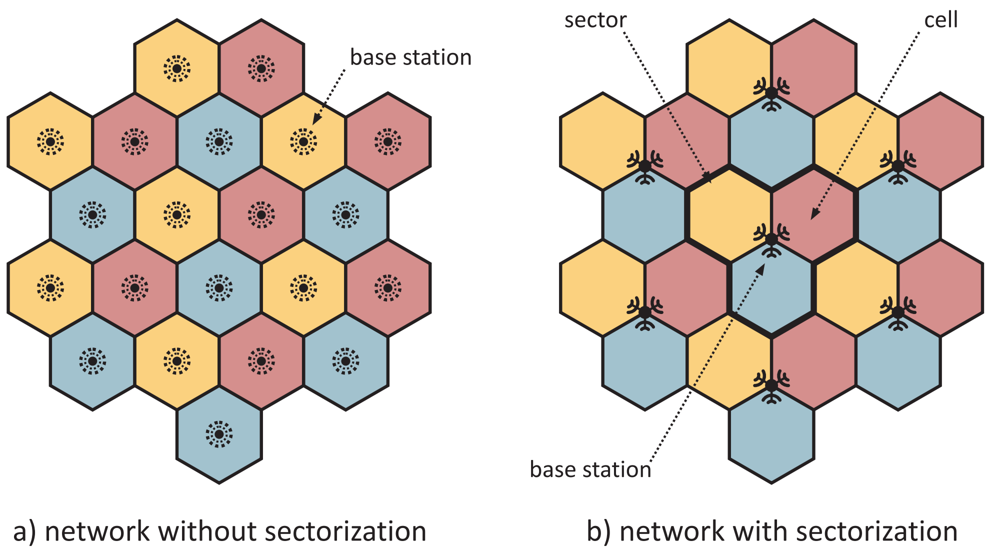
Distribution of operated territory into cells

Cellular structure of the GSM network is usually created by using macrocells with a diameter of up to few kilometres. Example of radio coverage of the territory based on the cellular principle is shown in previous figure. For cellular mobile network structure, a frequency planning is necessary. Frequency plan is working with three or seven frequencies. The same frequencies (f_1 to f_3, or f_1 to f_7) can be used in any cluster. The area of all three or seven cells cluster is approximately equal to the average interference zone.

2.3 Principle of sectorization

In the previous chapter you have learned that each cell of any mobile network is operated by one particular base station. However, for large area covered by mobile networks, this concept is not appropriate, especially in terms of a high number of base stations. Nevertheless, this number can be significantly reduced by using the principle of sectorization.

Let us now divide a single cluster in the previous figure into 21 smaller cells (see figure below - section a)). The number of available channels is not so changed, but there is increased number of base stations to 21. However, we can significantly reduce the number of base stations by principle of sectorization to 7. This could be accomplished under condition that the individual base stations are not placed in the centres of cells, but in the intersection points of three adjacent cells forming one sector (see figure below - section b)).



Principle of sectorization of cellular network

Three separated directional antennas with three transmitters and receivers will be used for each of these seven stations. The number of base stations in this case is the same as in the previous figure (section b)) with the distribution of the service area into cells, but configuration of a network is much more efficient due to operating characteristics (for example lower transmission power and increasing the number of mobile stations which can be simultaneously served).

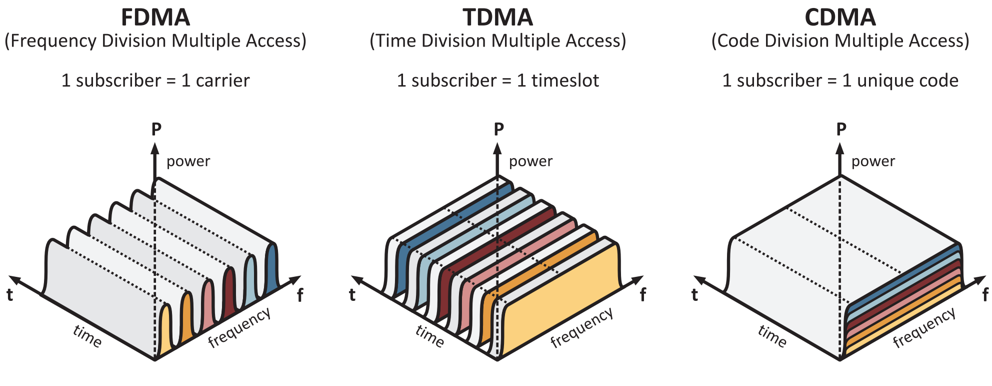
Small cells (with range of approximately 10 up to 500 m) will be necessary to use in the areas with high density of users. In areas with lower density it is sufficient to deploy cells with larger radius (with range approximately 1 up to 10 km) and for very lightly loaded areas can be the cell diameter even few kilometres.

2.4 Access Methods

In each cell it must be ensured that the connection between one base station and a higher number of mobile stations in each period of time might be established and realized. For this purpose the methods of the multiple access are used.

A dedicated frequency band for a given radio system can be accessed by one of the following methods:

- FDMA divides the allocated frequency band into the subchannels and then assigns these subchannels to each communication channel
- TDMA creates in a particular frequency subchannel the sequence of time slots and then its individual slots allocates to each communication channel on the principle of time multiplex
- CDMA processes data sequence on each side of the transmitting channel by coding process through unique coding policy, which is deliberately different from the coding policy of all the other channels. Thus, the signals of each channel can be transmitted in the same frequency band, which is without time resolution. The communication channels are distinguished from each other on the receiving side on the basis of a unique coding policy, which was used for the coding at side of transmitting channel
- OFDMA is a combination of time and frequency division multiple accesses. The available resources are split into subcarriers in a frequency domain and also into several time intervals in time domain. Then individual users have assigned not only one or several subcarriers but also a time interval for communication.



Multiple access methods

Above mentioned basic access methods are very often combined in practical applications (for example FDMA and TDMA).

The assigned frequency bands can be accessed in either *Time Division Duplex* (**TDD**) or *Frequency Division Duplex* (**FDD**) way. In case of TDD, the transmission in downlink (transmission to user) and in uplink (transmission from user) directions are separated in time domain but, the same frequency is assigned to both transmission direction. This means that the frequency is better utilized. On the other hand, the issue here is the high requirements on synchronization. In case of FDD, both transmission directions are separated in frequency, but the transmission/reception is simultaneous. The main disadvantage of this approach is that two antennas, one for receiving and one for transmitting, are necessary to be used. In most of countries, FDD duplex is utilized for its less complex implementation

2.5 Principle of automatic reconnection

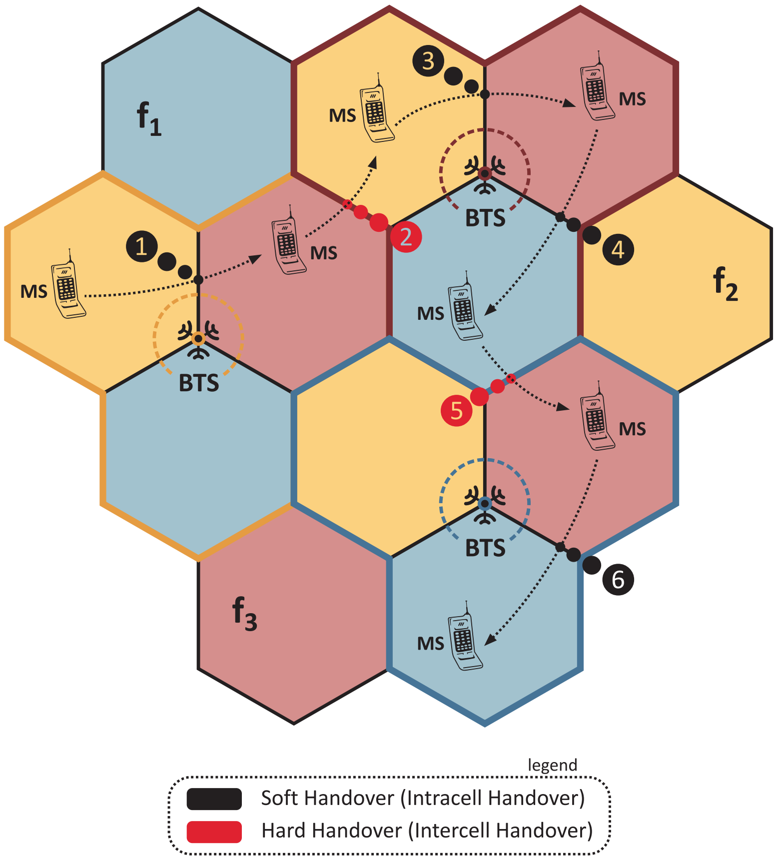
A mobile station communicates always with the nearest base station, more precisely with a station that provides the strongest signal for the mobile station. If the mobile station moves into a neighboring cell, it is automatically switched to the base station of the neighboring cell.

This operating mode is referred to as a handover.

The handover manages a change of a current serving station to a proper target station during user's movement across the cells boundaries. The major purpose of handovers in mobile networks is to ensure continuous connection with high QoS or balance load in network.

Tracking of current position of mobile station in the network enables continuous automatic connection established between the mobile and base station. This indication is stored in the mobile network registers, which allows routing of connection to the called party directly into the area where the station is currently located.

On the basis of frequency planning communication channels with different frequencies in the adjacent cells are always used. When the mobile subscriber moves across the border lying between two different cells, it is always necessary to retune user's mobile station.



Principle of handover

Basically, two types of handovers can be distinguished:

- Hard handover - if the hard handover is performed, a MS firstly closes all connections with the current serving station. As soon as the connections to the serving station are terminated, new connections with the target station are established. Therefore, this type of handover is also known as break-before-make since a short interruption in communication between the MS and the network is introduced. The duration of the handover interruption depends on the management message flow exchanged between the MS and the network. Thus the length of the interruption depends on several factors such as used wireless technology (e.g., LTE, or WiMAX), physical layer frame length, or network load. In general, the duration of the interruption varies from tens to hundreds of milliseconds in networks.

- Soft handover - enables simultaneous connection of a MS to several BSs. Consequently, no handover interruption is observed by users during communication. This handover is also known as make-before-break. The soft handover can be realized as a **MDHO** (*Macro Diversity HandOver*) or a **FCS** (*Fast Cell Selection*) also denoted as a **FBSS** (*Fast Base Station Switching*). In MDHO, the macro diversity combining of signals received from several BSs included in active set (in WiMAX denoted as diversity set) is performed. The significant drawback of this approach is high complexity and complicated implementation. In the case of FCS, the best frame simultaneously received from all stations included in active set is selected and processed. Even if the implementation is simpler comparing to MDHO, it is still essentially more complex than in the case of the hard handover. Therefore, the hard handover is considered as mandatory in mobile networks while other types of handovers are optional.

3 GSM Mobile Network – mobile network of 2nd generation

3.1 Fundamentals of GSM system

At present, GSM is still widely used digital cellular system. This system was built as a European open standard and its deployment has allowed to solve the international roaming, i.e., the operation of the same mobile station with one phone number in all countries that adopt this system.

An important element of GSM system is an identification system based on unique subscriber card called as **SIM** (*Subscriber Identity Module*).

The SIM card contains not only basic identification data, but also many other specific individual information, such as subscriber identification number, authentication keys, information on prepaid phone services or the participant list. Mobile station can then be used only with the activation of the specific operator's card. Nonetheless, the exceptions are emergency calls that can be performed without SIM card. Coding and encryption of transmitted information, which significantly complicates the possibility of eavesdropping is other important aspect from a user perspective.

Switching process begins in the moment when the active call begins to send the initial signalling data. One of the most important initial processes is to control of eligibility access of the mobile station to the network.

The MS sends its identification number **IMSI** (*International Mobile Subscriber Identity*) through BS and **BSC** (*Base Station Controller*) to exchange **MSC** (*Mobile Switching Centre*). The block **AuC** (*Authentication Centre*) sends towards the MS random number that is converted on the basis of individual data and algorithms of SIM card to another different number that is as original response sent back to the MSC exchange. Subsequently individual data are compared with the data of the database located in the VLR block. If there is a consensus then the mobile station has allowed access into the mobile network.

To ensure anonymity, the subscriber station is communicating with assigned provisional identification number, known as **TMSI** (*Temporary Mobile Subscriber Identity*), under which the MS is identified in the MSC. When switching the mobile station to another MSC, it is re-assigned with another TMSI. Self-transfer of user data can only start when these processes are finished.

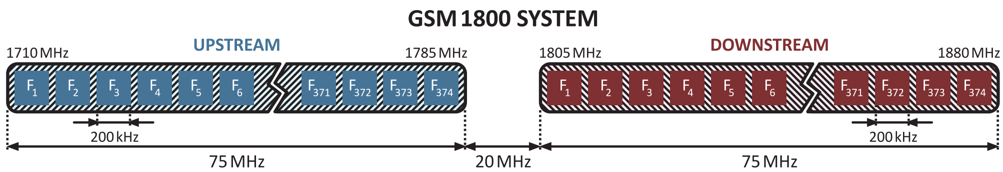
3.2 GSM system and its standards

Basic applications are implemented at 900 MHz band in GSM. Increase of traffic has led to development of other versions with multiple frequency bands. Thus, there are three standards, which differ primarily in used frequency range and number of allocated channels:

- GSM 900 – frequency band at 900 MHz, maximum capacity 2×124 channels, bandwidth of 2×25 MHz
- GSM 1800 – frequency band at 1800 MHz, maximum capacity 2×374 channels, bandwidth of 2×75 MHz
- GSM 1900 – frequency band at 1900 MHz, maximum capacity 2×298 channels, bandwidth of 2×75 MHz

The later versions of GSM system, that is GSM 1800 and GSM 1900, are sometimes referred to as **DCS** (*Digital Communication System* or *Digital Cellular System*) systems.

Allocation of frequency bands used by GSM 1800 in the European continent is shown in the following figure.



Frequency bands used by GSM 1800 systems and their separation

GSM 1800 system does not bring any major technological innovations but allows you to meet others subscribers interested in mobile communications, especially in cities. The system allows you to create either standalone network or network cooperating with the former GSM 900. Usually, there are combined macrocells of GSM 900 with the microcells of GSM 1800, which aim is to provide the services in areas with high concentrations of users (such as shopping centres or downtowns). However, the two-way (or dual mode) MSs have to be utilized in order to exploit both frequencies.

3.3 Services & Applications

GSM system enables the provision of telecommunications services (Teleservices) and transmission services (Bearer Services).

The telecommunications services may be classified into:

- telephony (including emergency calls and through roaming also in all other networks)
- message services such as **SMS** (*Short Message Services*) with the possibility to send a maximum of 160 characters between two points in both directions or with possibility to send the message to all mobile stations in the cell by means of **CBS** (*Cell Broadcast Service*) such as traffic or weather reports
- voicemail
- e-mail (service in connection with Internet electronic mail)
- bank services
- information services, etc.

Into the transmission services may be included in particular:

- duplex asynchronous data transmission with transfer rates ranging from 300 to 9 600 bps
- duplex synchronous data transmission with transfer rates ranging from 2400 to 9 600 bps

GSM services are continuously expanded and list of implemented services depends on network operator (provider). At present, these include increasing of the transmission rate up to 14.4 kbps, respectively with the changing of the encoding up to 21.4 kbps.

There are also a variety of services to increase participant comfort. For example, the immediate introduction of accounting services (Hot Billing) allows the use of prepaid cards and their subsequent service charge, and thus creates a group of anonymous users who do not pay flat monthly fees.

3.4 Architecture of GSM Network

The whole basic structure of the GSM system is shown in the figure below. The basic structure of the GSM system can be divided into three fundamental parts: Base Station Subsystem, Networks Switching Subsystem, and Operation Support Subsystem.

Base Station Subsystem (**BSS**)

The MSs communicate with BSs. Several BSs are assigned to a BSC, whose main task is to allocate and release the free radio channels for communication with MSs and ensure the correct handover. Operation of the system requires that each MS, which is in operation, provides the information to the system on its location, it means about the cell in which it is located. MS monitors also signals from nearest base stations and always selects the optimum BTS, through which the connection is established.

Network Switching Subsystem (**NSS**)

This subsystem includes in particular the MSC exchange, which is represented by a common type of telephone exchange that is supplemented by additional features resulting from the mobility of the MSs. These additional features are stored in various databases comprising:

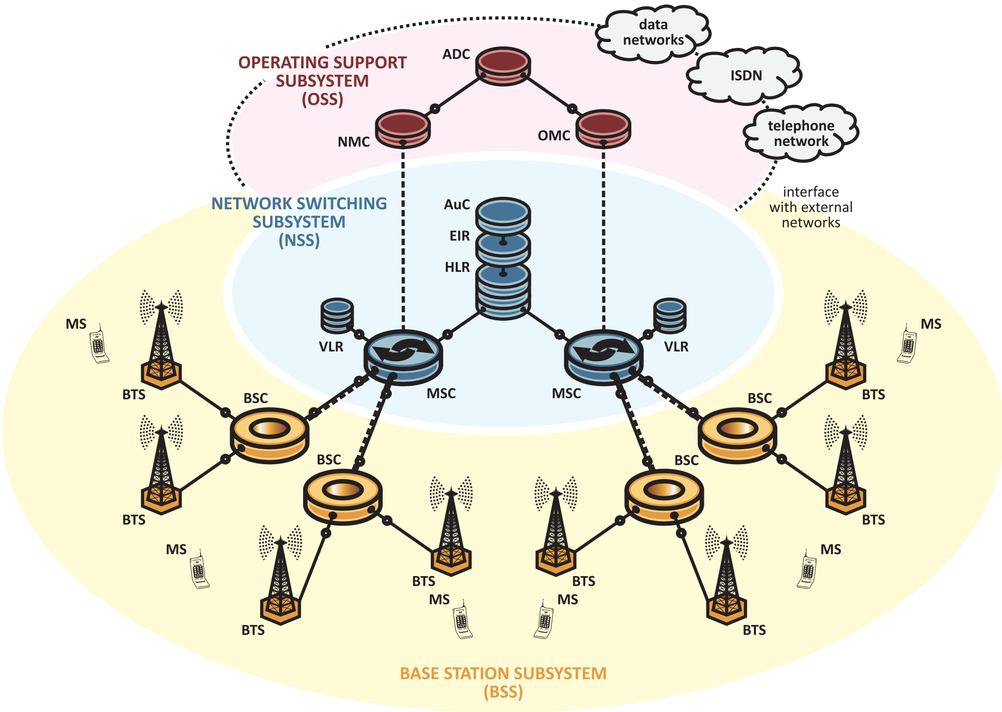
- **HLR** (*Home Location Register*) – keeps track of all participants in the area. Authentication (identification) of subscriber is provided by AuC. Each participant of the network is stored only in a single HLR.

- **VLR** (*Visitor Location Register*) – temporarily stores the latest information on position of the MSs in the range of the MSC. The VLR always requires and obtains data from the HLR and if the MS leaves the visited area, the MS's related data are always deleted from the VLR.

- **EIR** (*Equipment Identity Register*) - stores information about MSs (for example, list of authorized stations or stolen stations, etc.)

Operation Support Subsystem (**OSS**)

The OSS is responsible for the operation of BSS and NSS. It contains mainly supervisory block, **ADC** (*Administrative Centre*), addressing administrative tasks (e.g., report participation fees, billing, etc.), followed by a **NMC** (*Network Management Centre*) block providing overall management of information flow in the network, and operational and service block, **OMC** (*Operation and Maintenance Centre*), addressing the role of maintenance and operation of the network.

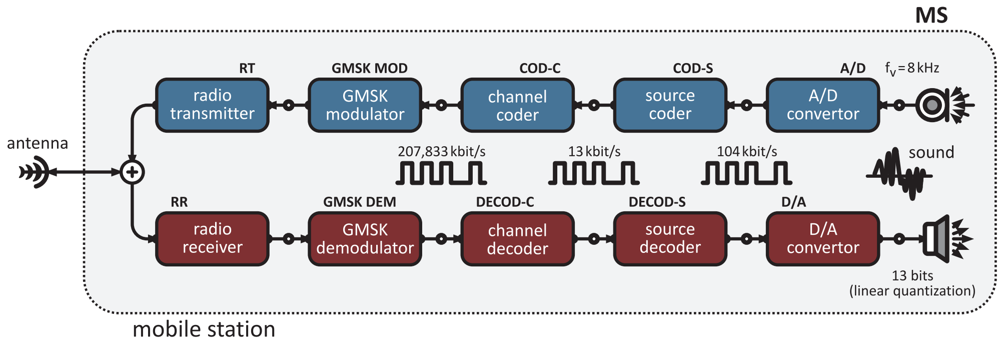


Architecture of GSM system

3.5 Structure and Functionality of Mobile Station

In the following two chapters the structure and operating principles of the MS and its coupling to the BS will be described.

The MS is composed of transmitting and receiving parts, a control microprocessor, SIM card and other accessories (handset, keyboard, display, etc.).



Block scheme of mobile station

The analogue voice signal on microphone output is digitized by A/D converter, which operates on the principle of PCM (8 kHz sampling rate, the sample of 13 bits, linear quantization, signal transmission speed $v_p = 8000 \times 13 = 104$ kbit/s). The next block, **COD-S** (*CODer of Source*), carries out speech synthesis on the principle of vocoder, also called as source coding and that is performed by **RPE** (*Regular Pulse Excitation*) excitation method, thereby reducing the transmission rate to a value $v_p = 13$ kbit/s.

In block **COD-C** (*CODer of Channel*) is implemented channel coding. This process ensures reduced risk of error generation in speech and protect digital signal against errors originating in the transmission. Additionally, there is interleaving, which purpose is to increase the signal robustness against bursts of errors. On the receiving side, the signal is inverted by regulation adopted interleaved sequence and subsequently it is transformed to the original sequence. If the several consecutive bits are received with errors (burst errors), these may be repaired by using implemented security processes.

In addition, part of the signal, which contains user information, is still subjected to the encryption process in order to prevent direct monitoring of the communications. After of all these modifications, transmission rate of one radio channel is $v_p = 22.8$ kbit/s.

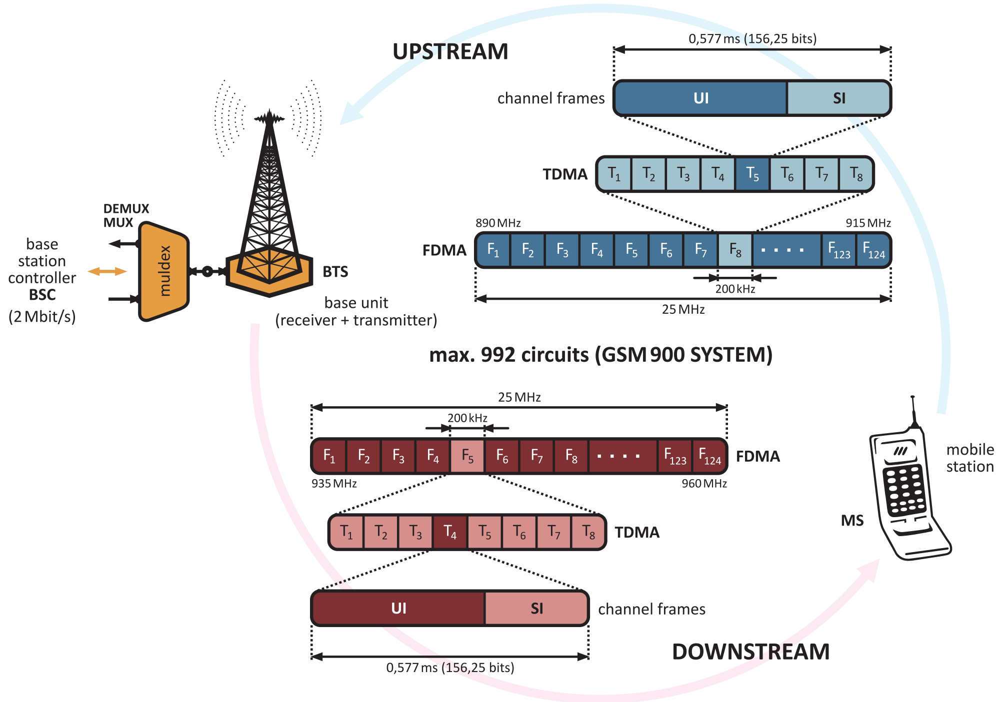
Afterward, the following sequence of integration of signal into **TDMA** frame is processed in **GMSK-MOD** (*Gaussian Minimum Shift Keying MODulator*) and the overall transmission rate is then $v_p = 270{,}833$ kbit/s (physical layer frames with length of 156.25 bits are transmitted every 0,577 ms).

After processing in other circuits of the *Radio Transmitter* (**RT**), the signal is radiated by the antenna of the MS. In terms of transmission power, the MSs are divided into five classes ranging from 0,1 to 20 W.

The signal transmitted from the antenna to the *Radio Receiver* (**RR**) block, is further processed in the demodulator (**DEM**) and both decoders (**DECODing-C**, **DECODing-S**). The D/A converter then transforms digital signal to analogue signal. Finally, the analogue signal is transferred into the handset.

3.6 Mobile Terminal and its Connection to Base Station

Two frequency bands with a width of 25 MHz with separation of 45 MHz are reserved for the transmission of signals in GSM 900. Signal in the direction from the BTS to the MS is transmitted in the frequency band from 935 MHz up to 960 MHz. The opposite direction is then carried in the band from 890 MHz up to 915 MHz. Access methods for the implementation of the radio circuit between the MS and BTS is based on a combination of **FDMA** and **TDMA** access methods. In both frequency bands, the band is divided into 124 sub-bands by FDMA (F_1 through F_{124}) with a nominal width of 200 kHz. In each frequency sub-band, the radio resources are then split into 8 time slots (T_1 to T_8) by using TDMA.



Principle block scheme of MS based on the standard GSM 900 and its relation to the BTS

Each channel frame (Burst) with a length of 156,25 bits and with duration of 0.577 ms contains bits including the user information (UI) and service information (SI). This means that the BTS can offer total capacity of up to 124 × 8 = 992 pairs of usable channels, it means, 992 radio circuits.

Synchronization is derived in the GSM from the clock generator with nominal frequency f_t = 13 MHz. The basic transmission rate is v_p = 13 000 000/48 = 270 833.33 bit/s. One TDMA frame (including 8 time slots) has duration equal to

4.615 ms (multi-frame is composed from 26 frames and its duration equals 120 ms).

3.7 Data Transmission in GSM Network and 2.5 Generation of Mobile Systems

Basic transfer rate between a MS and a GSM network is 13.2 kbps for voice call in both directions. This channel can be used not only for transferring calls but also for data transmission based on circuit switching, **CSD** (*Circuit Switched Data*), with nominal transfer rate of 9.6 kbps. Later, the transfer rate has been increased up to 14.4 kbps, mainly by reducing redundant information in the form of protective codes.

However, considerable increase in transfer rates is ensured by the systems belonging to 2.5 generation, either by using:

- transmission based on packet switching, known as **GPRS** (*General Packet Radio Service*), with the theoretical available bit rate up to 171 kbps or
- transmission based on circuit switching, known as **HSCSD** (*High Speed Circuit Switched Data*), with the theoretical available bit rate up to 115 kbps

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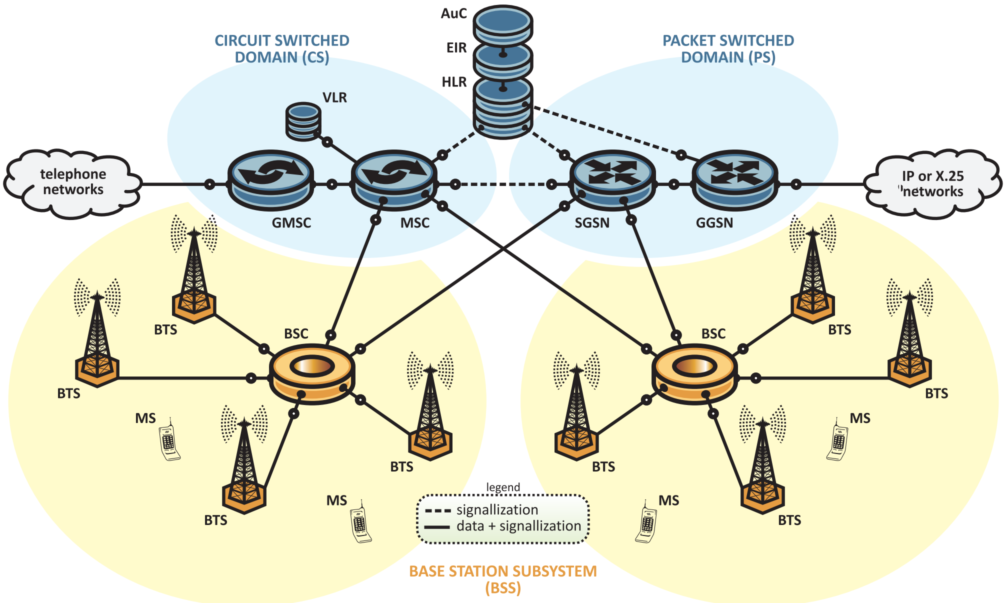
A further increase of transfer rates is possible by utilization of technique denoted as **EDGE** (*Enhanced Data for GSM Evolution*). Compared with conventional modulation in GSM system, there is used the modulation with a higher number of states, namely **8-PSK** (*Phase Shift Keying*). Theoretical achievable bit rate of data transmission using EDGE is 473.6 kbps.

Infrastructure of GSM network with 2.5G systems is supplemented by a data node, **SGSN** (*Serving GPRS Support Node*), which communicates with the radio part of GPRS network. For the data transfer to other packet networks, such as the Internet, a data gateway **GGSN** (*Gateway GPRS Support Node*) is subsequently implemented. The GGSN operates as a router. Simplified infrastructure of 2.5G mobile network can be seen in the following figure.

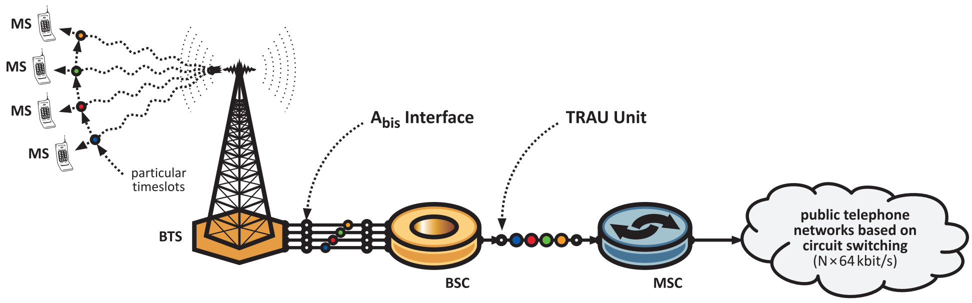


Simplified infrastructure of 2.5G mobile network

3.8 CSD Data Transmission in GSM Network

Digital cellular networks like GSM have been primarily developed for transferring of human voice. Nevertheless, even the human voice is transmitted in digital form. This makes it relatively easy to transfer general data instead of human voice. However, there are certain limitations associated mainly with the maximum achievable bit rates.

The single GSM radio channel has a maximum transfer rate of 33.8 kbps. Nevertheless for data transfer, the maximum available transfer rate is equal only to 9.6 kbps. The reason is that 11 kbps is used as service channel capacity and it is designed to ensure the functionality of the GSM network. The remaining capacity of 13.2 kbps is utilized to ensure the reliability of transmission, treatment errors and failures. This principle of data transfer is referred as CSD.



Data transfer by CSD

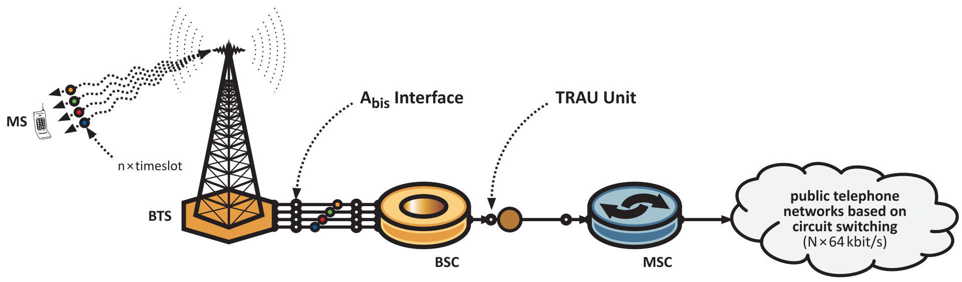
After some time it has been successfully tested, that the data transfers always do not require a highly robust protection mechanisms and it is possible to increase the available transfer rate for data transmission. Specifically, the data transfer rate has been increased to 14.4 kbps but in conjunction with the condition on availability of signal with high quality.

To increase the reliability of data transfer by adding bits to the transmitted data is not the only option. It is possible to use solutions based on the feedback between the sender and the recipient. If the recipient recognizes corrupted data, the sender transmits data again on the basis of request for retransmission. This process obviously requires that both sides agree on this procedure and it is therefore necessary to define the protocol. This protocol is called **RLP** (*Radio Link Protocol*).

3.9 HSCSD Data Transmission in GSM Network

As it was stated in the previous chapter, the maximum available data transfer rate that can be used within one time slot is 14.4 kbps.

The one way how can be further increased bit rate in an existing GSM network is to use multiple time intervals simultaneously for serving a user. This mode of data transmission is abbreviated as HSCSD. It is still based on principle of CSD but compared to conventional CSD it brings significant increase in transmission rate.



Data transmission through HSCSD

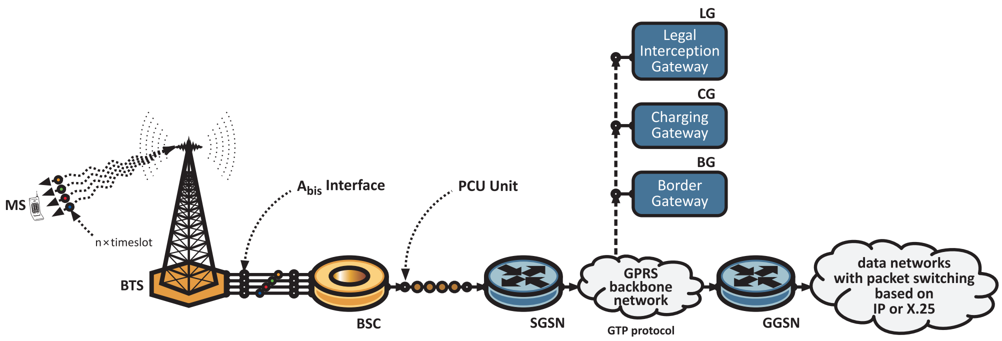
As it is shown in the previous figure, communication between the MS and the BS is performed through an associated group of time slots and transferred to the MS. The allocation of timeslots is dependent on the current number of available radio channels and on the capabilities of the MS itself. The data are transmitted by the BS and the BSC (A_{bis} interface) over channels with a bit rate of 16 kbps. Overall 32 channels with the transfer rate 64 kbit/s (i.e., 2 Mbps in total) are available for communication between the BS and BSC. Each of these channels is further divided into four subchannels with transmission rate of 16 kbps.

The channel with transfer rate of 16 kbps can be used to transmit voice (13 kbps) or common data (14.4 kbps). Data are sequentially merged into the standardized channel (64 kbps), which is done in the BSC in **TRAU** (*Transcoder and Rate Adaptation Unit*) unit. This unit is designed to convert encoded voice (with rate of 13 kbps) into a standard voice PCM channel (rate 64 kbps) or modify the data transfer rate to 64 kbps. Hence also the maximum transfer rate per one channel, which can be achieved using this technology, is 64 kbps.

Transfer, which is performed this way, can be in most cases asymmetrical. This means, for example, that in the direction from a MS to the network can be assigned three time intervals and in the direction to the MS can be assigned with only one interval. This method of capacity allocation of radio channel is used very often and it is suitable, for example, for connection to the Internet, where the data are transferred mainly from the network to the user. Standard, which defines the mode of HSCSD, divides the available modes up to 18 classes according to how many channels can be used for both directions.

3.10 GPRS Data Transmission in GSM Network

In order to increase the transmission rate in the original GSM network, which is solely circuit oriented, packet oriented transmission mode had to be introduced. For this purpose, it is necessary to supplement the original network by new equipments as it is shown in the figure below. This new network structure enables packet oriented transmission called as GPRS.



Data transmission through GPRS

Data node, SGSN, communicates with the radio part of GPRS network. For the data transfer to other packet networks, such as the Internet, a GGSN data gateway is used; this entity acts as a router. Access of user into defined networks is allowed through **APN** (*Access Point Name*). This way, the operator can enable access to given APN only for defined set of SIM cards and can thus to create a GPRS network for the private group of users, whose traffic is strictly separated from other traffic. From the information mentioned above, public data networks or private data networks can be created by this way. Transmitting data through the APN can be also a different way how to determine price for individual service of provider, such as **WAP** (*Wireless Application Protocol*) or **MMS** (*Multimedia Messaging Service*).

Access to the Internet is possible by WAP, which makes accessible the content of the web sites and information services on MSs through low-capacity channels and displays with limited resolution. The application of data communication and WAP allow utilization of GPRS (packet oriented transmission) and enhancement of transfer rates theoretically up to 192 kbps. However, the implementation of this service requires much more extensive and costly investments not only in the structure of the GSM network, but also in the MSs.

Within the GSM network with GPRS is already implemented cooperation between SGSN and BSC for efficient allocation of transmission resources, which means that transmission resources are allocated to the MS only if it has data to send or receive.

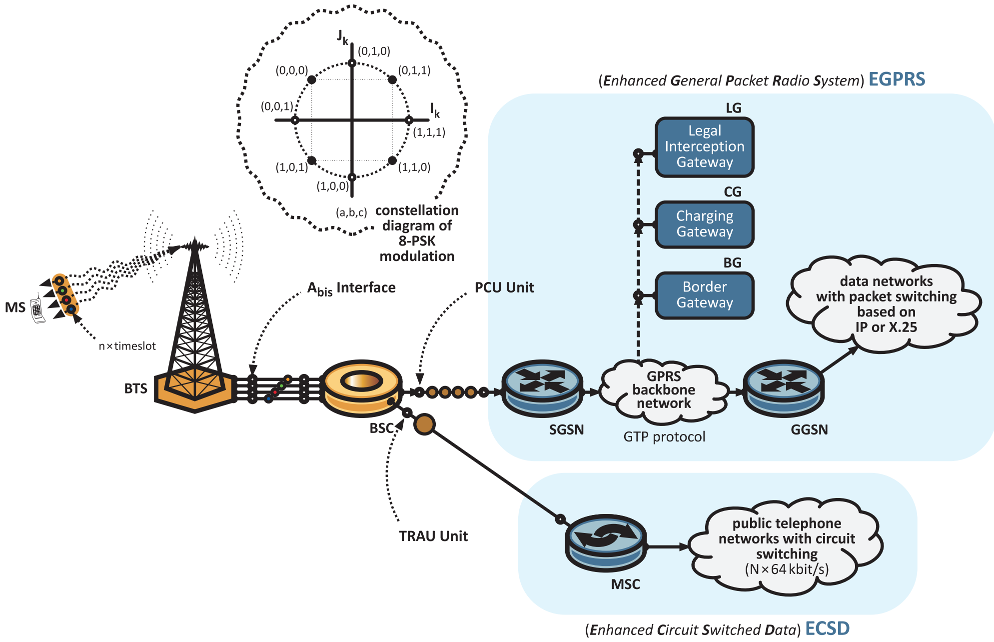
+ There are not permanently blocked transmission paths unlike in the conventional data transmission in GSM network using HSCSD.

Higher transfer rates of GPRS are enabled by association of more than one timeslots for a user and by selecting a suitable coding system for this channel. In this context, we refer to transmission using combination of 3+1, 4+1 or 4+2 timeslots (represents downlink + uplink). If asymmetrical data transmission is used in the GPRS, then the direction to the subscriber (downlink) is always of a higher transfer rate. The available transmission rates are strongly dependent on the specific location and BS load.

− A negative impact of delay can be observed in GSM network with GPRS. It is caused by data packets passing through packet oriented networks. The delay is strongly dependent on the length of packets. Short packets (up to 100 bytes) have the transmission delay from 0.5 up to 1 second depending on the network load. Moreover, packets with length of 1 kB can be delayed even few seconds.

3.11 EDGE Data Transmission in GSM Network

A data transmission technique in 2.75G of mobile networks, denoted as **EDGE** (*Enhanced Data Rates for GSM Evolution*), increases the network capacity and available transfer rates in the context of HSCSD and GPRS techniques. The improvement of data transmission through more potential coding or by utilization of more timeslots is no longer an option. The only possible method to further increase of the throughput is achieved by the utilization of more efficient modulation technique. Thus instead of the original **GMSK** (*Gaussian Minimum Shift Keying*) modulation, the EDGE uses eight level phase modulation, **8-PSK** (*Phase Shift Keying*),



Data transmission through EDGE

The symbol rate at one frequency channel with a width of 200 kHz is the same as before, i.e., 270.833 ksymbols/s. Nonetheless, by utilization of eight level modulation schemes, the bit rate is now three times higher comparing to the original binary modulation. The maximum theoretical reachable transfer rate is 473.6 kbps (59.2 kbps per time slot). However, this is available only if all 8 slots are employed and conditions for signal propagation are favourable.

Deployment of this technology requires implementation of a new type of transceivers to the network. This change must be applied in each cell, where new technology will be deployed. The next step is an adjustment related primarily to update its own software. Combination of EDGE technology along with GPRS and HSCSD technologies seems to be very convenient, especially for widespread transmission the new modulation method principle brings significant acceleration of data transfers. A common deployment arise principles called as **ECSD** (*Enhanced Circuit Switched Data*) and **EGPRS** (*Enhanced General Packet Radio System*). EDGE is generally considered as a last step in improvement of the original GSM system before the introduction of the third generation called as UMTS.

4 Universal Mobile Telecommunication System (UMTS)

4.1 Introduction

The second generation of telecommunication systems, such as GSM, enables easy transmission of voice traffic over wireless environment. Nevertheless, 2G networks are incapable to satisfy all data requirements as these are continually increasing and are driven by fast development of mobile applications demanding high data transmissions (video on demand, downloading of high quality images, etc.). To that end, the next generation of mobile networks, known as 3G, has been developed to meet these requirements.

In order to work toward global 3G mobile radio standard, the Third Generation Partnership Project (3GPP) had been founded in December, 1998. The 3GPP consists of members of the standardization bodies from all around the world such as Europe (ETSI), USA (ANSI) or Japan (ARIB). The 3GPP have been working on a common 3G radio standard, which is usually referred to as UMTS. Nonetheless, the 3G term is not applied on one particular technology or standard but it encompasses many technical specifications that have been gradually approved to enhance former UMTS in order to increase still increasing demands.

In the next chapters we will describe in more detail the frequency allocation, basic transmission principles and **WCDMA** (*Wideband Code Division Multiple Access*) access method adopted in UMTS. Further, evolution of 3G technology focusing especially on high speed data access is delivered. In addition, we will depict the services and applications used in 3G networks together with **QoS** (*Quality of Service*) classifications.

4.2 UMTS frequency allocation

The operational frequency bands of UMTS system is set to be around 2 GHz, which still ensures reasonable transmission characteristics, low signal attenuation and easy signal penetration to indoor environment. The assigned frequency bands depend whether UMTS works in so called *Time Division Duplex* (**TDD**) or in *Frequency Division Duplex* (**FDD).**

The UMTS can be used for operation with a minimum spectrum of 2 x 5 MHz for paired frequency band assigned for UMTS FDD and 5 MHz for unpaired frequency band dedicated for UMTS TDD. The assigned frequency bands are somewhat different for Europe and USA. In Europe, UMTS TDD has allocated frequency bands in region of 1900-1920 MHz for uplink and in region of 2010-2025 MHz for downlink. The UMTS FDD has allocated much wider bands due to the fact that FDD is supposed to be used preferably. Regarding the uplink direction, frequency bands varying from 1920-1980 MHz are allocated. On the other hand, frequency bands between 2110-2170 MHz are assigned for downlink.

The reason why uplink frequency bands are located at lower frequency than downlink is due to limited power and battery limitations of mobile terminals. The lower frequency bands are characterized by lower signal attenuation. Consequently, the coverage in uplink can be easily guaranteed while the transmission power of mobile terminals can remain at reasonable value. Note that maximal transmission power of base station is usually 43-46 dBm, while transmission power of mobile terminal is usually up to 23 dBm.

4.3 WCDMA

Since the TDMA and FDMA access at the air interface used in former 2G networks has not been sufficient for 3G networks, the UMTS adopted more sophisticated access method based on CDMA, explained in chapter 2.4. To be more specific, the WCDMA technique is utilized in UMTS networks. In WCDMA, user information bits are spread over a much wider bandwidth than in conventional CDMA case. This way, the capacity and data bit rates achieved by individual users in the system is increased. Similarly as in case of CDMA, user data are multiplied with quasi-random bits, called chips, derived from CDMA spreading codes. The UMTS uses the chip rate of 3.84 **Mcps** (*Mega chip per second*) and carrier bandwidth is equal to 5 MHz. The other distinguished characteristic of WCDMA based system is the frequency reuse factor 1. It means, the frequency used in all cells is the same (contrary to GSM, where each cell uses individual frequencies to minimize interference). Thus, radio resources are better utilized.

The utilization of WCDMA brings several important features into UMTS standard:

- Rake receiver – Radio channel propagation in mobile networks is characterized by multiple reflections, diffractions, and attenuation of the signal energy caused by obstacles such as buildings, hills, vegetation, etc. This results in so called multipath propagation and fading (strong signal attenuation at certain frequencies). WCDMA system combats these negative effects by rake receiver, which is able to effectively receive and combine individual signal's component

- Power control – One of the most important aspects regarding WCDMA is fast closed loop power control, on the uplink in particular. Otherwise, single user could block a whole cell if transmitting with high power. Consequently, in WCDMA based systems, the power received by all users should be the same at the base station. Note that in case of HSPA this is only partly true as explained later.

- Soft handover – Since the adjacent cells use the same frequency, the users can be simultaneously connected to two base stations (two different codes have to be used) at the same time. This result in handover between two cells without any interruption, also known as soft handover as already described earlier.

4.4 Standardization and UMTS evolution

The evolution of 3G networks had been initiated in 1999 by issuing the very first UMTS release known as Release 99. Since then several releases have been already approved as suggested in the following.

Release 99

This release is based on GSM network, thus UMTS is backward compatible with GSM and interoperation between those two is possible. In comparison to 2G networks, Release 99 brings completely new type of radio access network, known as **UTRAN** (*UMTS Universal Radio Access Networks*). The theoretical data bit rates achieved by this release are 2 Mbps in downlink and 384 kbps in uplink.

Release 4

Release approved in 2001 introduces several major changes to core network and GERAN. The main features are separation of transport bearer and control bearers in the **CS** (*Core Switched*) network, introduction of new interfaces in CS network, enabling of low chip rates and mainly, introduction of **IMS** (*IP Multimedia Subsystem*).

Release 5

The main improvement of Release 5 over former releases rests in introduction of **HSDPA** (*High Speed Downlink Packet Access*). The HSDPA increases the bit rates in downlink direction up to approximately 14 Mbps.

Release 6

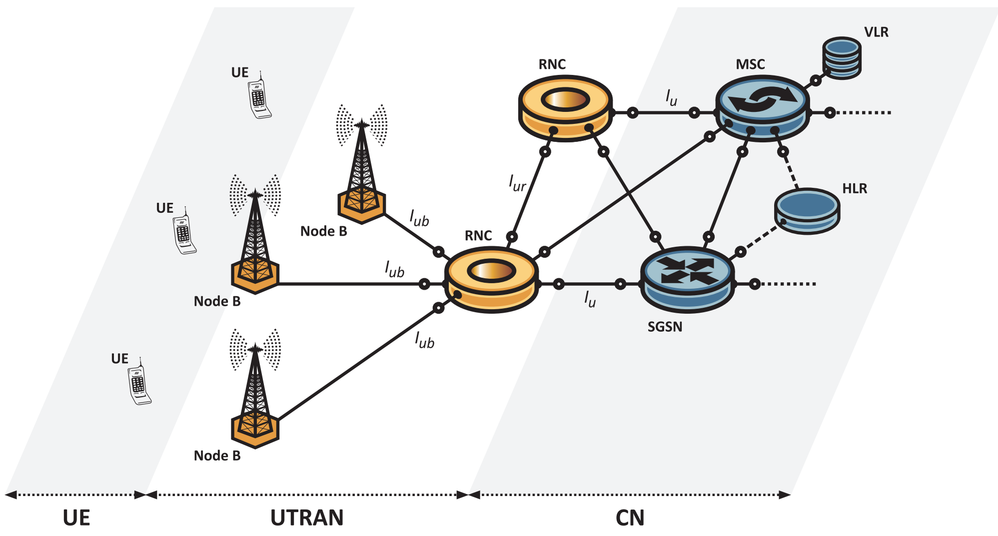
Similarly as previous release, Release 6 brings significant improvement in data transmission as it specifies **HSUPA** (*High Speed Uplink Packet Access*). The HSUPA enables bit rates in uplink direction up to 5.76 Mbps. In addition, significant work has been accomplished regarding IMS subsystem.

Release 7

In this release, HSPA+, also called as Evolved High Speed Packet Access, has been introduced. It further increases achievable bit rates both in uplink and downlink direction. The improvement is enabled by introduction of more effective modulation (64 QAM) and **MIMO** (*Multiple Input Multiple Output*) technique where more antennas at the receiver and the transmitter can be employed. Consequently, maximal theoretical bit rates are up to 42 Mbit/s in downlink and up 11.5 Mbit/s in uplink.

4.5 Network architecture

The UMTS network architecture is composed of three main parts as suggested in Figure 16. Note that the main blocks in network architecture remain the same also for newer UMTS releases; only their functions are gradually changed or enhanced. The UMTS network architecture is logically divided into three parts; **UE** (*User Equipment*), the UTRAN, and the **CN** (*Core network*). The individual parts are separated by interfaces defined by 3GPP standardization body. The main purpose of interfaces is to enable direct communication between individual entities to facilitate their optimal cooperation and coordination.



The structure of UMTS network

The first part of UMTS network architecture (i.e., the UE) includes two blocks:

- *Mobile Terminal* (**MT**) - Physical device handling all communication on Uu interface, i.e., mobile phone, laptop or PDA. The MT is the same device as Mobile Station (MS) defined in 2G networks.

- *UMTS Subscriber Identity Module* (**USIM**) – The card including user specific data such as identification, authentication algorithm, and other subscription information. Again, the USIM is analogical to SIM formerly used in 2G networks.

The second part of UMTS network architecture is represented by the UTRAN, which is the most modified when compared to 2G networks. The UTRAN consists of two network elements:

- Base station (NodeB) – The NodeB is analogical to the BTS in 2G network. To that end, the NodeB is composed of the pole, antenna part and necessary hardware and software. The main purpose of the NodeB is to enable the UEs to connect to the 3G network as it is connected by means of radio channel (i.e., via Uu interface). Hence, the NodeB control uplink and downlink radio paths and ensures radio channel functions.

- Radio Network Controller (RNC) – The RNC entity is analogical to BSC node formerly used in 2G. Similarly, the RNC is responsible for controlling of several connected NodeBs. The RNC and former BSC have several similar functions, such as responsibility for radio resources management (allocation of radio resources to the NodeBs, admission control, congestion control, etc.). Still, the RNC has larger responsibility when concerned with mobility management of individual users than the BSC. In 2G networks, the MSC and SGSN play more important role regarding mobility issues.

The interesting thing regarding the NodeB is the name selected for it. One can ask why exactly NodeB? In 2G networks (GSM), the base station is logically named as BTS, which stands for Base Transceiver Station. When the 3G network has been initially designed, individual entities in the network were referred to as Nodes, i.e., mobile equipment has been formerly known as NodeA, base station has been labeled as NodeB, etc., in order to differentiate names with 2G networks. All entities have been one by one renamed appropriately, but the name for base station remained as NodeB since no more suitable name had been found.

The last part of UMTS network is composed of the CN. The CN is logically divided into Circuited Switched (CS) and Packet Switched (PS) domains. Since the structure of CN is nearly the same as in case of 2G networks, only the main differences are listed here (detail description of individual network elements in CN can be found in section 3.4):

- As already mentioned, part of mobility management has been moved from the CN to the UTRAN in UMTS.
- Security features are enhanced in UMTS as new ciphering algorithms are adopted. The ciphering execution is performed in RNC instead of the CN.
- The speech handling function is done by CN instead of BSS as in case of 2G.

IP Multimedia Subsystem

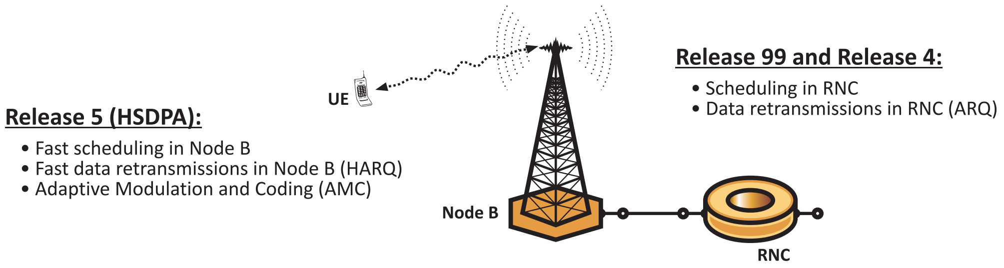
The important part of UMTS CN is an IMS. The IMS is important for flexible, rapid development and deployment of new 3G services. To be more specific, the IMS provides a standard framework for deployment of next generation IP-based application using wide range of integrated media, video, text, and data.

The IMS has been firstly introduced in UMTS Release 4 where its architecture has been separated from access network to provide independent service control. The UMTS Release 6 extends the IMS toward an access independent networks infrastructure. As a consequence, IMS can be integrated also with other access networks based GPRS or EDGE technologies.

The IMS standards adopted a Session Initiation Protocol (SIP). The SIP establishes IP connection between the UEs (e.g., for voice or video sessions) or IP connectivity between the UE and application servers.

4.6 HSDPA

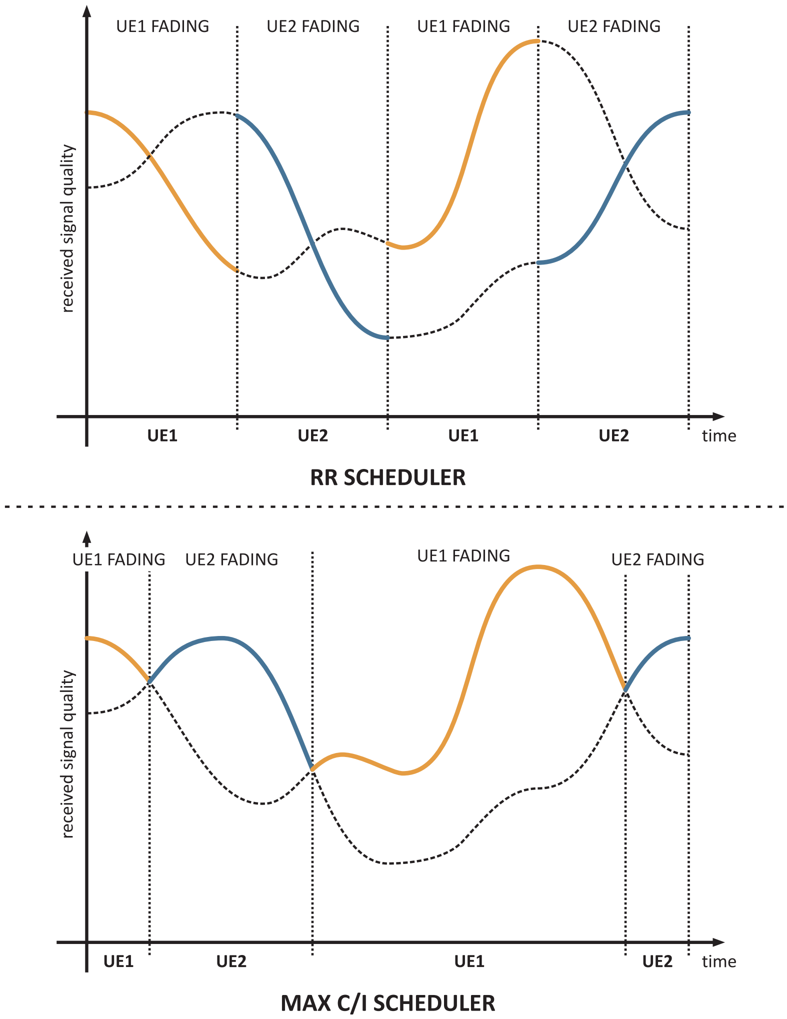
The HSDPA is newly introduced in UMTS Release 5. It adopts new techniques in order to significantly enhance data bit rates in downlink direction (data rate in uplink direction remains the same). As a result, the highest theoretical data bit rates per cell is increased from 2 Mbps up to 14.4 Mbps. Changes in networks are done especially in UTRAN and the key idea is to move several radio resource management procedures to NodeB instead of RNC as in prior UMTS releases (i.e., Release 99 and Release 4). The advantage of this step is that NodeB is much "closer" to the UE. Thus it can much more effectively react to varying radio channel quality. The procedures that have been moved to the NodeB are depicted in Figure 17.



The HSDPA enhancements to NodeB – Comparison with Release 99 and Release 4

Fast data scheduling

The important aspect regarding HSDPA is that most of the resources are shared between individual active users. Note that the rest of radio resources are dedicated for users with active voice or media stream (more will be described in chapter 4.8). The purpose of data scheduling is to assign current NodeB's radio resources to its users. The allocation period, also called as *Time Transmission Interval* (**TTI**), is set only to 2 ms when compared to UMTS Release 99 or Release 4 where the minimal TTI of 10 ms is used. This is the main reason why we are talking about "Fast" data scheduling. The scheduling interval could be shortened since the NodeB directly communicates with attached UE and receives up to date information regarding their requirements and channel quality. Several strategies can be adopted by NodeB to allocate effectively radio resource in downlink as illustrated in Figure 18.

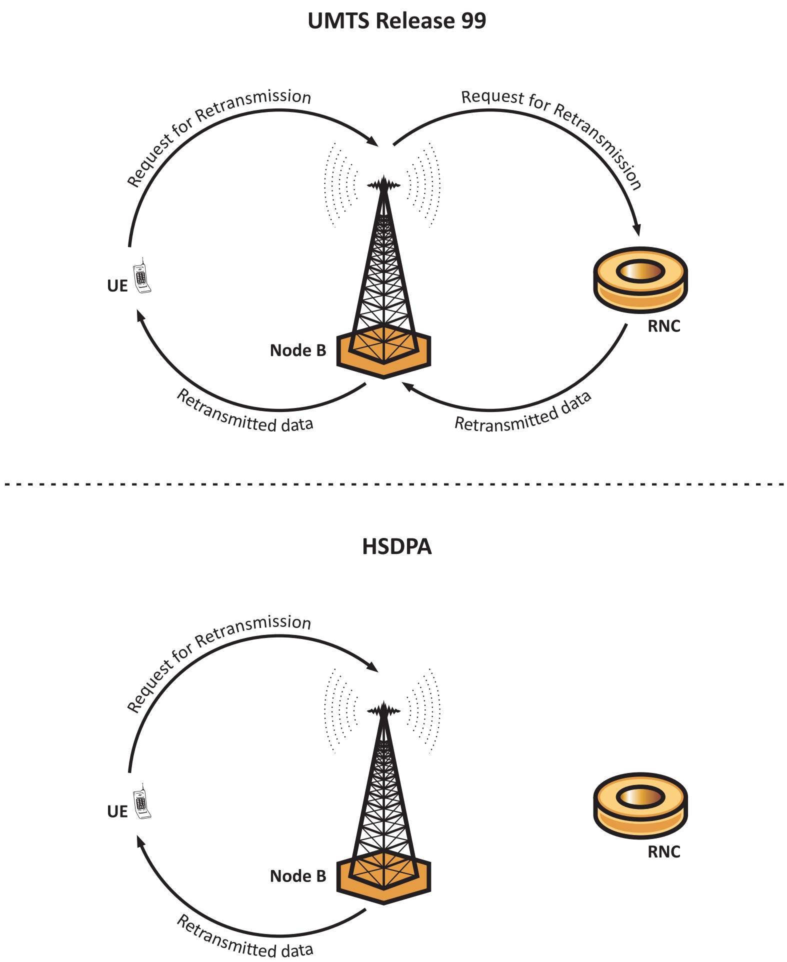


Scheduling methods in HSDPA

The most popular and the simplest scheduling algorithm is *Round Robin* (**RR**) where HSDPA users are scheduled with equal probability independently on radio channel conditions. The main disadvantage of this approach is that radio resources are not efficiently utilized (users with poor channel conditions are not able to transmit at high bit rates). The second scheduling method, *Maximum Carrier-to-Interference ratio* (**Max-C/I**), assigns all radio resources to users with highest channel quality and thus to maximize HSDPA cell throughput. Nevertheless this approach is not convenient as well, since users close to the NodeB are preferred to users with poor channel quality. The reasonable trade-off between both scheduling mechanisms mentioned above is to implement *Proportional Fair* (**PF**) method. The probability that the UE can receive data depends both on channel quality and amount of received data in the past. Consequently, if the UE is inactive for a long time, its priority is increasing.

Fast data retransmissions

If the UE is not able to decode received data packet correctly, it requests immediately its retransmission from NodeB. In prior UMTS releases, the request for retransmission is sent to RNC, which is responsible for this operation. In HSDPA, the retransmission procedure is handled solely by NodeB (that is why we are referring to a procedure as "fast" data retransmissions). The basic principle is depicted in the following figure.

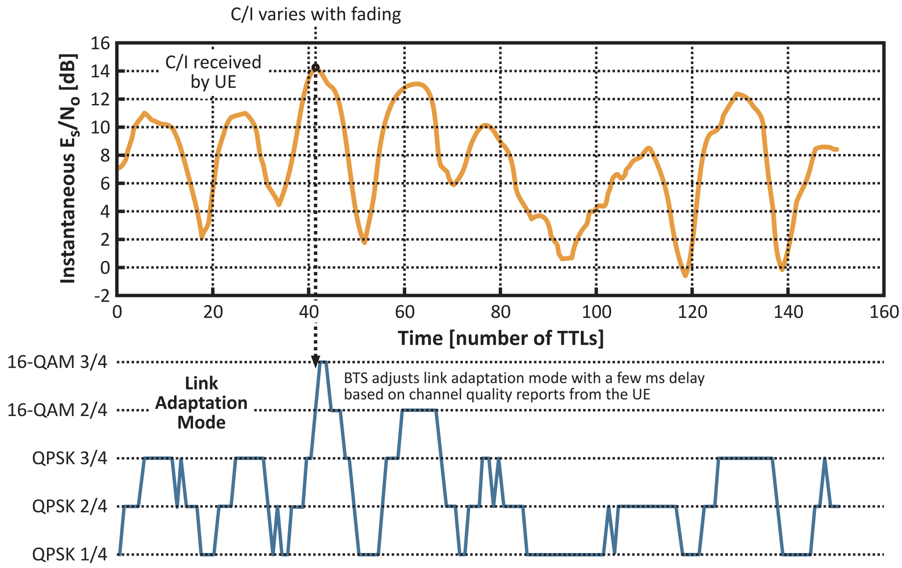


Fast data retransmission procedure in HSDPA

The additional novelty in HSDPA is the principle of a retransmission mechanism. While in original UMTS standard, simple *Automatic Repeat request* (**ARQ**) is assumed, HSDPA introduces its modified version, known as *Hybrid ARQ* (**HARQ**). The HARQ is able to temporarily store corrupted data in a buffer that could be additionally combined with newly received data packet to increase the probability of successful decoding. This way, the amount of retransmissions is effectively minimized and thus expensive radio resources are saved for other transmissions.

Fast link adaptation

The link adaptation means continuous parameters adjustment of transmission link depending on current quality of radio channel. The fast link adaptation is again enabled by moving this functionality to NodeB. In UMTS Release 99 and UMTS Release 4, link adaptation is mainly achieved by fast power control. On the other hand, HSDPA introduces new procedure known as *Adaptive Modulation and Coding* (**AMC**) that dynamically selects appropriate *Modulation and Coding Scheme* (**MCS**) used in downlink transmission. Consequently, if channel quality is poor, more robust MCS scheme is utilized resulting in lower data bit rates but guaranteeing low *Packet Error Rate* (**PER**). As soon as the radio channel characteristics are improved sufficiently, that is, higher Carrier to Interference ration (C/I), NodeB selects more efficient MCS in order to make data transmission more efficient (see Figure 20). Note that other important difference when compared to previous UMTS releases is that HSDPA supports also 16 QAM modulation scheme in addition to QPSK.



Fast link adaptation in HSDPA

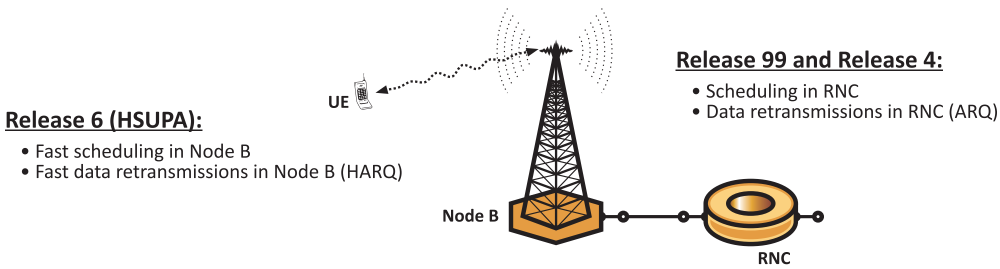
Evolution of HSDPA

The further enhancements to support higher data transmission in downlink are introduced in Release 7 where MIMO with 2x2 configuration is supported for the first time (more detailed explanation of MIMO principle will be described later in Section **Chyba! Nenalezen zdroj odkazů.**). This way, the former theoretical downlink transmission rate per cell 14 Mbps can be doubled to 28 Mbps. In addition, downlink data bit rates are also increased up to 42 Mbps by introduction of 64 QAM modulation in Release 7. More than that, the downlink throughput can be enhanced by DC method. The principle of DC is to aggregate more carrier (in most cases two carrier with resulting bandwidth equal to 10 MHz). As a result, theoretical data bit rate in downlink is up to 84 Mbps for Release 8.

The data bit rates mentioned above are only theoretical and the user will experience only a small fraction. This is due to several reasons. The first important aspect necessary to be taken into account is that the maximal theoretical data bit rates cannot be achieved in real system. The available capacity per one cell is strongly dependent on aspects, such as quality of radio channel (utilized MCS), maximum allowed transmitting power or signalling overhead required for management of control procedures. In addition, real capacity per one cell is further divided between individual active UEs. Other important restriction is related to UE capabilities. The UEs do not need to support all available bit rates. As a result, current networks are able to support only bit rates up to several Mbps per one user.

4.7 HSUPA

The HSUPA is firstly introduced in UMTS Release 6 and it increases maximum uplink bit rates up to 5.76 Mbps per cell. While HSDPA brings software modifications only to the NodeB and the RNC, HSUPA modifies also the UE. This is quite logical since the purpose of HSUPA is to increase capacity in uplink and thus some changes have to be applied to the UE as well. The principle of HSUPA is similar to HSDPA as it moves fast scheduling and fast data retransmissions to NodeB as indicated in Figure 21. While the principle of fast data retransmissions is analogical to HSDPA with implementation of HARQ, the fast scheduling approach is quite different (as explained bellow). In addition, in HSUPA there is no fast link adaptation to changing channel conditions as in case of HSDPA. Nevertheless, variation of channel is handled by fast power control.



The HSUPA enhancements to NodeB – Comparison with Release 99 and Release 4

Fast data scheduling

The most notable difference between HSDPA and HSUPA data scheduling is that in downlink (HSDPA), radio resources are shared between individual users whilst in uplink (HSUPA), all active users have assigned certain amount of resources simultaneously. Depending on UE capabilities, scheduling is done either every 10 ms (mandatory) or 2 ms (optional).

The principle of fast data scheduling in HSUPA is as follows. Whenever the UE has data to be transmitted, first, it needs to send request for radio resources allocation in uplink. After that, the NodeB assigns radio resources to the UE. Data scheduling in HSUPA is performed by means of enhanced management of UE’s transmission power.

Since WCDMA is adopted in UMTS, the system is mainly limited by maximal transmitting power and consequently by the interference. To that end, by radio resources is meant rather the amount of transmitting power that could be utilized by individual UEs than the amount of bandwidth (the maximal allowed received power by the NodeB is defined by the RNC). Consequently, the allocation of radio resources by means of fast data scheduling is done in a sense that the UE changes its transmitting power depending on current requirements. If high data bit rates are necessary, the UE has to increase its transmitting power. This, on the other hand, may result in decrease of transmitting power of other UEs as the summarized transmitting power of all UEs must be always kept below a predefined threshold. Similarly, if less amount of data needs to be sent by the UE, this particular UE can decrease its power to save radio resources in uplink.

Evolution of HSUPA

Similarly as in case of HSDPA, the HSUPA performance can be further improved by similar techniques as those contemplated for HSDPA. The most notable improvement is achieved by utilization of MIMO technique doubling maximal theoretical data bit rates approximately up to 11.5 Mbps.

The combination of HSDPA together with HSUPA technique is mostly labelled as HSPA that simply refers to high speed data access in both transmission directions. In addition, more recent HSDPA/HSUPA based on Release 7 is often referred to as HSPA+.

4.8 Services and QoS differentiation

The 2G mobile networks (GSM) enabled circuit-switched voice services with simple text messaging (SMS) and relatively expensive circuit-switched data capabilities (CSD, HSCSD). The 2.5G system evolution (GPRS) and 2.75G (EDGE) introduced packet-switched data services, such as web browsing, WAP or multimedia messaging service (MMS). These supported realistic data bit rates approximately up to 50 kbps for GRPS and up to 200 kbps for EDGE respectively. The achievable data bit rates were insufficient for more demanding multimedia services.

The 3G mobile technologies provide a wide range of new services since UMTS is supposed to support IP-based multimedia applications with high data bit rates up to several Mbps depending on current Release. Several services that 3G users can benefit from are summarized below:

- Streaming video on demand –Video content, such as movies, music, sport events, etc., can be streamed to UE.
- Real-time gaming – Games can be supported depending on UMTS version (i.e., which Release is implemented). Games requiring high data bit rates and low delays can be supported only by newer releases such as Release 5 and more recent ones.
- Downloading multimedia contents – User can easily download mp3 songs, photos, and other interactive content.
- Messaging services – While in 2G networks, only SMS or simple MMS messages (text together with picture) are available, 3G networks supports inclusion of short video, etc.
- Video telephony and conferencing – Simultaneous interaction of several mobile users who can communicate online using both voice and video.
- Location based services – Offering to users useful navigation services like navigation of users to final destination, notification of spot of interest (restaurant, sport centres, malls, sightseeing) or improved emergency services.
- Push to talk – By simple pressing a button on the handset, the user can immediately start talking to other users like in case of walkie-talkie.

In order to support above mentioned services in UMTS network, differentiation in QoS is defined. To put it simply, if the load of the system is high, some services needs to be prioritized in order to guarantee end-to-end application performance. To that end, UMTS introduces four QoS classes (see Table below) that are mainly distinguished by their packet delay susceptibility:

Service class	Description
Conversational class	High requirements on delay with symmetric or nearly symmetric traffic, and with dedicated amount of radio resources
Streaming class	Lower requirements on delay in comparison to conversational class, but still with dedicated amount of radio resources
Interactive class	High requirements on delay, no dedicated amount of radio resources
Background class	Low requirements both on delay and the amount of radio resources

The conversational class together with streaming class is used for supporting of real-time (RT) applications, such as voice (conversational class) or video streaming (streaming class). On the other hand, the interactive class and background class is designated to be used by non real-time (NRT) applications when so called Best Effort (BE) access is used. Consequently, these are designed for services such as online gaming, location based services (interactive class) or www browsing (background class).

5 Long Term Evolution (Advanced) - LTE(-A)

5.1 Introduction

The next evolution step of UMTS mobile networks is known as Long Term Evolution (LTE). The LTE is defined by 3GPP releases 8 and 9. Contrary to UMTS, LTE uses OFDMA and SC-FDMA (Single Carrier OFDMA) access for downlink and uplink respectively at physical layer instead of WCDMA used in UMTS. Therefore, the transmission features are quite different comparing to UMTS. Nevertheless, LTE is still considered as a part of 3G systems as it does not fulfil requirements defined by ITU for 4G networks. The first standard denoted as 4G is **LTE-A** (*Long Term Evolution - Advanced*) according to 3GPP Release 10, standardized in June 2011. It is evolution of former LTE Release 8 or Release 9 and it is based on the same principles as both LTE releases, but it is aligned with the set of requirements known as IMT-Advanced defined by ITU. LTE-A adds new features and capabilities on top of LTE. Comparing to LTE, LTE-A introduces carrier aggregation, inter-cell interference coordination, or enhances multiple antennas transmission (MIMO). All these improvements enable LTE-A to meet IMT-Advanced requirements for 4G mobile networks and offer peak data rates up to 1 Gbps. Besides LTE-A, also **WiMAX** (*Worldwide Interoperability for Microwave Access*) according to IEEE 802.16m, standardized by **IEEE** (*The Institute of Electrical and Electronics Engineers*) was approved as 4G technology for mobile networks. Nevertheless, WiMAX is not expected to be heavily deployed in Europe. Therefore, we focus only on LTE/LTE-A in the following chapters. First, let us give a short overview of the LTE/LTE-A evolution in terms of 3GPP releases with focus on major innovations with respect to the former releases.

Release 8

This release adopted OFDMA modulation for user's access. In addition, the cell can utilize wider channel bandwidth, mostly only in downlink, by means of DC achieving downlink capacity up to 84 Mbps (64 QAM together with DC).

Release 9

This release introduces femtocells, denoted as *Home eNodeB* (**HeNB**), to the networks architecture. It further enables support of **MBMS** (*Multimedia Broadcast Multicast Services*) and enhances **LBS** (*Location Based Services*).

Release 10

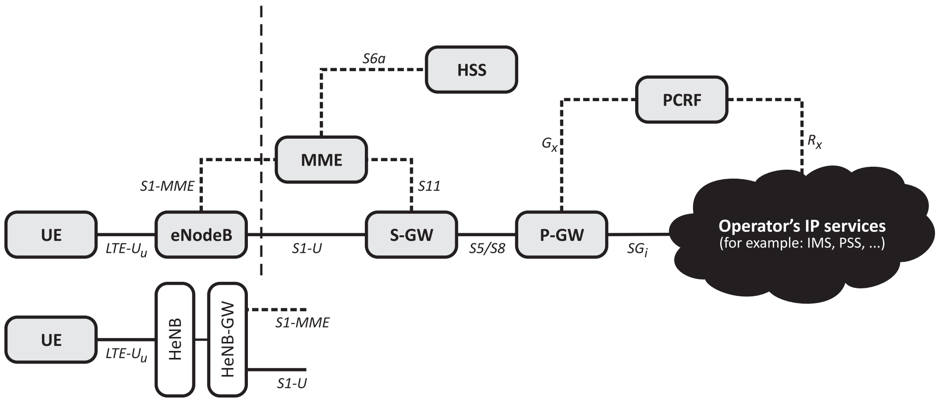
This release is the first 3GPP 4G compliant standard. New techniques such as carrier aggregation, enhanced downlink MIMO, uplink MIMO, or support of relays are introduced. To enable dense deployment of femtocells, advanced techniques for interference coordination are defined.

Release 11

This release is still under development. Contrary to the release 10, the new version should further enhance carrier aggregation, introduce *Cooperative MultiPoint communication* (**CoMP**), increase spectral efficiency, and ensure higher energy efficiency.

5.2 Network architecture

Network architecture of LTE is derived from the former GSM and UMTS architecture. Contrary, to both, the LTE is designed to support only packet-switched services. Circuit-switched services are no longer supported in network architecture since LTE. The network is composed of access network denoted as **E-UTRAN** (*Evolved Universal Terrestrial Radio Access Network*) and **EPC** (*Evolved Packet Core*) as shown in Figure 22.



Networks architecture of LTE

The access part, E-UTRAN, is composed of base stations denoted in LTE as eNodeBs (taken over from UMTS name NodeB, with "e" standing for "Evolved"). These eNodeBs are responsible for radio resource management and allocation of radio resources, mobility control, scheduling resources for both uplink and downlink, encryption of radio data transmission, or for connectivity to EPC. If femtocells are deployed, those are also a part of E-UTRAN. In this case, HeNB gateway (HeNB GW) can be included between HeNB and EPC to support large number of HeNBs.

The EPC is composed of several entities. The S-GW transfers all IP packets of all users in a network. It serves as a local mobility anchor for handover between eNodeBs. The S-GW routs and forwards packet to and from the users. This gateway accounts for inter-operator charging, e.g., in case of roaming. The second entity, the MME, controls and manages the signalling between UE and EPC including authentication, authorization, security control, establishment of connection between UE and network, roaming, and procedures related to the user's location management. Last, the P-GW is responsible for actions related to the quality of service and flow management. It means it filters user's packets, enforces QoS to guarantee required bit rates, or control service level in both downlink an uplink.

Besides these three entities, two logical functions are also a part of the LTE-A EPC architecture:

- **PCRF** (*Policy Control and Charging Rules Function*) defines rules for charging and policy control. It means, it defines actions and rules in case of problems of inconsistency between user's QoS profile and provided services.

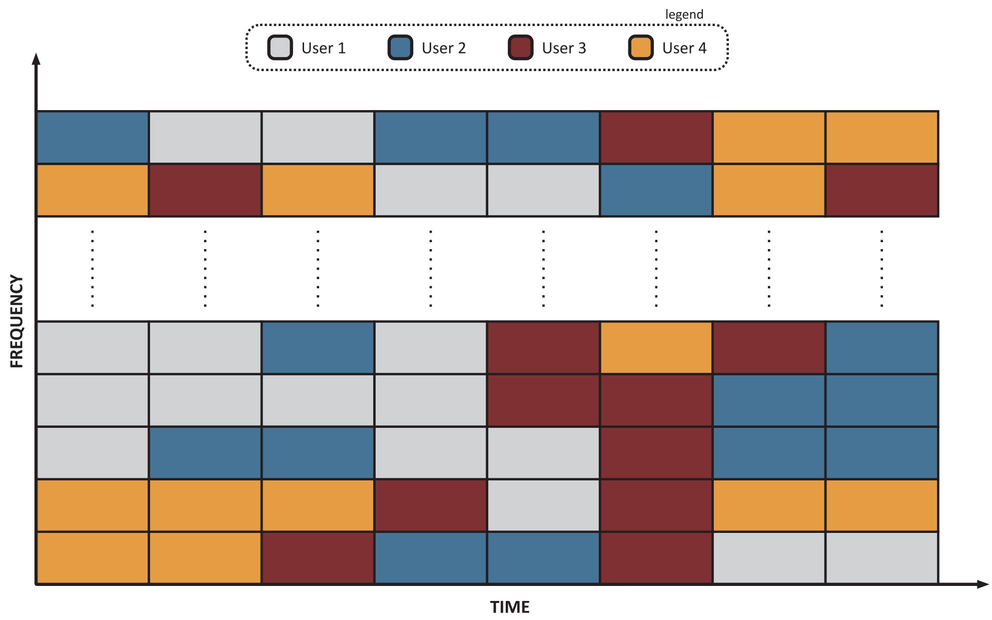
- **HSS** (*Home Subscriber Server*) contains user's subscription information such as QoS profile or roaming profile. It also keeps information about user home MME (i.e., the MME to which the UE is connected to)

5.3 LTE/LTE-A Physical layer

The physical layer communication in LTE/LTE-A is different to the one used in UMTS. It considers OFDMA and SC-FDMA techniques for access in downlink and uplink respectively. Also the larger spectrum of frequency bands is defined for LTE. At this time, 25 frequency bands are defined for 4G (17 paired bands for FDD and 8 unpaired for TDD). For example, bands at 2 or 2.6 GHz or bands around 3.5 GHz and under 1 GHz (700 - 900 MHz) are considered.

ODFMA is multiple access method exploiting OFDM. The OFDM is a multi-carrier transmission combining TDMA and FDMA (see Figure 23). The whole bandwidth is split into relatively densely spaced subcarriers. To enable dense spacing of subcarriers, orthogonal spectrum of the subcarriers must be ensured. It means, spectral maximum of one subcarrier overlaps with minimum of other subcarriers. The TDMA is considered by sharing each subcarrier by multiple users in time division manner. As a result, the whole available radio resources are split in time and frequency and can be shared by multiple users. A time interval at a subcarrier represents an OFDM symbol. Each symbol contains modulated data and the modulation can be different for each symbol. To avoid **ISI** (*InterSymbol Interference*), a *cyclic prefix* (**CP**) is introduced. The CP is composed by copying the last samples of the symbol and its purpose is to avoid of overlapping of individual symbols.

Note that contrary to UMTS, radio resources are understand rather as OFDMA symbols at physical layer in LTE/LTE-A.

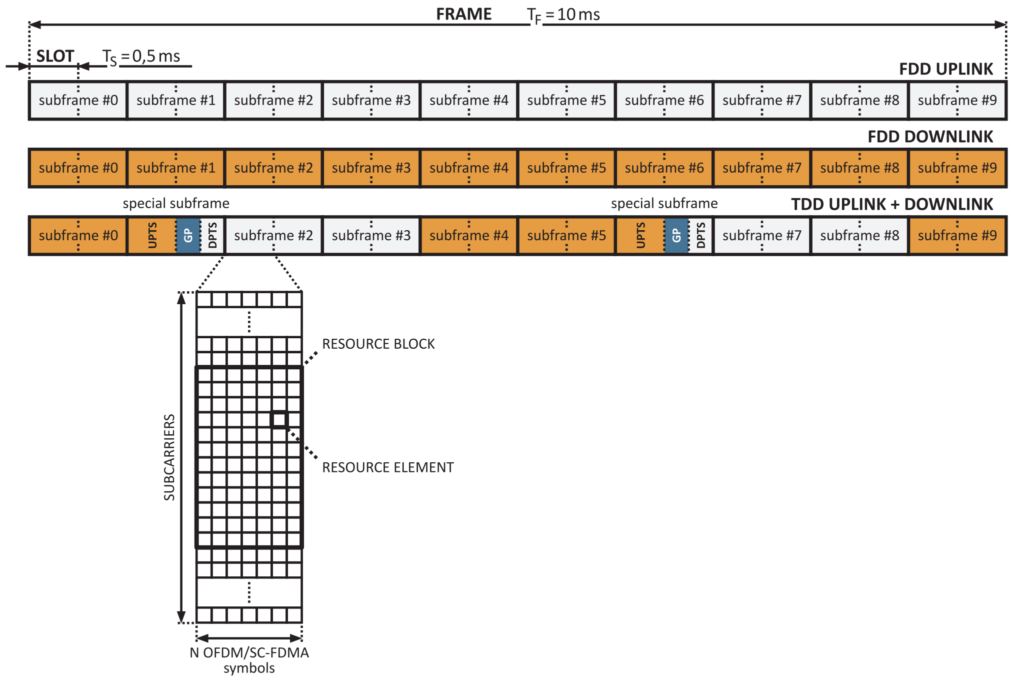


Resource allocation for users

OFDMA and resource allocation in LTE/LTE-A

A weakness of OFDMA consists in significant difference in power allocated at each subcarrier as user data are modulated independently over individual subcarriers. It means a subcarrier can be allocated with high power while other subcarrier power can be very low. Consequently, data at each subcarrier are modulated without considering information modulated on other subcarriers. This leads to high **PAPR** (*Peak to Average Power Ratio*), which negatively influences energy consumption. The SC-FDMA is used in uplink instead of OFDMA to reduce the PARP. In SC-FDMA, all data transmitted in the same time interval are modulated as a linear combination of this data symbols. Therefore, a symbol at subcarriers contains components related to information mapped onto other subcarriers. This way, a **PAPR** (*Peak to Average Power Ratio*) is lowered and consequently, it leads to minimizing interference and UE's battery consumption.

Like UMTS, LTE supports both FDD and TDD modes for data transmission as well. Two types of physical layer frames (labelled as Type 1 and Type 2) are defined as shown in Figure 24. Type 1 is applicable to FDD transmission in either full duplex or half duplex while Type 2 is intended for TDD. In both cases, the transmission is organized into frames with duration of 10 ms. Each frame is divided into ten subframes. In both types of frame, each subframe consists two slots with equal duration of 0.5 ms. The slots are composed of so-called resource blocks, which further comprise resource elements. The number of resource elements in a resource block is defined as a product of multiplication of a number of subcarriers per resource block and a number of symbols. Depending on the size of **CP**, the amount of symbols in one resource block is either six (if extended CP is applied) or seven (if normal CP is applied). The subcarriers in LTE(-A) systems are spaced equally with distance of 15 kHz and one resource block is composed of 12 subcarriers. Like in HSDPA, LTE/LTE-A uses adaptive modulation and coding. Hence, the actual modulation and coding rate are selected according to the quality of signal level. Therefore, the amount of bits carried in one resource element depends on selected Modulation and Coding Scheme (MCS). Three modulations are available in LTE/LTE-A: QPSK, 16-QAM, 64-QAM.



LTE-A TDD and FDD frame structures

In FDD frame, the different frequencies are considered for each direction. Therefore, ten subframes can be used for DL (downlink) as well as for UL (uplink) transmission simultaneously. This approach leads to equal distribution of radio resources in both directions if the same bandwidth is used.

In TDD frame, both transmission directions occupy the same frequency. To support various ratios of DL and UL, the LTE-A defines several different configurations for assignment of subframe to either DL or UL. The ratio of resources associated to DL and to UL can vary from 2:3 to 9:1. Generally, each subframe can be dedicated to DL transmission (in Table denoted as D subframe), to UL transmission (denoted as U subframe), or to combination of both. In the last case, the subframe is called as a Special (S) subframe. The first and sixth subframes are always assigned to DL direction regardless the selected configuration. Moreover, the second subframe is always dedicated to the S subframe and the third one is assigned to UL transmission. Based on the length of switch period between DL and UL, the seventh subframe can be associated to either D (10 ms switch period) or S subframe (5 ms switch period). The content of the other subframes depends on DL-UL configuration as defined in table.

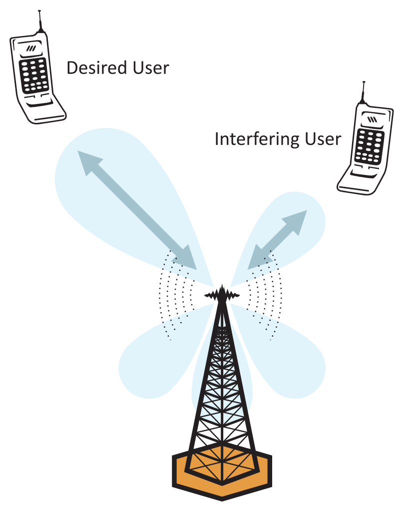
Assignment of subframe according to DL-UL configurations for TDD

DL-UL configuration	**DL-UL switch period [ms]**	**Subframe #**									
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0	5	D	S	U	U	U	D	S	U	U	U
1	5	D	S	U	U	D	D	S	U	U	D
2	5	D	S	U	D	D	D	S	U	D	D
3	10	D	S	U	U	U	D	D	D	D	D
4	10	D	S	U	U	D	D	D	D	D	D
5	10	D	S	U	D	D	D	D	D	D	D
6	5	D	S	U	U	U	D	S	U	U	D

The S subframe contains three parts: DL transmission part, known as Downlink Pilot Time Slot (DwPTS), Guard Period (GP) and UL transmission part denoted as Uplink Pilot Time Slot (UpPTS). The DwPTS part is usually occupied by DL data such as in conventional D subframe, only with reduced length. Its length varies between three and twelve symbols according to S subframe configuration. The UpPTS can consume either one or two SC-FDMA symbols and it is utilized for transmission of control channels only (i.e., no data transmission). The GP is scheduled right after the DwPTS and it is used for switching antennas from transmitting to receiving mode and vice versa. Thus no user's data can be transmitted during the GP. Its length determines the maximum supportable cell size as it is proportional to the signal propagation delay.

5.4 Beamforming

If only single antenna transmission is applied, the transmission for a user introduces interference for all other users in the same cell. If multiple antennas are used at the transceiver side, the transmission can be managed in a way to reduce interference by so-called beamforming. In case of the beamforming, direction of the transmission is controlled and adapted to focus the transmission power only to the desired user and to minimize interference to other users. It means, the transmitting power is aimed only to the direction of the receiver and the power in other directions is minimized. The beamforming is done by multiplication of the transmitted signal by complex weights to adjust magnitude and phase of individual antenna signals. To ensure transmission in a proper direction, the transmitter must be aware where the target terminal is located. Based on the knowledge of the direction, a set of complex weights is calculated or selected out of predefined codebook libraries defined by the standard. The direction can be derived from the channel quality reports collected by eNodeB from users. Nevertheless, the determination of the direction is not defined by LTE.



Principle of beamforming

5.5 Multiple antenna transmission

Multiple antenna transmission represents techniques using more than one transmitting and/or receiving antennas for communication. This approach enables to achieve higher bit rates due to creation of several parallel channels. Multiple antennas can also introduce diversity gain against fading effects if antennas are deployed relatively far from each other as the channel characteristics are not correlated in this case. Multiple antennas can be deployed either only on one side (receiver or transceiver) or on both sides. The former case is denoted as SIMO (Single Input/Multiple Output) or MISO (Multiple Input / Single Output) for multiple receiving antennas and multiple transmitting antennas respectively. The later situation with multiple antennas on both sides is known as MIMO (Multiple Input / Multiple Output). Note that in UMTS, only two antennas on both sides are assumed. In LTE and LTE-A, up to four and eight antennas can be deployed on one side for MIMO respectively. An assignment of modulated data (symbols) to individual antennas is performed by an antenna mapping block. As the transmissions are deployed at the same time/frequency resources, the parallel transmissions introduce interference. Therefore advanced signal processing must be applied in the similar way like for beamforming. Consequently, each stream from an antenna is multiplied by complex weights to change phase of the signal.

5.6 Carrier aggregation

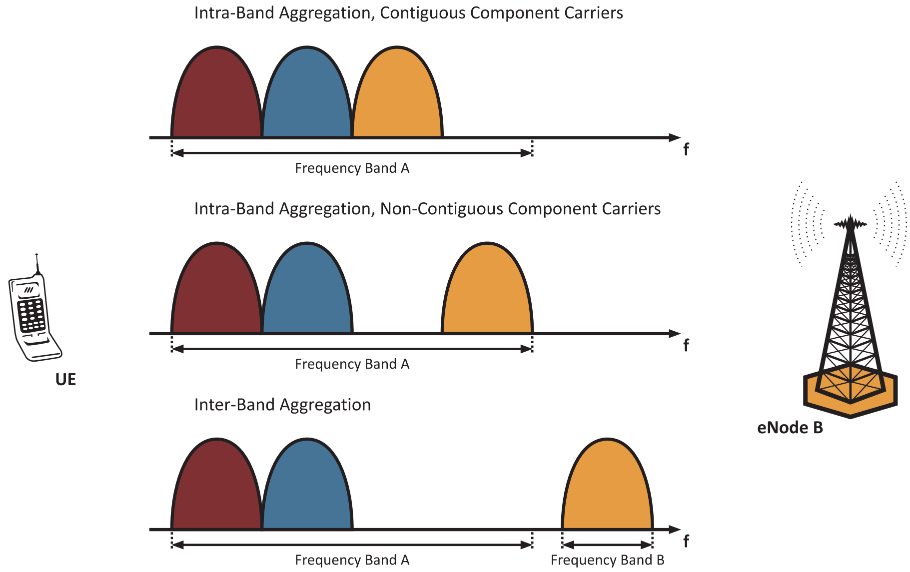
Even if LTE-A according to Release 10 reaches higher spectral efficiency comparing to former releases of LTE, the maximum peak rate of 1 Gbps required by IMT-Advanced for 4G mobile networks cannot be reached by using conventional frequency bands with width of up to 20 MHz. To meet this requirement, bandwidth must be increased. Therefore, LTE-A enables to merge radio resources across multiple bands (carriers) and perform parallel transmission to a UE. This approach is known as carrier aggregation. This technique is an enhancement of DC used in UMTS where only up to two carriers can be merged. In LTE-A, each carrier used for data transmission is referred to as a component carrier. LTE-A enables to use up to five component carriers in downlink and five in uplink for a transmission. It results in using bandwidth of up to 100 MHz. In both downlink and uplink, one component is denoted as a primary component and all other are secondary components. It means, each UE has to have one primary component and no or several (up to four) secondary components. The primary component is used for permanent signalling in idle mode (it is battery saving mode in case of no data communication). The secondary components are exploited for increase the bandwidth for data communication but not for signalling in idle mode. Assignment of primary and secondary components is UE specific, that is, it can vary from UE to UE.

There is no need for contiguous carriers to be assigned to a UE. Three types of carrier aggregation are enabled as shown in figure:

- Intra-band aggregation with contiguous components - this type is the easiest way of carrier aggregation since the band can be seen as one component from the UE point of view. Therefore, no special requirements are implied on either UE or HeNB (no additional transceivers or receivers are required).

- Intra-band aggregation with non-contiguous components - the signal cannot be seen as one transmission, thus more transceivers/receivers are required. This increases the cost of equipments.

- Inter-band aggregation (with non-contiguous components) - again, more transceivers/receivers are needed due to separation of the carriers in frequency.



Types of carrier aggregation

All UEs with carrier aggregation support can access all components in parallel. If a UE does not support carrier aggregation, it can access each carrier individually in conventional way of LTE to ensure backward compatibility.

The carrier aggregation is available for TDD as well as for FDD modes but the same mode must be applied for all components used by a UE. In addition, the same configuration of uplink and downlink subframes must be kept for TDD while configuration of special subframes can be different for individual components.

5.7 Services and applications in LTE/LTE-A

Like in former mobile communication standards, LTE/LTE-A enables multiple applications with various level of QoS (such as voice call and FTP downloading or video conference) running simultaneously. Clearly, voice or video calls requires lower delay jitter (i.e., fluctuation of packet delay) than FTP downloading to satisfy users with quality of connection. On the other hand, FTP downloading requires higher throughput and lower packet loss rate to minimize time of file download. To enable such various levels or quality requirements, LTE-A enables to define several various bearers with specific requirements of each one. To distinguish individual requirements, so called *QoS Class Identifier* (**QCI**) is introduced. The QCI defines nine classes according to a set of four transmission parameters as shown in table.

First, it indicates whether a bit rate is guaranteed (**GBR** - *Guaranteed Bit rate*) or not (Non-GBR). For GBR, a fixed amount of radio resources is permanently assigned to a service and it is not necessary to apply for these resources continuously. The amount of radio resources is set at the beginning of the connection and it corresponds to the maximum expected bit rate required by the given service. In case of Non-GBR, no resources are permanently reserved to the service. The amount of resources depends on the actual requirements of the service and amount of available resources in the network. Therefore, this type of services does not guarantee any bit rate.

The second important parameter defines priority of packets. This parameter indicates priority in case of processing. For example, if a node is congested, packets with higher priority are scheduled before packets with lower priority.

Next parameter is denoted as packet delay budget and represents maximum packet delay targeted by the given QCI.

The last parameter, packet error loss rate, is related to frequency of loosing packet during transmission. It represents amount of lost packets or packets received with an error that does not enable its further processing.

Service classes in LTE-A

QCI	GBR / Non-GBR	Priority	Packet delay budget	Packet error loss rate	Example of service
1	GBR	2	100	10^{-2}	Voice call (conversation)
2	GBR	4	150	10^{-3}	Video call (conversation)
3	GBR	5	300	10^{-6}	Streamed video (non-conversation)
4	GBR	3	50	10^{-3}	Real-time gaming
5	Non-GBR	1	100	10^{-6}	Signaling
6	Non-GBR	7	100	10^{-3}	Voice, video (live streaming)

7	Non-GBR	6	300	10^{-6}	Video (buffered streaming)
8	Non-GBR	8	300	10^{-6}	WWW, FTP, email, messaging
9	Non-GBR	9	300	10^{-6}	Like QCI 8, but with lower priority

5.8 Femtocells

As several studies show, most of the user's traffic is generated from indoor. The 4G networks are supposed to use not only low frequency bands (e.g., 800/900 MHz) but also frequencies above 2 GHz. Higher frequencies are related to worse propagation of the signal. This problem is emphasized especially for indoor environment. Therefore, indoor users are unable to reach sufficient signal quality corresponding to their requirements. Those problems are addressed by new concept of so-called femtocells (in LTE-A, denoted as HeNB). The femtocell can increase throughput for indoor users and, also, to offload macrocell by serving users receiving signal of low quality from a macrocell.

The femtocell is low cost base station, which is supposed to be deployed in user's premises such as a house or an office. The femtocell is connected to the backbone network (Internet) through a wired connection such as **DSL** (*Digital Subscriber Line*) or optical fiber. This backbone connection delivers user's data from the femtocell to a destination (a server or target user) and vice versa. The transmitting power of the femtocell is typically set to cover just indoor area to provide sufficient signal quality to close users.

The maximum transmitting power of femtocell is roughly up to 21 dBm. Thus, the femtocell's coverage is only in order of tens of meters.

The femtocells can provide three types of access:

- Open access: All users under the coverage of the femtocell can connect to it. Radio resources as well as backbone connection capacity are shared with the same priority for all users attached to the femtocell. A benefit of the open access consists in an opportunity to offload a macrocell by serving several outdoor users in areas with heavy traffic load or by serving users far from the macrocell. On the other hand, dense deployment of open access femtocells can lead to significant rise in amount of initiated handovers.

- Closed access: The femtocell with the close access admits only users included in so-called *Close Subscriber Group* (**CSG**) list. The CSG list defines users that can exploit the femtocell for access to the Internet. It is managed by a subscriber, that is, by a user who is in charge of the femtocell. Typically, only few users are assumed to be included in the CSG (roughly four or eight users are assumed for home femtocells). In case of the closed access, only limited amount of users listed in CSG shares radio as well as backbone resources of the femtocell. Therefore, higher quality can be quarantined to these users comparing to open access. On the other hand, the interference incurred by the femtocell to non-CSG users should be carefully managed to avoid an impairment of performance of non-CSG user.

- Hybrid access: The hybrid access is a combination of both open and closed accesses. In this case, the CSG list is defined and a part of femtocell transmission capacity (radio as well as backhaul) is accessible only by CSG users. The rest of the radio resources and backbone can be shared by other non-CSG users.

A critical issue related to femtocells deployment consists in increase of interference. The interference of femtocells can be classified to interference from femtocell to macrocell users (cross-tier interference) and femtocells interfere to other femtocells (co-tier interference). Level of negative impact of the interference depends also on allocation of radio resources among femtocells and macrocells. Two ways of radio resources allocation can be identified:

- Co-channel deployment: The frequency bands for macro and femtocells are overlapping. It means the same carriers are shared by macrocells and femtocells. In this case, the interference can be mitigated by proper control of femtocell transmitting power or by allocation of radio resources in order to minimize interference.

- Orthogonal (or dedicated) channel deployment: Different set of carriers is assigned for macrocells and femtocells. This way, cross-tier interference is eliminated. On the other hand, reuse of frequencies is lowered and radio resources are not utilized efficiently.

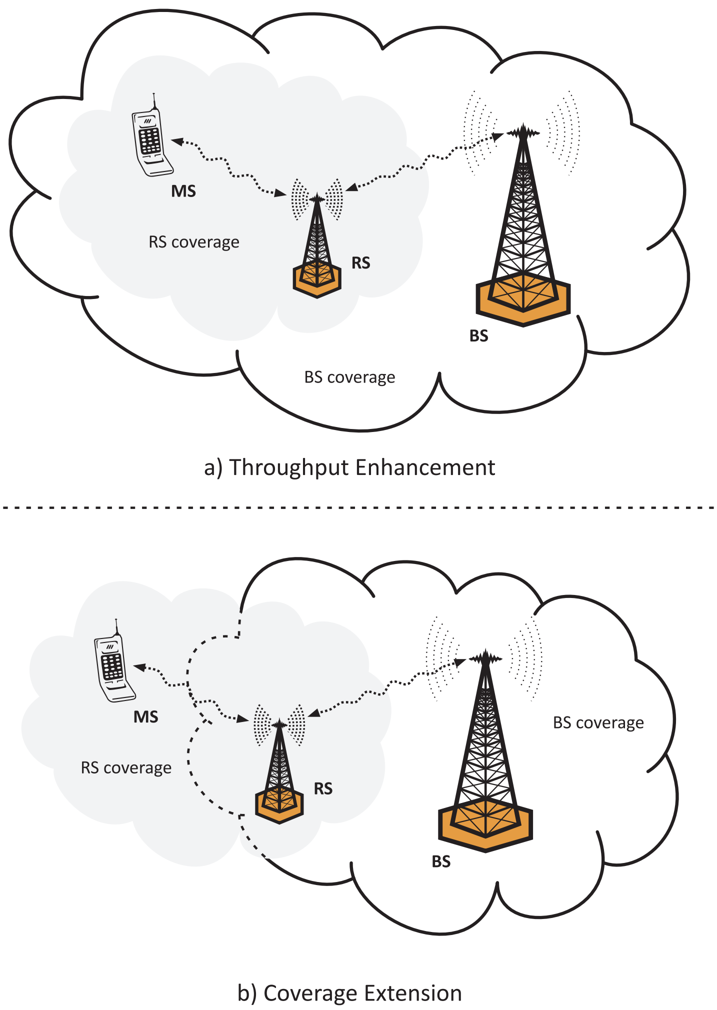
5.9 Relays

Along with femtocells, so-called relay stations are defined for the 4G networks. In general, the relays are simplified (and thus low cost) eNodeBs that relay data from the eNodeB to the user. The data exchange between UE and the network are routed via one or several relays.

Each part of communication (e.g., from UE to relay, from relay to eNodeB, etc.) represents one hop in communication. Therefore, communication using relays is also known as multihop communication.

The relays can be used either to extend coverage of an eNodeB or to increase throughput in specific area as shown in figure 27. For increasing the throughput, a relay is deployed within the coverage area of the eNodeB. The increase in throughput is reached as the distance between communicating nodes is decreased and signal level observed by target node or UE is thus increased. For enlarging the coverage, relay is deployed close to the edge of eNodeB cell. The relay can be deployed even outside the cell border experienced by UEs. This is due to the fact that the relay can be deployed in a place with direct visibility to the eNodeB and thus the communication channel between eNodeB and relay is of a higher quality comparing to the channel between eNodeB and UEs without direct visibility to the eNodeB.

In opposite to femtocells, relays are expected to be fully controlled by operators and their connection to the network is via wireless link shared with data connection of users served by the eNodeB.



Relays deployment and purpose

The relays can be classified according to the relays mobility to:

- Fixed relay - this relay is permanently installed at the same place without enabled mobility of the relay. The fixed relays are supported since Release 10 of LTE-A.
- Mobile relay - assumed to be deployed on moving vehicles such as bus or trains. In this case, the relay is supposed to perform often handover among eNodeBs. During the movement of the relay, handover among eNodeBs must be ensured even if the mobile relay serves UEs. The mobile relays are not supported in Release 10 but should be considered in Release 11.

6 Ad Hoc Networks

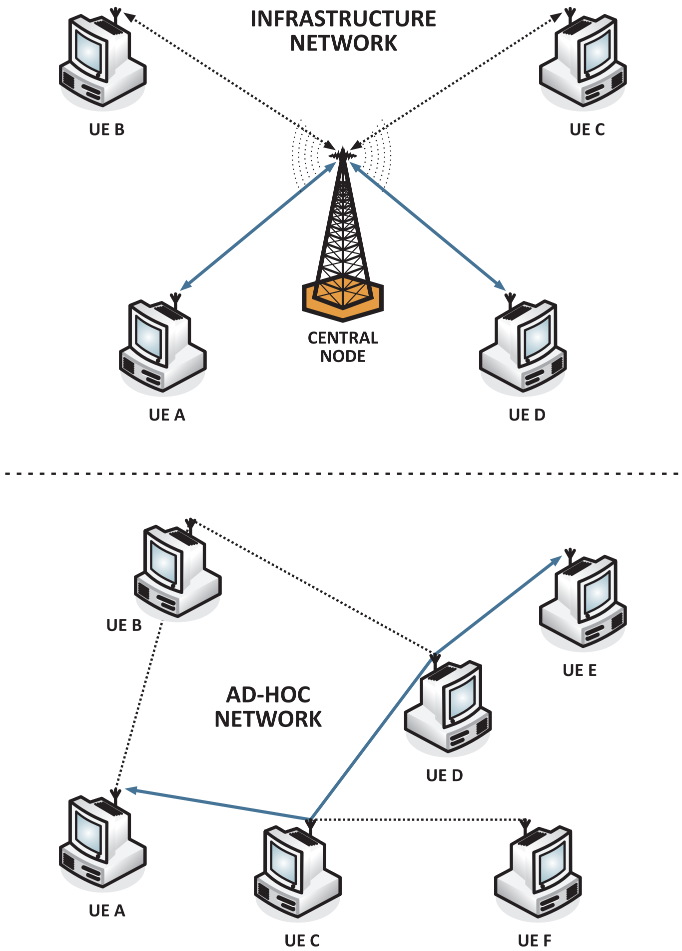
6.1 Ad Hoc Networks

The main characteristic of ad hoc network topology is that there is no central control unit as in case of infrastructure network. That means there is no base station as in case of conventional cellular network (GSM, UMTS, LTE) or no access points as in case of **WiFi** (*Wireless Fidelity*) technology, which should control and manage the network. In addition, while in cellular networks the mobile station send/receive data and control information solely to/from base station, in ad hoc network all mobile stations are equal and can communicate between themselves. The important notion is that ad hoc network is only of temporal duration, not permanent as in case of infrastructure network.

The meaning "ad hoc" is taken from Latin, which stands for "to this". To be more specific, ad hoc means something created without planning.

At the beginning, the idea behind ad hoc network was to connect several UEs that were in communication distance with each other (this is also definition of ad hoc network in WiFi based on IEEE 802.11 standards). In other words, only those UEs that were close enough can exchange data. In addition, formerly known ad hoc networks supported in WiFi were supposed to be mostly fixed, that is, the UTs were immobile.

Since the new trends in telecommunications industry is to allow all users mobility, network topology can change very often in sense that direct communication with individual UEs does not have to be available all the time. The networks supporting ad hoc topology together with users' mobility emerged and became known as a Mobile Ad Hoc Network (MANET). The MANETs are distinguished especially by mobility of terminals (that is why we call them "mobile" ad hoc networks) and by multihop communication. Like in case of relays in LTE-A, the multihop communication means that the source and destination stations do not have to be in communication distance. In order to send data between them, the packets have to be retransmitted through several neighbour stations. The comparison of ad hoc topology and infrastructure networks is depicted in Figure 28.



The comparison of ad hoc and infrastructure topologies

The advantages and disadvantages of ad hoc networks in comparison to infrastructure networks are stated below.

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- Simplicity in terms of network building since there are no central nodes
- Presence of external network is not necessary
- Easy to setup and simple to configure and install
- Cheap and fast deployment where needed

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- UE can communicate only with other UE, which is in communication distance (applied only to single hop fixed ad hoc networks based on WiFi standard)
- High interference could be generated as UEs are using the same frequency bandwidth

- Frequent topological changes due to UEs' mobility and necessity to keep track concerning the available path between individual UEs => high signalling overhead (applied only to MANET)
- Power constrains
- Security issues and protection against various attacks

The ad hoc and especially MANET networks can be used in many scenarios:

- Military communication and operations
- Emergency services (search and rescue operations, disaster recovery, policing and fire fighting)
- Commercial and civilian environments (dynamic database access, mobile offices, sport stadium, trade fairs, etc.)
- Home and enterprise networking (home/office wireless networking, conferences, meeting rooms, personal area network, construction sites, etc.)
- Education (University and campus, virtual classrooms)

In order to manage communication in ad hoc based networks and to address its many challenges, wireless **MAC** (*Medium Access Control*) protocols and appropriate routing protocol (in case of MANET) has to be implemented. These will be explained in the following chapters.

6.2 MAC protocols

The important issue regarding ad hoc networks is to design suitable MAC protocols, which main purpose is to control the access to radio channel. The main challenge is to avoid collision in transmission and to effectively utilize radio channel. The MAC protocols can be classified into contention-free and contention based:

- Contention-free protocols - The UE does not have to contend for access to the medium. Instead, some kind of controlled access, such as TDMA, FDMA, CDMA, polling or token based, is employed here. The main advantage of contention-free approach is that there are no collisions in the transmission. Consequently, these MAC protocols are meant to guarantee end-to-end delay and specific QoS requirements. The contention-free scheme is employed, for example, in Bluetooth technology where master-slave MAC mechanism is utilized.

- Contention-based protocols - The contention based protocols introduce specific contention mechanism where the UE has to contend for access to the medium. Basically, all contention methods are based on ALOHA and Slotted ALOHA schemes. In case of ALOHA, the UE accesses the medium as soon as it has data to transmit. Consequently, if two UEs send data at the same time, collision occurs and data retransmission has to take place. Although, Slotted ALOHA partially solves the collision problem, its applicability is still too limited.

The most common access methods used in contemporary wireless networks such as WiFi are based on **CSMA** (*Carrier Sensing Multiple Access*). As the name suggests, the UE firstly “sense” the medium before data are sent. If the medium is for the time being used, the UE has to wait until the medium is released. If the medium is free, the UE sends its data immediately. To further minimize the probability of collision, other methods were introduced, such as CSMA-CD (CSMA with Collision Detection) or CSMA-CA (CSMA with collision avoidance).

6.3 Routing protocols

The routing protocols are a building block of MANETs and thus lots of routing protocols have been proposed taking into account different metrics for selection of optimal routing path. In general, all routing protocols encompass three basic mechanisms:

- Route discovery – The purpose of this mechanism is to find optimal route between two stations that are exchanging some data packets. The most spread principle is based on so called flooding algorithm when source station transmits RREQ (Route Request) packet. When destination station receives this packet, RREP (Route Response) packet is generated and sent back to source station. The main issue is to find optimal routing path while signalling overhead of routing is minimized.

- Route selection – During routing discovery, several feasible paths could be found. Consequently, the main objective of route selection mechanism is to select the most appropriate routing path. The path selection can be based on many metrics, such as link capacity, the number of hops along the path, link reliability and its stability, etc.

- Route maintenance – When the proper path is selected, it is necessary to guarantee that it will be maintained at least as long as the communication between the stations is occurring. Nevertheless, due to UTs' mobility, significant changes in network topology can be observed. Thus, the objective of this mechanism is to find new routing path if the previous one is no longer available.

Classification of routing protocols

Routing protocols can be classified into several groups depending on their principle and applications. The most common routing protocols are "topology based" protocols that select proper path according to changing network topology. The topology based protocols can be further divided into:

- Proactive (also known as "Table driven") – The main principle of proactive protocols is to periodically update information regarding current network topology. All information is maintained in "tables" of individual nodes, hence we are called them as table driven protocol. The advantage is that when needed, proper path is already known and no delay is introduced by finding optimal routing path. The disadvantage is that significant signalling could be generated in order to maintain up to date information regarding network topology. The proactive protocols are mostly applicably only to small networks with low mobile UEs. The most common proactive protocols are **DBF** (*Distributed Bellman-Ford*), **DSDV** (*Distance Source Distance Vector*), **WRP** (*Wireless Routing Protocol*), or **OLSR** (*Optimized Link State Routing*).

- Reactive (also known as "On demand") – The reactive protocols do not periodically maintain information regarding individual routing paths if no data exchange is in progress. The path selection is activated only if data transmission is about to be initiated. This way, scarce radio resources are saved since the signalling overhead is low. On the other hand, it could be time consuming to find appropriate path when needed and high delay could be introduced. Among reactive protocols belong, for example, **DSR** (*Dynamic Source Routing*), **AODV** (*Ad-hoc On Demand Distance Vector*), or **SSR** (*Signal Stability Routing*).

- Hybrid – These protocols combine the advantages of proactive and reactive protocols. The most common known hybrid protocols are **ZRP** (*Zone Routing Protocol*) or **ZHLS** (*Zone-based Hierarchical Link State*).

Besides topology based protocols, the proper path can be selected also by means of position-based routing protocols (e.g., **DREAM** (*Distance Routing Effect Algorithm for Mobility*)) or by means of power-aware protocols which preserves UEs' battery life (e.g., **CPC** (*Cluster Power Control*)).

6.4 Security

Security issue is another important aspect concerning ad hoc networks since there is no security infrastructure. The attacks against ad hoc network can be categorized into two groups: passive and active. While the former typically involves "passive" eavesdropping, the latter one causes "active" attacks, such as modification or even deletion of exchanged data. The most common attacks in ad hoc networks are summarized below:

- **DDoS** (*Distributed Denial of Service*) – In most cases, jamming techniques to interfere with communication and thus preventing to use network.

- Impersonation – The attacker could gain access to the configuration system as super-user and could access or even destroy restricted data.

- Disclosure – The attacker can obtain confidential information by eavesdropping and take advantage from it.

- Routing attacks – Secure routing is difficult task due to ad hoc nature of network when malicious node can enter the network. The attacker can selectively drops the communication (selective forwarding attack), discard all packets (blackhole attack), etc.

The communication could be considered secured if the following is ensured:

- Availability – Guarantee that the communication would not be disrupted due to DDoS attacks.

- Authentication – Before the user is allowed to access an ad hoc network, it has to be properly authenticated (protection against impersonation).

- Confidentiality – Data packets have to be protected against any misuse.

- Integrity – It has to be guaranteed that data are not changed during their transmission.

Conventional methods for securing communication in network are through symmetric or asymmetric cryptographic keys. The asymmetric methods require a certification, which is difficult to employ in networks with lack of infrastructure such as ad hoc. Nevertheless, to guarantee safe certification process in ad hoc networks, Technique for Intrusion Resistant Ad Hoc routing algorithm (TIARA) has been proposed. Also, several secure routing protocols have been introduced to avoid most dangerous routing attacks, such as **SRP** (*Secure Routing Protocol*) based on the DST, or **SAODV** (*Secure AODV*).

6.5 Technologies enabling ad hoc

The most widespread technologies for wireless ad hoc networks are WiFi based on IEEE 802.11 and Bluetooth based on IEEE 802.15.

WiFi

WiFi is wireless network technology used especially for **LAN** (*Local Area Networks*) with small range (at most hundreds of meters) and provides high speed internet access to users inside households, offices, shopping centres, etc. The bit rates achieved by WiFi equipment depends on the supported standard. The most common bit rates are up to 54 Mbps per one access points (IEEE 802.11a/ IEEE 802.11g) and newly up to 300 Mbps (IEEE 802.11n).

The advantage of WiFi is especially in its low cost and that it can be used in licence-free ISM (Industry, Scientific and Medical) frequency bands. To be more specific, 2.4 GHz and 5GHz bands are usually dedicated for WiFi.

- 2,4 GHz – For most of the Europe countries is dedicated 13 channels, with bandwidth equal to 22 MHz at frequencies between 2.4 GHz to 2.484 GHz, are assigned. Note that for America, only 11 channels is assigned. The adjacent channels are partially overlapping. Consequently, only three channels could be found that do not interfere with each other
- 5 GHz – The band is further divided into three sub-bands:
    - 1. sub-band (5150 – 5250 MHz) – used inside buildings
    - 2. sub-band (5250 – 5350 MHz) – used inside buildings (the UEs must be able to control their power and to switch to another band if interference to radar occurs)
    - 3. sub-band (5350 – 5450 MHz) – the same rules as for the second sub-band are applied

Since WiFi is wide spread technology, allocated radio channels become often overloaded. Consequently, lots of modifications in optimal sharing of radio channels and the access to it have to be specified. That is why many versions of WiFi have been approved so far in order to avoid collisions, to improve its throughput, and capacity.

All equipments based on IEEE 802.11 are tested by **WECA** (*Wireless Ethernet Compatibility Alliance*) organization, which was renamed in 2002 on WiFi alliance. The WiFi alliance tests whether the equipments fulfil all necessary requirements or not.

The IEEE 802.11 standards define MAC layer that is together with **LLC** (*Logical Link Control*) a part of second layer of **RM-OSI** (*Reference Model - Open System Interconnection*) model. The purpose of MAC layer is to handle an access to the radio channel, fragmentation, and defragmentation of data packets, specifications of control frames, etc. In addition, WiFi standards specify several types of physical layers differing especially in various modulation techniques (frequency hopping, direct sequence spread spectrum or OFDM). As already mentioned, WiFi can use infrastructure or ad hoc topology (optionally also mesh, as it will be described later).

Bluetooth

Bluetooth specifies short range wireless communication. It is characterized by low transmitting power and low price. The purpose of the Bluetooth is to replace metallic interconnection of various electronic devices such as mobile phones, headsets, laptops, etc.

The basic transmission rate of the Bluetooth is 1 Mbps, but it can be extended up to 3 Mbps for version 2.0 with **EDR** (*Enhanced Data Rates*). In addition, high speed transmission up to 24 Mbps can be reached by Bluetooth v3.0 + **HS** (*High Speed*). This version enables to exploit WiFi bands for Bluetooth communication. Last version, Bluetooth v4.0 is focused on lowering power consumption. Thus, this version is also denoted as **LE** (*Low Energy*).

In all version of Bluetooth, **FHSS** (*Frequency Hopping Spread Spectrum*) is utilized to cope with the problem of interference and fading. In FHSS, all devices use a frequency hopping pattern of pseudo-randomly selected a carrier frequency for transmission of a packet. The carrier frequency is hopped 1600 times per second.

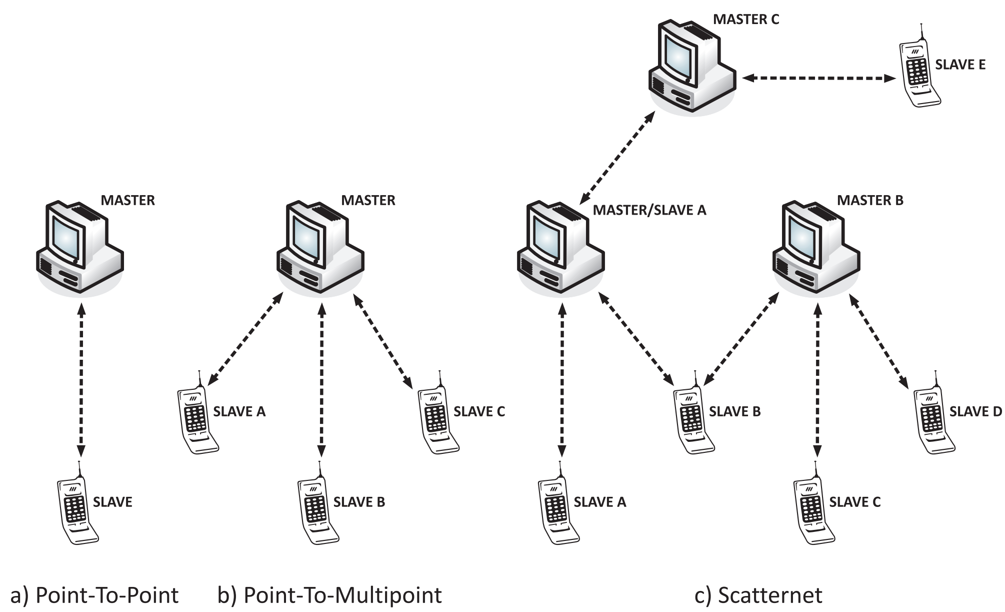
Physical layer transmission of Bluetooth is performed in license-exempt frequency bands of 2.4 GHz with bandwidth of 83.5 MHz. Specifically, frequencies between 2400 MHz and 2483.5 MHz are occupied. In basic transmission rate and EDR, the bandwidth is split into 79 transmission channels and two guard bands. Lower and upper guard bands are of 2 and 3.5 MHz bandwidth respectively. Both basic rate and EDR use full duplex and TDD transmission scheme.

The data are modulated with a binary **GFSK** (*Gaussian Frequency Shift Keying*) for basic 1 Mbps transmission in Bluetooth v1.0. For higher bit rates in later versions, **PSK** (*Phase Shift Keying*) based modulations are used. The modulation **π/4-DQPSK** (*pi/4 Rotated Differential Quadrature Phase Shift Keying*) and **8 DPSK** (*8 phase Differential Phase Shift Keying*) used in EDR mode enable to reach 2 and 3 Mbps respectively. To reach 24 Mbps, so called **AMP** (*Alternate MAC/PHYs*) operation must be enabled. After establishing EDR radio channel, AMP finds alternative wider band and shift the transmission to this band.

A different physical channel and access to the channel are defined to enable low power consumption. Contrary to basic rate and EDR, only GFSK modulation is considered by Bluetooth LE. The same bandwidth but with different subchannelization pattern is used. The band is split into 40 channels separated by 2 MHz with 2 and 1.5 MHz lower and upper guard bands. For access the channels, either TDMA or FDMA can be used. Three channels are used for "Advertisement" and 37 channels for data communication. In advertisement channels, devices can indicate what they intend to perform (i.e., setting a connection or preparing for data transmission)

Bluetooth supports direct point-to-point as well as point-to-multipoint communication. All communicating devices sharing the same channel is denoted as a piconet. In each piconet, a device is master to all others and the other devices are slaves to the master one. The device initiating the communication is labelled as the master. All slave devices must be synchronized to the master's clock and must follow the same hopping pattern as the master. The communication is able only between master and slave, but not between two slave devices. Up to seven slaves can be active in each piconet for basic data rate and EDR. Beside the seven active devices, additional up to 255 non-communicating devices (in parked state) can be also included in the piconet. In Bluetooth LE, the number of active devices is limited by amount of master's radio resources.

A device can be included in more than one piconet. In this case, the piconets mutually overlap each other and the topology is denoted as scatternet. The scatternet is composed of several piconets while even in scatternet, each piconet still contains only one master device. Each device can be the master only in one piconet in frame of the scatternet. In all other piconets, the device must take over the role of the slave. Therefore, the role of master and slave can be switched among devices in scatternet to avoid the situation when a device is the master in two piconets. If two piconets partly overlapping and having different masters, different pattern of frequency-hopping is used in each piconet. Therefore, a device participating in more than one piconet must apply time multiplexing and access individual piconets one-by-one.



Topology for Bluetooth

6.6 Wireless sensor networks

A **WSN (***Wireless Sensor Network*) consists of nodes organized into cooperative network and utilizes ad hoc network topology. Basically, the WSN is similar to the MANET networks in many aspects. The main reason is that both the WSN and the MANET are distinguished by self configuration nature, nodes are connected by means of wireless connection, and the communication between nodes is enabled by multihop networking. Still, the WSN and the MANET networks are different in several fundamental aspects as shown in table.

Comparison of MANET and WSN

Aspect	MANET	WSN
Utilization and applicability	Used by human beings for data transmission, mobile applications, etc.	Embedded in the environment, used for data gathering (e.g., measuring of temperature, ecological monitoring, etc.) and event detection (e.g., detecting of fire, intrusion detection, etc.)
Size of the devices	Handsets, PDAs, laptops	Small sensors (e.g., millimetres)
Amount of nodes in the network	Up to tens or up to hundreds nodes	Up to thousands or more nodes
Main reason for topological changes	Users' mobility	Failure of the node

Similarly as in case of MANET, the most critical issues of WSN are a proper MAC protocol and effective routing of data. The MAC protocols formerly used in MANET networks are not suitable due to WSN specifics. The main objective is to use protocol that consumes a little power, avoid collisions and has low memory requirements.

6.7 Mesh network

Mesh network is a perspective topology combining characteristics of infrastructure and ad hoc networks. Similarly as in infrastructure network, the mesh network is composed of central nodes (e.g., base stations, access points, etc.) that provide connectivity, for example, to the Internet. Nevertheless, the main difference in comparison to cellular networks is that in mesh network the UE can attach to other UE. These UEs can then communicate between themselves without intervention of the central node. When compared to ad hoc and MANET networks, the mesh network topology is supposed to be permanent and individual nodes in the network are mostly fixed. Still, mesh network also inherent the paradigm of multihop communications when data can be sent via several nodes to reach their destination. Consequently, optimal routing path (if UE is out of central node's range) has to be found by means of specific routing protocols similarly as in case of MANET or WSN.

The mesh topology is supported in WiFi technology by IEEE 802.11s amendment. While legacy user equipments (i.e., equipments supporting already existing standards such as IEEE 802.11a/b/g) can use mesh, it can not serve as an intermediate node between the UE and the access point. In addition, the mesh topology is also considered in WiMAX technology based on IEEE 802.16d standard where fixed topology is assumed. Nevertheless, more recent standards supporting mobility (IEEE 802.16e/j/m) excludes mesh and use only infrastructure topology such as PMP and relay due to high complexity of mesh implementation.

The advantages of mesh are the following:

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- Increase of coverage since UEs out of access point coverage can connect to it with help of their neighbours (i.e., no additional access points have to be installed).
- High flexibility and reliability (several possible routes to access point can be found).

The disadvantages of mesh are following:

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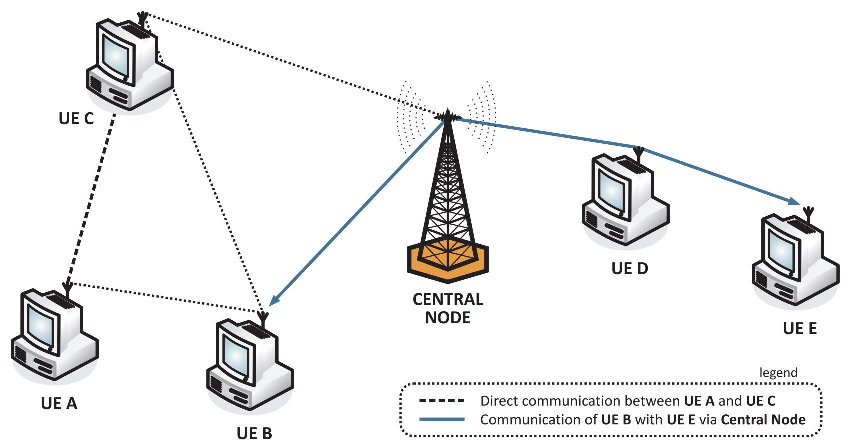
- The UE can be used for data transmission of other UE. Thus, intermediate UE will consume more battery energy.
- Throughput degradation if the same data are transmitted more than once between source and destination stations due to multihop communications.
- Higher overhead than in case of infrastructure network (it is necessary to maintain available routes).



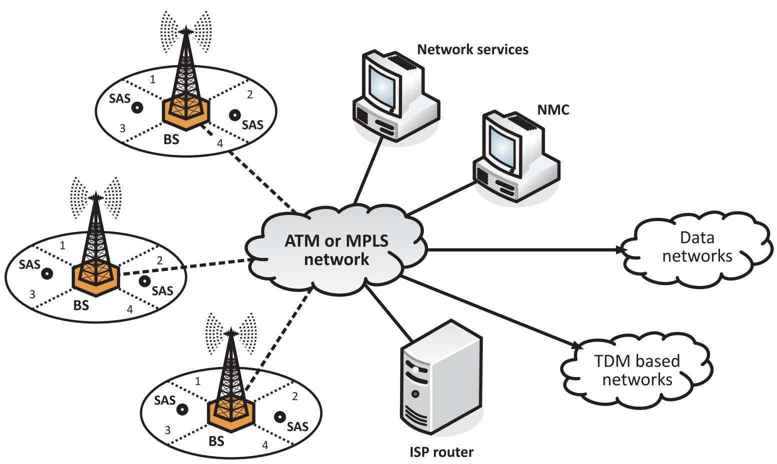
An example of mesh structure and communication between individual stations

7 Multipoint Distributive Systems

7.1 Local Multipoint Distributive System (LMDS)

A **LMDS** (*Local Multipoint Distributive System*) is a broadband wireless access technology originally designed for Digital Television Transmission. It has been conceived as a fixed wireless, point-to-multipoint technology for utilization in the last mile. The LMDS commonly operates on microwave frequencies across 26 GHz and 29 GHz bands. In the United States, frequencies from 31.0 to 31.3 GHz are also considered.

The LMDS systems are based on fixed wireless group access of fixed terminal stations, abbreviated as **SAS** (*Subscriber Access System*), to BS through radio interface. The SASs are equipped with directional antennas oriented to the BS.



Network structure of LMDS system

Throughput capacity and reliable distance of the link depends on common radio link constraints and used modulation method - either PSK or **AM** (*Amplitude Modulation*). Distance is typically limited to approximately 2 or 3 km due to rain fade attenuation constraints. Deployment links of up to 8 km from the **BS** are possible under specific circumstances such as point-to-point systems that can reach slightly farther distances due to increased antenna gain.

Although some operators in Europe use LMDS systems to provide access services among different networks, LMDS is more commonly used for high-capacity links for interconnection of backhauls of networks such as GSM, UMTS, WiMAX or Wi-Fi.

Number of sectors of the BS and their allocation is dependent on number of access places and on total data capacity routed to specific BS. Physical distribution of sectors is not naturally unchangeable and its modification is interconnected with increasing utilization of network. Basic allocation of territory and its coverage is implemented by two channels (A and B) as dual frequency plan.

We use the dual frequency plan, because we have serious problems with reciprocal interference among sectors. There are several techniques how to reduce reciprocal interference. The first condition is a utilization of direction quality of the BS’s antennas serving specific sector. The second condition is that SAS's parabolic antennas with narrow direction are aimed to the serving BS. Other way, how to reduce interference is allocation of different channels for each sector of the BS or the modification of wave polarization of radio signal in particular sectors to horizontal or to vertical wave.

7.2 Multichannel Multipoint Distributive System (MMDS)

A **MMDS** (*Multichannel Multipoint Distribution Service*), also known as **BRS** (*Broadband Radio Service*) or somewhere called as **Wireless Cable,** is a wireless telecommunications technology primarily used for broadband networking or, more commonly, as an alternative method for reception of cable television programs.

The MMDS systems use microwave frequencies at 2.1 GHz and frequencies from 2.5 GHz up to 2.7 GHz. Receiving of television programs respectively data signals delivered by MMDS is done with a rooftop microwave antenna. The antenna itself is attached to a transceiver, which makes possible receiving and transmitting of the microwave signal and converting them into frequencies compatible with standard TV tuners (much like on satellite antennas where the signals are converted down to frequencies more compatible with standard of cable TV). Some antennas can use an integrated transceiver. Digital TV channels can be then decoded with a standard cable set-top box or directly for TVs with integrated digital tuners. Common data signals can be received by the help of cable modem, supporting a **DOCSIS** (*Data Over Cable Service Interface Specification*) standard, connected to the same antenna and the transceiver.

The frequency band of MMDS systems is separated into 34 MHz channels, which were auctioned off like frequency bands for other distribution systems. The basic concept is primarily focused on how to allow transmissions with high capacity for providers through several huge distribution channels and how to multiplex them into several television and radio programs, and later general data onto each channel using digital technology. Similarly like with digital cable channels, each channel is capable to transfer up to 30 Mbit/s with **64-QAM** modulation, respectively 42 Mbps with **256-QAM** modulation. Due to **FEC** (*Forward Error Correction*) and other overhead, the real throughput is roughly 27 Mbps for **64-QAM** and 38 Mbps for **256-QAM**.

The newest MMDS band plans to prepare changes to channel size and to licensing in order to accommodate new WIMAX TDD on fixed and mobile equipments.

The LMDS and MMDS systems have adapted the DOCSIS from the specifications of cable modems. The version of DOCSIS was modified for wireless broadband as **DOCSIS+**.

Robustness of data transport is accomplished under MMDS by encrypting traffic flows between the broadband wireless modem and the **WMTS** (*Wireless Modem Termination System*) located in the base station of the provider's network using **Triple DES** (*Triple Data Encryption Standard*).

The MMDS systems provide significantly greater range than LMDS systems. The MMDS systems may become obsolete because of the newer 802.16 WiMAX standard approved since 2004. The MMDS systems were sometimes expanded to *Multipoint Microwave Distribution System* or *Multichannel Multipoint Distribution System*. All three phrases refer to in the principle to the same technology.

8 Location Based Services

8.1 Location Based Services

The knowledge of geographical location of users enables to exploit new services in areas of emergency, health, entertainment or work. The geographical location of the user is represented by user's spatial coordinates.

A service exploiting geographic information to serve a user is **LBS** (*Location Based Services*).

An example of location based services is a navigation service, tracking of people or vehicles (e.g., Latitude), geocaching, or tourist guide. Another important application of LBS is location in case of emergency (for example, location of car crash, injured person, etc.). In this case, the location of the person can be done automatically and required assistance can be navigated to the place without need for user intervention.

The concept of LBS requires architecture composed of five types of equipments:

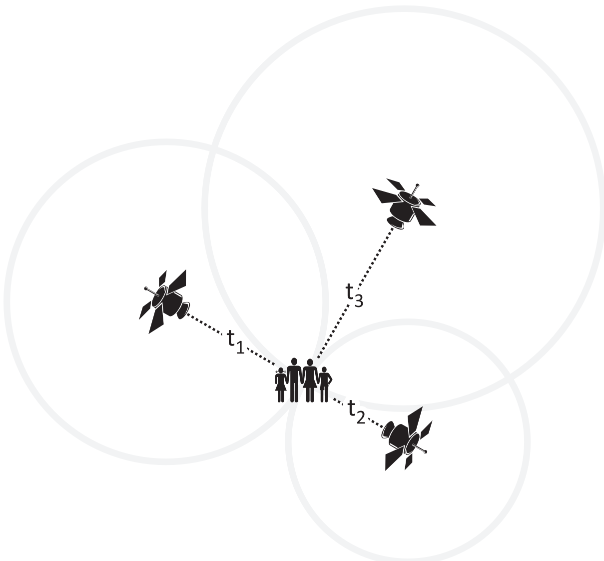
- User device - provides interface between a user and the LBS system. The user device is usually a mobile equipment such as mobile phone, laptop, tablet, navigation device, etc.
- Communication network - is responsible for delivery of user's data and requests from the user device to the service provider and backward delivery of the obtained information from the system back to the user.
- Location (positioning) component: determines the user geographical position.
- LBS provider - offers services to the users and it is responsible for the processing user's requests.
- Content provider: supply LBS provider with a content, which is not stored by the LBS provider; it can be, for example, a map source, position of objects, information related to the objects, information on public transport, etc.

For efficient and profitable exploitation of LBS, the determination of the location by Location component is critical issue. Without precise location, the service ability of LBS would be limited. The location can be derived by several ways. The most common way is to exploit either a satellite positioning system or a mobile network. Both major approaches are described in the next sections.

8.2 Satellite positioning systems

Possibility of using satellites for location has been discovered by a team led by Dr. Richard B. Kershner during monitoring radio transmission of the first satellite - Sputnik. The team observed the frequency of the transmitted signal changes as satellite moves due to Doppler effect. Based on this frequency changes, they were able to determine the position of the satellite. Reversing this approach and knowing position of the satellites, the position of users or objects is known.

The satellite systems for determination of position are based on measurement of a propagation time of a signal transmitted by the satellites. The propagation time, t, is then converted to the distance, s, between the satellite and the user according to formula s=c*t, where c is the speed of light in vacuum ($3 \cdot 10^8$ meters per second). Knowing the position of the satellite and the time of propagation, exact position is determined as a sphere of points located in the same distance from the satellite position. It means each satellite enables to determine a sphere of potential locations of the user. Exact position of an object is defined by four parameters: latitude, longitude, altitude, and time. For deriving all four parameters, four spheres must be known. Then the user location is represented by intersection of all four spheres. In ideal case with no error in estimation of the distance between the satellite and the user, all spheres intersect in one point. If the distance is not derived precisely, intersection of four spheres defines an area of potential locations of the user. The size of this area is proportional to the errors introduced by distance measurement and it can be reduced by considering information from additional satellites.



Principle of determination of user position

This general principle is exploited by the most common navigation systems such as American GPS system, European Galileo system, Russian Glonass system, or Chinese COMPASS.

GPS

The GPS is global navigation satellite system developed by United States Department of Defense for military purposes. The GPS project started in 1973 but the first experimental satellite was lunched in 1978. Now, the GSP provides positioning, navigation, and timing services not only for military but also for civilian use all around the world. It is composed of three segments: space, control, and user.

- Space segment: The space segment consists of at least 24 satellites flying roughly 20200 km above the Earth with orbital period of 11 hours and 58 minutes. The satellites are deployed in six orbits with inclination of 55°. The minimum number of satellites is derived from condition to ensure visibility of at least four satellites from each point at the Earth. Those additional satellites can increase accuracy of the system. In 2011, the former constellation of satellites was slightly changed to improve coverage and three satellites were included to the basic constellation of 24 satellites. It means 27 satellites are now considered as a baseline configuration. In last years, 31 actively operating satellites and three or four spare satellites are flying around the Earth. All satellites are transmitting signals with information required for derivation of user position.

- Control segment: The control segment is created of ground facilities for monitoring, controlling and managing the space segment satellites. The core of the control system is a master control station, in Colorado in United States. The master control station is responsible for management of all control procedures. To avoid the system shut down in case of the master control station failure, an alternate one is deployed in California. The master control station receives and analyses information from satellites (e.g., status information, transmitted navigation information, etc.) and navigation information from 16 ground monitor stations. The monitor stations monitor navigation transmission of the satellites and track the satellites. Based on the collected information, the master control station sends commands to satellites to ensure correct and reliable services (e.g., to change the satellite position). The commands, information for the satellites, or other data are sent via ground antennas. In control systems, 12 ground antennas are deployed not only in America but worldwide.

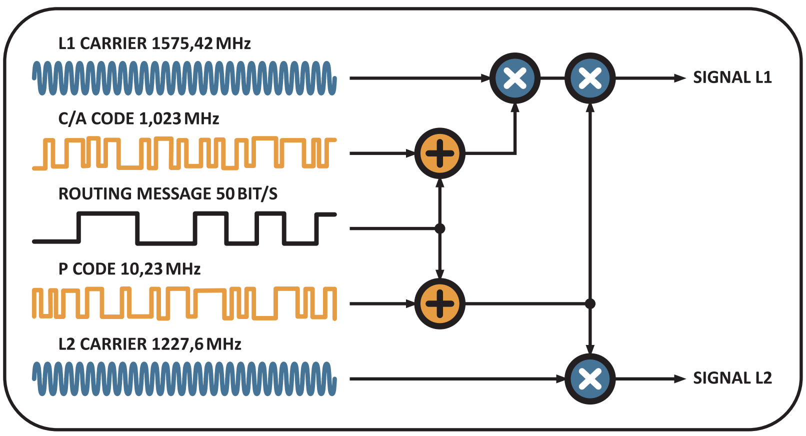
- User segment: The user segment is represented by a GPS receiver, which processes information received from satellites. The GPS receiver contains a GPS chip with radio part, digital signal processor, memory, control part, and interface to a host unit. The GPS provides processed navigation information to the host unit in form of NMEA (National Marine Electronics Association) protocol.

The satellites transmit three types of signal. The first one, P (precision code), is ciphered code with 10.23 MHz pseudorandom code sequence. This one can be replaced by the second one, Y code, in case of need of anti-spoofing transmission mode. The last one is C/A (coarse/acquisition code) used for civil purposes or for acquisition of the P(Y) codes. The C/A is composed of 1.023 MHz pseudorandom non-ciphered sequence. The navigation message, i.e., the message with information for determination of users' position, is transmitted with speed of 50 bps and it is added to the C/A code and P(Y) code. Both resulting sequences are modulated to so called L1 carrier at 1575.45 MHz. Next, sequence resulting from adding navigation message to a P(Y) code is modulated to L1 and in addition to L2 carrier at 1227.6 MHz. Both outputs L1 and L2 are then transmitted to users' GPS receivers. Besides, additional signals L2C (1227 MHz), L5 (1176 MHz), and L1C (1575 MHz) can be transmitted for minimizing positioning error for commercial, safety-of-life, and interoperability with other systems respectively. Since all satellites transmit at the same carrier frequency, CDMA multiple access technique is considered to enable reception of signal from more satellites.

The overall navigation message is transmitted in frames with 1500 bits length. Each frame is composed of five subframes containing different information such as, satellite position, health, clock correction, Ionospheric delay effect, constellation status or information on satellites orbits. To transmit the whole navigation message, 30 seconds is required (1500 bit transmitted with 50 bps).

As a time required for delivery of complete information on all satellites is too long (i.e., the time between device turn-on and determination of user position), a mobile communication networks can be exploited in combination with satellites navigation systems for delivery of some information by using mobile networks instead of conventional delivery directly by satellites. This approach is known as Assisted GPS. This way, the process of GPS initiation is speed up.

The accuracy of the positioning is influenced by several aspects: measurement of signal arrival time, atmospheric effects (especially ionospheric effects), multipath propagation, satellite position update, clock synchronization and amount of visible satellites. The most significant error is introduced by ionosphere; however, it can be minimized by considering several signals (e.g., L1 and L2) as an impact of ionosphere influences individual frequencies differently. The overall error is in order of meters for the GPS (typically, up to 8 meters). The accuracy can be further enhanced by differential GPS using ground nodes with exactly defined position to precise the position derived from satellites.



Transmitted signals by GPS satellites

Galileo

The European alternative to GPS is system Galileo. In space segment, 27 operational and 3 spare satellites are assumed to be placed over three orbits with inclination of 56°. The elevation of the satellites is 23 222 km above the Earth with orbital period of 14 hours. Control segment contains two control centers in Oberpfaffenhofen, Germany and Fucino, Italy. Further, two **LEOP** (*Lunch and Early Operations*) centers are in Toulouse, France and Darmstadt, Germany. Analogically to GPS, also 5 telemetry, tracking and control stations, 40 sensor stations, and 10 uplink stations for monitoring and controlling the satellites will be deployed worldwide.

The satellites, like in GPS, transmit several navigation signals (E1, E5a, E5b, E6). The first signal, E1, carries 1.023 MHz non-encrypted ranging codes and navigation data modulated at 1575.42 MHz. This signal with 125 bps rate is available for civil usage (denoted as Open services), commercial services and safety-of-life services. The second and third signals, E5a and E5b, are analogical to the E1; however, E5a and E5b both are modulated to 1176.450 MHz and E5b 1207.140 MHz respectively and both are of 10.23 MHz code sequence. The signal E5a carries data for navigation and timing functions. Contrary to E1 and E5b, the E5a uses more robust modulation and thus only 25 bps rate is supported and it is available only for open services. The last signal, E6, is designated for commercial purposes and thus it is encrypted. This signal is carried at 1278.750 MHz with 5.115 MHz of code sequence. The supported bit rate is 500 bps.

Other navigation system

The principle of the determination of user position is analogical also for other navigation systems such as:

- **GLONASS** (*GLObalnaja NAvigacionnaja Sputnikovaja Sistema*) - It assumes 26 satellites placed over three orbits in full constellation. An interesting fact is that the system was not originally compatible with GPS or Galileo due to using of FDM, i.e., each satellite transmits at different frequencies. However, from 2011, Glonass supports also CDMA signals for compatibility purposes. The ground system is located in Russia.

- COMPASS, also known as BeiDou 2, is Chinese navigation system, currently developed for national purposes, but expected to be extended to worldwide navigation (with 35 satellites) in next years.

- **IRNSS** (*Indian Regional Navigational Satellite System*) consist of 7 satellites in space segment, 3 in geostationary orbit and 4 in geosynchronous orbit to cover Indian subcontinent.

8.3 Location by mobile networks

The mobile networks enable to estimate user's position based on knowledge of the base stations positions. Several approaches to location exploiting mobile communication networks can be distinguished. The approaches differ in parameters used for the derivation of the position.

First way is known as *Cell Of Origin* (**COO**). It exploits an identification of the base stations (Cell ID) to derivation of user position. The user position corresponds to the coordinates of its serving base station. Of course, an error in derived position is equal to the cell radius.

Higher accuracy is reached by *Time of Arrival* (**TOA**), which utilizes not only Cell ID but also ability to measure a delay between transmission of a signal by a mobile station and its reception by a base station. Since the signal is propagated with the speed of light, the distance of the user from the base station is calculated easily in the same way like in the satellite systems.

If signal is received from more than one base station, the TOA can be extended to *Time Difference of Arrival* (**TDOA**). In this case, the signal propagation time from all neighbouring base stations is confronted and the user position is derived as an intersection of estimated distances from each base station. In TDOA, the time of arrivals are processed by the network. If a mobile station is in charge of the data processing, the algorithm is denoted as *Enhanced Observed Time Difference* (**E-OTD**). The E-OTD requires special equipment, *Location Measurement Unit* (*LMU*), in the network. The LMU ensures accurate timing and accuracy of the data.

If directional antennas are used in either base station or mobile station, a direction from which the signal is received can be triangulated in the similar manner as in the TDOA. This method is denoted as **Angle of Arrival** (**AOA**).

9 Future trends in mobile communications

Future mobile networks are assumed to consider new techniques and algorithms to enable even higher bit rates and to ensure full and seamless mobility. Therefore, it should reach higher spectral efficiency by cooperation among cells. It is expected that future mobile networks will be deployed densely and cells will be of low radius, especially in densely populated areas. To maximize advantage of dense deployment of cells, those cells should be interconnected with high speed optical fiber to avoid of bottleneck in the wired part. The future networks incline to the heterogeneity. Terminals should be able to exploit signals from various technologies. Nevertheless, not even very advanced radio transmission techniques will be able to reach user's requirements. Thus, also bandwidth must be enlarged. This supports evolution of aggregated spectrum techniques among fragmented spectrum and cognitive radio approach. To save radio resources, direct communication between users should be ensured by future mobile networks. Another aspect that attracts increasing attention is energy efficiency. The goal of the future mobile networks is to be more environmental friendly and to reduce power consumption. For this purpose, the network must contain entities for intelligent distribution of radio resources to save energy more efficiently.