TeleFOT

Field Operational Tests of Aftermarket and Nomadic Devices in Vehicles

Large Scale Collaborative Project

7th Framework Programme

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Communication technologies (state-of-the art and trends)

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LIST OF ABBREVIATIONS

ABBREVIATION	DESCRIPTION
3GPP	3rd Generation Partnership Project
AAS	Adaptive Antenna System also Advanced Antenna System
ACL	Asynchronous Connection-Less
AMC	Adaptive Modulation and Coding
BE	Best Effort
BS	Base Station
BSC	Base Station of Control
BTS	Base Transceiver Station
CCI	Co-Channel Interference
CDMA	Code Division Multiple Access
CINR	Carrier to Interference + Noise Ratio
CP	Cyclic Prefix
DL	Downlink
EBU	European Broadcasting Union
EDGE	Enhanced Data rates for GSM Evolution
EIRP	Effective Isotropic Radiated Power
ETSI	European Telecommunications Standards Institute
EUL	Enhanced Uplink
EVDO	Evolution-Data Optimized or Evolution-Data only
FBSS	Fast Base Station Switch
FDD	Frequency Division Duplex
FFRS	Fractionnal frequency reuse scheme
FFT	Fast Fourier Transform
FHSS	Frequency Hopping Spread Spectrum
FIC	Fast Information Channel

ABBREVIATION	DESCRIPTION
FOMA	Freedom of Mobile Multimedia Access
FRS	Frequency reuse scheme
FTP	File Transfer Protocol
FUSC	Fully Used Sub-Channel
GFSK	Gaussian Frequency Shift Keying
GGSN	Gateway GPRS Support Node
GMSC	Gateway Mobile Switching Centre
GPR	Ground Penetrating Radar
GPRS	General packet radio service
GSM	Global System for Mobile communications
HARQ	Hybrid automatic repeat-request
HCR	High Chip Rate
HHO	Hard Hand-Off
HLR	Home Location Registers
HLR	Home Location Register
HO	Hand-Off
HSCSD	High-speed circuit-switched data
HSDPA	High Speed Downlink Packet Access
HS-DPCCH	Uplink High Speed-Dedicated Physical Control Channel
HS-DSCH	High-Speed Downlink Shared Channel
HSPA	High-Speed Packet Access
HS-PDSCH	High Speed-Physical Downlink Shared Channel
HS-SCCH	High Speed-Shared Control Channel
HSUPA	High Speed Uplink Packet Access
IEEE	Institute of Electrical and Electronics Engineers
IMT	International Mobile Telecommunications
ISDN	Integrated Services Digital Network

ABBREVIATION	DESCRIPTION
ISI	Inter-Symbol Interference
LOS	Line of Sight
LTE	Long Term Evolution
MAC	Media Access Control
MAN	Metropolitan Area Network
MAP	Mobile Application Part
MBS	Multicast and Broadcast Service
MDHO	Macro Diversity Hand Over
MIMO	Adaptive Multiple Input Multiple Output
MOT	Multimedia Object Transfer Protocol
MSC	Mobile Switching Centre
MU	Mobile Unit
NLOS	Non Line-of-Sight
nrtPS	Non-Real-Time Packet Service
OFDM Multiplex	Orthogonal Frequency Division
OFDMA	Orthogonal Frequency Division Multiple Access
OSI	Open Systems Interconnection
PHS	Personal Handyphone System
PSTN	Public Switched Telephone Network
PUSC	Partially Used Sub-Channel
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RNC	Radio Network Controller
RRM	Radio Resource Management protocol
RTG	Receive/transmit Transition Gap
rtPS	Real-Time Packet Service
SCO	Synchronous Connection-Oriented

ABBREVIATION	DESCRIPTION
SDP	Service Discovery Protocol
SF	Service Flow
SFN	Single Frequency Network
SGSN	Serving GPRS Support Node
SIM	Subscriber Identity Module
SISO	Single Input Single Output (Antenna)
SMS	Short Message Service
SNIR	Signal to Noise + Interference Ratio
SNR	Signal to Noise Ratio
S-OFDMA	Scalable Orthogonal Frequency Division Multiple Access
SS7	Signaling System 7
STC	Space Time Coding
TD-CDMA	Time Division – Code Division Multiple Access
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TD-SCDMA	Time Division - Synchronous Code division Multiple Access
TPEG	Transport Protocol Expert Group.
TRU	Transceiver Unit
TTG	Transmit/receive Transition Gap
TTI	Transmission Time Interval
UEP and EEP	Unequal / Equal Error Protection
UGS	Unsolicited Grant Service
UL	Up Link
UMTS	Universal Mobile Telecommunications System
USIM	Universal Subscriber Identity Module
UTRAN	UMTS Terrestrial Radio Access Network
UWB	Ultra Wide Band
VLR	Visitor Location Register

ABBREVIATION	DESCRIPTION
VoIP	Voice over IP
W-CDMA	Wideband Code Division Multiple Access
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access

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EXECUTIVE SUMMARY

The Deliverable D2.6.1 *Communication technologies (state-of-the art and trends)* is the first deliverable of WP2.6 Technology and Service Observatory. This work package analyses the available technology and infrastructure equipment in the market in order to have a proper insight into what constitutes the potential and limitations of new ICT solutions for TeleFOT project.

The main objective of TeleFOT is to create a European wide user community for long term testing and assessing mobile driver support functions and services while driving. To do this several Field Operational Tests (FOTs) will be set up all around Europe. These FOTs will be focused on the impact of technically mature ICT systems, and will evaluate safety, user acceptance, efficiency and deployment aspects that will be evaluated and compared among the different FOTs.

The main aim of this document to provide the project and the different FOTs with relevant and up to date information about the different communication technologies available in the market and that can be potentially used in the different FOTs of the project.

Through the different sections and subsections of the deliverable, the different technologies are analysed in terms of their status on the market and compiled in the present document. The study of the communication technologies will include:

- Cellular communications. These will be fundamental for the deployment of services, that is why is so important a deep study on the characteristics and functionalities they offer. GSM, GPRS, EDGE, UMTS (HSDPA, HSUPA, LTE), etc. IP Multimedia Subsystem (IMS) will also be covered, this subsystem will facilitate the deployment and management of multimedia services
- Nomad technologies. Useful for local communication in some scenarios. WiFi (IEEE 802.11), WiMAX (IEEE802.16)
- Adhoc communications. Despite they are not mature enough nowadays, Ad hoc communications will be very important for safety services deployment shortly.
    - Vehicle to Vehicle (V2V), Vehicle to Infrastructure (V2I) and Vehicle to Person (V2P)
    - Routing technologies, including protocols for unicast, multicast and geocast communications
    - Gateways to other networks discovery mechanisms. Needed for the integration with Internet

- Short range communications. Needed for on board communications. NFC, Bluetooth, UWB, ZigBee
- Broadcast communications. Including RDS-TMC, DAB and DVB

The common methodology adopted by the TeleFOT Technology Observatory for the reporting suggests considering different aspects – such as description, maturity, availability, price, usability in TeleFOT, etc - when assessing technical issues.

For each of the technologies studied, a summary description, where a table including technical information - such as Bandwidth, Frequency, Coverage, Standardization, etc – is presented, and a detail description with a more extended description of the characteristics of each technology is presented.

INTRODUCTION

TeleFOT is a Large Scale Collaborative Project under the Seventh Framework Programme, co-funded by the European Commission DG Information Society and Media within the strategic objective "ICT for Cooperative Systems".

Officially started on June 1st 2008, TeleFOT aims to test the impacts of driver support functions on the driving task with large fleets of test drivers in real-life driving conditions.

In particular, TeleFOT assesses via Field operational Tests the impacts of functions provided by aftermarket and nomadic devices, including future interactive traffic services that will become part of driving environment systems within the next five years.

Field Operational Tests developed in TeleFOT aim at a comprehensive assessment of the efficiency, quality, robustness and user friendliness of in-vehicle systems, such as ICT, for smarter, safer and cleaner driving.

The main objective of this Deliverable is to provide an overview of the situation of the communications technologies available in the market.

This deliverable is written as part of the outcomes of the “observatory study” performed in WP 2.6 the main objective of this technology observatory is to have a proper insight into what constitutes the potential and limitations of new ICT solutions, infrastructure equipment, on board devices and innovative functions and services that TeleFOT can find in the market.

More concretely D2.6.2 will compile the work done in tasks T2.6.1. This task is focussed on the study of the communication technologies available. This document will therefore try to summarize the current situation of this type of technologies, by presenting, for each one of the technologies studied, its characteristics, current maturity, availability, price, etc.

This document has been structured in five different sections (apart from the introduction). Each one of these sections will compile a set of technologies which have been grouped depending on their general characteristics.

The first section deals with the Cellular technologies, including under this group the 2G, 3G and 4G communication technologies. GSM, GPRS, UMTS, HSDPA, HSUPA and LTE will be analysed and presented in this section.

The second section, Nomad technologies, will present the nomadic networks which are networks that support direct connection of users to the system. That is, instead of connecting two remote locations, such as a control centre and a base station, these nomadic networks connect individual users (computers, smartphones, pdas, etc) to a network. These technologies include IEEE 802.11 (WiFi) and IEEE 802.16 (WIMAX) which are common standards for such networks, and are presented under this section.

The next section in the document deals with Ad-hoc network technologies. An ad-hoc network can be defined as a network composed by fixed and mobile terminals or bya only mobile terminals, which do not depend on a pre-existing infrastructure and is deployed in a wireless environment.

There is also a section dedicated to short range communication technologies. This type of technologies are needed for onboard communications, as they are very usefull to eliminate cable connections between electronic devices, both portable and fixed, located close to each other, while maintaining high levels of security.

Last but not least there is a section dealing with broadcast which are used point to multipoint communications, where a sending node sends some kind of information to a multitude of receiving nodes simultaneously, without reproducing the same transmission node by node.

All the communication technologies studied and analysed within task T2.6.1 are presented in this deliverable with a similar structure: a summary description, where a table including technical information - such as Bandwidth, Frequency, Coverage, Standardization, etc –, and a detail description with a more extended description of the characteristics of each technology.

CELLULAR COMMUNICATIONS

Ever since its inception in the late 70s, the cellular communications have meant introduced great changes in our daily activities, to the point that cell phones have become a primary tool both for people and business, makes them feel safer and more productive.

Although the cell phone was designed for voice only, due to technological limitations at that time, cellular technology today is able to provide other services such as data, audio and video. This evolution have been performed into several steps named as generations:

The first generation (1G) appeared in the early 80s, it was an analogue technology and was designed strictly for voice transmission. The link quality of voice was very low, low-speed [2400 baud], the transfer between cells was highly inaccurate, have low capacity [based on FDMA, Frequency split a Multiple Access] and security did not exist. The predominant technology of this generation is AMPS (Advanced Mobile Phone System).

The second generation (2G) came in the early 90s, and unlike the first was characterized by being completely digital. The 2G system uses sophisticated encryption and protocols supporting higher data rates for voice and limited data communications. The predominant technologies: GSM (Global System for Mobile Communications), IS-136 (also known as TIA/EIA-136 and ANSI-136) and CDMA (Code Division Multiple Access) and PDC (Personal Digital Communications), the latter used in Japan.

The generation 2.5G provides extended features and additional capabilities to 2G systems such as GPRS (General Packet Radio System), HSCSD (High Speed Circuit Switched Data) and EDGE (Enhanced Data Rates for Global Evolution), among others. European carriers and U.S. moved to 2.5G in the late 90s/early 2000s.

The third generation (3G) brought the convergence of voice and data wireless Internet access, multimedia applications and high data transmissions. The protocols used in 3G systems support higher data speeds for applications focused beyond voice such as audio (MP3), moving video, video conferencing and fast Internet access, just to name a few. UMTS (Universal Mobile Telephone Service) is a good example of 3G technologies.

The fourth generation of mobile communications, or 4G [1], is based entirely on IP communications to provide speeds of 100 Mbps in motion and 1 Gbps without moving, maintaining the quality service of the communication. 4G is not a defined technology or standard, but a collection of technologies and protocols to allow for maximum throughput with the cheapest wireless network. This convergence of technologies arises from the need to group the different standards used to define the scope of operation of each one of them and also to integrate all the communication options in a single device transparently to the user with the objective of ensuring the quality service and meeting the minimum requirements for the transmission of multimedia messaging services, video chat, mobile TV or voice and data services anytime, anywhere. In summary, the 4G system should be able to dynamically share and utilize network resources saving user requirements.

The following paragraphs provide an overview of the long range wireless communication including their role, performances and evolution from the 2G (Second Generation), represented by GSM (Global System for Mobile Communication) and GPRS (General Packet Radio System sometimes defined as 2.5 Generation)) to the 3G (Third Generation) represented by UMTS (Universal Mobile Telecommunications System) and will go beyond 3G with LTE.

In particular GSM and GPRS provide a complete overview of the GSM possibilities that represents the wireless digital communication widely adopted in Europe and supported in North America and in many other world areas.

Today studies on 4G wireless communication systems, have been started with the goal of providing higher data rates and expanded multimedia services allowing the following main capabilities:

- 4G systems should fuse elements of current cellular systems with nomadic wireless-access systems and personal-area networks in a seamless layered architecture that is transparent to the user
- Data rates of 100 Mbps for mobile applications and 1 Gbps for nomadic applications
- To use worldwide common spectrum and open under a global standardization.

2. GSM - Global System for Mobile Communication

2.1. Summary

The following table shows the main characteristics of GSM:

Description	GSM- An open, digital cellular technology used for transmitting mobile voice and data services
Maturity	This technology is totally proven, and it's available worldwide
Availability	Available worldwide. Western Europe, supported in North America and in many other areas.
Price	Tariff based on circuit use
Bandwidth	14,4 Kbps
Frequency	900/1800 MHz
Application to vehicular environment	Information services, safety messages, etc
Infrastructure requirements	Interface adapter depending on the application
Standardization	Global standardisation available

Use in TeleFOT test site	Yes depends on the application requirements
How to incorporate this technology to TeleFOT test site?	Blutooth capabilities and other phisycal interfaces available on current commercial terminal would make it not difficult

Table 1 – GSM

GSM (stands for Global System for Mobile communications) is the most widely used digital mobile phone system and the most spread standard for mobile phones in Europe.

Originally defined as open European standard for digital mobile phone networks supporting voice, data, text messages and roaming in many countries. GSM is now one of the 2G digital wireless standards in the world. The GSM is present in over 160 countries and according to the GSM Association, have more the majority of the total digital wireless market.

The mail reasons for this wide spread of GSM use are the following main features:

- Worldwide roaming.
- Operational model, very simple and similar to the wireline network.

The amount of GSM users has gone over the three billion users in the world, and it is still growing.

2.2. Detailed description

The GSM has been designed as the second generation of the mobile network (2G), in order to achieve a solution capable of increasing the quantity of the data payload being transported. The GSM system is based on digital modulation techniques like TDMA (Time Division Multiple Access) [GSM1]. The adoption of these techniques allowed the inclusion of a growing amount of users in a specific range of frequencies. In addition, encryption techniques were introduced in order to ensure the communication privacy. It represents an enhancement, if compared with the analogical systems, where the conversations could be heard simply using a scanner.

The telephony leading service carried by the GSM system was accompanied by the "*vocoder*", a digital encoding and decoding system. In addition to the voice services, new features are installed, such as the data rate up to 9.6 Kbit/sec. However, the most interesting and popular service promoted by the GSM is the SMS (Short Message Service).

The basic requirements that have driven the GSM system concept development and its implementation are:

- the low cost of the devices;
- the devices' user friendliness;

- international roaming guaranteed;
- the optimized usage of the spectrum;
- the compatibility with the digital wireline network.

The current system respects these requirements. Moreover, the 200 kHz bandwidth of the radio channels with TDMA (Time Division Multiple Access) allows up to 8 users per channel allocation.

2.2.1. GSM Architecture

A GSM system consists of standardized interfaces and functional entities which can adapt to any cellular network system with any mobile station.

The following figure shows the architecture of the GSM system. As it is depicted below, each mobile device is connected to a Base Station (BS). Several Base Stations, in small groups, interact with a Base Station of Control (BSC), usually hosted in a Base Transceiver Station (BTS) managed by the mobile service provider. Moreover, each BSC is connected to Mobile Switching Centres (MSC) interacting with the Home Location Registers (HLR). The mobile unit then automatically switches from the current channel tothe new channel maintaining the communication (handover). Further functions are provided by VRL (Visitor Location Register, that stores information about Base Stations that are active for the visitor avoiding frequent access to HRL) and by EIR (Equipment Identity Register is a database with information to be compared with IMEI (International Mobile Equipment Identity) to carry out further checks. [GSM2]

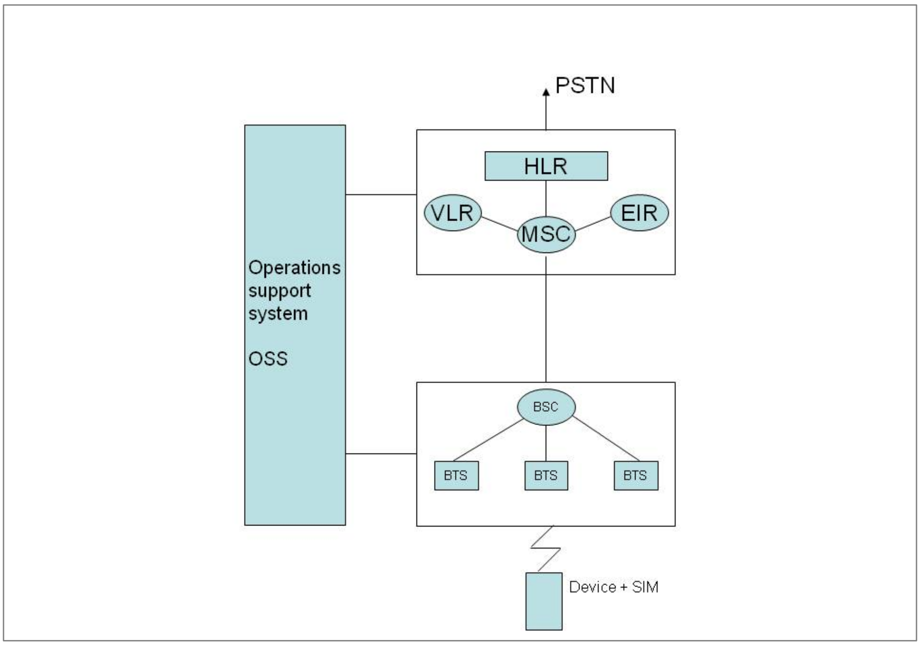

Figure 1 - GSM Architecture

In addition to this, the GSM architecture includes an Authentication Centre, involved in the service payment operations. During the authentication, the SIM (Subscriber Identity Module) plays a fundamental role: it allows the unambiguous identification of the user, permitting the seamless mobile terminal substitution, because keeping the same identity in the net.

3. GPRS - General Packet Radio System

3.1. Summary

The following table shows the main characteristics of GRS

Description	GPRS (General Packet Radio Service) is a very widely deployed wireless data service, available now with most GSM networks
Maturity	This technology is totally proven
Availability	Availability very near to GSM
Price	Tariff on data volume exchanged
Bandwidth	40 Kbps
Frequency	900/1800 MHz
Application to vehicular environment	Relevant for implementing mobile Telefot applications
Infrastructure requirements	Possible additional interfaces depending on the application/service to be implemented
Standarization	Worldwide
Use in TeleFOT test site	Yes. Used in the sites where the on-board device is sending information to an of-board device.
How to incorporate this technology to TeleFOT test site?	It depends on the service to be implemented. No particular obstacle

Table 2 – GPRS

After the advent of the GSM, the need to have higher bit rate channel brought the research towards the GPRS (General Packet Radio System) [GPRS1]. This technology offers a much higher bit rate than the GSM, getting to 172 Kbit/sec and, at the same time, it enables the access to many internet services via browser.

3.2. Detailed description

Unlike the GSM, based on the circuit switching, the GPRS is based on the packet switched service, allowing a more efficient usage of the available digital channels.

Using the circuit switching, the service pricing depends on the time and the distance, whereas the GPRS data transfer is typically charged per traffic transferred bytes, regardless of whether the user actually is using the connectivity. This difference is the base of the always-on connectivity.

The always-on connectivity, in addition to a higher bit rate, enables many applications supported by innovative devices, like PDAs and Smart phones, allowing the creation of internet services.

The GPRS solution is the result of software updating of the GSM system, and this is the reason why its cost is affordable: GSM and GPRS substantially operate on the same infrastructure and use the same Base Stations. In the following figure, the GPRS architecture is shown. [GPRS2]

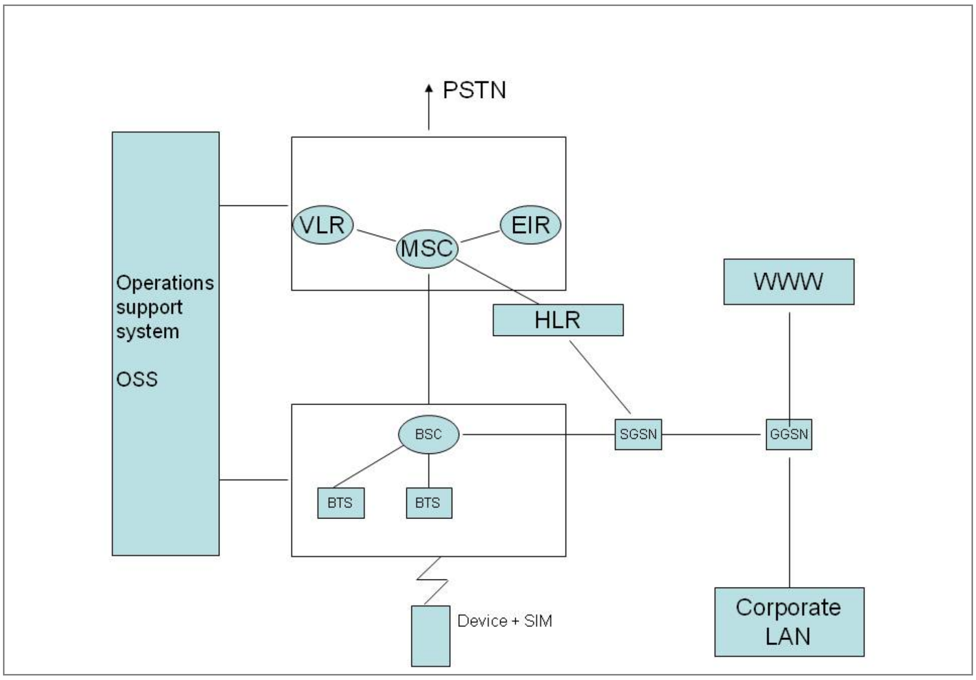

Figure 2 - GPRS Architecture

As already mentioned, the GPRS infrastructure is not completely separated from the GSM: the BTS and its relative control are the same with the addition of a new hardware component, called Packet Control Unit. The additional elements are basically two: the Serving GPRS support Node (SGSN) and the Gateway GPRS Support Node (GGSN). The SGSN manages the IP packets' routing, the addresses and it keeps active the sessions during the handover. The GGSN is basically a gateway, router and firewall in one block that interconnects the mobile network and the Internet, including the Corporate LANs. The figure above shows other functional blocks already described in previous paragraphs

4. EDGE – Enhanced Data rates for GSM of Evolution

4.1. Summary

The following table shows the main characteristics of EDGE:

Description	EDGE (Enhanced Data GSM Environment) is a faster version the Global System for Mobile (GSM) wireless service designed to deliver data at rates up to 384 Kbps and enable the delivery of multimedia and other broadband applications to mobile phone and computer users. The EDGE standard is built on the existing GSM standard, using the same time-division multiple access (TDMA) frame structure and existing cell arrangements.
Maturity	Mature
Availability	Available all over Europe
Price	Different tariffs depending on use and company
Bandwidth	Up to 384 Kbps
Frequency	900/1800 MHz
Infrastructure requirements	The same as for GSM, interface adapter depending on the application
Standarization	Global standardisation available
Use in TeleFOT test site	Yes, depending on the application
How to incorporate this technology to TeleFOT test site?	It depends on the service to be implemented. No particular obstacle

Table 3 – EDGE

Further enhancements to GSM networks are provided by Enhanced Data rates for GSM Evolution (EDGE) technology, which provides up to three times the data capacity of GPRS.

EDGE allows the delivery of advanced mobile services such as the downloading of video and music clips, multimedia messaging, high-speed Internet access and e-mail . EDGE uses the same structure, as today's GSM networks, which allows it to be overlaid directly onto an existing GSM network. For many existing GSM/GPRS networks, EDGE is a simple software-upgrade.

Due to the very small incremental cost of including EDGE capability in GSM network deployment, virtually all new GSM infrastructure deployments are also EDGE capable

4.2. Detailed description

EDGE is the next step in the evolution of GSM and TDMA. The aim of this new technology is to provide higher transmission rates, better spectral efficiency, and facilitate new applications and increased capacity for the mobile user. With the introduction EDGE in GSM phase 2 +, existing services such as GPRS and HSCSD are improved by offering a new physical layer. EDGE is introduced into the specifications and existing descriptions rather than creating new ones.

EDGE is a method to increase data speeds on the radio link GSM. While GPRS allows data rates of 115kbps and theoretically 160 kbps at the physical layer, with the implementation of EDGE system would be able to achieve rates of 384 and theoretically 473.6kbps. Basically, EDGE only introduces a new modulation technique and new channel coding that can be used interchangeably to transmit voice and data packet switching and circuit. EDGE, therefore, is an addition to GPRS and can not work separately. This will can increase the applications available to access the Internet wirelessly e-mail and file transfers. [EDGE1]

4.2.1. EDGE Architecture.

EDGE Architecture is the same as in GPRS, with some minor changes. The packet control unit can be placed either in the base station, in the base station controller, or in GPRS support node. The unit control center is always placed on the base station. The controller of the radio link is must extend to EDGE to avoid the collapse of the radio protocol

The main changes from GPRS to EDGE are a software update in the base station controller and the addition of an element called EDGE TRU (Transceiver Unit) in the Base Transceiver Radio.

GPRS and EDGE share the same package management protocols, and therefore they behave in the same way. By using EDGE, the same time slot can support a higher number users. This decreases the number of radio resources required to support the same traffic, freeing up the system capacity to support more traffic of voice or data. Like GPRS, EDGE can be seen as an element that increases the capacity of the system when necessary.

4.2.2. EDGE technology.

EDGE uses the GPRS standard techniques to provide significant improvements.

GPRS and EDGE share the same symbol transmission rate (270 KSIM/s), despite of that the modulation bit rate is not the same. Because of this, EDGE can transmit 3 times more bits than GPRS during the same period of time. The EDGE data transmission rate is 384kbps. This rate corresponds to 48 kbps per slot time, assuming a terminal 8 time slots.

4.2.2.1 EDGE modulation technique.

EDGE is specified to reuse the channel structure, channel width, encoding channel and the other existing mechanisms from GPRS and HSCSD. The modulation selected for EDGE was 8PSK or 8-phase shift keying, as this type of modulation meets all the requirements desired for EDGE.

8PSK modulation has the same characteristics in terms of generating adjacent channel interference than GMSK modulation. This makes it possible to integrate the EDGE channels into an existing frequency plan and allocate new EDGE channels in the same way as it is done for GSM channels.

The modulation method used is a linear method in which 3 consecutive bits are mapped to a symbol at the I/Q plane.

The symbol rate remains constant, however, using this modulation per each symbol three bits are sent, instead of just 1. Therefore, the transmission rate is increased by a factor of 3. Under good conditions for transmission, this modulation is optimal, in the case of bad conditions, and due to the fact that the bits are much closer in this type of modulation, this leads to errors in receiving the correct symbol. To avoid this in the case of bad transmitting conditions, the extra bits are used to add code to detect errors. [EDGE2]

4.2.2.2 Package Management

EDGE has the capability to retransmit a packet that was not correctly decoded with a more robust coding scheme. with GPRS, if a packet was not successfully received it is retransmitted with the same encoding scheme. In EDGE there is an algorithm that decides the level of reliability with which the link should work and adapts the codification to that level. The result of this is the re-segmentation. The packages sent with a low protection error index can be retransmitted with a higher error protection, if the conditions change and is necessary to ensure the data transition.

4.2.2.3 Address window.

Prior to the transmission of a sequence of coded packages or radio blocks through the Um interface, the transmitter must assign an identification number for each packages. After the transmission of a sequence of packages, the transmitter asks the receiver if the packets were received correctly by sending an acknowledge/unacknowledged report. This report provides the transmitter information about the identification number of the packages that were not received correctly and require retransmission. With EDGE, the number of possible assignments is 2048 and the addressing window can be 1024.

5. UMTS– Universal Mobile Telecommunications System

5.1. Summary

In the following table, a summary with the most important characteristics is given:

Description	Universal Mobile Telecommunications System. It is a system based in WCDMA protocol. [Wideband Code Division Multiple Access]
Maturity	This technology is totally proven, and it’s working at its maximum rate all over Europe
Availability	Available in densely population regions
Price	N/A
Bandwidth	21 Mbps
Frequency	900 MHz
Infrastructure requirements	3G modem
Standarization	3GPP
Use in TeleFOT test site	Yes in the test sites using devices with SIM card
How to incorporate this technology to TeleFOT test site?	It depends on the service to be implemented. No particular obstacle

Table 4 – UMTS

5.2. Detailed description

UMTS (Universal Mobile Telecommunications System) is a communication standard used in the third generation mobile telephony. UMTS provides broadband on mobile communications and broadcast a significant amount of data over the network. With the third generation video conferencing is possible, download videos, exchange e-card, horseback 'virtual' for homes for sale, etc ... everything from the phone.). [UMTS1]

The UMTS system offers a wide range of different applications and services with different qualities of service, QoS (Quality of Service). These services and applications can be classified into different categories considering its QoS:

- Conversational or voice traffic
- Streaming (continuous flow)

- Interactive
- Background (delayed, background, secondary)

The biggest difference between these four types of traffic is the users delay perception, being the first one the most sensitive to the delay in the communication and the background the least one. In the next paragraphs there is a brief description of each of these four types of traffic:

Conversational: This type of communication is based in real time services generating symmetric traffic, such as voice or video telephony, where the information is very sensitive to delays. The most important premises are to preserve the temporal relationship between the various entities participating in communication, and maintaining a constant and very low delay in the communication flow.

Streaming: it is oriented to a stream communication in real time such as the video transfer or the connection to a distributor to get music on your mobile. Such applications are very asymmetric and less delay-sensitive conversational services. These services are generally unidirectional services. The user receives an audio or video signal but the flow is mainly in one direction. In this case the delay is not that important, but if it exists, it is advisable to remain constant.

Interactive services: These kind of communication is intended for applications where the end user is requesting data while he is connected, such as access to Web pages, online games, consultation basis data and access to servers. One of the main characteristics is that the user makes the request waits for a certain period of time response (variable and moderate delay), and this information is transmitted with low bit error rates.

Background communication: This type of communication is intended for the end user, usually a machine, is sending and receiving data files, for example e-mail applications, SMS, downloads, databases, etc. In these cases the time delay is not very important, yet it must guarantee the integrity of the data, so the bit error rates should be very low.

5.2.1. UMTS Architecture

UMTS architecture is divided into three main parts: the air interface (access technique used), the UMTS Terrestrial Radio Access Network (UTRAN) and core network. The next Figure shows these elements.

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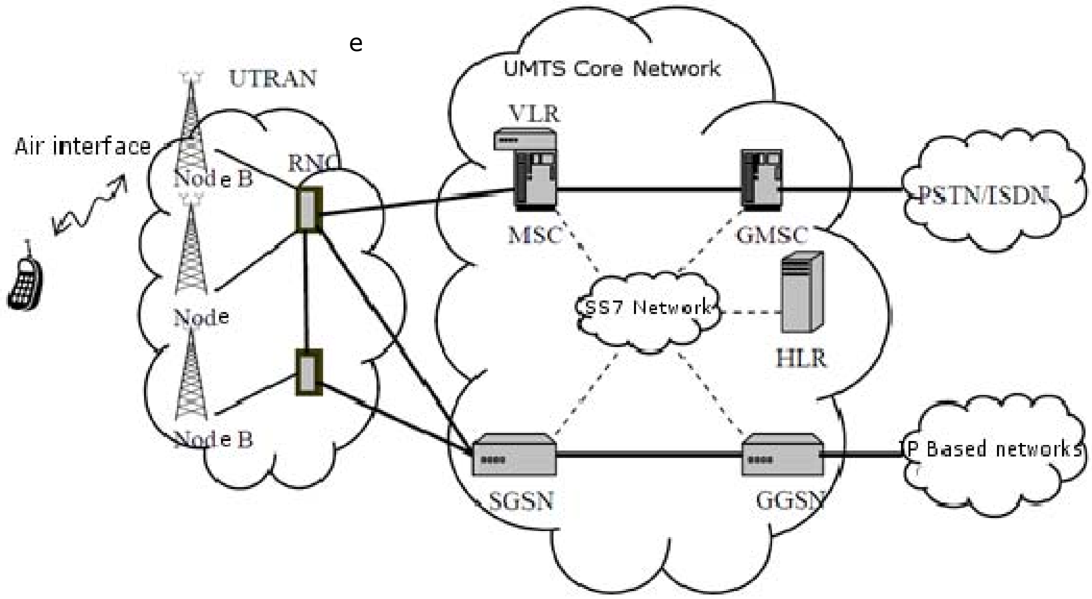

Figure 3 - UMTS Architecture

MSC: Mobile Switching Centre

GMSC: Gateway Mobile Switching Centre

SGSN: Serving GPRS Support Node

GGSN: Gateway GPRS Support Node

HLR: Home Location Register

ISDN: Integrated Services Digital Network

UTRAN: UMTS Terrestrial Radio Access Network

PSTN: Public Switched Telephone Network

SS7: Signaling System 7

RNC: Radio Network Controller

VLR: Visitor Location Register

The Radio Access Network (UTRAN) includes Base Stations (Node B) and Radio Network Controller (RNC). The UTRAN is responsible for the mobility management, the mobility of terminals is transparent to the core network. It is also responsible allocating resources through the air interface and provide support on Quality of Service (QoS, Quality of Service) requested.

The core network is composed by the circuit switched domain (CS, Circuit Switching) and the packet switched domain (PS, Packet Switching). The Circuit switching involves the MSC / VLR and GMSC, and it establishes the connections to telephone networks. The packet switching domain comprises the SGSN and the GGSN, and routes the information packets to IP based networks. The following paragraphs describe an overview of the elements of the network and its functions:

Base Station (Node B), is mainly responsible for the conversion, reception and data transmission over the air interface to the user terminals. Performs error correction bit-rate adaptation, modulation at radio level, and it also sends reports measurements to the RNC.

RNC is an ATM switch that can multiplex / demultiplex packets and users circuits together. The RNC is connected together through the Iur interface, solves problems of allocation of radio resources, controls the congestion autonomously and the transfers the users from the coverage of one base station to another. The RNCs perform a monitoring of the network and issue appropriate alerts and reports when needed.

SGSN is responsible for session management data to produce charges information and legal interception. They Route the packets to the respective RNC, establish and finalyse the sessions, and also defines QoS patterns available for each session.

GGSN works as an input/output router, it contains a firewall and IP address assignment methods. It can send service requests to intranets and notify service charges to the users.

MSC is a Mobile switching centre and also contains a Visitors Location register. The MSC is responsible for the operation of circuit switching. It is in charge of the connection management (setting circuits), mobility management (called and location of users) and is also responsible for some security features and generation of CDRs (Call Detail Record) for billing purposes.

GMSC is a border node that processes inbound and outbound network external such as public switched telephone network (PSTN) to traffic on switching circuits. For incoming calls it is in constant communication with the MSC after having questioned the HLR, and then establishes the connection.

5.2.2. Technology

UMTS combines three different air interfaces, GSM's Mobile Application Part (MAP) core, and the GSM family of speech codecs.

5.2.2.1 Air interfaces

UMTS provides several different terrestrial air interfaces, called UMTS Terrestrial Radio Access (UTRA). All air interface options are part of ITU's IMT-2000. In the currently most popular variant for cellular mobile telephones, W-CDMA (IMT Direct Spread) is used.

Please note that the terms W-CDMA, TD-CDMA and TD-SCDMA are misleading. While they suggest covering just a channel access method (namely a variant of CDMA), they are actually the common names for the whole air interface standards.

Non-terrestrial radio access networks are currently under research.

W-CDMA (UTRA-FDD)

W-CDMA uses the DS-CDMA channel access method with a pair of 5 MHz channels. In contrast, the competing CDMA2000 system uses one or more arbitrary 1.25 MHz channels for each direction of communication. W-CDMA systems are widely criticized for their large spectrum usage, which has delayed deployment in countries that acted relatively slowly in allocating new frequencies specifically for 3G services.

The specific frequency bands originally defined by the UMTS standard are 1885–2025 MHz for the mobile-to-base (uplink) and 2110–2200 MHz for the base-to-mobile (downlink).

W-CDMA is a part of IMT-2000 as IMT Direct Spread.

UTRA-TDD HCR

UMTS-TDD's air interfaces that use the TD-CDMA channel access technique are standardized as UTRA-TDD HCR, which uses increments of 5MHz of spectrum, each slice divided into 10ms frames containing fifteen time slots (1500 per second). The time slots (TS) are allocated in fixed percentage for downlink and uplink. TD-CDMA is used to multiplex streams from or to multiple transceivers. Unlike W-CDMA, it does not need separate frequency bands for up- and downstream, allowing deployment in tight frequency bands.

5.2.2.2 Radio access network

UMTS also specifies the UMTS Terrestrial Radio Access Network (UTRAN), which is composed of multiple base stations, possibly using different terrestrial air interface standards and frequency bands.

UMTS networks are often combined with GSM/EDGE, the latter of which is also a part of IMT-2000.

Security includes two procedures: integrity and ciphering. Integrity validates the resource of message and also makes sure that no one (third/unknown party) on radio interface has not modified message. Ciphering makes sure that no one listens your data on air interface. Both integrity and ciphering will be applied for SRBs where as only ciphering will be applied for data RBs.

5.2.2.3 Core network

With Mobile Application Part, UMTS uses the same core network standard as GSM/EDGE. This allows a simple migration for exiting GSM operators. However, the migration path to UMTS is still costly: while much of the core infrastructure is shared with GSM, the cost of obtaining new spectrum licenses and overlaying UMTS at existing towers is high.

The CN can be connected to various backbone networks like the Internet, ISDN. UMTS (and GERAN) include the three lowest layers of OSI model. The network layer (OSI 3) includes the Radio Resource Management protocol (RRM) that manages the bearer channels between the mobile terminals and the fixed network, including the handovers.

5.2.3. Interoperability and global roaming

UMTS phones (and data cards) are highly portable—they have been designed to roam easily onto other UMTS networks (assuming the providers have roaming agreements in place). In addition, almost all UMTS phones are UMTS/GSM dual-mode devices, so if a UMTS phone travels outside of UMTS coverage during a call the call may be transparently handed off to available GSM coverage. Roaming charges are usually significantly higher than regular usage charges.

Most UMTS licensees consider ubiquitous, transparent global roaming an important issue. To enable a high degree of interoperability, UMTS phones usually support several different frequencies in addition to their GSM fallback. Different countries support different UMTS frequency bands – Europe initially used 2100MHz while the most carriers in the USA use 850Mhz and 1900Mhz. T-mobile has launched a network in the US operating at 1700MHz (uplink) /2100MHz (downlink), and these bands are also being adopted elsewhere in America. A UMTS phone and network must support a common frequency to work together. Because of the frequencies used, early models of UMTS phones designated for the United States will likely not be operable elsewhere and vice versa. There are now 11 different frequency combinations used around the world—including frequencies formerly used solely for 2G services.

UMTS phones can use a Universal Subscriber Identity Module, USIM (based on GSM's SIM) and also work (including UMTS services) with GSM SIM cards. This is a global standard of identification, and enables a network to identify and authenticate the (U)SIM in the phone. Roaming agreements between networks allow for calls to a customer to be redirected to them while roaming and determine the services (and prices) available to the user. In addition to user subscriber information and authentication information, the (U)SIM provides storage space for phone book contact. Handsets can store their data on their own memory or on the (U)SIM card (which is usually more limited in its phone book contact information). A (U)SIM can be moved to another UMTS or GSM phone, and the phone will take on the user details of the (U)SIM, meaning it is the (U)SIM (not the phone) which determines the phone number of the phone and the billing for calls made from the phone.

6. HSDPA - High Speed Downlink Packet Access

6.1. Summary

In the following table, a summary with the most important characteristics is given:

Description	High Speed Downlink Packet Access
Maturity	Technology under development, nowadays its bandwidth is 7,2 Mbps
Availability	Available in densely population regions.
Bandwidth	14,4 Mbps
Frecuency	2100 MHz

Infrastructure requirements	3G Modem
Standarization	3GPP
Use in TeleFOT test site	Yes

Table 5 – HSDPA

HSDPA stands for High speed downlink packet access. HSDPA is the new 3G communication technology included in 3GPP Release 5. HSDPA allows the user to achieve higher data rates, and capacity in the downlink while on the move.

In order to increase the performance of the communication system, HSDPA introduced new characteristics in the system, such as AMC, which allows adjustable bit rate according to the quality of the channel, HARQ, responsible for retransmitting packets with errors, and a layer in the BS for Medium Access Control. [HSDPA1]

One of the main features of UMTS/HSDPA was the implementation of new channels which operate in parallel with DCH from Release 99. Three new channels were introduced:

6.2. Detailed description

HSDPA mobile technology is specially designed to improve UMTS technology. In fact, HSDPA is known as 3.5G and will serve as a transition between the 3G mobile to the fourth generation (4G).

HSDPA has been designed so that the new protocols are compatible with the predecessors in order to incorporate improvements and new applications in a more easy way. In the 3G telephony, HSDPA uses the infrastructure already deployed for UMTS, as it did with GSM. Obviously, implementing HSDPA involves a series of changes, both in adding new elements in the UMTS network architecture and the implementation of new protocols of operation, but those would be easy to implement.

HSDPA is designed to achieve higher bandwidth on UMTS / WCDMA with a new channel shared among all users on the downlink. The new channel improves both spectral efficiency and speed of data transfer up to 14.4 Mbps. That is, the spectral efficiency increases HSDPA on WCDMA, the same way that it increased it on GSM or GPRS (General Packet Radio Services) [HSDPA 2]

The introduction of this new transport channel known as HS-DSCH impacts several protocol layers; the most significant changes are in the physical and MAC layers. The following features enable the high throughput capabilities of HSDPA:

- HSDPA introduces an Adaptive Modulation and Coding (AMC) scheme, whereby modulation method and coding rate are selected based on information about channel conditions provided by the terminal and the Node-B. In the downlink

HSDPA supports 16QAM as a higher-order modulation method for data transmission under good channel conditions, in addition to QPSK which is already specified for use in WCDMA.

- HSDPA uses a Hybrid Automatic Repeat reQuest (HARQ) protocol to handle re-transmissions and to guarantee error-free data transmission. HARQ is a key element of the new MAC entity denoted as MAChs, which is located both in the Node-B and in the User Equipment (UE).
- A fast packet scheduling algorithm, which allocates the HS-DSCH resources (such as time slots and codes) to the different users, is also implemented as part of Node-B functionality.

6.2.1. New transport and physical channels

Downlink Channels

- HS-DSCH (High-Speed Downlink Shared Channel): Common transport channel for U-plane traffic.
- HS-SCCH (High-Speed Shared Control Channel): Common control channel including information such as user equipment identity.

Uplink Channels

- HS-DPCCH (High-Speed Dedicated Physical Control Channel): Feedback channel for HARQ ACK/NACK messages as well as for channel quality information.

The HS-DSCH is a transportation channel for user data downstream. This channel is mapped to channel HS-PDSCH physical layer. The main characteristics:

- The lack of fast power control. Instead, select the appropriate combination of codes, coding and modulation rates are used.
- Support of higher order than the DCH. With 16QAM the number of bits per symbol is doubled compared to QPSK.
- The plot of 10 ms for WCDMA is divided into 5 subframes of 2 ms, also called transmission time intervals (TTI). In the channel HS-DSCH uses a maximum of 15 parallel codes, which can be appointed to a user during a TTI, or may be divided among multiple users.
- The use of physical layer retransmissions and retransmissions combined. - The lack of soft-handover.
- Lack of control of information in the physical layer on HSPDSCH, instead this is carried out in the HS-SCCH.
- In the multicode operation only using a factor spread of 16. - With HSDPA turbo coding is used only.
- There is no discontinuous transmission (DTX) at the canal. The HS-PDSCH is conveys all or nothing in the 2 ms TTI. An important property of the HS-DSCH is the dynamic nature of the distribution of resources, enabled by the short period of assignment, which is 2 ms. With the HS-PDSCH, once there is data to transmit,

there HS-DSCH transmission to the user in question, resources are assigned to another user (2 ms)

6.2.2. Hybrid automatic repeat-request (HARQ)

Data is transmitted together with error correction bits. Minor errors can thus be corrected without retransmission.

The user equipment monitors the HS-SCCHs looking for an indication that there is about to be some data destined for it. When there is data indicated, the HS-SCCH contains enough information to enable the UE to decode the data on the HS-DSCH. The timing of the 2 channels is arranged so that the signalling can be decoded in time to allow decoding of the user data.

When the user data block is received on the HS-DSCH, it is checked to see if it has been received correctly. If yes, an ACK (positive acknowledgement) is sent on the uplink HS-DPCCH. If not, it is stored at the receiver to allow combination with a repeat attempt, and a NACK (negative acknowledgement) is sent on the uplink HS-DPCCH. Even if the retransmitted packets are corrupted, their combination can yield an error-free packet. Retransmitted packet may be either identical (Chase combining) or different from the first transmission (incremental redundancy). The function attempting to correctly receive a particular data block is referred to as a HARQ process. [HSDPA 2]

The system may have several HARQ processes running concurrently for a user equipment

6.2.3. MAC layer

HSDPA not only introduces new transport and physical layer channels, but also has an impact on higher-layer protocols, including the MAC layer.

Different types of MAC entities are identified for different classes of transport channels. In 3GPP Rel. 99, dedicated and common transport channels are differentiated, and consequently, the MAC layer contains a MACd and a MAC-c entity. The introduction of HSDPA made it necessary to define a new entity called the MAC-hs. As opposed to Rel. 99 specifications, where the MAC layer is implemented in the RNC, the MAC-hs is used in Node-B to take into account the requirement for a high-performance implementation of the standard.

The Node-B MAC-hs is responsible for handling layer-2 functions related to the HS-DSCH, and includes the following functions:

- Handling of the HARQ protocol, including generation of ACK and NACK messages.
- Re-ordering of out-of-sequence subframes. Note that this is actually a function of the RLC protocol; however, this protocol layer is not implemented in the Node-B for the HS-DSCH. Therefore, the MAC-hs must take over some of the critical tasks of the RLC. Subframes may arrive out-of-sequence as a result of the re-transmission activity of the HARQ processes.
- Multiplexing and de-multiplexing of multiple MAC-d flows onto/from one MAC-hs stream.
- Downlink packet scheduling.

6.2.4. Roadmap

The first phase of HSDPA has been specified in 2002 by the 3rd Generation Partnership Project (3GPP). Phase one introduces new basic functions and is aimed to achieve peak data rates of 14.0 Mbit/s (see above). The High Speed Downlink Shared Channels (HS-DSCH), the adaptive modulation QPSK and 16QAM and the High Speed Medium Access protocol (MAC-hs) in base station have been also introduced in HSDPA.

The second phase of HSDPA was specified in the 3GPP release 7 in 2007 and was named HSPA Evolved. It can achieve data rates of up to 42 Mbit/s. It introduces antenna array technologies such as beam forming and Multiple-input multiple-output communications (MIMO). Beam forming focuses the transmitted power of an antenna in a beam towards the user's direction. MIMO uses multiple antennas at the sending and receiving side.

Further releases of the standard have introduced dual carrier operation, i.e. the simultaneous use of two 5 MHz carrier. By combining this with MIMO transmission, peak data rates of 84 Mbit/s can be reached under ideal signal conditions.

After HSPA Evolved, the roadmap leads to E-UTRA (Previously "HSOPA"), the technology specified in 3GPP Release 8. This project is called the Long Term Evolution initiative. The first release of LTE offers data rates of over 320 Mbit/s for downlink and over 170 Mbit/s for uplink using OFDMA modulation.

7. HSUPA– High Speed Uplink Packet Access

7.1. Summary

In the following table, a summary with the most important characteristics is given:

Description	High Speed Uplink Packet Access
Maturity	Technology under development, nowadays its bandwidth is 84 Mbps
Availability	Available in densely population regions
Bandwidth	84 Mbps
Frecuency	2100 MHz
Infrastructure requirements	3G Modem
Standarization	3GPP
Use in TeleFOT test site	Yes

Table 6 – HSUPA

HSUPA or High-Speed Uplink Packet Access is a data access protocol for mobile networks with a high data transfer rate of (up to 5.76 Mbit / s). Known as Generation 3.75 is an evolution of HSDPA (High-Speed Downlink Packet Access, also known as 3.5G). HSUPA is defined in Universal Mobile Telecommunications System standard published by Release 6 3GPP. This technology offers a substantial improvement in speed for the upstream from the terminal to the network. [HSDPA 2]

7.2. Detailed description

The standard for mobile communications HSUPA (High-Speed Uplink Packet Access), defined in Release 6 of 3GPP UMTS published in December 2004, increase the transfer rate in the uplink to 7.2 Mbps.

Many similarities between the two standards are complementary resulting in the HSPA (High-Speed Packet Access), commonly known as 3.5 G or 3.75 G

Logically, these features have required some adaptation given the obvious differences between the uplink, where the communication is from several terminals to a single base station and the downlink, where the situation is just the opposite. [HSDPA 2]

7.2.1. Architecture

To increase capacity and performance in the uplink, HSUPA uses two basic technologies already used previously in HSDPA:

- - Fast scheduling
- - Fast ARQ with Soft Combining Hyb

HSUPA also includes the possibility of using a 2 ms TTI, allowing faster adaptation in the downlink, enabling better utilization of available resources.

All these improvements are implemented on WCDMA in a new transport channel called E-DCH (Enhanced Dedicated Channel).

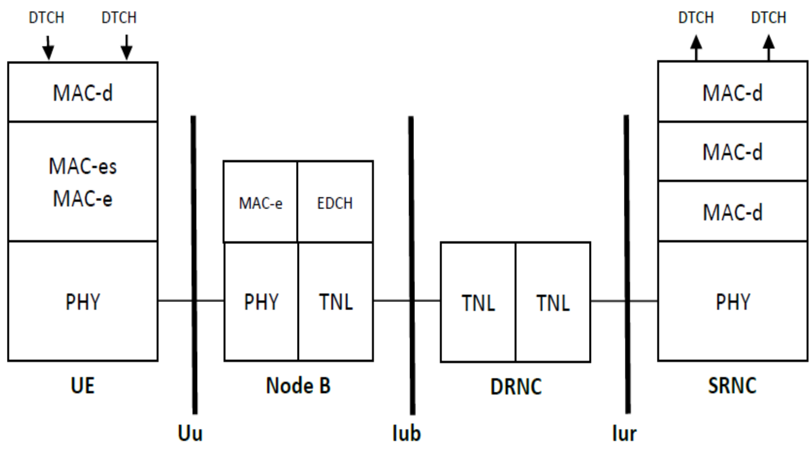

Figure 4 - HSUPA Architecture

To support the new capabilities of the uplink, the nodes are modified from those used in WCDMA Release 99:

- User Equiment: a new MAC-es/MAC-entity appears. This entity will be responsible for handling the HARQ retransmissions, the allocation of resources, and the TFC selection of the channel E-DCH.
- Node B: appears a new entity MAC
- S-RNC: a new MAC entity appears, it will be used to ensure the proper order in the data stream and allow the combination of information received from different B Nodes in case of soft handover.

7.2.2. Transport Channel

Enhanced Dedicated Channel is the new uplink transport channel where improvements are implemented with respect to the Release 99 uplink. The processing of this new transport channel is similar to the DCH channel, with only two differences:

- Each equipment can only have one single E-DCH channel active at any time, whereas previously it was possible to multiplex information on different DCH channels in parallel. In any cas the MAC layer is able to multiplex different services over a single channel E-DCH.
- Hybrid-ARQ is supported

The processing of the transport layer maps the information contained in the E-DCH channel on one or several E-DPDCH channels for transmission over the physical layer. An extra E-DPDCH channel containing all the control information neded to receive the data channel properly, is transmitted in parallel.

This function is performed in parallel with the processing of the uplink DCH channel R99. Therefore, both transport channels can be used simultaneously on the same terminal, the channel DCH being limited to a maximum speed of 64kbps when E-DCH channel is configured. [HSDPA 3]

7.2.3. Physical channels

The definition of the E-DCH introduces five new physical layer channels, two of them will be used for the uplink and three for the downlink:

Uplink Physical channels:

- E-DPDCH (Enhanced Dedicated Physical Data Channel): This is the physical channel used by E-DCH for the transmission of user data.
- E-DPCCH (Enhanced Dedicated Physical Control Channel): Control channel associated with the E-DPDCH providing information to the Node-B on how to decode the E-DPDCH.

Downlink Physical channels

- E-AGCH (Absolute Grant Channel): Provides an absolute power level above the level for the DPDCH (associated with a DCH) that the user equipment should adopt.
- E-RGCH (Relative Grant Channel): Indicates to the UE whether to increase, decrease or keep unchanged the transmit power level of the E-DCH.
- E-HICH (HARQ Acknowledgement Indicator Channel): Used by Node-B to send HARQ ACK/NACK messages back to the UE.

The E-HICH has a similar function to HSDPA's HS-DPCCH, namely that it is used to provide HARQ feedback information (ACK/NACK). However, it does not contain CQI information, since HSUPA does not support Adaptive Modulation and Coding.

The Node-B contains an uplink scheduler for HSUPA. However, the goal of the scheduling operation is completely different compared with that of HSDPA. The aim of HSDPA is to allocate HS-DSCH resources (in terms of time slots and codes) to multiple users, the goal of the uplink scheduler is to allocate only as much capacity (in terms of transmit power) to the individual E-DCH users as is necessary to ensure that Node-B does not have a "power-overload".

The two physical scheduling channels E-RGCH and E-AGCH tell a user equipment how to regulate the power level transmitted. In case of the E-RGCH, the user equipment is configured to either increase or decrease one step the power level transmitted, or to keep the current power level unchanged. In the case of the E-AGCH, the Node-B provides an absolute value for the power level of the E-DCH at which the UE should transmit

7.2.4. Roadmap

After HSUPA the 3GPP is working on further advancing transfer rates. LTE provides up to 326.4 Mbit/s for downlink and 86.4 Mbit/s for uplink. LTE-Advanced, in development as a minor update to LTE networks, supports maximum download rates of over 1 Gbit/s.

8. LTE – Long Term Evolution

8.1. Summary

The following table shows a summary of the main characteristcs of LTE technology:

Description	LTE (Long Term Evolution) is a new standard of the 3GPP regulations. LTE is, sometimes, defined as an evolution of the 3GPP standard UMTS (3G) or as a new concept of the architecture (4G). In fact LTE will be the key to the uptake of mobile Internet. Services such as data transmission over 300 meters and high-definition video, will be used widely in the mature phase of the technology, thanks to OFDMA

	(Orthogonal Frequency-Division Multiple Access) modulation schema used in LTD.
Maturity	Not mature. Tests of the technology where done during 2009 and first real deployments started in the last quarter of 2010
Availability	Available only in some countries (Norway, Sweden, Finland, Austria, Poland and Germany in Europe)
Price	N/A
Bandwidth	100 Mbps on the downlink, and up to 50 Mbps.
Frequency	The LTE standard can be used with many different frequency bands: 900, 1800, 2600 MHz in Europe
Application to vehicular environment	Yes, by using the capabilities provided by the network operator
Infrastructure requirements	Integrated in mobile network operators
Legal issues	Some regulation uncertainties, mainly due to spectrum availability, still to be solved
Standardization	3GPP
Use in TeleFOT test site	No

Table 7 – LTE

LTE is the next step in the user experience, enhancing more demanding applications such as interactive TV, mobile video blogging, advanced gaming, and professional services. Data rates are significantly higher. LTE supports a full IP-based network and harmonization with other radio access technologies. LTE reduces the cost per Gigabyte of data delivered, which is essential to address the mass market..

The LTE or Long Term Evolution arises from the need to meet the growing demand of users and networks, and. This technology, based on the use of IP protocols, was tested during the second part of 2009 and the first commercial deployments have started to come to the market in 2010. [LTE1]

The novelty of LTE is the use of radio interface based on OFDMA for the downlink (DL) and SC-FDMA for uplink (UL). The modulation chosen by the 3GPP standard makes the different antenna technologies (MIMO: Multi Input, Multi Output) have greater ease of implementation, this provides a much better (sometimes even quadrupling) data transmission efficiency.

Comparing LTE with the previous cellular systems, it has introduced several new technologies that enable LTE to operate with a higher spectrum use efficiency and provides a much higher data rates.

- OFDM (Orthogonal Frequency Division Multiplex): OFDM enables high data bandwidths to be transmitted both with a high efficiency and providing a high degree of resistence to reflections and interference.
- MIMO (Multiple Input Multiple Output):The use of MIMO technology allows the system to use multiple signals arising from the many reflections as an advantage to increase the throughput. Without MIMO these reflections act as interferences to the main signal.
- SAE (System Architecture Evolution): LTE brings a very high data rate and a low latency. This makes necessary to evolve the system architecture in order to achieve the improved performance. One of the measures taken to improve the architecture is the transfer of several functions from the core network to the periphery for its processing. This provides a much "plane" network architecture. This helps to reduce latency times and to route data more directly to its destination.

Thanks to the inclusion of these new technologies, some of the improvements that LTE brings to the cellular communications are, among others:

- High spectral efficiency
  - OFDM in Downlink, Robust against multipath interference and High affinity to advanced techniques such as Frequency domain channel-dependent scheduling and MIMO
  - DFTS-OFDM("Single-Carrier FDMA") in Uplink, Low PAPR, User orthogonality in frequency domain
  - Multi-antenna application
- Very low latency
  - Short setup time & Short transfer delay
  - Short HO latency and interruption time; Short TTI, RRC procedure, Simple RRC states
- Simple protocol architecture
  - Shared channel based
  - PS mode only with VoIP capability
- Simple Architecture
  - eNodeB as the only E-UTRAN node
  - Smaller number of RAN interfaces, eNodeB « MME/SAE-Gateway (S1), eNodeB « eNodeB (X2)
- Compatibility and inter-working with earlier 3GPP Releases and other systems, e.g. dema2000
- FDD and TDD within a single radio access technology
- Efficient Multicast/Broadcast
  - Single frequency network by OFDM

8.2. Detailed description

8.2.1. System Architecture

The evolved architecture comprises E-UTRAN (Evolved UTRAN) on the access side and EPC (Evolved Packet Core) on the core side. One of the major advantages of E-UTRAN is that allows to reduce the cost and complexity of the needed equipment, by removing the control node. Therefore, the functions of radio resource control, QoS control and mobility have been built into the new Node B, called Node B. envolved (ENB). [LTE1]

The Evolved Packet Core is the core network of the LTE architecture and comprises the following elements:

- Serving Gateway (S-GW): serves as the local mobility anchor for data carriers (service flows) and retains information about the carriers when the UE (user equipment) is in idle mode. The S-GW performs some administrative functions in the visited network such as collecting charging information (e.g. volume of sent and received user data).
- PDN Gateway (P-GW): responsible for allocation of IP address to the UE, QoS enforcement and flow-based charging according to PCRF rules. The P-GW filters downlink user IP packets into different bearers depending on QoS classification. The P-GW serves as a mobility anchor for non-3GPP technologies (e.g. CDMA2000, WiMAX).
- Mobility Management Entity (MME): this is the control node that processes signaling between the UE and the core network. The MME manages establishment, maintenance and release of bearers. It also manages the establishment of the connection and security between the UE and the network.
- Policy Control Enforcement Function (PCRF): responsible for policy control decision making. The PCRF provides QoS authorization.
- Home Subscriber Server (HSS): The HSS contains the subscriber profile such as QoS profile, access restriction and roaming capabilities. Furthermore, the HSS holds information such as to which MME the UE is attached. The HSS may include an Authentication Center (AuC) which generates authentication vectors and security keys.

For LTE Radio Access Network Evolved there is a single element, eNB (evolved Node B) acting as interface with the user terminal. Some of its main functions is to manage radio resources, user and control information encryption / decryption, and compression / decompression of headers downlink packet / ascending in the user plane.

The eNBs are interconnected with each other by means of the X2 interface. The eNBs are connected by the S1 interface to the EPC (Evolved Packet Core). On the other hand, the eNB connects to the MME (Mobility Management Entity) by means of the S1- MME interface and to the Serving Gateway (S-GW) by means of the S1-U interface. The S1 interface supports a many-to-many relation between MMEs / Serving Gateways and eNBs. [LTE2]

The following figure shows a typical LTE architecture described in the previous paragraphs.

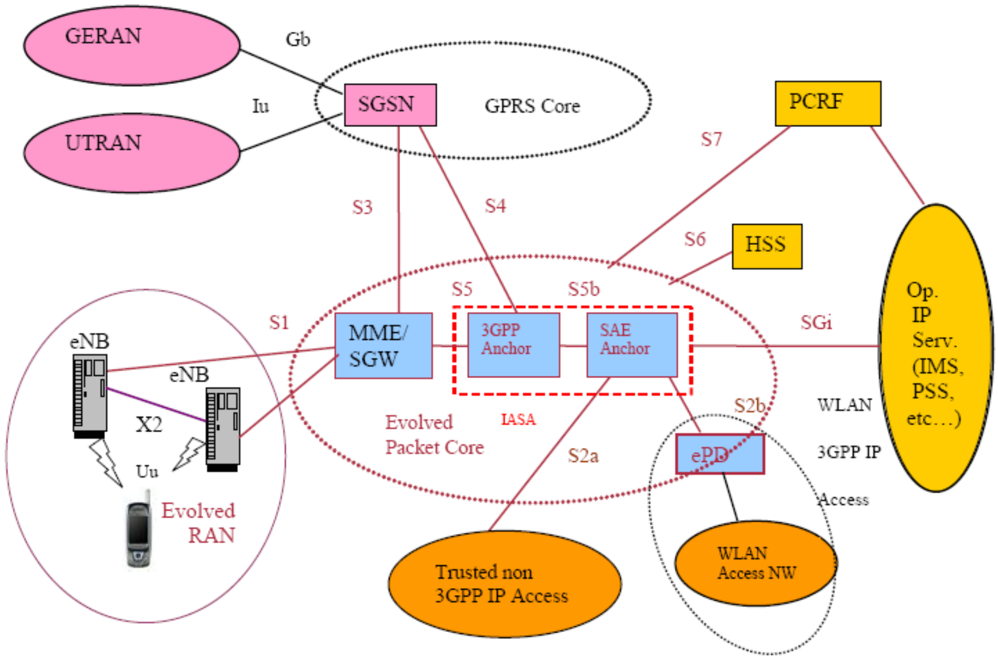

Figure 5 - LTE Architecture

8.2.2. LTE Physical layer

LTE physical layer multiple access scheme is based on Orthogonal Frequency Division Multiplexing (OFDM) with a cyclic prefix (CP) in the downlink, and on Single-Carrier Frequency Division Multiple Access (SC-FDMA) with a cyclic prefix in the uplink. LTE supports two duplex modes: Frequency Division Duplex (FDD), supporting full duplex and half duplex operation, and Time Division Duplex (TDD). For FDD, radio frame structure type 1 is used and has duration of 10ms and consists of 20 slots with a slot duration of 0.5ms. Two adjacent slots form one sub-frame of length 1ms.

For TDD, radio frame structure type 2 is used and consists of two half-frames with a duration of 5ms each and containing each 8 slots of length 0.5ms and three special fields (DwPTS, GP and UpPTS).

The capabilities of the eNodeB are quite different from the previous systems. Therefore the LTE Physical Down Link and Up Link are quite different. The following sections describe DL and UL physical layer.

Despite of the differences, the LTE DL and UL share the generic frame structure. LTE specifications define both FDD and TDD modes of operation. The following picture defines the frame structure applying both to the DL and UL for FDD operation.

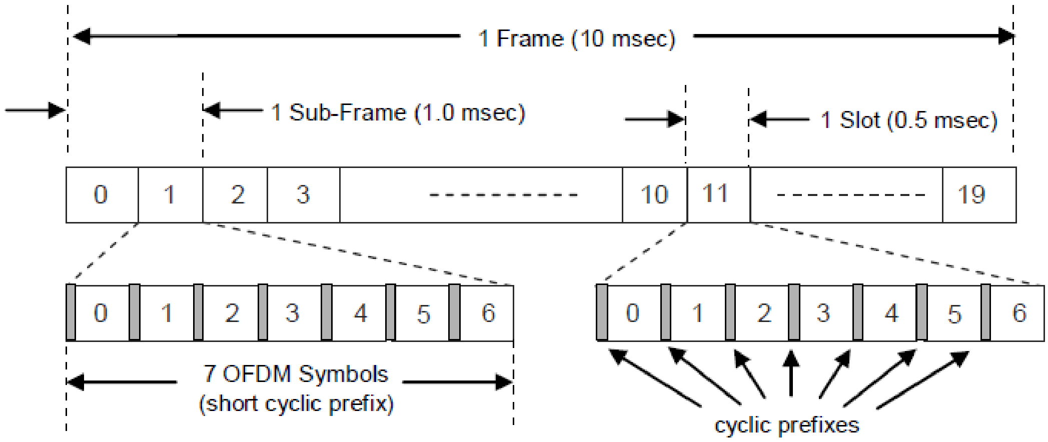

Figure 6 - LTE Generic Frame structure

8.2.2.1 Down link

The LTE PHY specification is designed to accommodate bandwidths from 1.25 MHz to 20 MHz. OFDM was selected as the basic modulation scheme because of its robustness in the presence of severe multipath fading. Downlink multiplexing is accomplished via OFDMA.

The DL supports physical channels, which convey information from higher layers in the LTE stack, and physical signals which are for the exclusive use of the PHY layer. Physical channels map to transport channels, which are service access points (SAPs) for the L2/L3 layers. Depending on the assigned task, physical channels and signals use different modulation and coding parameters. [LTE3]

Downlink Multiplexing

OFDMA is the basic multiplexing scheme employed in the LTE downlink. OFDMA is a new-to-cellular technology. In OFDMA, groups of 12 adjacent subcarriers are grouped together on a slot-by-slot basis to form physical resource blocks (PRBs). A PRB is the smallest unit of bandwidth assigned by the base station scheduler.

Referring to the following figure, a two dimensional (time and frequency) resource grid can be constructed to represent the transmitted downlink signal. Each block in the grid represents one OFDM symbol on a given subcarrier and is referred to as a resource element. Note that in MIMO applications, there is one resource grid for each transmitting antenna.

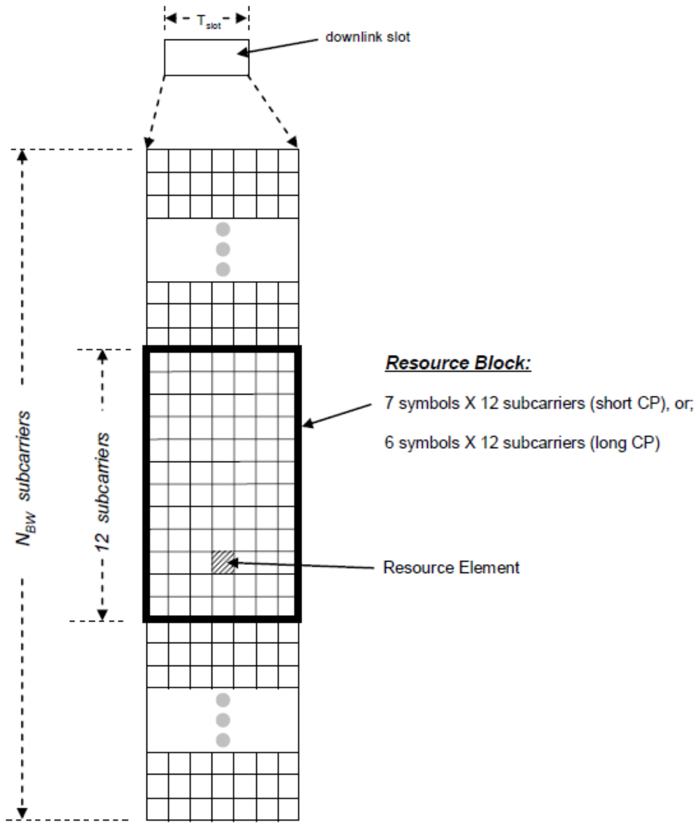

Figure 7 – OFDMA downlink resource grid

Physical Channels

Three different types of physical channels are defined for the LTE downlink. One common characteristic of physical channels is that they all convey information from higher layers in the LTE stack. This is in contrast to physical signals, which convey information that is used exclusively within the PHY layer.

LTE DL physical channels are:

- Physical Downlink Shared Channel (PDSCH)

  This channel is used for data and multimedia transport. Therefore it is designed for very high data rates. Modulation options of this channel include QPSK, 16QAM and 64QAM. Spatial multiplexing is also used in the PDSCH. Spatial multiplexing is exclusive to the PDSCH.

- Physical Downlink Control Channel (PDCCH)

  The PDCCH conveys UE-specific control information. Robustness rather than maximum data rate is the main consideration for this channel. QPSK is the only available modulation format. The PDCCH is mapped onto resource elements in up to the first three OFDM symbols in the first slot of a subframe.

- Common Control Physical Channel (CCPCH)

  The CCPCH carries cell-wide control information. Robustness rather than maximum data rate is the main consideration for this channel. QPSK is therefore the only available modulation format. In addition, the CCPCH is transmitted as close to the centre frequency as possible. CCPCH is transmitted exclusively on the 72 active subcarriers cantered on the DC subcarrier. Control information is mapped to resource elements (k, l) where k refers to the OFDM symbol within the slot and l refers to the subcarrier. CCPCH symbols are mapped to resource elements in increasing order of index k first, then l.

Physical channels are mapped to specific transport channels. Transport channels are SAPs for higher layers. Each physical channel has defined algorithms for:

- Bit scrambling
- Modulation
- Layer mapping
- CDD precoding
- Resource element assignment

Transport channels are included in the LTE PHY and act as service access points (SAPs) for higher layers. Downlink

Transport channels are:

- Broadcast Channel (BCH)
  - Fixed format
  - Must be broadcast over entire coverage area of cell
- Downlink Shared Channel (DL-SCH)
  - Supports Hybrid ARQ (HARQ)
  - Supports dynamic link adaption by varying modulation, coding and transmit power
  - Suitable for transmission over entire cell coverage area
  - Suitable for use with beamforming
  - Support for dynamic and semi-static resource allocation
  - Support for discontinuous receive (DRX) for power save
- Paging Channel (PCH)
  - Support for UE DRX
  - Requirement for broadcast over entire cell coverage area
  - Mapped to dynamically allocated physical resources
- Multicast Channel (MCH)
  - Requirement for broadcast over entire cell coverage area

- o Support for MB-SFN
- o Support for semi-static resource allocation

Layer mapping and pre-coding are related to MIMO applications. Basically, a layer corresponds to a spatial multiplexing channel. MIMO systems are defined in terms of Ntransmitters x Nreceivers. For LTE, defined configurations are 1x 1, 2 x 2, 3 x 2 and 4 x 2. Note that while there are as many as four transmitting antennas, there are only a maximum of two receivers and thus a maximum of only two spatial multiplexing data streams.

For a 1 x 1 or a 2 x 2 system, there is a simple 1:1 relationship between layers and transmitting antenna ports. However, for a 3 x 2 and 4 x 2 system, there are still only two spatial multiplexing channels. Therefore, there is redundancy on one or both data streams. Layer mapping specifies exactly how the extra transmitter antennas are employed.

Precoding is also used in conjunction with spatial multiplexing. Recall that MIMO exploits multipath to resolve independent spatial data streams. In other words, MIMO systems require a certain degree of multipath for reliable operation. In a noise-limited environment with low multipath distortion, MIMO systems can actually become impaired.

Physical Signals

Physical signals use assigned resource elements. However, unlike physical channels, physical signals do not convey information to/from higher layers.

There are two types of physical signals:

- Reference signals used to determine the channel impulse response (CIR)

  Reference signals are generated as the product of an orthogonal sequence and a pseudo-random numerical (PRN) sequence. Overall, there are 510 unique reference signals possible. A specified reference signal is assigned to each cell within a network and acts as a cell-specific identifier.

  Reference signals are transmitted on equally spaced subcarriers within the first and third from- last OFDM symbol of each slot. UE must get an accurate CIR from each transmitting antenna. Reference signals are sent on every sixth subcarrier. CIR estimates for subcarriers that do not bear reference signals are computed via interpolation.

- Synchronization signals which convey network timing information

  Synchronization signals use the same type of pseudo-random orthogonal sequences as reference signals. These are classified as primary and secondary synchronization signals, depending how they are used by UE during the cell search procedure. Both primary and secondary synchronization signals are transmitted on the 72 subcarriers centered around the DC subcarrier during the 0th and 10th slots of a frame (recall there are 20 slots within each frame).

8.2.2.2 Uplink

The LTE PHY uses Single Carrier - Frequency Division Multiple Access (SC-FDMA) as the basic transmission scheme for the uplink. SC-FDMA is essentially a multi-carrier scheme that re-uses many of the functional blocks included in the UE OFDM receiver signal chain. The principle advantage of SC-FDMA over conventional OFDM is a lower PAPR (by approximately 2 dB) than would otherwise be possible using OFDM.

Modulation Parameters

In FDD applications, the uplink uses the same generic frame structure as the downlink. It also uses the same subcarrier spacing of 15 kHz and PRB width (12 subcarriers). Downlink modulation parameters (including normal and extended CP length) are identical to the uplink parameters. Subcarrier modulation is, however, much different.

In the uplink, data is mapped onto a signal constellation that can be QPSK, 16QAM, or 64QAM depending on channel quality. However, rather than using the QPSK/QAM symbols to directly modulate subcarriers (as is the case in OFDM), uplink symbols are sequentially fed into a serial/parallel converter and then into an FFT block. The result at the output of the FFT block is a discrete frequency domain representation of the QPSK/QAM symbol sequence.

The discrete Fourier terms at the output of the FFT block are then mapped to subcarriers before being converted back into the time domain (IFFT). The final step prior to transmission is appending a CP. It is interesting to note that while the SC-FDMA signal has a lower PAPR in the time domain, individual subcarrier amplitudes can actually vary more in the frequency domain than a comparable OFDM signal.

Multiplexing

Uplink PRBs are assigned to UE by the base station scheduler via the downlink CCPCH. Uplink PRBs consist of a group of 12 contiguous subcarriers for a duration of one slot time.

Uplink Physical Channels

Uplink physical channels are used to transmit information originating in layers above the PHY. Defined UL physical channels are:

- Physical Uplink Shared Channel (PUSCH)

  Resources for the PUSCH are allocated on a sub-frame basis by the UL scheduler. Subcarriers are allocated in multiples of 12 (PRBs) and may be hopped from sub-frame to sub-frame. The PUSCH may employ QPSK, 16QAM or 64QAM modulation.

- Physical Uplink Control Channel (PUCCH)

  As the name implies, the PUCCH carries uplink control information. It is never transmitted simultaneously with PUSCH data. PUCCH conveys control information including channel quality indication (CQI), ACK/NACK, HARQ and uplink scheduling requests. The PUCCH transmission is frequency hopped at the slot boundary for added reliability.

Uplink Physical Signals

Uplink physical signals are used within the PHY and do not convey information from higher layers. Two types of UL physical signals are defined: the reference signal and the random access preamble.

- Uplink Reference Signal

  There are two variants of the UL reference signal. The demodulation signal facilitates coherent demodulation. It is transmitted in the fourth SC-FDMA symbol of the slot and is the same size as the assigned resource. There is also a sounding reference signal used to facilitate frequency dependent scheduling. Both variants of the UL reference signal are based on Zadhoff-Chu sequences.

- Random Access Preamble

  The random access procedure involves the PHY and higher layers. At the PHY layer, the cell search procedure is initiated by transmission of the random access preamble by the UE. If successful, a random access response is received from the base station. The random access preamble format consists of a cyclic prefix, a preamble and a guard time during which there is no signal transmitted. Random access preambles are derived from Zadoff-Chu sequences. They are transmitted on blocks of 72 contiguous subcarriers allocated for random access by the base station. In FDD applications, there are 64 possible preamble sequences per cell.

Uplink Transport Channels

As in the DL, uplink transport channels act as service access points for higher layers. Characteristics of UL transport channels are described below.

- Uplink – Shared Channel (UL-SCH)
  - Support possible use of beam forming
  - Support dynamic link adaption (varying modulation, coding and/or Tx power)
  - Support for HARQ
  - • Support for dynamic and semi-static resource allocation
- Random Access Channel (RACH)
  - • Supports transmission of limited control information
  - • Possible risk of collision

8.2.3. LTE deployment

LTE standardization, which covers FDD and TDD modes, is complete, and 3GPP Release 8 is the basis for initial LTE deployments. Infrastructure solutions offer an easy upgrade path to LTE. With the HSPA mobile broadband eco-system in place, LTE is the natural migration choice for GSM/HSPA network operators, and there is also a roadmap for CDMA operators to evolve to LTE as the clear mobile broadband system of choice.

8.2.3.1 LTE spectrum allocation

LTE can be deployed in existing 2G or 3G bands, and in new spectrum such as 2.6 GHz now being allocated in many regions, and the 700 MHz band. Initial deployments in Japan use 800 MHz, 1.5 GHz and 1.7 GHz (operator-dependant). There is strong demand in Europe and elsewhere to access new spectrum from the Digital Dividend (800 MHz), which enables LTE to be deployed more efficiently over large geographical areas, and improve in-building coverage. There is also high interest in using re-farmed spectrum LTE, e.g. 900, 1800 MHz bands as regulators adopt a technology-neutral approach.

Most LTE commitments and deployments use the paired spectrum (FDD) mode. The LTE TDD mode for unpaired spectrum is complementary and valuable in several markets. LTE TDD also provides a future-proof evolutionary path for TD-SCDMA, another 3GPP standard, which is widely deployed in China.

LTE TDD is the perfect choice for providing high speed mobile broadband access in unpaired spectrum. It is an integral part of the 3GPP standards implementing a maximum of commonalities with LTE FDD and offering comparable performance characteristics with similarly high spectral efficiency.

Within the globally assigned IMT bands for mobile (broadband) communication, significant spectrum resources are suitable for LTE TDD. The largest contiguous bands are at 2.3 GHz (100 MHz) and within the 2.6 GHz band (e.g. 50 MHz according to the CEPT band plan). Due to the recognized demand for radio technologies for unpaired bands and based on the commonalities as explained above, LTE TDD can exploit global economies of scale similar to LTE FDD, with a short time to market. It is likely that LTE TDD will become a globally accepted technology, and provide an excellent evolution path for TD-SCDMA and WiMAX™ networks.

The growing number of LTE TDD operator commitments, system maturity, and the expanding eco-system are detailed later in this report.

8.2.3.2 LTE implementation

According GSA (the Global mobile Suppliers Association), LTE made excellent progress in the recent years, and especially during 2010 and deployments are foreseen to be very fast in 2011. The rapid increase in mobile data traffic experienced over the past 3 years and which is currently supported primarily by HSPA and HSPA+ systems is driving a lot of interest in deploying LTE as quickly as possible, attracting global industry support. In fact, LTE operator commitments are developing faster than they did for HSPA and confirm LTE as the fastest developing mobile system technology ever.

At this moment (April 2011) 17 commercial LTE networks are already available in the market, and, at least 73 LTE networks are foreseen to be in commercial service by the end of 2012. The following figure shows the list of the commercial LTE deployments available and the date of launching:

Country	Operator	Launched
Norway	TeliaSonera	15.12.09
Sweden	TeliaSonera	15.12.09
Uzbekistan	MTS	28.07.10
Uzbekistan	UCell	09.08.10
Poland	Mobyland & CenterNet	07.09.10
USA	MetroPCS	21.09.10
Austria	A1 Telekom Austria	05.11.10
Sweden	TeleNor Sweden	15.11.10
Sweden	Tele2 Sweden	15.11.10
Hong Kong	CSL Limited	25.11.10
Finland	TeliaSonera	30.11.10
Germany	Vodafone	01.12.10
USA	Verizon Wireless	05.12.10
Finland	Elisa	08.12.10
Denmark	TeliaSonera	09.12.10
Estonia	EMT	17.12.10
Japan	NTT DoCoMo	24.12.10

Figure 8 – Commercial LTE network launches – March 24, 2011

A part from the above mentioned LTE operators and networks, many other operators have strong commitments in LTE deployments all around the world, so that 196 operators in 75 countries are investing in LTE at the moment. [LTE1]

The following figure shows the situation of LTE commitments around the in the countries around the world depending on the deployment level: commercial service, deployments on-going, pre-commitment or trial phases

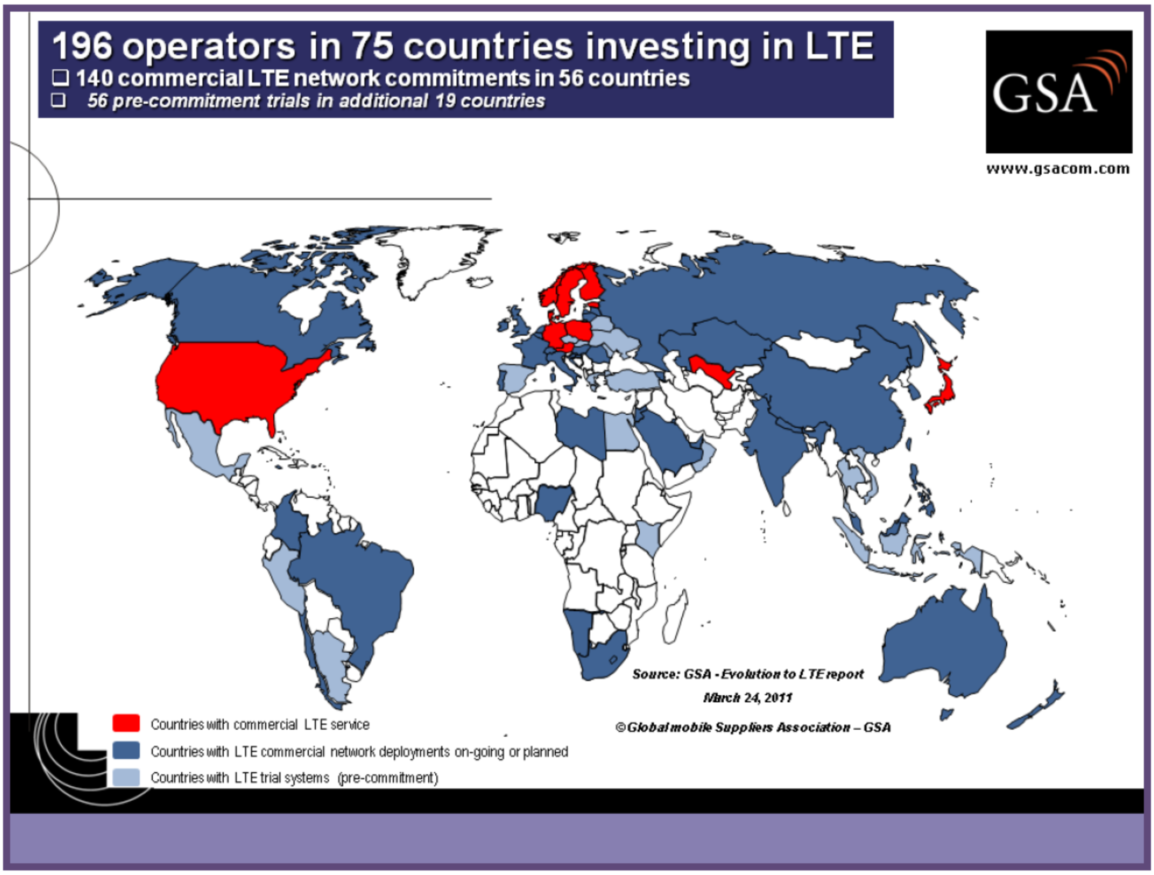

Figure 9 –LTE commitment around the world [LTE1]

NOMAD TECHNOLOGIES

The so called nomadic networks are networks that support direct connection of users to the system. That is, instead of connecting two remote locations, such as a control centre and a base station, these nomadic networks connect individual users (computers, smartphones, pdas, etc.) to a network. These networks are designed to provide a low level of mobility to the users.

IEEE 802.11b / g (WiFi) and IEEE 802.16 are common standards for such networks, and can be called nomad technologies.

To be completely portable, the user device must be small, low power and the antennas used should be small and omnidirectional. The effects of loss of signal through walls or other objects reduce the area covered by these type of networks.

Despite its short coverage range, the nomadic networks are very common in the market, offer and have become the first step taken to provide the users with high speed access data in many public and private areas.

9. IEEE 802.11 (WiFi)

9.1. *Summary*

In the following table, a summary with the most important characteristics is given:

Description	Wireless LAN (Local Area Network) based in Ethernet protocol
Maturity	Highly robust and proven technology.
Availability	Available everywhere.
Price	Free to use.
Bandwidth	11 Mbps
Frequency	2.400 GHz
Coverage	10-30 meters from the access point indoor. 100 meters outdoors.
Infrastructure requirements	Omnidirectional antenna and wifi receiver like a card wifi receiver
Legal issues	Maximum transmission (Tx) power must be less than 100mW. [Tx max power < 100mW]
Standarization	802.11
Use in TeleFOT test site	No

Table 8 – WiFi

9.2. Detailed description

The IEEE 802.11 (ISO / IEC 8802-11) is an international standard that describes the characteristics of a wireless local area network (WLAN). WIFI (Wireless Fidelity contraction in Spanish Wireless Fidelity) is the name initially given to this certification by WECA (Wireless Ethernet Compatibility Alliance), the agency responsible for certifying that the equipment meets the 802.11 standard. Due to an abuse of language and for marketing reasons the name of the standard today is confused with the name of the certification. Thus a WiFi network is actually a network under the 802.11 standard. [WIFI1]

With WiFi it is possible to create wireless local area networks to broadband. This allows them to communicate WIFI laptops, desktops, personal assistant (PDA) and even peripheral with a broadband connection (11 Mbit / s) within a radius of several tens of meters inside (usually between 20 and 50 meters). Outdoor range may be several hundred meters under optimal conditions tens of kilometres.

Therefore the Internet access providers begin to offer wireless Internet access in areas with enough concentration of users (train stations, airports, hotels, trains, etc.).. These areas of Internet access are called "hot spots".

Apple iBooks in 1999 were the first computers for the general public that the technology proposed integrated WiFi (with the name of Airport), then followed by other brands. Since 2003, we also show models of PCs built around Intel Centrino technology, which allows integration. The other models have yet to be equipped with an extension card.

Now a days an IEEE 802.11 device is installed in many personal computers, video game consoles, Smartphones, printers, and other peripherals, and virtually all laptop or palm-sized computers.

In the following table, the different modifications on the 802.11 standards are shown:

802.11 network standards								
802.11 Protocol	Release	Freq. (GHz)	Bandwidth (MHz)	Data rate per stream (Mbit/s)	Allowable MIMO streams	Modulation	Approximate indoor / outdoor range	
							(m)	(m)
–	Jun 1997	2.4	20	1, 2	1	DSSS, FHSS	20	100
a	Sep 1999	5	20	6, 9, 12, 18, 24, 36, 48, 54	1	OFDM	35	120
		3.7[y]					--	5,000
b	Sep 1999	2.4	20	5.5, 11	1	DSSS	38	140
g	Jun 2003	2.4	20	6, 9, 12, 18, 24, 36, 48, 54	1	OFDM, DSSS	38	140
n	Oct 2009	2.4/5	20	7.2, 14.4, 21.7, 28.9, 43.3, 57.8, 65, 72.2[z]	4	OFDM	70	250
			40	15, 30, 45, 60, 90, 120, 135, 150[z]			70	250

- [y] IEEE 802.11y-2008 extended operation of 802.11a to the licensed 3.7 GHz band. Increased power limits allow a range up to 5000m.
- [z] Assumes Short Guard interval (SGI) enabled, otherwise reduce each data rate by 10%.

Table 9 – 802.11 Network Standards

The 802.11 standard defines the OSI model layers for a wireless link using electromagnetic waves, as follows: Physical Layer (PHY) which offers 3 types of information codes; the data link layer, composed of two sublayers: -the logical link control (Logical Link Control, or LLC); -the media access control (Media Access Control or MAC).

The physical layer defines the electrical specifications and the type of signals used for data transmission, while the data link layer defines the interface between the device and the physical layer, especially a similar access method to the one used in the Ethernet standard and the communication basis between the different base stations. The proposed 802.11 standard actually defines three physical layers to alternative modes of transmission:

In a WiFi network you can use any protocol, just as in an Ethernet network.

9.2.1. Operation modes

Two different operation modes can be identified in a WiFi network:

Infrastructure mode is a mode that allows you to connect computers equipped with a WiFi network card using one or more access points (AP) that act as connectors (e.g., Hub / Switch wired network). This mode is used especially by firms. The implementation of such a network requires setting terminal (AP) at regular intervals in the area to be covered by the network. The terminals and computers must be configured with the same SSID (network name) to communicate. The advantage of this mode is that it ensures a step required by the AP, which verifies who enters the network. Instead, the network cannot grow unless you put more terminals.

The **"Ad-Hoc" mode** allows direct communication between computers equipped with a WiFi network card, without using other additional equipment such as an Access Point (AP). This mode is ideal for connecting computers between them quickly without supplementary material (e.g., exchange of files between laptop on a train, sharing Internet access at home, in the street, coffee, etc.).. The implementation of a network of this type is limited to configure the devices in Ad-Hoc mode (as opposed to Infrastructure mode), selecting the channel (frequency) and an SSID (network name) common to all. The advantage of this mode is that it eliminates expensive supplemental materials and is easier to implement. With the addition of a dynamic routing program (e.g., OLSR, AODV, etc..) Network grows automatically with the connection of new equipment. put more terminals.

9.2.2. Wi-Fi certification

Wi-Fi technology builds on IEEE 802.11 standards. The IEEE develops and publishes these standards, but does not test equipment for compliance with them. The non-profit Wi-Fi Alliance formed in 1999 to fill this void — to establish and enforce standards for interoperability and backward compatibility, and to promote wireless local-area-network technology. As of 2009 the Wi-Fi Alliance consisted of more than 300 companies from around the world. Manufacturers with membership in the Wi-Fi Alliance, whose products pass the certification process, gain the right to mark those products with the Wi-Fi logo.

Specifically, the certification process requires conformance to the IEEE 802.11 radio standards, the WPA and WPA2 security standards, and the EAP authentication standard. Certification may optionally include tests of IEEE 802.11 draft standards, interaction with cellular-phone technology in converged devices, and features relating to security set-up, multimedia, and power-saving.

9.2.3. Advantages and limitations

Wi-Fi allows the deployment of local area networks (LANs) without wires for client devices, typically reducing the costs of network deployment and expansion. Spaces where cables cannot be run, such as outdoor areas and historical buildings, can host wireless LANs.

As of 2010 manufacturers build wireless network adapters into most laptops. The price of chipsets for Wi-Fi is lower and lower, making it an economic option for constructing a network and is being included in even more devices. Wi-Fi has become widespread in corporate infrastructures.

Different competitive brands of access points and client network-interfaces can inter-operate at a basic level of service. Products designated as "Wi-Fi Certified" by the Wi-Fi Alliance are backwards compatible. "Wi-Fi" designates a globally operative set of standards: unlike mobile phones, any standard Wi-Fi device will work anywhere in the world.

Wi-Fi is nowadays broadly used worldwide both in industry or company and home environments. The current version of Wi-Fi Protected Access encryption (WPA2) as of 2010 is considered secure, provided users employ a strong passphrase. New protocols for quality-of-service (WMM) make Wi-Fi more suitable for latency-sensitive applications (such as voice and video); and power saving mechanisms (WMM Power Save) improve battery operation.

Spectrum assignments and operational limitations do not operate consistently worldwide. Most of Europe allows for an additional 2 channels beyond those permitted in the U.S. for the 2.4 GHz band. (1–13 vs. 1–11); Japan has one more on top of that (1–14). Europe, as of 2007, was essentially homogeneous in this respect. A very confusing aspect is the fact that a Wi-Fi signal actually occupies five channels in the 2.4 GHz band resulting in only three non-overlapped channels in the U.S.: 1, 6, 11, and three or four in Europe: 1, 5, 9, 13 can be used if all the equipment on a specific area can be guaranteed not to use 802.11b at all, even as fallback or beacon. Equivalent isotropically radiated power (EIRP) in the EU is limited to 20 dBm (100 mW).

9.2.4. Range

Wi-Fi networks have limited range. A typical wireless router using 802.11b or 802.11g with a stock antenna might have a range of 10-30 meters from the access point indoor and 100 meters outdoors. The new IEEE 802.11n however, can exceed that range by more than two times. Range also varies with frequency band. Wi-Fi in the 2.4 GHz frequency block has slightly better range than Wi-Fi in the 5 GHz frequency block. Outdoor ranges - through use of directional antennas - can be improved with antennas located several kilometres or more from their base. In general, the maximum amount of power that a Wi-Fi device can transmit is limited by local regulations.

Wi-Fi performance decreases roughly quadratically as distance increases at constant radiation levels.

Due to reach requirements for wireless LAN applications, Wi-Fi has as fairly high power-consumption compared to some other standards. Technologies such as Bluetooth (designed to support wireless PAN applications) provide a much shorter propagation range of less than 10 meters and so in general have a lower power-consumption. Other low-power technologies such as ZigBee have fairly long range, but much lower data rate. The high power-consumption of Wi-Fi makes battery life in mobile devices a concern.

Researchers have developed a number of "no new wires" technologies to provide alternatives to Wi-Fi for applications in which Wi-Fi's indoor range is not adequate and where installing new wires (such as CAT-5) is not possible or cost-effective. For example, the ITU-T G.hn standard for high speed Local area networks uses existing home wiring (coaxial cables, phone lines and power lines). Although G.hn does not provide some of the advantages of Wi-Fi (such as mobility or outdoor use), it's designed for applications (such as IPTV distribution) where indoor range is more important than mobility.

Due to the complex nature of radio propagation at typical Wi-Fi frequencies, particularly the effects of signal reflection off trees and buildings, algorithms can only be predict Wi-Fi signal strength generally for any given area in relation to a transmitter. This effect does not apply equally to long-range Wi-Fi, since longer links typically operate from towers that broadcast above the surrounding foliage.

9.2.5. Mobility

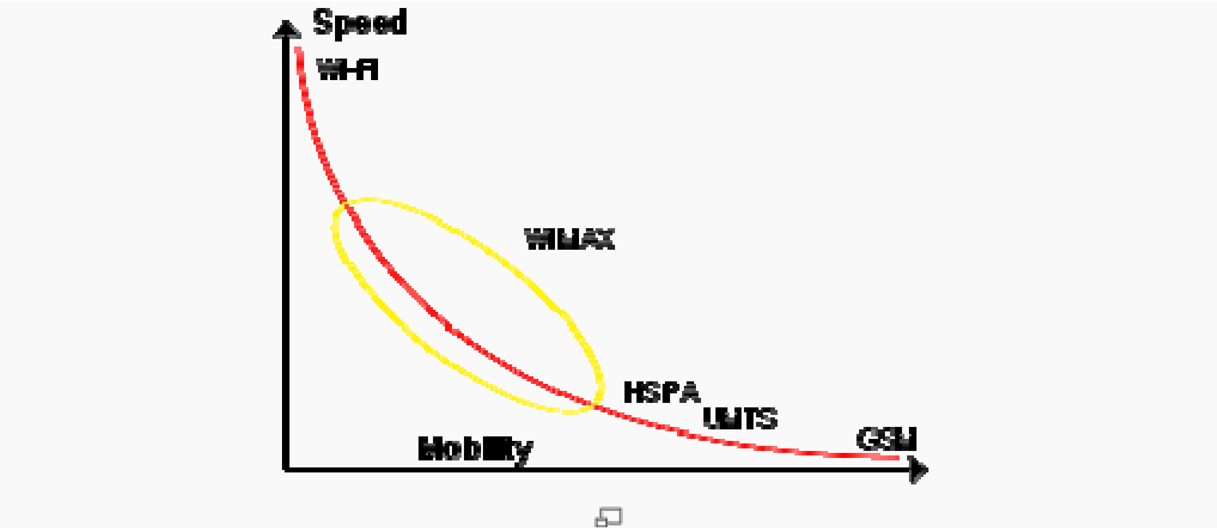

Figure 10 – Speed vs. Mobility of wireless systems: Wi-Fi, HSPA,UMTS, GSM

The very limited practical range of Wi-Fi essentially confines mobile use to such applications as inventory-taking machines in warehouses or in retail spaces, barcode-reading devices at check-out stands, or receiving/shipping stations. Mobile use of Wi-Fi over wider ranges is limited. Due to the low speed mobility allowed in WiFi environments, other wireless technologies are more suitable for mobility applications. As illustrated in the graphic above technologies such as HSPA or UMTS allow a higher speed for mobility applications.

10. 802.16e - Mobile WiMAX

10.1. Summary

The following table shows a summary of the most important characteristics of Mobile WiMAX:

Description	WiMAX (Worldwide Interoperability for Microwave Access) is a wireless digital communications system, also known as IEEE 802.16, that is intended for wireless "metropolitan area networks"
Maturity	Mature and proven technology.
Availability	Available all over Europe.
Price	Depending on the device and use.
Bandwidth	Up to 70 Mbps
Frequency	2-11Ghz, 10-66GHz
Coverage	Up to 10 Km with LOS, up to 4 Km with NLOS.
Application to vehicular environment	Yes
Infrastructure requirements	Omnidirectional antenna and WiMAX receiver
Standarization	IEEE 802.16
Use in TeleFOT test site	No

Table 10 – Mobile WiMAX

Mobile WiMAX is a broadband wireless access technology based on IEEE 802.16-2004 and IEEE 802.16e-2005 air-interface standards. The WiMax forum developed mobile WiMAX system profiles that define the mandatory and optional features of the IEEE standard that are necessary to build a mobile WiMAX compliant air interface which can be certified by the WiMAX Forum.

10.1.1. 802.16 standard characteristics

The wireless transmission uses the air interface, this leads to problems of attenuation and distortion by multiple factors such as vegetation, buildings, rain and vehicles that move and change unpredictably. The 802.16 standard recognizes this and includes mechanisms to make robust links to online sight (LOS, Line-Of-Sight), obstructed line of sight and without line of sight (NLOS Non Line-Of- Sight). [WIM1]

The medium access control (MAC) layer provides different types of QoS (quality of service) depending on different transmission needs. Voice and video require low latency but tolerate some error rate. In contrast with the data, which do not tolerate errors, but the latency is not critical. The standard accommodates voice, video and other transmissions of data using appropriate characteristics of the MAC layer, since it is more efficient than doing it in upper layers.

The standard supports adaptive modulation, which balances different data rates and link quality in an efficient way. The modulation method can be adjusted almost instantly as optimal data transfer. The adaptive modulation allows the efficient use of bandwidth.

802.16 supports two duplexing schemes, frequency and time (FDD and TDD, respectively). FDD (Frequency Division Duplex) is used extensively in cellular phones, this system requires two channels; a transmission channel and a reception channel with a distance between them to avoid interference. TDD (Time Division Duplex) provides a flexible scheme, where uplink and downlink transmission channel is the same, since they are not simultaneous but sequential. A TDD system can dynamically allocate uplink and downlink bandwidth depending on traffic requirements.

The 802.16 standard has evolved and there have been numerous versions. Some of them are summarized below:

- 802.16-2004: this version was developed to work in unlicensed bands and licensed bands between 2 to 11GHz with LOS and NLOS systems for Point-to-Multipoint and Point-to-Point communications. Was published in July 2004. Formerly known as IEEE 802.16d or Review "d". Replaces all previous versions of the standard (802.16, 802.16a, 802.16c). This standard is used for fixed applications.
- 802.16e: This standard supports mobility. Also known as 802.16 - 2005 the year of its publication in December 2005. This standard is used for mobile applications.

10.2. Detailed description

10.2.1. 802.16e (Mobile WiMAX) characteristics

Mobile WiMAX is the solution for mobile, nomadic, portable and fixed broadband through radio access technology and flexible network architecture. Mobile WiMAX air interface uses in SOFDMA (Scalable Orthogonal Frequency Division Multiple Access) channels that supports scalable bandwidth from 1.25 to 20 MHz The most relevant characteristios of Mobile WiMAX are:

- The use of OFDMA: The mobile WiMAX air interface uses Orthogonal Frequency Division Multiple Access (OFDMA) as the radio access method for improved multipath performance in non-line-of-sight (NLOS) environments.
- High data rates: The use of multiple-input multiple-output (MIMO) antenna techniques along with flexible subchannelization schemes, adaptive modulation

and coding enable the mobile WiMAX technology to support both peak downlink and uplink high data rates. Thanks to these techniques Mobile WiMAX supports up to 63 Mbps downstream and 28 Mbps upstream by Industry 10 MHz channels

- Quality of Service: The fundamental premise of the IEEE 802.16 medium access control (MAC) architecture is QoS. It defines service flows which can be mapped to fine granular IP sessions or coarse differentiated-services code points to enable end-to-end IP based QoS.
- Scalability: The mobile WiMAX technology uses scalable OFDMA (S-OFDMA) and has the capability to operate in scalable bandwidths from 1.25 to 20 MHz to comply with various spectrum allocations worldwide.
- Security: The mobile WiMAX incorporates the most advanced security features that are currently used in wireless access systems. These include Extensible Authentication Protocol (EAP) based authentication, Advanced Encryption Standard (AES) based authenticated encryption, and Cipher-based Message Authentication Code (CMAC) and Hashed Message Authentication Code (HMAC) based control message protection schemes.
- Mobility: The mobile WiMAX supports optimized handover schemes with latencies less than 50 ms to ensure real-time applications such as Voice over Internet Protocol (VoIP).

10.2.2. Physical layer description

10.2.2.1 OFDMA Structure

OFDM is a multiplexing technique that divides the bandwidth into multiple frequency channels. This system divides the data in several low data rate subcarriers and each one is modulated and transmitted over channels separate among them in an orthogonal way. OFDM provides high spectral efficiency and higher tolerance to interference, in fact it can support NLOS because it has more multipath tolerance and better coverage than other multiplexing techniques.

OFDMA is a multiple access scheme that allows multiplexing of several users in a subchannel. OFDMA is a multi-user version of OFDM.

IEEE 802.16e Wireless MAN OFDMA is based on the concept of scalable OFDMA (SOFDMA). Scalability is supported by adjusting the size of Fast Fourier Transform (FFT, Fast Fourier Transform), while the fit subcarrier frequencies of 10.94 kHz apart.

10.2.2.2 TDD structure

The initial WiMAX Mobile standard only includes Time Division Duplexing (TDD) although in the subsequent versions it of this standard Frequency Division Duplexing (FDD) was also introduced. Despite of that The TDD mode is preferred for the following reasons:

- It enables a dynamic allocation of DL and UL resources to support efficiently asymmetric DL/UL traffic (adaptation of DL:UL ratio to DL/UL traffic).

- It ensures channel reciprocity for better support of link adaptation, advanced antenna techniques such as transmit beam-forming or MIMO.
- Unlike FDD, which requires a pair of channels, TDD only requires a single channel for both downlink and uplink providing greater flexibility for adaptation to varied global spectrum allocations.
- Transceiver designs for TDD implementations are less complex and therefore less expensive.

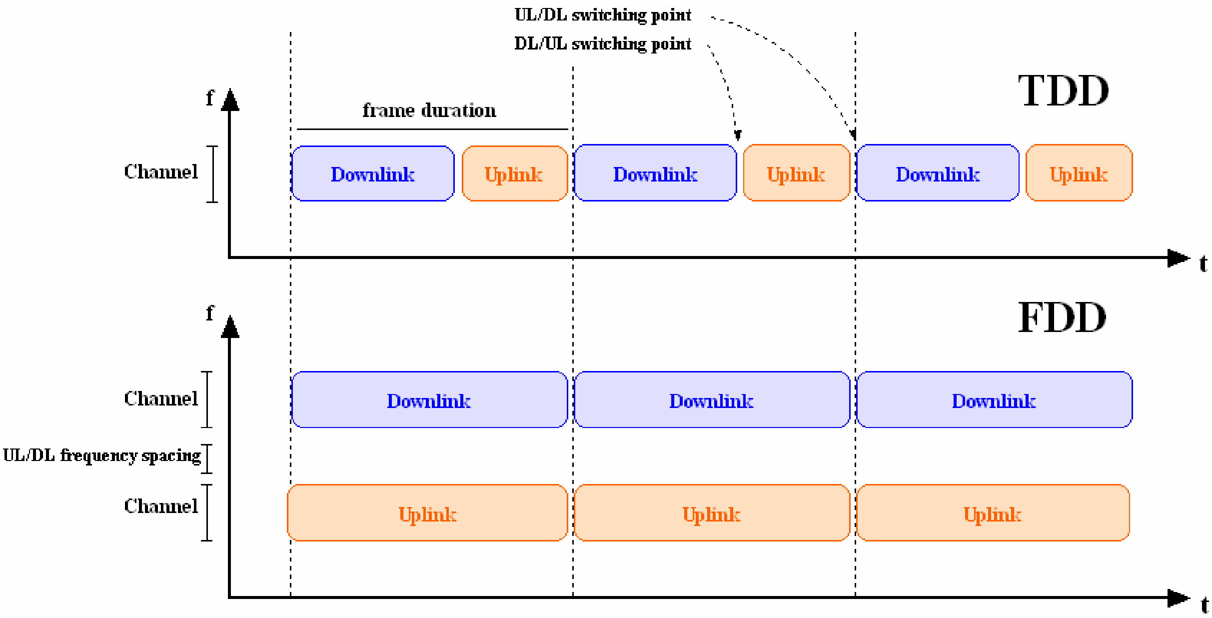

Figure 11 – TDD versus FDD

10.2.2.3 Other features

Adaptive modulation and coding (AMC), automatic repeat request Hybrid (HARQ) and fast feedback channel (CQICH) were introduced to Mobile WiMAX to enhance coverage and capacity.

Mobile WiMAX supports QPSK, 16QAM and 64QAM, which are used in downlink and uplink, 64QAM being optional for the uplink. The best transfer rates achieved using 64QAM modulation.

10.2.1. MAC layer description

10.2.1.1 QoS

Mobile WiMAX implements QoS control all through the link supports the use of MPLS (Multiprotocol Label Switching) system that allows the prioritization of the different packets sent. This standard supports a great variety of services and applications with different QoS requirements, such as VoIP, audio or video streaming, FTP, data transmission, etc.

10.2.1.2 Mobility

The battery life of mobile devices and the handoff are two critical aspects in mobile applications. The Mobile WiMAX standard also deals with these issues as they are crucial for facilitating the mobility of the user equipment:

- Efficient Energy Administration: Mobile WiMAX supports two types of operations for the energy administration: Sleep Mode and Idle Mode. These modes provide help the device to see battery power and contribute to the autonomy of the batteries.
- Handover: The standard supports three types of handoff, Hard Handoff (HHO), Fast Base Station Switching (FBSS) and Macro Diversity Handover (MDHO). Of these, the HHO is mandatory while FBSS and MDHO are two optional modes.

10.2.1.3 Security

Mobile WiMAX incorporates the best security techniques available. The security features included in the standard are device and user authentication, management protocols, traffic encryption, control and management of message protection and protocol optimization for fast handoffs.

10.2.2. Mobile WiMAX architecture

The WiMAX architecture is based on an ALL-IP platform. That means that the Packet Switching is present in all the architecture (end-to-end).

The WiMAX network provides the flexibility to accommodate a wide range of different implementations, such as:

- Radio coverage and enough capacity both for densely or non-densely populated areas.
- For urban, suburban and rural areas.
- It supports different types of topology.
- Coexistence of fixed, nomadic and mobile services in the same network.
- Licensed and unlicensed bands, although it is unlikely to make a mobile deployment in unlicensed bands, due to low control over the band and the interference affecting the transmission.

The WiMAX Forum's Network Working Group (NWG) has developed a network reference model to serve as an architecture framework for WiMAX deployments and to ensure interoperability among various WiMAX equipment and operators. An example of this network reference model is shown in the following figure:

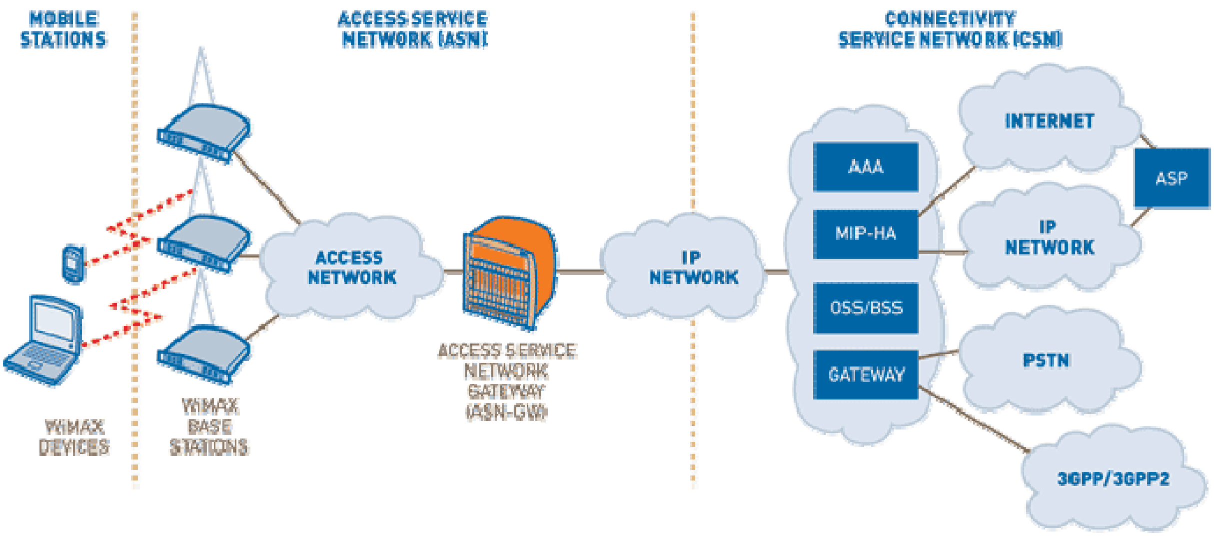

Figure 12 – WiMAX IP-based architecture

The network reference model specifies a unified network architecture for supporting fixed, nomadic, and mobile deployments and is based on an IP service model. The simplified illustration above of an IP-based WiMAX network architecture. The overall network may be logically divided into three parts:

- Mobile Stations (MS) or user equipment used by the end user to access the network.
- The access service network (ASN), which comprises one or more base stations and one or more ASN gateways that form the radio access network at the edge.
- Connectivity service network (CSN), which provides IP connectivity and all the IP core network functions.

The network reference model developed by the WiMAX Forum NWG defines a number of functional entities and interfaces between those entities. Fig below shows some of the more important functional entities.

- Base station (BS): The BS is responsible for providing the air interface to the MS. Additional functions that may be part of the BS are micro mobility management functions, such as handoff triggering and tunnel establishment, radio resource management, QoS policy enforcement, traffic classification, DHCP (Dynamic Host Control Protocol) proxy, key management, session management, and multicast group management.

- Access service network gateway (ASN-GW): The ASN gateway typically acts as a layer 2 traffic aggregation point within an ASN. Additional functions that may be part of the ASN gateway include intra-ASN location management and paging, radio resource management and admission control, caching of subscriber profiles and encryption keys, AAA client functionality, establishment and management of mobility tunnel with base stations, QoS and policy enforcement, foreign agent functionality for mobile IP, and routing to the selected CSN.
- Connectivity service network (CSN): The CSN provides connectivity to the Internet, ASP, other public networks, and corporate networks. The CSN is owned by the NSP and includes AAA servers that support authentication for the devices, users, and specific services. The CSN also provides per user policy management of QoS and security. The CSN is also responsible for IP address management, support for roaming between different NSPs, location management between ASNs, and mobility and roaming between ASNs.

AD-HOC NETWORK TECHNOLOGIES

Some of the wireless networks that exist nowadays are composed spontaneously by a set of mobile terminals without dependence on any infrastructure. These type of networks are leading to a new generation of networks, making it possible to develop a set of new services and applications in different fields and scenarios where until a few years was not possible to provide such connectivity. As an example VANET (Vehicular Ad-Hoc Networks) networks can be named. VANET networks are composed by different vehicles of a particular scenario that communicate to each other in order to exchange information to improve comfort and security of the drivers.

In general an ad-hoc network can be defined as a network composed by fixed and mobile terminals or by only mobile terminals, which do not depend on a pre-existing infrastructure and is deployed in a wireless environment. One of the most interesting characteristics of these type of networks is the possibility of deploying an isolated and independent network between devices without needing base stations, fixed routers etc.

11. NGN (New Generation Network) and IMS (IP Multimedia Sybsystem)

11.1. Summary

The following table shows the main characteristics of NGN

Description	In order to grant compatibility and interoperability between technologies, the IP Multimedia Subsystem (IMS) specifications can be followed. IMS is the Next Generation Networking (NGN) architecture for implementing IP based services. IMS is based on the concept of All-IP, i.e., all access technologies use as network protocol IP, which allows global addressing and routing, and over the IP layer the IMS layer grants interoperable services. Furthermore, the future standard for long-range wireless systems (4G) will integrate Cellular and BWA technologies seamlessly and will support also IMS with seamless handovers, which will grant interoperability among technologies.
Maturity	Mature
Availability	Available in some European countries, such as in Ireland
Price	N/A
Bandwidth	N/A
Frequency	N/A

Coverage	N/A
Application to vehicular environment	Yes, by using the capabilities provided by the network operator
Infrastructure requirements	Integrated in mobile network operators
Legal issues	Solved
Standardization	ETSI, ITU-T, 3GPP
Use in TeleFOT test site	Not directly
How to incorporate this technology to TeleFOT test site?	Using the capabilities provided by the network operator

Table 11 – NGN and IMS

11.2. Detailed description

The Next Generation Network (NGN) model, standardized thanks to collaborations among different organisms (ETSI, ITU-T, 3GPP), has as aim to provide advanced multimedia services (and integrating traditional services) regardless of the access technology used.

By using an architecture with independent layers (access, transport, control, capabilities and applications) and defining open interfaces, the NGN model obtains a real convergence of the different networks (3G/2G, WiFi, WiMAx, PSTN, broadband, etc.) allowing a fast spread of services in the environment of mobility. [ADHOC1]

NGN architecture divides into five different layers, each of them with different functionalities. Its layers are: Access Layer, Transport Layer, Control Layer, Capability Layer and Application Layer.

Access Layer, with different access communication technologies: WiFi, WiMax, 3G/2G, etc. wired technologies also belong to this layer (e.g. ADSL).

The **Transport Layer** allows media and signalling conversion among networks. It is usually based on IP/MPLS (Multiprotocol Label Switch) packet commutation technology that allows ensuring quality of service through packet categorization. This way, a common backbone is available for data, voice and video IP packets.

The **Control Layer** is based on IMS (IP Multimedia Subsystem) standardized by 3GPP and will allow to manage sessions among users and user-application sessions regardless of the access mechanism that is being used, giving support to mobility scenarios, charging, control of services, etc. In order to provide these functionalities, IMS is based on standard protocols (SIP, Diameter, Radius, SDP). Together with IMS, we will also find in this layer the SS7 signalling system, traditionally used by the operator to provide intelligent network services.

The **Capability Layer** provides horizontal functionalities that are interesting for the final user applications (e.g. ID Management, Policy Management, Messaging, Location, Broadcast/Multicast, User Profile, Terminal Profile, etc.) in a secure and safety way. Capabilities export open and standardized interfaces using several protocols (http, SOAP, SIP, etc.). This way, we get a fast and simple development and deployment of complex services. [ADHOC3]

The **Application Layer** represents the final user services that can or cannot make use of the capacities during their development.

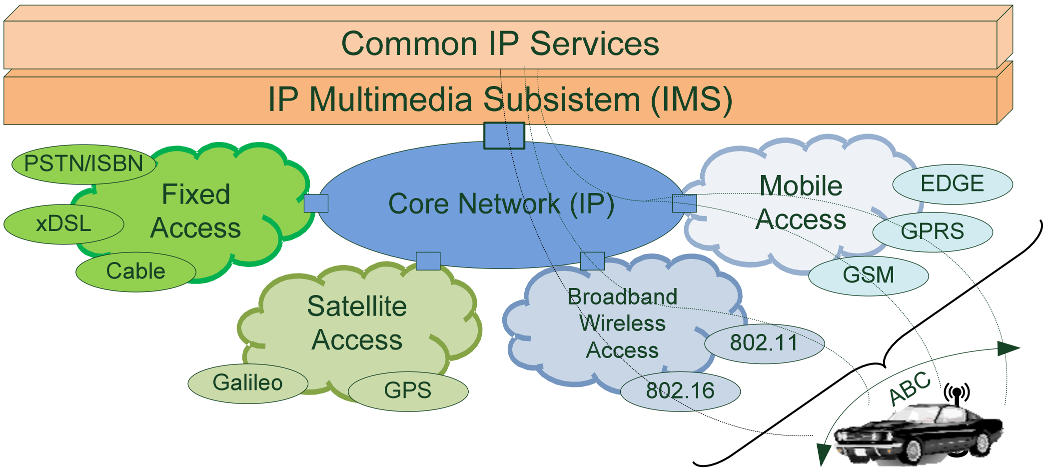

Figure 13 - NGN Architecture

11.2.1. Control Layer for NGN Networks

Following the NGN referent model, the control layer can be built over the IMS subsystem (IP Multimedia SubSystem). IMS, which is mainly standardized by 3GPP, allows providing services to users on an all-IP network, regardless of the access technology used, thanks to the NGN layer model that separates access and transport layers from the control layer.

From the user's point of view, he is provided with the possibility of establishing user-user and user-service communications in which he can exchange all kind of personalized and controlled multimedia content (voice, images, video, messaging). It is also possible to combine the different services and content (e.g. to exchange a video in a voice communication) and all of that ensuring the mobility of the user among different technologies/access systems because IMS has inherited the mobility, location and access control, common in cellular networks.

From the point of view of the operator, the all-IP system is much more flexible and simple to keep and manage, making more dynamic the service launch and reducing its cost. Also, since it is a session control system valid for any access technology (3G, WiFi, ADSL, WiMaX), it optimizes inversion without subordinating the evolution and inclusion of new technologies to any special manufacturer because it is an open standard.

Among the available functions in IMS, we can find interfaces to manage sessions, control access, manage mobility, control services, charging, etc.

IMS uses SIP as protocol in its communications with applications and users and only requires to have guaranteed the client's IP connectivity. This way, it is separated from the type of access used during the session.

11.2.1.1 Identification in IMS

The following identifiers are used in IMS:

- **IMPI (IP Multimedia Private Identity).** It is the user's private identity represented by a unique global identifier assigned by the operation and only visible to the IMS network control nodes. IMPI follows the Network Access Identifier format (e.g. user129@movistar.es) as well as authentication, authorization, charging, etc.
- **IMPU (IP Multimedia Public Identity).** It is the user's (or service) public identity, *i.e.*, the identifier through which the user (or service) is known by other entities or users. One user can have several IMPUs. IMPU follows the SIP URI format (e.g. sip: user@movistar.es) and is used to establish sessions. One user can be registered with the same IMPU from different terminal or with multiple IMPUs from the same terminal.

In the case of IMS accesses with UICC (Universal Integrated Circuit Card), identifiers can be derived and/or obtained directly from the UICC. If the access is done without UICC, IMPI and IMPU are provided to the IMS client as part of the configuration process.

11.2.1.2 Elements of the IMS Architecture

Figure 5 depicts a scheme of the interconnection of the IMS control layer, the rest of layers and the basic nodes in their core among which we can point out the following:

- **P-CSCF (Proxy Call Session Control Function**): First contact point between the user and the IMS subsystem. Among its functions there are:
  - o To route register messages to a specific I-CSCF.
  - o To provide part of the state control to all that sessions belonging to users registered in the system.
  - o To authorize or unauthorized quality of service resources for a certain session.
- **I-CSCF (Interrogating Call Session Control Function):** It is an element that obtains, from the HSS, the S-CSCF location corresponding to each user.
- **S-CSCF (Serving Call Session Control Function):** A node in which there is the most part of the system intelligence. Among its functions, we can point out:

- To store register information in the HSS platform (URI-Location relation).
- To provide an important part of the state control in sessions of registered users.
- To make service requests to the higher-level AS by using the call filter criteria defined in each user's profile.

- **HSS (Home Subscriber Server).** It is a database with information about the user that stores:
  - User profile: ID, IP address, access licenses, available services, call filter criteria, etc.
  - User location (S-CSF were it is accessible)
  - User state.
- **SLF (Subscription Location Function):** Database that stores the HSS in which there is information about the requested user. This element is not necessary when the system has only one HSS.

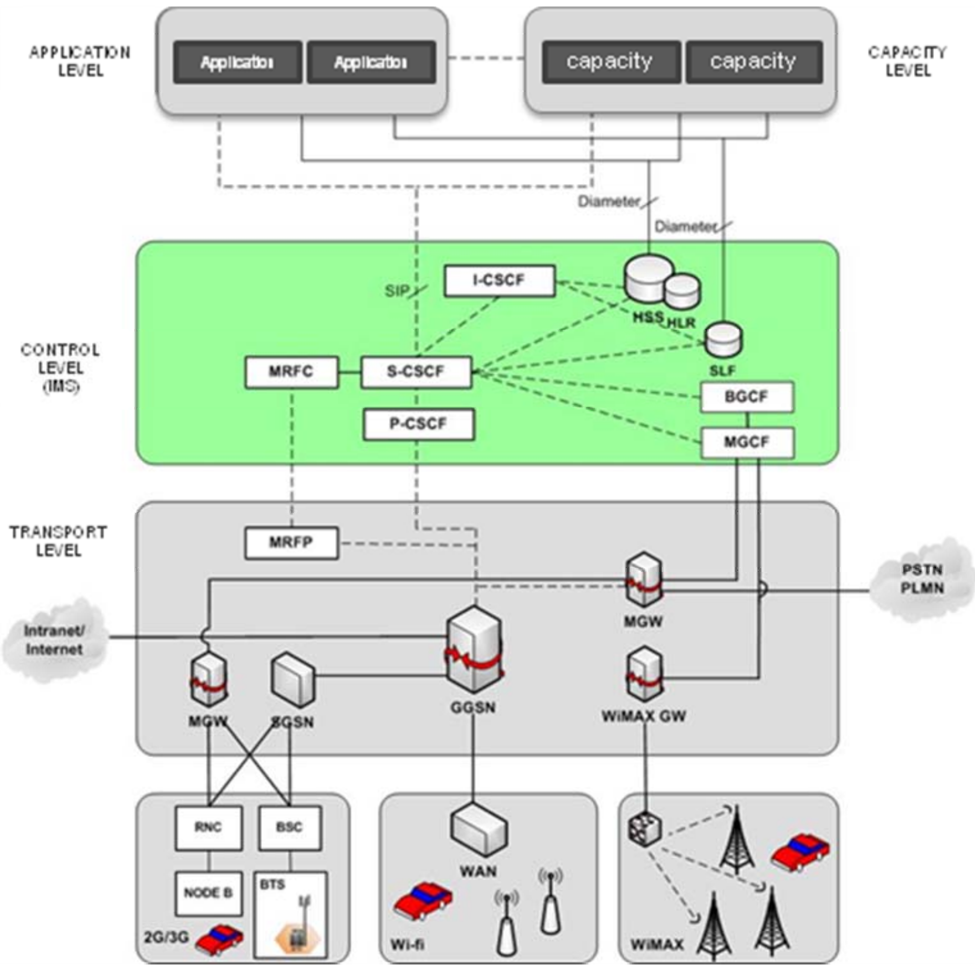

Figure 14 - IMS Control Layer Interconnection

11.2.1.3 *Register in IMS*

Before starting using IMS services, the user must register in the IMS Core so that the system elements update the information they have about his state.

Figure 6 shows a summarized scheme about how the register is made with the collaboration of several Core nodes.

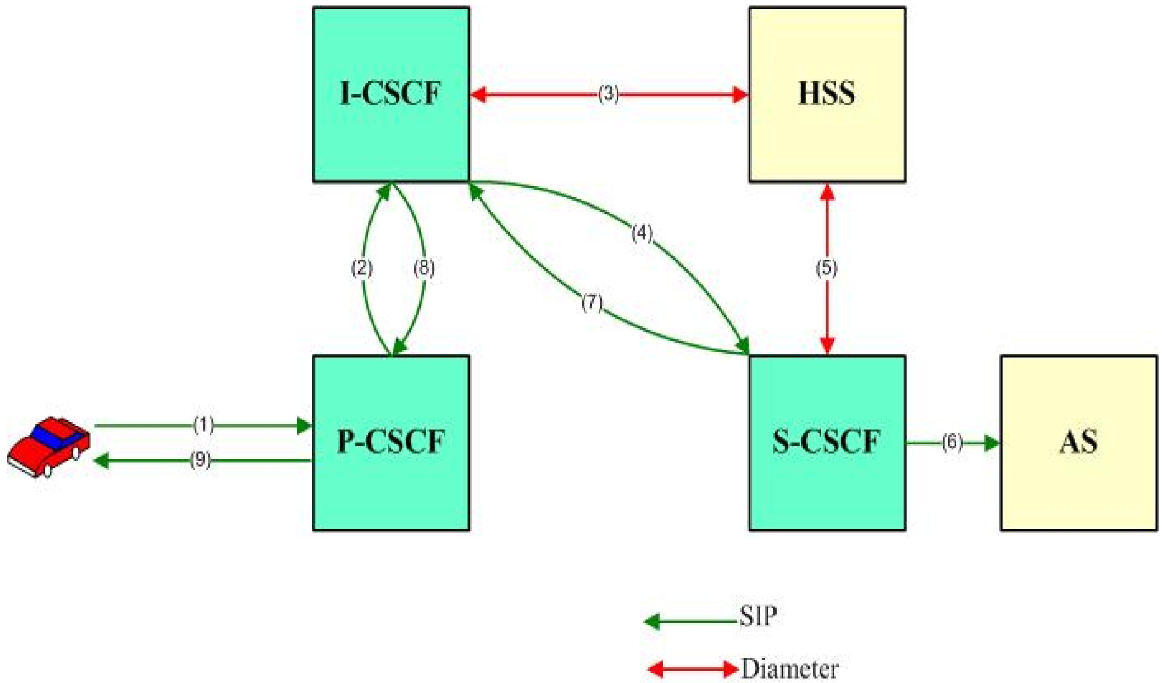

Figure 15 - IMS Registration scheme

The process is the following:

- The user sends a registration request including one (or some) of his public identities (IMPUs) to the login point of the IMS subsystem, P-CSCF (1).
- This request is forwarded to I-CSCF (2) which obtains from the HSS (3) the S-CSCF in which the user has to register.
- Once the request arrives to the chosen S-CSCF (4), this S-CSCF acts as registrar, and for that, it downloads the user authentication data from the HSS (5) and sends him a challenge (7), (8) y (9) so that he can authenticate himself.
- It may happen that the S-CSCF has to give notice of the user registration to some application that requests it (6) as it happens in the case of presence control applications.

- Once the user has been correctly authenticated, the S-CSCF completes the user's registration and notifies the HSS updating the information it has about him.

11.2.1.4 Access to IMS Services

Once the user has been registered, he can start using the IMS services. Let's see, as example, what control traffic is necessary to be managed by the IMS Core so that two registered users can establish a communication (e.g. a video call between Car1 and Car2).

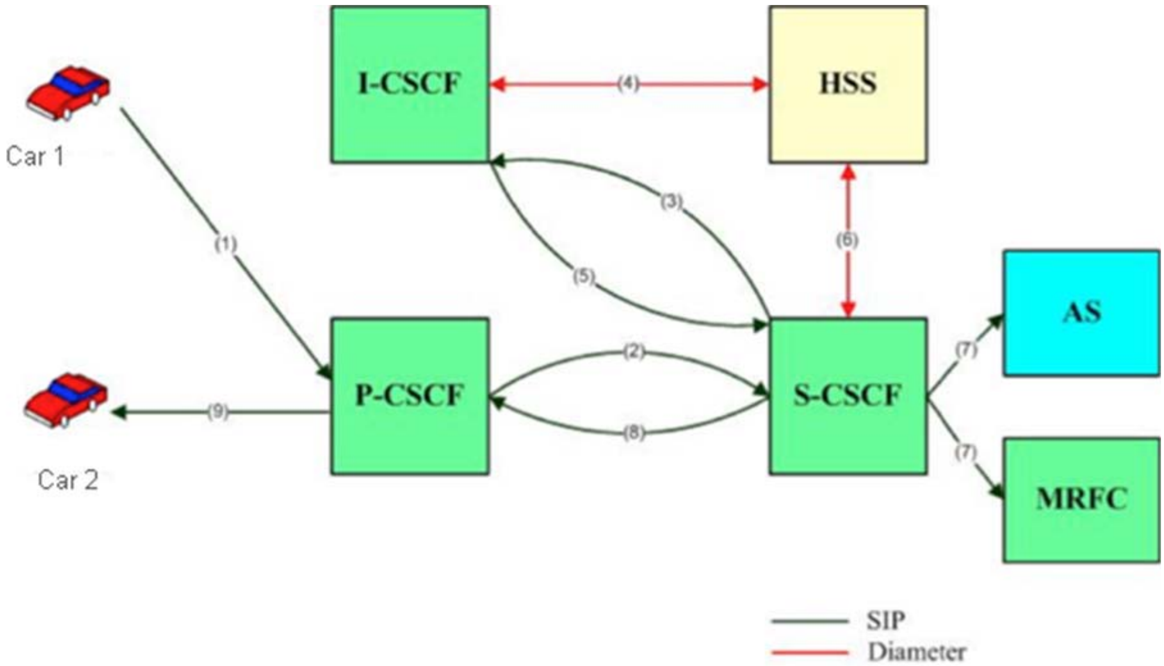

Figure 16 - IMS Services

The process is the following:

- Car 1, who initiates the video call, sends a request to establish the call including the public identity (IMPU) of Car 2 with which it wants to communicate. Additionally, a series of parameters to negotiate the media to be used in the communication (video, audio, etc.) are also included as well as codec for each media used.
- As it happens with the registration request, the contact point of the user with the IMS Core is P-CSCF. (1) P-CSCF sends the request to the S-CSCF assigned to Car1. (2) Which consults the I-CSCF (3) about which is the S-CSCF assigned to Car2, and that information is provided by the HSS (4).

- If both users share the IMS Core, I-CSCF forwards the video call establishment request to the S-CSCF of Car2 (5) which will download, if necessary, information about Car2 from the HSS (6).
- Depending on the operation, S-CSCF will probably need some media resources (in this case, video) and for that, it will send a request to the MRFC, or notify some application (7).
- Once this has been made, it broadcast the request to establish the requested communication between the cars to the P-CSCF corresponding to Car2 (8).
- Finally, Car2 receives the request form Car1 (9) replying in the appropriate way and starting the arranged communication.
- The data communication (in this case the video call) does not pass through the IMS Core, it is established Car1-Car2 directly.

11.2.2. Capability Layer

The **capability layer** provides horizontal functionalities that are interesting for the final user applications (e.g. ID Management, Policy Management, Mailing, Location, Broadcast/Multicast, User Profile, Terminal Profile, etc.) in a secure and safety way. Capabilities export open and standardized interfaces using several protocols (http, SOAP, SIP, etc.).

Using these capabilities, service development becomes simpler since they only need to implement the specific logic for the functionality offered to the final user whereas common functionalities can be offered by the capabilities.

The presence of available capabilities in the architecture does not force services to use them since their use is on demand. Even within a same service, certain functionalities can be delegated to capabilities and others; can have been implemented within the service itself.

The relation among the different capabilities, final user services and management systems, as well as certain functionalities necessary for them to be accessible in a secure and safety way (e.g. register, discovery, capabilities access control, user privacy, policy management) are defined by different standardized architectures. All architectures, which are compatible among them, complement each other offering the most suitable solution according to the type of service and necessities of the capabilities and users. [ADHOC1]

The relevant standardized capabilities architectures and the relation among them are:

- OMA Service Environment from Open Mobile Alliance.
- OSA/ Parlay and Parlay X from 3GPP and Parlay Group.
- Liberty Alliance from Liberty Project Forum.

Some of the most relevant capacities/skills that can be useful in an ITS scenario are the following:

- Location;
- Broadcast/Multicast;
- Identity Supplier;
- Charging;
- DRM;
- Sensors.

12. VEHICULAR AD-HOC NETWORK TECHNOLOGIES

12.1. Summary

The following table shows the main characteristics of these technologies:

Description	Vehicular Ad-Hoc Network Technologies (VANET)
Maturity	Experimental
Availability	Prototypes
Price	N/A
Bandwidth	N/A
Frequency	N/A
Coverage	N/A
Application to vehicular environment	Yes
Infrastructure requirements	Roadside units to improve the coverage
Legal issues	N/A
Standardization	IEEE, ETSI
Use in TeleFOT test site	NO
How to incorporate this technology to TeleFOT test site?	N/A

Table 12 – Vanet technology

12.2. Detailed description

Road safety and efficiency services will be provided using several communication technologies, which will involve vehicle-to-vehicle (V2V, also known as car-to-car) and vehicle-to-infrastructure (V2I) communications, as well as communications using future, specific roadside equipment, each of which is called Road-Side Units (RSU). These RSUs will form a network of hot spots, and will allow for vehicle-to-roadside (V2R) communications. Note that V2V is especially targeted for cooperative safety among cars, and that V2R is targeted for both cooperative safety and dissemination of traffic information in safety and efficiency scenarios. Typically, V2V and V2R communications are achieved using ad-hoc networks composed of vehicles and RSUs.VANET (Vehicular Ad hoc NETworks) is a particular case of Ad hoc networks. An Ad hoc network consists of a group of nodes provided by a wireless interface and which are able to communicate with each other without needing any infrastructure. One of the biggest benefits of Ad hoc networks is that one node can send information to another node which is outside its coverage area via multi-hop communication. Thus, when the receiving node is out of the sender's range, the intermediate nodes act as bridge between them and so, data travel from node to node until the destination is reached. Another important characteristic of Ad hoc networks is that nodes may move and this makes communication difficult. Ad hoc networks can be classified according to node's mobility, their process capacity, energetic efficiency, etc. [ADHOC1]. This would be that classification:

- **Mobile Ad hoc Networks (MANET).** It consists of a group of nodes with a random mobility pattern. Energetic efficiency is very important as well as memory and computational capacity, which are limited, and so, routing protocols cannot be very complex. Existing protocols mean that there is connectivity among all nodes, on the contrary, the packet is ruled out.
- **Wireless Mesh Networks (WMN).** In this type of networks it is assumed the fact that nodes are static and act as base station, which set up the multi hop routing. Processing, energy or memory capacities are not a problem for nodes.
- **Wireless Sensor Networks (WSN).** Sensor Networks consist of small devices with limited memory, capacity and energy. Therefore, it is very important for these kinds of networks that there are simple protocols. These networks are used to control one environment (temperature, humidity, etc.).
- **Vehicular Ad-Hoc Networks (VANET).** In this type of networks the nodes (vehicles) move at high speed. In this case, the nodes (vehicles) have a restricted mobility patron due to the fact that vehicles move on roads with a topology, norms, etc. A big advantage of using vehicles as communication nodes is the disponibility of a big processing and memory capacity, good communication possibilities and no energy problems. Moreover, vehicles have nowadays on board navigator and, therefore, the vehicle's location and streets on the map in the vehicle's surroundings are well known by the system and the user. Even so, nodes also have problems and unexpected events on this kind of networks. Navigators use maps that are not completely precise because they may be outdated or a street be suddenly closed for repair. [ADHOC2]

12.3. Routing algorithms

One of the most critical parts of the deployment of ad-hoc networks is the routing mechanisms. Many routing protocols for Ad hoc networks have been developed. Nodes mobility, topology instability, lack of pre-established organization and wireless communications operation make routing algorithms developed for fixed networks unable for these networks. In an Ad hoc network, routing algorithms have to create routing structures in a distributed and automatic way. In this section we will see a description of some of the most important routing protocols for Ad hoc networks.

There are many ways of classifying the various routing algorithms for Ad hoc networks. For example, depending on the moment where routing structures start to be created, we will talk of proactive protocols or reactive protocols. Proactive protocols are those that try to have information as precise and updated as is possible about the different possible paths to get to the various network nodes regardless of the routing necessities. On the other side, reactive algorithms obtain routing information only when it is necessary. Normally, operations to form routes are not initiated until it is necessary to send or receive a message. On the other hand, two protocol families, according to the type of information exchanged, can be identified: link-state routing protocol and distance-vector routing protocol. In an algorithm based on the link-state protocol, nodes exchange information about their neighbours with the rest of nodes on the network. However, in a distance-vector protocol, each node exchanges distance or cost information to the rest of the network nodes with its direct neighbours.Routing Protocols

Many multicast algorithms for Ad hoc networks are based on or are extensions of unicast or broadcast routing protocols. In order to classify the different routing protocols, they have been divided into proactive and reactive algorithms since this information is critical to evaluate the possibility of adaptation to vehicular networks.

12.3.1. Proactive Algorithms

Proactive algorithms are known for having a periodic working, regardless nodes' routing necessities. For this reason, nodes carry out activities intended to collect useful routing information even though it would not be necessary at that moment. Routes are calculated before being necessary. Tables' update may be periodic or determined by events.

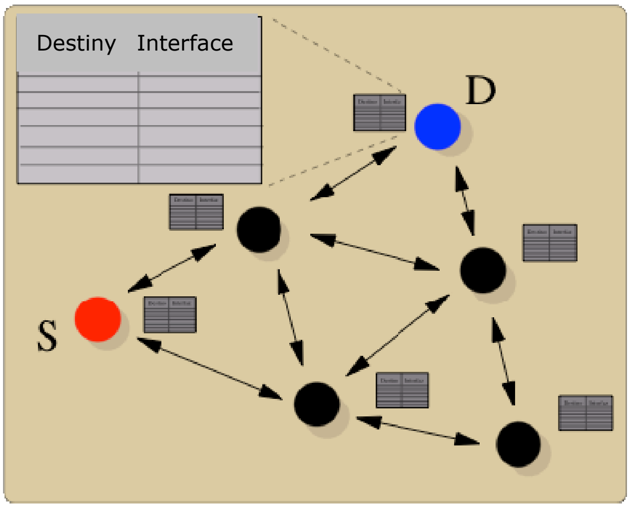

Figure 17 - Scheme of operation of a routing protocol for ad hoc networks

There are several proactive algorithms available, some of them are indicated below:

- **DBF (Distributed Bellman-Ford) Algorithm:** This algorithm aims to find the shortest route (in accordance with a certain metric) to any destination. To obtain that, it is based on the creation and maintenance of routing tables. Each node must have its own routing table with entries like <destination, distance, successor> being the neighbour the chosen node to be the next hop in the route towards the destination node. Their creation and updating is made in three different steps:
    - Start condition. Each node starts with a routing table which has entries only for directly reachable neighbours, i. e., at the initial moment each node knows how to get to their neighbours.
    - Send step. There are periodical <destination, distance> tuples' exchanges among immediate neighbours. The sending frequency may be configured depending on the size and the dynamic of the network. The more frequent the exchanges are, the more precise tables will be and the faster they will react to changes in the topology.
    - Receiving step. When receiving a new tuple, nodes check out whether the distance to its destination plus the cost of reaching the neighbour sending the tuple is less than the distance existing in the routing table for the same destination. If this happens, the table updates and goes to the send step.

- **DSDV (Dynamic Destination-Sequenced Distance-Vector) Algorithm:** it is a modification of DBF in order to remove the looping and count-to-infinity problems. Tables can be transmitted completely or partially and periodically or when any change happens. Every entry in the table will contain not only the distance to destination and the next hop but also a sequence number. One route is better than another when it has a bigger sequence number or, if there is an equal sequence number, when the distance is shorter.
- **WRP (Wireless Routing Protocol)**: It is a protocol based on distance vectors that improves DBF by removing the count-to-infinity problem. It is designed to be executed at the link level of a wireless network. As the table update messages may lose or become corrupted due to the nature of wireless communications, WRP introduces the use of reply messages (Acknowledgement Messages, ACK) to get reliable updates on the basis of retransmissions. Every node maintains four different tables: a distance table, a routing table, a link cost table and another one to manage ACK replies and retransmission called Message Retransmission List Table. It contains the sequence number of the received table update messages, a retransmission counter and a list of the sent updates.
- **OLSR (Optimized Link State Routing Protocol):** This is a proactive protocol, resulting from minimizing the update overhead of links on the link state algorithm. The optimization consists, principally, of reducing the size of the link tables exchanged and the number of retransmissions needed during the flooding periods. The key of this algorithm is the use of multipoint relays (Multipoint Relay, MPR). The MPR are nodes selected to broadcast messages to the network. By using only MPR nodes, instead of a whole flooding, the number of broadcast messages to reach all nodes in the network diminishes.

12.3.2. Reactive Algorithms

Reactive algorithms try to minimize every unnecessary information exchange. Routes are calculated only when necessary and therefore, nodes, which do not participate, do not try to have updated information at all times. To establish a communication it is necessary to carry out some kind of query to build the path.

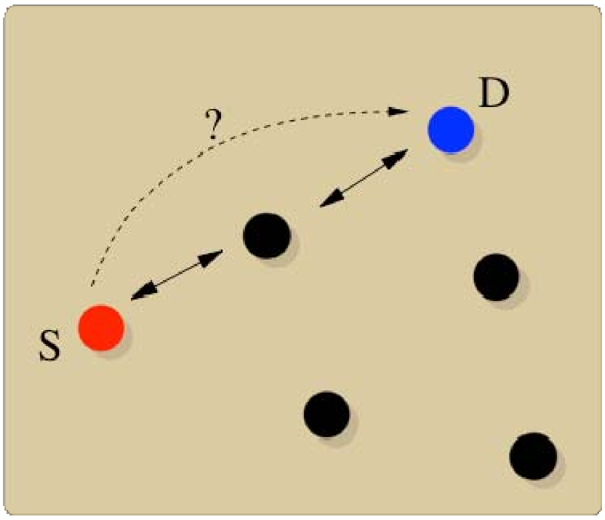

Figure 18 - Reactive algorithms scheme

The following paragraphs show some of the reactive algorithms available:

- **DSR (Dynamic Source Routing protocol)**: This protocol is specifically designed to be used in multi-hop wireless ad hoc networks with mobile nodes. DSR allows the network to be completely self-organizing and self-configuring with no need for an administrative infrastructure. DSR uses the Source Routing mechanism. Messages include the whole list of nodes needed to get to the destination. This protocol consists of two parts, route discovery and route maintenance. Both, nodes belonging to a route and adjacent nodes, keep in their tables the complete routes to the destinations. These routes are learned when receiving the messages directly or listening in promiscuous mode the different protocol messages sent by neighbour nodes. When a node S wants to send a message to another node, D, and does not know the route to get to D, it must discover the route first. To do that, it sends a ROUTE REQUEST message, containing both addresses, to all neighbours. When receiving this message, every node will check whether it already has or not an entry for the searched destination. If it does not, it will forward the message causing a controlled flooding; but if it does, it will reply with an ROUTE REPLY message that will travel unicast back. All nodes taking part in the ROUTE REPLY forwarding will add their address to the message, creating this way the complete route and sending it to the destination. The advantage of this protocol is that intermediate nodes do not need to know how to route a message due to the route is included in the header.

- **AODV (Ad-hoc On-demand Distance Vector)**: AODV is a routing protocol that makes use of broadcasting messages and alert to its presence at pre-established regular intervals or make use of a technique called Link Layer Feedback in order to archive that nodes would learn their neighbours and would keep their neighbours' tables updated reflecting changes in the near topology due to the appearance, disappearance or mobility of nodes. Unlike DSR, this protocol maintains the route state in every intermediate node. These tables keep updated along time removing unnecessary entries. AODV uses sequence numbers for each destination in order to avoid processing messages with out-of-date information and the creation of loops.
- **TORA (Temporally Ordered Routing Algorithm protocol)**: TORA is a routing protocol for Ad-Hoc networks in which they have tried to uncouple the control message generation from changes produced on the network topology. TORA builds a multipath structure without loops that is used as the basis to send traffic towards a certain destination. At the same time, this built structure can be reused to send messages towards other destinations from other different sources apart from the initial source. Although it is a reactive protocol, it can also be proactive. TORA is a distributed protocol based on link reversal routing algorithm and Lightweight Mobile Routing (LMR). TORA is designed to discover on-demand routes, provide multiple routes towards a destination, establish routes quickly and minimize communication overhead thanks to its location algorithm that reacts to topology changes. In TORA, long routes are more important than the shortest (shortest-path routes) but harder to discover routes. This way, the route discovery overhead reduces because, if it already exists one, then a shorter one is not searched.
- **DYMO (Dynamic MANET on-demand):** This protocol is an evolution of the AODV protocol. As most of reactive protocols, DYMO consists of two basic operations: route discovery and route maintenance. During the discovery operation, the source node starts flooding the network with a Route Request message (RREQ) so that it reaches the desired destination. During this flooding, nodes forwarding the message store who is the message's predecessor. When the destination node receives the RREQ, it replies with a Route Reply (RREP) message that is sent unicast towards the source node. Exactly like before, intermediate nodes store information about the inverse path. When the message reaches the source node, it means that communication routes in both directions have been established between both nodes. In order to react to changes in the network topology, nodes keep the routes and monitor the link status. When a packet for a route which link is not valid anymore is received, a Route Error (RERR) message is created and sent to the source. When receiving the RERR message, the source initiates the discovery process again in case it needs to continue sending messages.

12.3.3. Multicast Algorithms

Multicast routing is a subset of the routing problem. The advent and development of multiuser applications have led to the need for reliable, cost effective multicast mechanisms. The majority of the ad hoc network applications are in the multiuser domain. Hence the issue of multicast routing becomes an object of interest as well as concern. All the problems associated with routing are applicable to multicasting as well, but the demands are much higher, in the sense that multi-point delivery should be guaranteed.

Multicast operations in mobile networks are generally used for dissemination of important and confidential information. The multicast operation is also used as a means for synchronization of operations between various mobile hosts. Multicast algorithms are hence expected to ensure a reliable message delivery and in most of the cases, the source of message transmission is to be informed of the success of the message delivery.

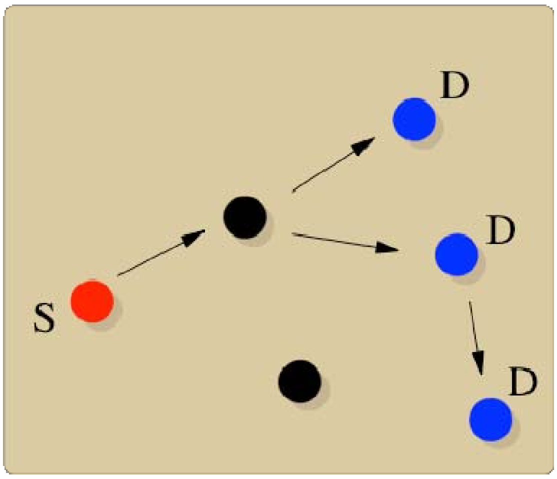

Figure 19 - Scheme of the multicast algorithm in ad hoc networks

Some of the multicast algorithms that can be used in the ad-hoc networks are as follows:

- ODMRP (On-Demand Multicast Routing Protocol): ODMRP is a multicast routing protocol for ad hoc networks with mobile nodes. It uses a mesh multicast scheme, instead of the typical tree network scheme, with restricted flooding inside the mesh network. A small set of nodes (Forwarding Group) do the broadcast. It uses procedures that are typical of reactive protocols in order to build routes and maintain multicast groups. ODMRP adapts well to wireless Ad-Hoc networks with mobile nodes where the bandwidth is limited and topology changes fast and frequently. ODMRP builds Shortest Path Tree (SPT) or relay mesh for each multicast group. The greater the number of sources and receivers is, the more nodes from the relay set shall be used in different SPT and therefore, the rate of destinations reached grows. However, this adaptive capacity involves a high network overhead regarding control messages.

- **CAMP (Core-Assisted Mesh Protocol)**: This protocol builds and maintains a multicast mesh for the distribution of information to each multicast group. Every mesh is a topology network subset in which there is at least one path between each source and each destination in the multicast group. CAMP ensures that the shortest path between sources and receivers are part of the mesh. Similarly to CBT, it uses the core node concept as the management centre of multicast groups. The main difference with CBT is that the use of these nodes is not mandatory.
- **MAODV (multicast extension of AODV)**: Also known as Multicast AODV, this protocol pretends to build shared bi-directional multicast trees, which connect multiple sources and destinations for each multicast group. These trees remain whereas there are members of the multicast group connected to some part of the tree. Each multicast group has a leader node which is in charge of maintaining the value of the group sequence number which is also the root of the multicast tree. This number is used to ensure the accuracy of the routing information. One node becomes a leader after trying several times to join a multicast group by flooding the network with join messages with no success.
- **AMRIS (Multicast Protocol for Ad hoc Wireless Networks)**: AMRIS is an on-demand multicast protocol which builds shared trees to support multiple senders and receivers in a same multicast group. Every node has an id called msm-id which is dynamically calculated by each node itself during the initialization phase. The msm-id value of each node must be higher than the one of its logic parent node in the multicast tree. The initialization phase is initiated when a special node called Sid is elected among the different senders. Sid shall have the smallest msm-id in the multicast group and, therefore, it shall be the root of the tree. The msm-id allows nodes that have broken off from the tree due to topology changes, to re-join the tree without causing loops. Every node broadcasts periodically a message to neighbouring nodes to signal their presence and containing its msm-id. AMRIS consists of two phases: tree initialization mechanism and tree maintenance mechanism. The first one consists of flooding the network with an announce message from the Sid node. When receiving these announces messages, each node calculates its own msm-id adding to it a random value that replaces the original one in the message. The periodic exchange of presence messages allows each node to determine what neighbour is its parent node in the multicast tree. A node may receive announce messages from different neighbours. After comparing the msm-id values and other metrics included in the header, it may change its parent if it decides that the new one is a better one.
- **ADMR (Adaptive Demand-Driven Multicast Routing**): ADMR does not use periodic control messages neither it depends on any protocol from an inferior level to execute this or other operations. ADMR builds different multicast trees for each receiver and multicast group. Having the receivers as the root of the multicast trees, paths created from the sources are the shortest ones in number of hops. The ADMR protocol is hard to adapt to sensor networks due to it requires a high memory level to store the status of nodes.
- **DSR-MB**: Is a specific extension of DSR for small Ad-Hoc network environments with high mobility, flooding may be more efficient than creating and maintaining structures for multicast or broadcast packet forwarding. When a node receives a multicast or broadcast message for the first time, it just forwards the message to all its neighbours. There is no need for nodes to store any type of status. It

consists of a very simple strategy that only has use in certain and very concrete scenarios. Therefore, this protocol cannot be considered for general multicast use.

- **MMRP (Mobile Mesh Routing Protocol)**: MMRP is a part of a family that also includes MMLDP y MMBDP. A flexible and extensible Ad-Hoc network may be build using these three protocols. MMRP is based on the link state protocol, i.e., on the exchange of the information that each node has about its neighbours. MMRP creates messages containing information about its neighbours and which are sent in a limited broadcast. These messages are not sent to the entire network but to a limited number of hops. The concept is to make use of the possible spatial area in communications. The size of this area, in other words, the maximum number of hops in which a message can be transmitted to, are part of the protocol parameters and must be configured depending of the type of network.
- **MMARP (Multicast Routing for MANET Extensions to IP Access Networks)**: MMARP is a protocol especially designed for mobile Ad-Hoc networks. It is fully compatible with the standard IP multicast model and allows multiple mobile nodes with standard IP to take part in multicast communications. Communication with fixed nodes acting as connection bridges for mobile networks is performed by mobile nodes one hop away from the fixed ones. These nodes are called Multicast Internet Gateways (MIGs). The MMARP uses a mesh-based distribution structure, similar to the one used by ODMRP, which offers good protection against the mobility of the nodes. MMARP uses a hybrid approach to construct and maintain that distribution mesh. Multicast routes among mobile nodes are established on-demand, as reactive algorithms; however, multicast routes to the multicast sources in the fixed network are established proactively.
- **MURU (Multihop Routing Protocol for Urban Vehicular Ad-Hoc Networks)**: MURU is a reactive protocol especially thought for urban scenarios. The idea is to try to predict cars' movement on the basis of roads and speed limits. This allows to reduce control overhead of the route discovery process. MURU introduces a new metric called EDD (Expected Disconnection Degree) to measure the quality of each one of the possible routes. It tries to take into account the probability of breakage of a certain route. Its performance is not compared with VANET specific protocols but, in view of how it operates, it does not seem to be very efficient or scalable. This is due to the need of broadcasting Route Request (RREQ) messages.

12.3.4. Geographic Unicast Routing Protocols

Routers existing before wireless networks consisted of a series of communication ports (also called interfaces), which linked them to other routers. These physical links created paths between different points in the network. The routers through which traffic was routed to get from the origin to the destination determined the path between two nodes in the network. The possibility -or the lack of it- of generating traffic between two different routers was directly related to the existence or not of a physical path, which could connect one of the ports in a router to a port in the other router. In any case, the space situation of the routers was not related to the chances of communication between them. In fact, two routers located in the same room could perfectly be isolated one from the other.

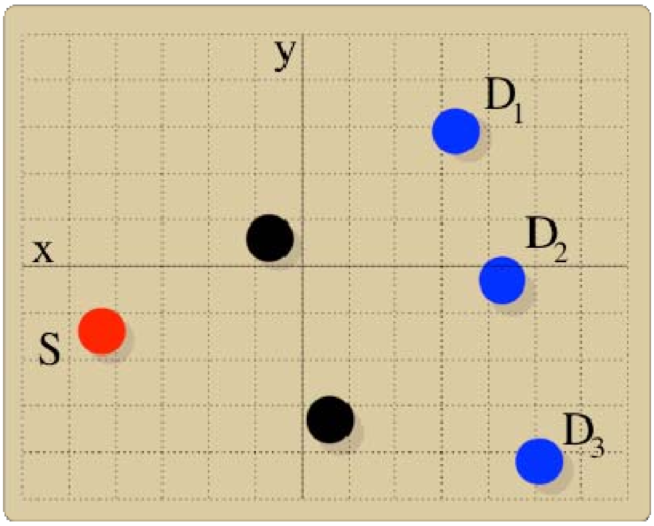

Figure 20 - The Geographic routing protocol makes use of node positions

The way of identifying nodes in the network has a direct influence on the routing protocol architecture. The messages sent must contain both the origin address and the destination address. A router's basic function reduces to processing incoming messages according to the destination address showed in their headers. When receiving a message, an interface suitable for the destination address included in the message must be selected. For this purpose, routers count on pair tables <destination, interface>. Although these tables can be defined manually, they are normally built by distance vector or link state routing protocols.

The development of wireless communications changed the situation drastically. Two routers with wireless communication interfaces have the possibility to communicate directly to each other if the distance between them is less than the coverage area of the radio interface at the routers and the environment conditions allow it. In this case, the physical location of the routers is directly related to the creation of paths between network nodes. When one of these routers sends a message, although there is only one wireless interface, due to it is a shared environment, all nodes within the coverage area of the sender receive messages. In the practice, we could talk about as many ports/interfaces as receivers in the coverage area. Therefore, in principle, the number of interfaces is unknown and dynamic at all times although it is possible to deploy a wireless network with a predefined structure.

Taking into account that, in architecture of this kind, nodes' name or nodes' identifier won't serve to determine the path to follow to get to them, it is necessary to use their own position in the space as a node identifier. Using coordinates in a Cartesian space as node identifier, the function of the router changes in the same way as the information to be included in the header of a message. Therefore, messages must include both origin and destination position. Routers must know their own position in the same coordinate space in order to calculate the relative position of the message's destination regarding routers themselves.

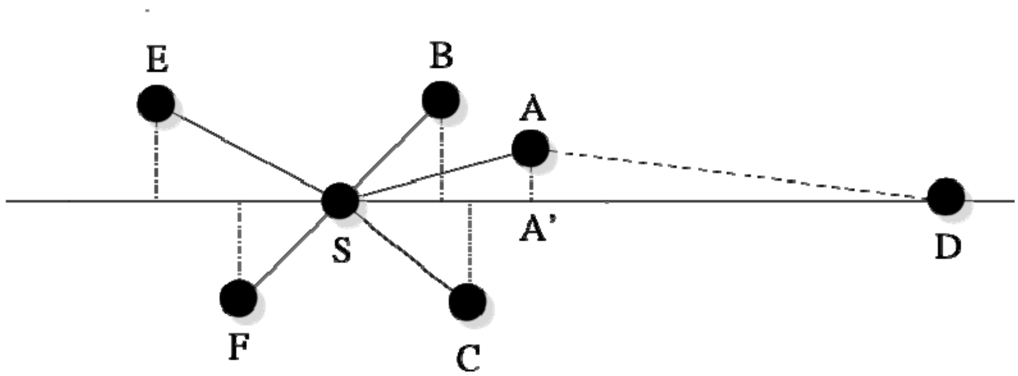

Figure 21- Different ways of measuring progress

At the end of the 80s works in geographic routing or based in coordinates begin. The concept of progress is the basis of all these works. The previous figure shows any given node in the network, **S**, which wants to send a message to another node, **D**, which is far enough to not be reached directly from **S**. It also shows a set of nodes: **A, B, C, E**, and **F** which are neighbours of **S**, i. e., those nodes which are within the coverage radius of **S** radio interface. The progress obtained when selecting a neighbour as the next hop is measured as the segment projection going from **S** to the neighbour placed in the line existing between **S** and destination, **D**. In the figure, **A'** appears as the point in the line going between **S** and **D** which match with the projection of **A** on that line. Progress may be positive if it goes in the direction to the destination and negative in the other case. In the figure, nodes **A, C**, and **F** provide a positive progress whereas nodes **B** and **E** give a negative one.

In geographic routing algorithms or coordinate based algorithms, it is not necessary to exchange information with neighbour nodes about the routing tables. The most important thing is to know the position of the destination node and also of neighbour nodes and the node making the decision. This decision is made on the basis of geometric criteria. Positions must be defined in a coordinate system that may or not correspond with the classic R^2. The working of the geographic routing algorithm can be divided into four phases.

- **Obtaining the location of the destination node**. This information is usually included in the header of the message. The originating node must place this information in order to be available for the possible intermediate nodes in the path towards the destination. The location of the destination node is necessary to determine the next hop in phase 3.

- **Obtaining the neighbours location**. This information is obtained through a mechanism, which periodically sends short messages between neighbours. Neighbours location is fundamental to determine which one will be the next hop in the path.
- **Determining the next hop**. Selecting the neighbour is the key of every geographic routing algorithm. Its entry parameters are the locations of the neighbours, destination and current nodes. It gives back the suitable neighbour to act as the next relay in the path towards the destination.
- **Deliver the message to the next hop.** Once the next hop is chosen, the message must be delivered to it.

Geographic routing is based on the location and identification of nodes. Therefore, three factors take part when using any algorithm based on this technique. Three solutions to the three posed problems are described below.

- **Nodes may know their own location.** One node can know its own location in several ways. Location may be calculated with a triangulation system by measuring the power of the signal received from certain fixed nodes. Nodes may include GPS (Global Positioning System) devices. A node location does not need to be real. Systems, which allow the assignment of virtual coordinates in a distributed way, have been developed. There are also distributed location systems and triangular-based. On the other hand, some works present routing algorithms that do not need coordinates.
- **Nodes may know their neighbours location**. Taking into account that one node can know its own location, it is very simple to obtain the location of all its neighbours. Being a neighbour means being within the coverage radius, and therefore, direct communications are possible. Using the shared environment nature of wireless communications, one node can periodically broadcast short messages containing its own location. These messages, commonly known as **Beacons**, allow each node to know its nearest neighbours location.
- **Nodes may know the destination location.** Including the destination location in the header of the message obviously allows intermediate nodes to know it. The problem is in determining the destination location in the origin node.

The two existing types of geographic routing are as follows: greedy forwarding and face routing. Both of them have the aforementioned four phases and use locations to decide the next hop. The difference is in which situation one or the other is used. Greedy forwarding is only applied when there are neighbours providing a progress towards the destination whereas face routing can always be applied.

12.3.5. Greedy forwarding

Most of geographic routing algorithms consist of modifications of the neighbour selection function. When a node running a routing algorithm receives a message which destination is not it itself, it has to choose one of its neighbours and forward the message to it so that the routing can continue its path until it reaches the destination.

Taking the aforementioned notion of progress or its slight variations as basis, a selection function, which chooses the best candidate among those neighbours offering progress, can be developed. Choosing a neighbour that does not offer progress or does it negatively may end in a loop that has to be avoided. The problem is that this can only be possible when there are neighbours offering progress.

Greedy forwarding can be applied as long as there are neighbours offering progress. Taking this idea as basis, some of the algorithms using this schema are presented in the following paragraphs:

- **Greedy Scheme:** This algorithm is one of the first algorithms using geographic routing. In it, the neighbour selection function consists of simply choosing the nearest neighbour to the destination. It is a greedy algorithm but it is easy to find examples where this algorithm goes into a loop. There is a simple reason for that, progress has been mistaken for proximity to the destination. The fact that a neighbour is the nearest node to the destination does not mean that it is nearer than the node itself and so, it could choose as next hop a neighbour which progress would be negative. The next figure shows an example in which there is a loop. Independently of the node from which the message is sent towards D, choosing the nearest neighbour originates a loop.

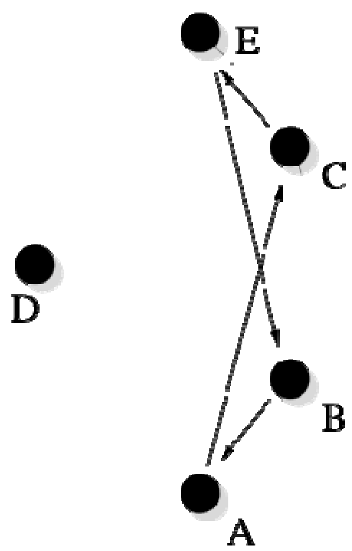

Figure 22- Loop in a greedy scheme

- **CR (Compass Routing)**: Is another variant with a greedy scheme. Compass Routing, is an algorithm based on the angle measurement. The angle formed between S and N is compared with the angle formed between S and D. The selection function chooses as next hop the neighbour with the smallest angle. Like in Greedy Routing. This algorithm can also go into loops.
- **MFR (Most Forward Progress)**: In order to solve the looping problem appearing in both of the aforementioned algorithms, the MFR was designed as the first algorithm that really uses the concept of progress. In particular, the selection function chooses as next hop the neighbour who gives the greatest progress towards the destination. This algorithm does not go into loops like the above mentioned do but, loops made of two nodes may happen when one of the nodes has the other one as its only neighbour. These type of loops are avoided by making each node save the neighbour to which it has just sent the message. If immediately afterwards it receives a message from that neighbour then the message is ruled out and the routing aborted.
- **Nearest Forward Progress (NFP)**: This algorithm is a modification of the MFR algorithm in order to save energy so that, instead of choosing the node that offers the greatest progress, the one offering the least
- **GG (Gabriel Graph).algorithm**: In the Gabriel Graph (GG) planarization algorithm, every node keeps all those neighbours in such a way that the circle which diameter is formed by the edge linking the neighbour to it does not contain any other neighbour.
- **Relative Neighbourhood Graph**: Nodes keep all neighbours for which the intersection of the circles that are in their location, with a radius equal to the distance between the nodes, has no more neighbours. This algorithm removes more neighbours than GG because the area not admitting neighbours is greater.
- **Local Delaunay Triangulation**: To apply the Local Delaunay Triangulation each node, after obtaining a planar vision with GG, conserves all the triplets of nodes from its local sub-graph for which the circle, which centre is the triplets' circumcentre and which circumference contains the three points, does not contain any other neighbour from the neighbourhood.
- **GPSR (Greedy Perimeter Stateless Routing)**: GPSR combines both greedy routing and perimeter routing. It is based on FACE-2 and consists of an algorithm working in planar graphs. It uses MFR greedy routing as long as it is possible, that is, as long as there are nodes offering progress towards the destination. When it arrives at an empty area or local minimum, as authors call it (a node where no neighbour closer to the destination can be found apart from itself), it starts the FACE-2 perimeter routing. This means that messages must have a header showing which routing mode is being used at every moment. This way, when a node receives the message, it knows what selection function has to apply. The perimeter mode ends when a node located nearer to the destination than the node that changed to that mode is found. The idea behind this algorithm is to

combine the efficiency of the greedy forwarding with the effectiveness of the perimeter forwarding.

- **AFR (Adaptive Face Routing) and GOAFR+**: AFR is an extension of FACE-1 in which, by an imaginary ellipsis with focus points on the origin and destination nodes, the faces through which the message can be routed are enclosed.

12.3.6. Geographic Routing in VANETs

Geographic routing uses nodes' locations as their addresses, and forwards packets (when possible) in a greedy manner towards the destination. One of the key challenges in geographic routing is how to deal with dead-ends, where greedy routing fails because a node has no neighbour closer to the destination. Geographic routing is scalable, as nodes only keep state for their neighbours, and supports a fully general any-to-any communication pattern without explicit route establishment. However, geographic routing requires that nodes know their location. While this is a natural assumption in some settings (e.g., sensornet nodes with GPS devices), there are many settings where such location information isn't available. Some geographic routing algorithms are presented hereby:

- **GSR (Geographic Source Routing)**: is a geographic algorithm to route unicast packets in VANET networks. GSR is presented as a solution to the problem of the standard geographic algorithms, especially those of local minimum recovery in VANET environments. The GSR algorithm consists on determining through which junctions the message should pass through to reach the destination. The vertices represent junctions and the links, the adjacent streets. The message's transit between the junctions in the path is done greedily assuming that there are not obstacles. On this point, authors obviate the fact that there may not be enough vehicles between two junctions as to forward the message.
- **SAR (Spatial Aware Routing)**: SAR is an algorithm that works like GSR following a pre-calculated list of junctions through which messages can flow towards the destination. The list of junctions of the found path is as well introduced in the packet header and this one is routed greedily between junctions. However, when it comes a situation in which a point where there are no vehicles (nodes) is reached, three techniques are proposed to keep routing:
    - Keeping the packet in the vehicle for a prudential time and periodically try to restart the routing when finding new neighbours (other vehicles). If the packet is kept too much time or the buffer becomes full, then it is ruled out.
    - Using greedy forwarding normally as long as it is possible and until the source routing could be taken up again.
    - Finding an alternative route to get to the next hop of the source routing by using the source routing to reach it.
- **MDDV (Mobility-Centric Data Dissemination Algorithm for Vehicular Networks)**: MDDV is a geographic routing algorithm, opportunistic and path-

based. As in other algorithms, a road map acknowledge is assumed and a heuristic function is used to label the graph links that represent it. In particular, each street is given a weight related to the number of lanes it has. If streets are well dimensioned with respect to the traffic of vehicles that use them, then the higher the number of lanes, the higher the estimated vehicle density. Therefore, connectivity shall be better in that streets and messages could be routed more easily.

- **A-STAR** is an algorithm which includes, on the one hand, a strategy to determine feasible paths to route packets from the knowledge of the city bus routes. On the other hand, a new recovery strategy is described in the case that packets arrive at an empty zone (local maximum). A-STAR is similar to GSR and also to Terminate Remote Routing (TRR) since it also calculates in advance the path that a packet follows and includes in the header certain points in the path called anchor points through which the packet must pass. As in GSR, greedy routing is also used to carry the message from an anchor point to another. A-STAR it is a protocol based on spatial information, that is, it uses a global map to calculate routes.
- **GPCR (Greedy Perimeter Coordinator Routing)**: presents an improvement of GSR, which consists of avoiding the need of knowing a priori the city map and without using source routing. It is based on the idea that city topology, formed by streets and junctions, is already planar. Authors designate nodes located in the area of junctions as "coordinators". When selecting the next hop during the greedy routing, preference must be taken for coordinators since they are more likely to follow the paths than nodes located in the middle of a segment between junctions. Coordinators select the street to continue routing the packet by choosing among their neighbours that node offering the largest progress towards the destination. The recovery strategy consists of applying the right hand rule on street maps, turning at junctions.
- **VADD (Vehicle-Assisted Data Delivery)**: VADD is a routing protocol for Inter vehicular communication systems based on "store and forward". The difference between VADD and the previous proposals of this type is that this one uses information about the path, speed, road speed limits as well as the city topology in order to decide when to store and when to forward. The function that selects the "next forwarder" does it taking into account the time estimation towards each neighbour's destination. The followed strategy consists of forwarding the packet whenever a better next hop is found. When planning routes, streets with a higher estimated speed are chosen and the route is re-planned at every hop. If at a certain junction a next hop is not found in the direction selected as the most promising one, then the packet is not forwarded, the current vehicle keeps it and waits for the next opportunity to forward it. Once a node selects the direction to forward the packet at a junction, it chooses the neighbour acting as the next hop. There are two strategies for this operation: location-based and direction-based.
    - The location-based strategy chooses the closest neighbour to the destination. This strategy may create routing loops, which cannot be removed completely. Only those loops with n size, recording the n previous hops, can be avoided but the useful alternatives to route the packet are lost even with n=1. These are situations in which a loop beginning may be profitable in the long term.

- o The direction-based strategy chooses that neighbour which movement tends to approach the destination and, among them, it chooses the closest one to the destination. This strategy does not create routing loops.

12.3.7. Geocasting Protocols

The goal of a geocasting protocol is to deliver data packets to a group of nodes that are with a specified geographical area, i.e., the geocast region. In an ad hoc environment, there are numerous scenarios which would benefit from geocast communication (e.g., to broadcast emergency information within a mile radius of a fire or to broadcast a coupon for coffee within a block of a Starbucks). Geocasting, a variant of the conventional multicasting problem, distinguishes itself by specifying hosts as group members within a specified geographical region, i.e., the geocast region. In geocasting, the nodes eligible to receive packets are implicitly specified by a physical region; membership in a geocast group changes whenever a mobile node (MN) moves in or out of the geocast region

Three geocasting protocols are explained in the following paragraphs:

- **GeoTora, MGRP(Mesh-Bashed Geocast Routing Protocol) and GAMER (Geocast Adaptive Mesh Environment for Routing):** These protocols use a traditional unicast mesh-based protocol in Ad-Hoc networks to reach some node belonging to the multicast area. In the case of GesTORA, TORA is modified and used to carry messages to the multicast area. In particular, it is proposed a modification to route in anycase so a message sent by the source could reach "any" node in the multicast area. Using TORA as basis has the problem of the overhead produced by the flooding which is necessary to maintain routes between the source and destinations. On the other hand, MGRP and GAMER, try to reduce the flooding needed to maintain routes by using the forward zone. In other words, they execute a broadcast that is limited to a predefined area in which both the origin node and the whole multicast zone must be included.

- **LBM (Location-Based Multicast) and GeoCast:** These two protocols face the problem of broadcasting the same information to a group of destinations in a different way from the traditional one, which is based on the use of distribution trees. In particular, they define the group of receivers as those nodes, which are located in a certain area. Therefore, instead of spreading and maintaining a distribution tree, authors propose to broadcast a message to some point of the area (normally the centre) using geographic routing and, once it is there, to broadcast the message with some kind of limited flooding. LBM propose two ways of broadcasting the message to the zone occupied by the multicast region. The first one consists of doing a limited broadcast. That limitation is given by the smallest rectangle including both the muticast area and the origin node. This scheme presents a high overhead of controlled messages. In the second scheme, each node determines whether it must or not forward the message by comparing the location of the node from which it has received the message and its own location in relation to the centroid of the multicast area. The node broadcasts the message if the current situation shows a distance reduction; if there is no reduction it does not broadcast it. This scheme presents fewer overheads but it also has less reliability because there are less nodes taking part in the task of carrying the message towards the multicast area.

- **Abiding Geocast:** This protocol proposes a geocast mechanism with a theoretical model which is not very realistic and has very strong presumptions. The innovation in this scheme is that more than trying to broadcast a message to nodes of a certain area and at a given time, its aim is to broadcast the message to all nodes within the geocast area before a lifetime associated to that geocast message. This mechanism would not work if the destination region were not connected.

13. DSRC: Dedicated Short Range Communications

13.1. Summary

The following table shows the main characteristics of DSRC:

Description	Dedicated Short Range Communications (DSRC) is a short to medium range (1000 meters) communications service that supports both public safety and private operations in roadside-to-vehicle and vehicle-to-vehicle communication environments by providing very high data transfer rates where minimizing latency in the communication link and isolating relatively small communication zones is important
Maturity	Mature
Availability	Yes
Price	Various depending on the application
Bandwidth	6 to 27 Mbps
Frecuency	Europe: in August 2008 the European Telecommunications Standards Institute (ETSI) has allocated 30 MHz of spectrum in the 5.8GHz band for ITS.
Coverage	Estimated range of about 1000 m.
Application to vehicular environment	All use applications in this standard refer to ITS sector, and usually can fit in one of the following 5 categories: public safety, traffic management, traveller information/support, freight transport or transit
Infrastructure requirements	Not necessary.

Standarization	All of the DSRC activities within ISO are taking place within TC204. There are, within TC204, two different working groups developing DSRC, or DSRC-like, communications standards. In addition to the communications standards themselves, many other working groups are developing standards for applications that utilize DSRC for communications: Working Group 15 and Working Group 16
Use in TeleFOT test site	No
How to incorporate this technology to TeleFOT test site?	Using an On Board Unit with this radio interface

Table 13 – DSRC

13.2. Detailed description

Dedicated Short Range Communications (DSRC) is a general purpose rf communications link between the vehicle and the roadside. More specifically, it is a short to medium range communications service that supports both Public Safety and Private operations in roadside to vehicle and vehicle to vehicle communication environments. DSRC is meant to be a complement to cellular communications by providing very high data transfer rates in circumstances where minimizing latency in the communication link and isolating relatively small communication zones are important. [DSRC

13.2.1. DSRC Overview

European Directive 2004/52/EC deals with the interoperability of electronic road toll systems in the Community. The Directive sets a target date of July 2006 for international agreement on the definition of the European Electronic Toll Service (EETS).

Expert Group 1 (EG1) on microwave technologies was established by the European Commission to provide analyses on the inclusion of microwave DSRC technologies at 5.8 GHz to be used for the EETS, in support of the European Directive 2004/52/EC

13.2.1.1 Main characteristics

Deployment of electronic fee collection (EFC) in Europe is predominantly based on the European DSRC 5.8 GHz technology. Whereas the deployed EFC systems in Europe have much in common, they also display major differences e.g. in terms of technology and charging principles (i.e. whether it is based on network-, distance- or zone/congestion).

The tariff principle is one of the main reasons for differences in the classification parameters used and how the fee is calculated. The security architecture is also one of the main reasons for the differences, e.g. access to data stored in the OBU and the different security mechanisms to protect the integrity of the data.

Heavy goods vehicle (HGV) distance-based charging and urban road user charging are two relatively new market segments. The first implementations of HGV charging systems in Europe are found in Switzerland, Austria and Germany, and are based on different principles and technologies. The Austrian, German and Swiss OBUs are very different from one another. The former is a monolithic OBU, whereas the two latter are relatively complex (DSRC, external power, GPS, tachograph, vehicle movement sensor etc.) and need mounting by authorised agents.

In general, the current European EFC OBU served exclusively by DSRC 5.8 GHz technology has the following typical characteristics:

- Focused design: EFC for single lane and multi-lane environments.
- Inexpensive end-user equipment: mass-produced, inexpensive (approximate price 15-20 EUR) with a product life cycle of 2-4 years.
- High speed: predictable and reliable performance in constrained low speed toll lanes to mainline speeds (up and beyond the authorised speed limit). Claimed transaction error rates are typically less than 1 in 10,000 in all environments.
- Well-defined dedicated communication zone: vehicle-to-roadside communication link over a distance of typically less than 10-15m
- Self-mounted: OBUs are designed to be distributed through retail outlets, automated vending machines and by post. This ensures high market penetration with limited (or no) operator installation support.
- Harsh environment use: traffic and transport, capable of operation between extremes of air temperatures from parked vehicles in direct sunlight to sub-zero temperatures
- Autonomous: No interface to the vehicle. The OBU is simply fixed to the windscreen with a proprietary holder
- Low lifetime cost: long battery life, from 3 to 6 years is typical for an OBU with a simple human machine interface (HMI).
- High volume: around 10 millions3 OBUs have been issued in Europe with typical project batch sizes between 50,000 and 200,000. Start up volume batch sizes are sometimes greater based on forecast of initial adoption rates.
- Simple to use: simple HMI including an audible indicator and, in some cases, chipcard reader, an array of light emitting diodes, liquid crystal display interface.

Whilst elaborating on interoperability of all EFC systems operating in Europe, with their sometimes different requirements with respect to in-vehicle technologies (DSRC, GPS, SM/GPRS, chipcard reader, tachograph, gyro vehicle sensors …), it may be worthwhile to consider clustering of OBU requirements into groups in order to provide suitable solution for different groups of users / operators.

13.2.1.2 Technologies and systems

There are two main EFC 5.8 GHz technologies that are deployed in Europe, the European standard and the Italian Telepass technologies.

The **European standard EFC 5.8 GHz technology** is based on the CEN Standards for DSRC that define layers 1, 2 and 7, and that provide an application layer to the application users. The CEN standards support two parameter sets (known as L1-A and L1-B). Practical compatibility of these to parameters sets was proven in 2003 through laboratory and field tests carried out by the Federation of French motorway and toll facility companies (ASFA) and the Norwegian Public Road Administration (NPRA), see also Annex F. The EFC standard defines a generic transaction model, EFC functions and application data and the rules for addressing data. The associated EFC test standard [EFC AID Tests] defines OBU conformance test procedures.

Telepass technology, based upon UNI-10607 standard, is deployed in the nation-wide Italian EFC system. The Italian specifications [UNI DSRC, UNI AID] largely mirror the communication architecture of the European standard for DSRC and EFC. In addition it should be noted that:

- The Portuguese 5.8 GHz "low data rate" system is currently migrating to become "CEN DSRC EFC" compliant; and
- Systems using other technologies than microwaves at 5.8 GHz are outside the scope of EG 1; the current national Slovenian 2.45 GHz system (that is migrating to CEN DSRC EFC at 5.8 GHz), and the German HGV Tolling System

International interoperability between EFC systems based on the CEN 5.8 GHz technology exists between Switzerland/Austria4 was also demonstrated in the French/Spanish commercial pilot in the PISTA project. The Nordic countries also strive to achieve interoperability during 2005, through the NORITS initiative. The interoperability of these systems is all based on the CARDME / PISTA5 charging transaction specification, or dialects thereof and makes use of a common set of standardised functions and data.

The German HGV OBU provides a DSRC 5.8 GHz link according to CEN DSRC that is not used in the German HGV system but that is intended for interoperability with other European EFC systems after the German HGV OBU has been updated with the appropriate set of EFC application data and functions.

The MEDIA project – encompassing Alpine EFC operators (France, Italy, Slovenia, Austria and Switzerland) - is currently studying the prospect of introducing an interoperable service for Heavy Goods Vehicle (HGV) based on central account charging and a dual-protocol OBU.

13.2.1.3 Interoperability

EFC interoperability, using DSRC at 5.8 GHz, on a technical and procedural level involves both the DSRC protocol and the EFC application itself. Focusing on the interface between the OBU and the RSE - the most crucial interface concerning interoperability between existing DSRC-systems in Europe - both the DSRC and EFC application should be taken into account when defining the EETS using DSRC at 5.8 GHz.

There are two main EFC 5.8 GHz technologies that are deployed in Europe, one according to the European CEN standards and the other one according to the UNI-10607 standard.

A key issue to decide when designing the technical concept for interoperability is whether it is more effective:

- to update the RSEs providing them with additional application and communication capabilities to handle the existing and mixed population of OBUs, or
- to define a new “enhanced” OBU that is supported by all existing RSEs in Europe, or
- to agree on a common European solution, based on the CEN standards, that is associated with the ETTS that should be supported by all RSEs in Europe, not precluding the OBUs and RSEs to support additional services locally at their own discretion.

It is of course also possible to combine the approaches above. EG1 address this issue in its investigation of suitable concepts

14. IEEE 802.11p - WAVE

14.1. Summary

Description	Inside the IEEE 802.11 family, the 802.11.p standard is the one that results to be the most interesting one in the automobile sector. This standard, also known as WAVE (Wireless Access for the Vehicular Environment), has the mission to define improvements on the 802.11 family to be applied in the Intelligent Transportation Systems (ITS). These improvements go towards making communications more robust within an environment as aggressive as the vehicular environment, where the degree of terminal mobility and the number of interference sources is very high
Maturity	Non mature
Availability	Only prototypes
Price	No commercial products
Bandwidth	6, 9, 12, 18, 24 and 27 Mbps
Frecuency	Europe: In August 2008, the royalty-free spectrum and services dealing with safety and efficiency was allocated in Europe: 30 MHz at the 5.8-GHz band (5.875GHz – 5.905GHz).
Coverage	Estimated range of about 300 m and 6 Mbps.
Application to vehicular environment	All use applications in this standard refer to the vehicular sector, since it is its specific use, and they are particularly related to road safety, traffic control, commercial transactions and interesting information while travelling (routes, traffic, directions, etc.).

Infrastructure requirements	Not necessary. However, Road Side Units with this technology could be used to extend the range and to allow V2R (Vehicle to RoadSide) and V2I (Vehicle to Internet) communications
Standarization	Nowadays, the level of the specification standardization is still in draft state. It is important to clarify that this standard is considered as an amendment regarding the technical standards of the original IEEE 802.11 family. The workgroup in charge of carrying out this standardization is the IEEE 802.11p. In Europe, besides the IEEE, standardization and regulation efforts are driven by the C2C-CC1, the Communications Architecture for Land Mobile (CALM) working group of the International Organization for Standardization and of the International Telecommunications Union, and the ITS technical committee of the European Telecommunications Standards Institute (ETSI TC ITS), created in late 2007.
Use in TeleFOT test site	No
How to incorporate this technology to TeleFOT test site?	Using an On Board Unit with this radio interface

Table 14 – IEEE 802.11

14.2. Detailed description

Inside the IEEE 802.11 family, the 802.11.p standard is the one that results to be the most interesting one in the automobile sector. This standard, also known as WAVE (Wireless Access for the Vehicular Environment), has the mission to define improvements on the 802.11 family to be applied in the Intelligent Transportation Systems (ITS). These improvements go towards making communications more robust within an environment as aggressive as the vehicular environment, where the degree of terminal mobility and the number of interference sources is very high.

The main difference that this standard offers regarding the use of mobile networks is to improve the latency in transmissions that do not have very big range. To this, it is necessary to add the fact that it also allows sending both unicast messages and broadcast messages principally in applications related to road safety of small data transactions such as payments or exchange of local information. [WAVE]:

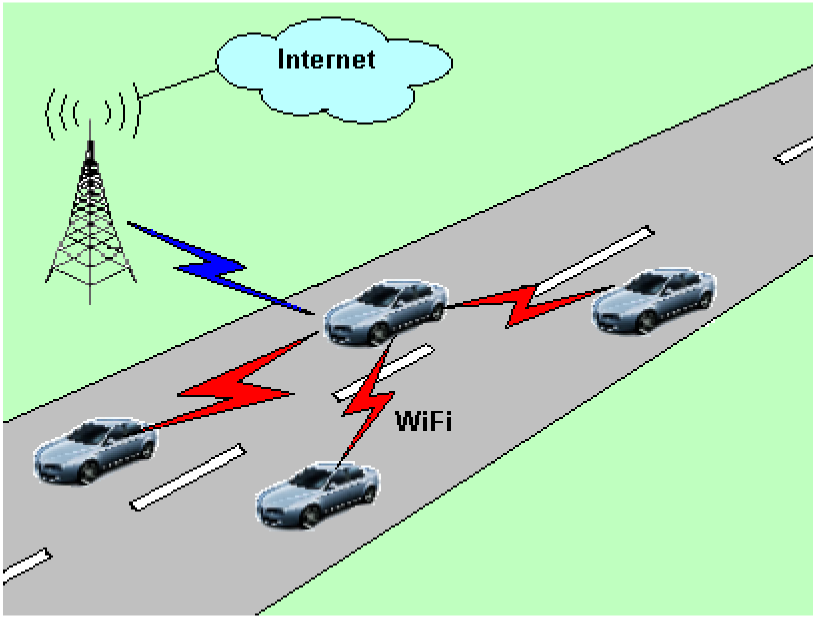

Figure 23- Example of sending a broadcast message in an application related to road safety.

In this section you will find a summary of the principal characteristics of 802.11p standard, covering the physical and MAC levels and also the state of the standardization process carried out by the Institute of Electrical and Electronic Engineers (IEEE). To conclude with this section, we will make a brief description of the forecast of the future of this protocol pointing out the interest that the automobile industry has in its application in the field of ITS, examples of use cases and emerging products.

14.2.1. Physical layer characteristics

The characteristics of the 802.11p physical layer are based on the 802.11a PHY layer using an OFDM modulation with 64 sub-carriers. Some of the differences between 802.11a and 802.11p are the following:

- Frequency assignment, channels division and channels spacing. In Europe and Japan it is used the 5.9 Ghz band, and the 6 Ghz band is used in the USA where, with a 75 MHz bandwidth, it is divided into 7 channels with 10 MHz each one and a guard band at the lower end of the 5 MHz band. The central channel is used to spread control messages whereas the rest of channels are used to offer service. Optionally, two adjacent channels intended to offer service may be used as an only channel at 20 MHz. European Authorities (CEPT) are working on a channels' frequency and distribution similar to the one provided for by the FCC in the USA.
- Power output. Four types of power (EIRP -Effective Isotropic Radiated Power-) have been defined in order to support wide ranges of transmission by establishing a maximum allowable power up to 44.8 dBm (30W). This value is reserved for transmissions among emergency vehicles. For the rest of cases it will be use a typical of 33 dBm.

- Estimated range of about 300 m and 6 Mbps.
- Tolerance for multi-path propagation effects. Mechanisms of tolerance to multi-path propagation effects in vehicular environment are increased because it is there where signals can reach destination directly or through reflections on the road.

The different characteristics of the IEEE 802.11p physical layer are described in a summary table below.

Bandwidth	75 MHz Frequency range [5.850,5.925] GHz
Channels	7 non-overlapping channels with 10 MHz frequency bandwidth.
Guard channel	5 MHz, at the lower end of the frequency band.
Modulation	BPSK OFDM. QPSK OFDM. 16-QAM OFDM. 64-QAM OFDM. The first 128 bits are always BPSK coded.
No. of sub-carriers	52 (48+4) One symbol consists of 48 data sub-carriers and 4 pilot sub-carriers.
Data rate	6, 9, 12, 18, 24 y 27 Mbps. From which 3 Mbps are used at the preamble.
Preamble duration	32 µs.
Symbol duration	8 µs.
Guard time	1.6 µs.
FFT period	6.4 µs.
Separation between sub-carriers	0.15625 MHz.
Code rate	½, ⅓, ¾.
Power	From 33 dBm (≈2 W). [EIRP] Up to 44.8 dBm (≈ 30 W). Only in the USA

Table 15 - IEEE 802.11p physical layer characteristics

14.2.2. MAC level characteristics

The 802.11p MAC layer implements the channel access mechanism provided by 802.11e standard, called EDCA (Enhanced Distributed Channel Access), to prioritize channel access including Listen Before Talk processes and random back-off policies. The channel access prioritization defines four types of traffic (Access Category): generic or background (AC_BK), best effort (AC_BE), voice (AC_VO) and video (AC_VI) traffic; assigning to each of these types a FIFO queue. This classification is made on the basis of the quality of service (QoS) required by the application, for example, the access background category refers to beacons and 'hello' broadcast messages which are periodically transmitted by systems in a 802.11 network. The best-effort category refers to applications which do not guarantee a good quality of service, whereas the rest of the access categories, audio and video, differentiate from the others by having higher restrictions regarding latency and throughput. The Access Category Index (ACI) would identify each one of these access categories in the MAC layer, with a magnitude among 0 and 3. Traffic would address to one queue or another depending on this index.

Each type of traffic has an independent access to the environment. Using the appropriate EDCA parameters carries out the traffic differentiation, which is implemented by priorities. These parameters are:

- AIFS (Arbitration Inter-Frame Space). Minimal space of time where the environment must be free.
- Contention window (CW).
- Transmission opportunity (TXOP) limit. Limit of time, which a station has to transmit after accessing to the environment. If its value is zero it means that only a simple MSDU (MAC Service Data Unit) transmission is permitted.

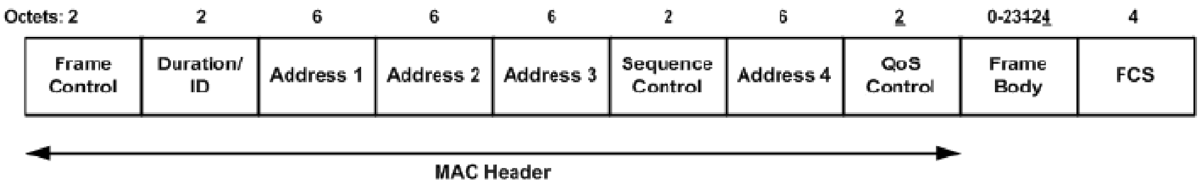

Figure 24- General frame format used for 802.11

The MAC frame format consist of a series of fields organized in a fixed way for all frames. The first three fields (Frame Control, Duration/ID, and Address 1) and the last field (FCS) appear in all types of frames. Address 2, Address 3, Sequence Control, Address 4, QoS Control, and Frame Body fields do not always appear depending on the type of frame.

- Types of frames:
  - o Control. For example:
    - ▪ Request to Send (RTS);
    - ▪ Clear to Send (CTS);

- Acknowledgment (ACK);

o Management. For example:
- Beacons o WAVE Service Announcement (WSA);
- Association Request;
- Association Response;
- Reassociation Request;
- Reassociation Response;
- Probe Response.

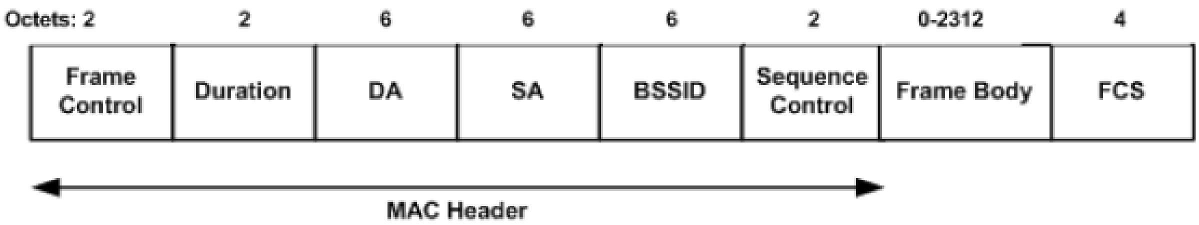

Figure 25- Management frame format

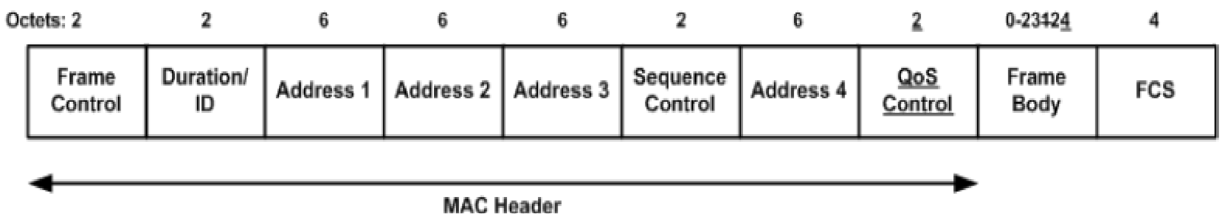

Figure 26 - Data frame format

14.2.3. Level of standardization

Nowadays, the level of the specification standardization is still in draft state. It is important to clarify that this standard is considered as an amendment regarding the technical standards of the original IEEE 802.11 family. The workgroup in charge of carrying out this standardization is the IEEE 802.11p.

The 802.11p Task Group is still active. Per the official IEEE 802.11 Work Plan predictions the approved 802.11p amendment is scheduled to be published in November 2010.

Worldwide, the spectrum regulation for a specific band devoted to road safety and efficiency services is progressing

In Europe, besides the IEEE, standardization and regulation efforts are driven by the C2C-CC, the Communications Architecture for Land Mobile (CALM) working group of the International Organization for Standardization and of the International Telecommunications Union, and the ITS technical committee of the European Telecommunications Standards Institute (ETSI TC ITS), created in late 2007.

The C2C-CC groups vehicle manufactures and automotive equipment suppliers with a threefold mission: to create and establish an open European industry standard for V2V communication systems based on WiFi components and to guarantee European-wide inter-vehicle operability; to enable the development of active and cooperative safety applications by specifying, prototyping and demonstrating this V2V system; to push the harmonization of V2V communication standards worldwide, and to develop realistic deployment strategies and business models to speed up the market penetration. Besides, through the Pre-Drive C2X project, the C2C-CC is estimating the impact on traffic safety and mobility of cooperative systems for road safety and efficiency.

CALM is an international initiative to define and standardize a set of wireless communication protocols and air interfaces for a variety of scenarios in ITS spanning multiple communication modes and transmission methods (e.g., WiFi, cellular). These scenarios include road safety and efficiency.

The ETSI TC ITS receives inputs from relevant European players and initiatives, such as the eSafety Forum and the COMeSafety action. Its general objective is to promote and monitor the implementation of the recommendations identified by the eSafety working group and to support the development, deployment and use of eSafety systems. COMeSafety integrates and harmonizes the use of communication technologies for road safety and efficiency addressed in major European R&D projects, such as CVIS, COOPERS or SAFESPOT. COMeSafety has developed a framework that will be standardized through ETSI TC ITS.

In August 2008, the royalty-free spectrum for ITS systems and services dealing with safety and efficiency was allocated in Europe: 30 MHz at the 5.9-GHz band (5.875GHz – 5.905GHz). At the physical layer (PHY) 10-MHz channels are used.

14.2.4. Future forecast

WAVE is another IEEE standardization project promoted by the Department of Transportation of the United States and by an important number of car manufacturers whose aim is to create a communications network, which allows the information exchange between cars and the road infrastructure.

WAVE shall improve functionality and shall provide benefits related to safety and reliability to future ITS communications.

According next figure the WAVE (Wireless Access Vehicular Environment) protocol stack consist on the IEEE P1609 standards family, which specify the higher system layers, IEEE 802.11p protocol, which describes the system at layers 1 and 2, whereas IEEE P1556 is developing safety.

Going a little more into details, the WAVE protocol stack is the following:

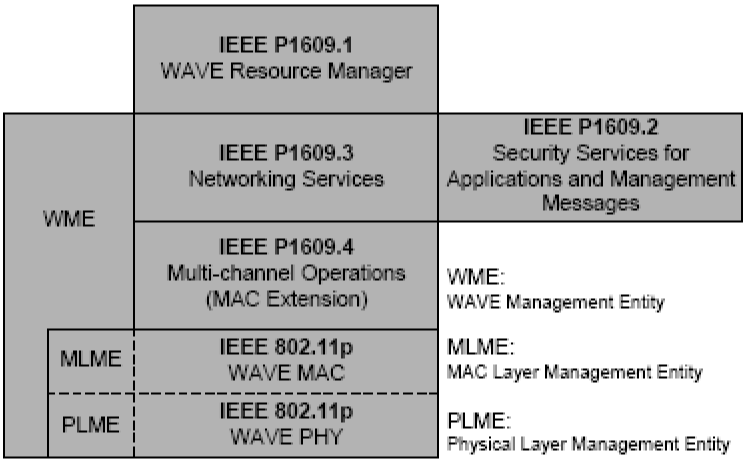

Figure 27 - WAVE protocol stack

- IEEE P1609.1. It describes the application's architecture and interfaces with the lower layers. It also specifies the types of devices that may be supported by the On Board Units (OBUs).
- IEEE P1609.2. It defines messages formats and security process.
- IEEE P1609.3. It defines network and transport layer services.
- Concerning the MAC layer, the IEEE1609.4 standard provides and enhancement of the MAC layer defined in 802.11p to manage operations in a multichannel system. It specially specifies an scheme to coordinate operations between control and service channels by using a global synchronization and taken the UTC as reference.

The most important thing in the vehicular environment is to have highly dynamic network topologies capable to adapt to the fast nodes movement. Due to these characteristics, many of the projects researching into vehicular networks are based on ad-hoc network formation through 802.11 technologies.

14.2.5. Most common uses in the vehicular sector in particular

All use applications in this standard refer to the vehicular sector, since it is its specific use, and they are particularly related to road safety, traffic control, commercial transactions and interesting information while travelling (routes, traffic, directions, etc.).

Companies like Daimler-Chrysler, General Motors, Hitachi, Motorola, Nissan and Toyota among others have participated in the development of this technology contributing with pilot projects applied to their vehicles.

Avis, one of the biggest multinationals in car renting, has planned to offer their clients the possibility to rent vehicles with a Wi-Fi router which would be able to create an 802.11 network and have Internet through 3G networks. This way their clients would be able to access to the Internet from the vehicle or its surroundings by using a laptop or a PDA.

14.2.6. Products / Prototypes

Some automobile manufacturers have started implementing their own solutions based on the standard and some other companies, as Arada Systems, announce products implementing the 802.11p standard.

MARK IV Industries and Tecnocom have developed products addressed to the ITS market. Particularly, these products have been defined as OTTO and MCNU respectively. They cosiest of two communication OBUs based on the 802.11p protocol.

Other companies offer HW platforms along with SW develop kits in order to develop ITS applications.

SHORT RANGE COMMUNICATIONS

Short range communication technologies aim to eliminate cable connections between electronic devices, both portable and fixed, while maintaining high levels of security. The main features of this type of technologies are their reliability, low consumption and low cost and, obviously, their short range. Bluetooth and Zigbee are a couple of examples of well-known and broadly used short range communication technologies.

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15. Bluetooth

15.1. Summary

The next table shows the main characteristics of Bluetoth technology

Description	Bluetooth is a short-range wireless connectivity global standard
Maturity	Mature
Availability	Available worldwide.
Price	Cheap
Bandwidth	723 Kbps (in one direction) + 57.6 Kbps (in opposite direction) / 432.6 Kbps symmetric connection
Frequency	2.4 GHz (unlicensed ISM Band)

Range	10 cm / 10 m /100 m
Application to vehicular environment	Useful to connect on board devices without needing to wire them
Infrastructure requirements	None. Wireless connection between devices
Standardization	Global standardisation available
Use in TeleFOT test site	Yes depends on the application requirements
How to incorporate this technology to TeleFOT test site?	Bluetooth capabilities are available on current commercial terminal would

Table 16 – BLUETOOTH characteristics

Bluetooth is a short-range wireless connectivity global standard. This technology, in many cases, replaces the cables, needed to connect devices: printers, laptops, personal computers, PDAs, keyboards, joysticks, mouse and all the devices can be connected each others, delimitating a short range network called Personal Area Network (PAN).

15.2. Detailed description

The Bluetooth is based on the Service Discovery Protocol (SDP) that makes one Bluetooth device able to determine which services the other devices within the PAN offer. A device implementing the SDP can work both as a server and as a client. In the first case, it can be asked by other devices about the available services. In the second case it enquiries other devices. [BLUT1]

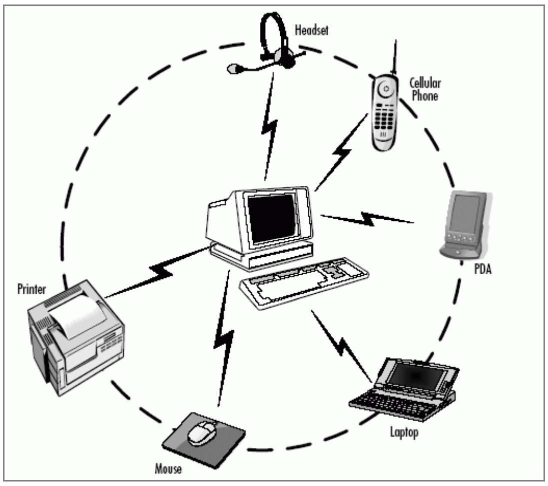

Figure 28- Example of PAN

The above figure shows a typical Bluetooth network. Each device contains information about the services and the protocols it is able to support. Other devices can use this information in order to determine the possible interactions with the network. From the operational point of view the basic functions of the Bluetooth are:

- the enquiry, allowing to know the surrounding devices;
- the discovery and the browsing, for finding services and devices, their settings and requested connectivity parameters.

Other relevant technical features that have driven the Bluetooth development are:

- worldwide working;
- peer-to-peer connection supported;
- both the data and the voice traffic enabled;
- low power consumption;
- small size radio transmitters;
- low cost and large affordability.

The following table summarizes the some additional Bluetooth features complying the requirements above.

Maximum power transmission	1 mW (0 dBm) / 2,5 mW / 100mW
Technology	Spread Spectrum
Duplex	TDD full Duplex transmission (duration of slot 625 μs)
Transmission method	Frequency hopping (1600 Hps, 79 channels, each 1 MHz wide)
Modulation	GFSK
Maximum nr. of voice channels	3 per piconet
Maximum nr. of data channels	7 per piconet
Number of supported devices	8 per piconet, 10 piconet in the covered area
Security	Encryption and authentication request
Voltage	0 volt / 3 volt
Module average cost	5 $
Module size	About 30 mm^2

Table 17 – BLUETOOTH additional characteristics

The frequency band is 2.4 GHz in the ISM band. In the USA and in the most part of Europe the frequency of 83.5 MHz is allowed and used to allocate 79 RF channels, each 1 MHz wide. This band is free and interference problems can occur.

The equipment can belong to one of the 3 power classes:

- Class 1: wide range devices (up to 100 meters) and maximum transmission power of 20 dBm (100 mW).
- Class 2: ordinary distances devices (10 meters) and maximum transmission power of 4dBm (2.5 mW).
- Class 3: short range devices (10 cm) and transmission power of 0 dBm (1 mW).

The frequency hopping technique requires that the Bluetooth devices that want to communicate have to know and follow the same hopping sequence. The Bluetooth devices can work in 2 different ways:

- Master
- Slave

Among master and slave devices, there are not differences, neither at the circuit nor at hardware level. The master decides the hopping sequence. The slaves that want to communicate with a master have to synchronize time and frequency on the master hopping sequence. A group of slaves working together and synchronized with the same master forms the so called *Piconet.* One piconet may include one single master and maximum 7 active slaves, where active slave means one unit keeping synchronized with the piconet master. The Bluetooth standard includes some operative modes allowing to virtually extending the range of one piconet. Some devices, in order to limit the power consumption, can temporarily dissociate from the piconet, but periodically receiving information from the master, and being able to actively come back anytime in the piconet, if there is room. [BLUT2]

Besides controlling the hopping sequence, the master dominates the transmission that means it decides when the devices of its own piconet can transmit. With regard to this, Bluetooth distinguishes 2 kind of traffic:

- Voice
- Data

The master decides when the slaves can transmit, assigning some slots to the voice traffic and some others to the data traffic. Regarding the data traffic, one slave can only transmit a reply to one dedicated master transmission; in particular, if the master, during a data slot, transmits one packet to one specific slave, next slot is automatically kept for a possible data transmission of that specific slave, regardless of the presence of data to transmit.

In the voice traffic, any slave transmits within slots reserved to the master, regardless they have been recipients of the previous transmissions. Anyway the communication always occurs between one master and one slave, so that whenever one slave sends a packet to another one from within the same piconet, it transits through the master device. This time slot division between 2 devices is called Time Division Duplex (TDD).

Since the number of active slaves composing a piconet is maximum 7, it is possible extend the coverage making a bigger network, built by more piconets linked together. This kind of network is called *Scatternet*. [BLUT3]

The connection between one master device and one slave device is called *link*, and Bluetooth defines 2 different types:

- Asynchronous services or ACL (Asynchronous Connection-Less): this link exists when a connection between one master and one slave has been established. One master can have an undetermined number of ACL links with many slaves, but only one per slave. An ACL link sets a packet switching communication between master and slave: the packets are sporadically exchanged, i.e. when there are data to send from the Bluetooth stack higher levels. Per each slot, the master chooses which slave must send or receive, and this makes possible both asynchronous services and short-time services. Moreover, special types of packets exploit the retransmission schemas and the error control techniques, in order to improve the transmission efficacy. A slave can reply during the following slot using an ACL packet exclusively if it has been directly addressed. Due to the peculiarity of the RF transmission, all the slaves under the range of one master listen its transmissions. If one listening slave fails its own address decoding in a given packet, it is not authorized to use the following slot for sending. A further option is the broadcast transmission of packets that can be receives from all the listening slaves.
- Synchronous services, or SCO (Synchronous Connection-Oriented): this is a symmetric link among master and slave, reserved frequency and periodic data exchange as dedicated slots. A SCO sets a circuit switching between master and slave. One master can support up to 3 SCO links with 3 different slaves at once. One slave can support 3 SCO links with only one or two different masters. The packets sent by a SCO link can never be retransmitted. One slave can always reply to a SCO packet, within the dedicated slot, also if its address has not been decoded. Each device will have to schedule the ACL traffic, in order to respect the slot dedicated to the SCO traffic.

16. Zigbee

16.1. Summary

The following table shows the main characteristics of Zigbee:

Description	ZigBee is a technology associated to wireless personal area network (WPAN) and, specifically, to sensor nets, due to its specific characteristic of low energetic consumption and low implementation cost.
Maturity	Mature
Availability	Available everywhere
Price	Around 10 euros
Bandwidth	250kbps at 2.4GHz, 40kbps at 915MHz and 20kbps at

	868MHz.
Frecuency	868 MHz in Europe, 915 MHz in USA y 2,4 GHz everywhere
Coverage	10-75m
Application to vehicular environment	Sensor networks
Infrastructure requirements	Usually used in mesh network so it would be necessary a gateway to centralize communications.
Legal issues	Solved
Standarization	IEEE 802.15.4
Use in TeleFOT test site	NO
How to incorporate this technology to TeleFOT test site?	Sensor wireless networks inside and outside the car

Table 18 – ZIGBEE

16.2. Detailed description

ZigBee is a technology associated to wireless personal area network (WPAN) and, specifically, to sensor nets, due to its specific characteristic of low energetic consumption and low implementation cost. [ZGB1]

16.2.1. Technical characteristics

ZigBee works with several ISM frequency bands according to which its transfer speed is set:

Frequency	***Band***	***Scope of application***	***Binary speed***	***Number of channels.***
2.4 GHz	ISM	Global	250 Kbps	16
868 MHz	ISM	Europe	20 Kbps	1
915 MHz	ISM	America, Australia	40 Kbps	10

Table 19 – ZigBee Frequencies

Its maximum range varies between 10 – 100 m, depending on the environment. ZigBee has been optimized with a very improved radio design to get low costs, which makes it adequate for industrial uses. Its objective is applications that require safe communications with a low data-sending rate and maximization of the useful life of its batteries. Its principal characteristics are: low consumption (slave nodes scheme in sleep mode), mesh network topology (or star/tree network) and easy integration (nodes can be created with few electronic resources).

ZigBee uses physical level and access level (MAC) defined in the IEEE 802.15.4, whereas the rest of layers from the protocol stack are defined by ZigBee Alliance. ZigBee Alliance is a non-profitable organization, with more than 100 companies, most of them semiconductor manufacturers, that has the objective of promoting ZigBee's development and implementation as a low cost wireless technology. Standing out companies such as Invensys, Mitsubishi, Honeywell, Philips and Motorola work on the establishment of a standard system of communication, via radio and bi-directional, applicable to domotic devices, building automation (immotic), industrial control, PC peripheral, toy business, medical sensors etc. This alliance's members justify this standard development to fill the gap existing underneath Bluetooth regarding transmission capacity and low consumption since Bluetooth standard is unable to be an optimum election for sensor nets.

ZigBee allows implementing several network topologies: star (the most common one), mesh and cluster tree, although no device in the market is known which is able to support this last one. From a functional point of view, ZigBee nodes may be active (FFD: Frequency Division Duplexing), that is, with full functionality, or limited capacities. The active ones normally operate as net coordinator or traffic routing nodes (wide networks), while the passives one, due to their simplicity and reduced cost, work normally as sensors/actuators.

This way, ZigBee, compared with other wireless technologies, increases the range of devices connected to the same network. Every routing node may be in charge of up to 255 terminal nodes, and at the same time, all the routing nodes may be part of the network depending on the coordinating node which administers the interactions with the rest of nodes up to 65,536 theoretical nodes. Most of the time, the transceiver of these terminal nodes is slept in order to save consume less energy than other wireless technologies. However, the routing nodes must be always awake which may cause disequilibrium in the different network terminals' consume. ZigBee technology is mostly used in the home wireless automation or domotic.

Range	10-75 m
Modulation	B-PSK; OQPSK
Speed of transmission	Up to 250 kbit/s
Power of transmission	1 mW
Batteries life	From 100 to 1000 days
Safety	AES algorithm

Table 20 – ZigBee Characteristics summary

16.2.2. Standardization

Similarly to Bluetooth, the way ZigBee works is also based in a profile concept. The present problem is that not all the profiles proposed in the standard are defined and approved yet, and therefore, there is no device counting with all the advantages that this technology theoretically presents.

Actually, most of devices named as ZigBee that are nowadays in the market are in fact implementations of standard 802.15.4 which do not follow the profiles defined by ZigBee Alliance but others with a proprietary character defined by the manufacturers themselves.

IEEE 802.15.4 is the standard on which ZigBee's physical and MAC layers are based. Its most important characteristics are: network flexibility, low cost and low energy consumption, although the data transmission capacity is not very big. Some of the IEEE 802.15.4 characteristics are described in the following table:

Frequency bands and data transmission range	868 MHz: 20 kb/s; 915 MHz: 40 kb/s; 2.4 GHz: 250 kb/s.
Latency	less than 15 ms.
Channels	868/915 MHz: 1/10 channels. 2.4 GHz: 16 channels.
Addressing modes	Every chip has the 64 IEEE bits addressing modes
Access channel	CSMA-CA
Security	128 AES
Network	Up to 2^{16} devices
Temperature range	From -40° to +85° C

Table 21 – ZIGBEE aditional characteristics

Network synchronism is obtained by beacon frames that keep all nodes synchronized without listening to the communication channel continuously and optimizing this way its energetic consumption.

16.2.3. Application to the vehicular environment

Using as basis the 802.15.4 in its physical and MAC layers, different implementations which doesn't exactly follow the Zigbee protocol in its superior layers have been developed, however they have permitted uses where consumption, latency and improvement of technology flexibility have been remarkable permitting the building of auto-configurable mesh or tree cluster networks fed with AA batteries that last up to one year and adding the necessary elements to introduce actuators in the typical sensor networks. Their most common uses are as sensor networks easy to install and auto-configurable especially on the outside.

Nowadays, in the ZigBee Alliance there are a few tens of certified products and platforms. It is predicted the development of more profiles which allow the suitable standard application in all those fields where it may be useful such as:

- Building control and monitorization: access control, environmental conditions, security, illumination control;
- People/patients monitoring;
- Electronic of consumption: remote control, TV, DVD;
- Home's control: access, watering, illumination, heating;
- Internet of things: information services, interaction with objects.

This technology will be used in the vehicular environment for data transmission of sensors distributed inside de car and even in its environment making its use feasible for the exchange of small information with the context for control, security and maintenance applications.

Nowadays, there are already several commercial solutions of on-board equipment designed for the vehicular environment that count with Zigbee interface. They are usually fleet control or data acquisition equipments that communicate to each other by using Zigbee with an additional sensors network, which measures vehicle's parameters. It is reasonable to think about spreading this use to other on-board control applications.

17. NFC - Near Field Communication

17.1. Summary

The following table shows the main characteristics of NFC:

Description	Near Field Communication (NFC) is a short-range wireless communication technology designed for a simple, intuitive and safe communication among electronic devices.
Maturity	Mature
Availability	Available only on a limited number of devices
Price	Around 2 euros
Bandwidth	106, 212 and 424 Kbps.
Frecuency	13,56 MHz
Coverage	Up to 20 cm

Application to vehicular environment	Communications between devices inside the vehicle such as identification, Electronic key, electronic payments
Infrastructure requirements	Reader device.
Legal issues	Solved
Standarization	NFCIP-1
Use in TeleFOT test site	NO
How to incorporate this technology to TeleFOT test site?	Electronic payments, driver identification, access control…

Table 22 – NFC

17.2. Detailed description

Near Field Communication (NFC) is a short-range wireless communication technology designed for a simple, intuitive and safe communication among electronic devices.

17.2.1. Technical characteristics

NFC works in a 13.56MHz frequency band (available globally without licence) and works through an induction process of magnetic fields. You can obtain a transfer rate up to 424 Kb/s. It offers an intuitive, simple and safe way of communication among electronic devices, both in reading and writing way. [NFC1]

Communication among two devices with enable NFC happens when these devices are in a distance between 0 and 20 cm. Communication is possible with just a "touch" between them in order to make an electronic payment, identify an element, control an access or to activate and make easy the configuration and initialization from another communication channels with bigger capacity such as Wi-Fi or Bluetooth.

Given that the transmission level has a very short-range, transactions based on NFC are totally safe in an inherent way. This situation of extreme proximity provides the user a feeling of process control that is absent in other kinds of communication.

It is expected that it could be used by an enormous variety of devices such as cell phones, PDAs, laptops and digital cameras given the great acceptance from the manufacturers and the telecommunication and industrial operators from the sector in general.

The possibilities of this technology are very wide. As an example we can mention: toll payment, access control in sport and spare time events, use of the device as an integrated key (homes, cars, hotel parkings…).

17.2.2. Standardization

NFC technology is an open technology platform, developed by Philips and Sony and standardized in ISO 18092, ISO 21481, ECMA (340, 352 AND 356) and ETSI TS 102 190.

- ISO/IEC 18092/ECMA-340/ETSI TS 102 190 (Near Field Communication Interface and Protocol-1 (NFCIP-1)): this standard defines the passive and active communication modes for the interface and NFC protocol. It specifies modulation, codification, speed of transmission and format schemes of the RF interface frames but also initialization schemes and conditions required for the control of data collision in the initialization phase.
- ISO/IEC 21481/ECMA-352/ETSI TS 102 312 (Near Field Communication Interface and Protocol-2 (NFCIP-2)): this standard defines the mechanism needed to detect and select one out of the three working modes available with a frequency of 13,56 MHz, according to NDCIP-1 standard: NFC, PCD (Proximity Coupling Device) y VCD respectively (Vicinity Coupling Device).

The NFC Forum association (founded by NXP, Sony and Nokia) also promotes the use and standardization of the NFC technology pursuing the compatibility among the various manufacturers. Nowadays, the NFC Forum is made up of more than 130 members.

17.2.3. Application to the vehicular environment

NFC is mostly used for ticketing and automatic payment applications, in a short range of 20cm at maximum. This range limits in a certain way the applications where NFC could participate in a vehicular environment. Some practical examples of use are: vehicular identification or identification of some of its components to carry out an access control or a diagnosis, use of the NFC mobile terminal such as vehicle electronic key or access to available infrastructures in certain points of the road network (information points, rest areas, toll points…).

18. UWB - Ultra Wide Band

18.1. Summary

Description	UWB-Ultra-wideband communications is fundamentally different from all other communication techniques ,it is based on the use of short-time pulses (sub-nanosecond) to communicate between transmitters and receivers. The short pulses generate very wide bandwidth (GHz) in frequency domain.
Maturity	Not yet
Availability	Not yet (mainly for regulatory issues)
Price	NA-however transceiver technology is very simple :it will require low cost
Bandwidth	Very high: however to protect the existing radio services the emission mask is positioned between 3-10 Ghz for commercial applications
Frecuency	Is carrier less
Coverage	10-200m
Application to vehicular environment	To be used in automotive industries for collision avoidance road assistance
Infrastructure requirements	UWB is a carrier less system ,it is not necessary to have transmitter block and receiver recovery stage. The front end of a UWB system is very simple and cheaper
Legal issues	Not Solved; European authorities are considering these aspects already defined (with several limitations) in USA and Canada. Only allowed to be used indoors
Standarization	IEEE 802.15.3a
Use in TeleFOT test site	NO
How to incorporate this technology to TeleFOT test site?	To be evaluated considering the legal restrictions

Table 23 - UWB

Wireless communication is steeply growing, but the radio frequencies are limited and they are quickly running out. Several techniques have been studied to solve this problem, like the Ultra Wide Band technology (UWB) that enables high speed transmissions and low power, using an extremely wide band. The UWB uses an accurate timing system in order to emit electromagnetic pulses in pseudo-random time slots. In this way, the UWB signal looks like a noise, making the system safe and hard to decode, and overcoming many interferences problems, typical of the traditional wireless communications. [UBW1]

18.2. Detailed description

A UWB system can be employed for setting up high-rate communications within high performance multi-users networks and in geo-located applications using very precise accuracy (up to 1 cm). As a consequence of the robustness in the multi-path, these systems are suitable for wireless indoor applications, for example, to create networks in an "ad hoc" solution. Moreover, the particular used waveform makes easy implementing multiple access time-slicing techniques. The used power is extremely low - and by consequence the level of interceptions - making the UWB especially suitable for military applications.

Among the potential applications, the GPR (Ground Penetrating Radar) allows to reveal an identify targets hidden behind obstacles and barriers (like foliage or walls) or underground. This means that human target can be revealed as well, in urban contexts or in buildings, at a great distance (e.g. a pedestrian on the zebra crossing at hundreds of meters).

Studies carried out under FCC (Federal Communications Commission) report high performances in indoor context: a real data rate of 100 Mbps (Intel) (up to 450 meters are possible), comparing with the 802.11g frequencies that offer a data rate of 54 Mbps. In few years UWB data rate will be reached up to 1 Gbps.

This speed get the wireless communication near the most common wired LAN (Fast Ethernet), and in the future will reach higher performance in terms of mobility, flexibility and scalability in wireless network.

The most part of the UWB devices are aimed to military and research scopes, although the market offers some products enabling higher distance communication.

A significant application of this technology is the road safety (communications among vehicles and traffic alerts). The functional core is a very small UWB radar, working in a 500 MHz C-band; the power peak is 0.2 W (average power 4 μW) and the range is 30 meters in presence of other vehicles; the accuracy overcome is of 1 m. Unlike the RF used in road communication that uses low power (peak 0.3 W, average power 500 μW) and 250 MHz in L-band to reach the throughput of 128 kb/s in a range higher than 200m, the UWB offers better precision and multipath immunity.

BROADCAST COMMUNICATIONS

The Broadcast communication technologies Broadcast, are those technologies used for the broadcast transmission, that is, they are used for point to multipoint communications, where a sending node sends some kind of information to a multitude of receiving nodes simultaneously, without reproducing the same transmission node by node. There are several communication technologies that can be used as broadcast communications depending, among others, on the type of information to be transmitted. Some of these technologies are defined below.

19. RDS-TMC (Radio Data System-Traffic Message Channel)

19.1. Summary

Description	The Traffic Message Channel (TMC) is a technology for providing traffic and travel delay information to drivers. The complementary Radio Data System (RDS) broadcasts digital information carrying TMC updates via FM radio waves.
Maturity	Mature
Availability	Available all over Europe
Bandwidth	Extreamely low
Frecuency	80 – 104MHz
Application to vehicular environment	Can be used for invehicle traffic messages broadcasting
Infrastructure requirements	Receiver device with RDS- TMC decoder
Legal issues	Solved
Use in TeleFOT test site	Not foreseen but can be used

19.2. Detailed description

The RDS (Radio Data System) is an open technology developed by members of the EBU (European Broadcasting Union) in the last 25 years. Since it was developed up to date, its use has increased incrementally, especially within services TTI (Traffic and travel information) under the use of RDS-TMC.

RDS-TMC service is used for the dissemination of Traffic information via radio, TTI (Traffic and Travel Information). It was developed under the TMC-Forum and the TISA (Traveller Information Services Association), with the aim of standardizing the distribution of traffic information in Europe. [RDSTMC1]

As we shall see later for the dissemination of TMC messages (Traffic Message Channel) RDS (Radio Data System) technology will be used. The messages will contain properly coded information, basically the location of the events and its descriptions. The set of points of interest of the road network will form the location table.

19.2.1. Technical overview

RDS (Radio Data System) uses the FM subcarrier for sending information, such as identification of the transmitting station, the frequency in which it is transmitting or the frequencies of the closest repeater stations to ensure that the receiver is tuned into the most powerful station of all those that provide coverage to a certain point. [RDSTMC2]

TMC (Traffic Message Channel) is based on the use of the transmission capacity of the RDS to broadcast information messages on the traffic situation. The messages are encoded and transmitted digitally along with the FM broadcast, decoded and interpreted in the vehicle by the car radio device and presented to the driver via a voice synthesizer or a display panel.

RDS is, therefore, the mechanism by which data packets are sent to a receiver through the sub-carrier frequency modulation. On this very low capacity channel the system can transmit digitally encoded TMC informational messages about:

- The nature, severity and evolution of the traffic problems that may arise, both urban and intercity
- Aid to navigation by means of the recommendation of alternative itinerary when conflict situations appear on existing or planned routes.
- Additional data about the area in which the car is circulating, as well as tourist information.

To issue a TMC message, the traffic information system collects data from various sources that provide information on traffic incidents (police, control, road sensors, etc). The information is collected at the station, either directly from sources or through the organization responsible.

Upon reception of this information in the radio station, the information is filtered and only the relevant information for the area of coverage of each station or repeater is encoded and broadcasted. The traffic information properly coded is embedded into the signal emitted by radio station and broadcasted together with the FM signal.

A vehicle with a car radio system able to receive RDS-TMC will decode and translate the message and provide it to the driver via a small message board, the radio display or a through a voice synthesizer.

Thus, TMC can provide the driver much more information than the information provided using a RDS channel in a “conventional” way.

The messages are repeated every 5 minutes. To decide which information is transmitted in a cycle, each message is assigned a priority parameter. An urgent message can be sent immediately while long-term events can be included or not in a cycle according to the space available.

19.2.1.1 TMC message structure

A standard TMC message provides the following information:

- Event Description (11 bits), corresponding to the code phrase that represents the traffic information, weather, general information
- Location: (16 bits) indicates the area, road segment or the exact point of the event.
- Address and Extension (4 bits) indicates the direction and length of the affected segment (e.g. queue length).
- Duration (3 bits) indicates the estimated duration of the event, 8 different values that can be taken from various tables that include various sizes of "grain" (from the quarter of hour to the whole day).
- Alternative itinerary (1 bit) indicates whether or not advised drivers to use an alternative route default.

A part from this information there is some other implicit information that the decoder can deduce from the code received. This implicit information is stored in the vehicle equipment and consists of:

- Type and number of road
- Road segment
- Area, region and country
- Pre-assigned alternative route
- Urgency of presentation: extreme, normal or low
- Number of affected direction: only the direction of movement or both.
- Duration: Determines whether the event is going to evolve or will have long life.
- Status: kind of sentence that must be presented (information, warning, danger, or no show)

19.2.1.2 TMC message codification

The RDS-TMC uses an intensive codification scheme that optimizes the use of small data channel available. The scheme is based on the definition of a list of verbal descriptions of common traffic situations and verbal descriptions of the locations where these situations occur (regions, junctions, roads, towns, or relevant points) each of which correspond to a single code. The codes also correspond to phrases that describe the estimated duration of events or indicators of the magnitude of the events.

The sentences or encrypted messages correspond to seven different categories:

- Traffic Situation
- State Highway

- Weather conditions
- Causes
- Effects
- News
- Recommendations

In addition to these messages the TMC-RDS messages also contain information on the quantifiers of the previous events or situations including:

- Length
- Speed
- Time
- Temperature
- Weight

The code tables are used both by the transmitter and the receiver to transform the information coded in the corresponding sentences and information and vice versa. The use of codes offers some advantages such as that the messages are independent of the language used. Therefore the system can be used throughout Europe and the messages are presented to the driver always in the native language.

19.2.1.3 The Geographic data base

RDS-TMC system requires the development of geographic databases encoding the different locations that identify where an event has occurred. Codification these locations means assigning 16 bits codes to each of them

In a country can exist up to 64 different databases each containing up to 65,536 locations. Different databases are used to provide detailed local and regional coverage in urban environments, while locations include national road network and the main European corridors:

- Local: This basic level includes urban and suburban locations. The information at this level will only be broadcasted on local stations covering the area where information is affecting. The information will only be interpreted correctly if the vehicle device contains the appropriate data base for the decoding.
- Regional level: This level includes the locations referred to primary and secondary roads or regional level. Information will be provided to vehicles moving around the region concerned.
- National level: This level includes those events in meaningful ways for the long-distance movement. This information will be disseminated, in principle, throughout the country concerned. All national vehicles receiving this message should be able to decode the location codes of this category.
- International level: this level includes events in major link roads between different countries. The messages referred to these locations will be broadcast both in the country where the event occurred as the neighbor countries. All European vehicles should be able to decode these messages.

Each of the different levels of information include location names. These can be classified into:

- Country
- Region
- Areas
- Road segment
- Specific locations

There are two different ways to perform the encoding of locations, one tries to use some internal structure in the representation of 16 bits (65,536 codes) and another one in which there is no internal structure.

19.2.1.4 The RDS-TMC protocol

The transmission of TMC messages is divided into groups of 104 bits. Each group is composed by 4 blocks of 26 bits, 10 of which are aimed at checking and error handling. A RDS group is the minimum data packet that can be defined. The signal is transmitted using RDS subcarrier of 57 KHz. a transmission rate of 1187.5 baud, equivalent to 11.4 groups per second. In practice, however, it is not possible to transmit more than 15 to 30 messages per minute, due to the sharing of the RDS signal with other functions.

To ensure the correct transmission of information through a noisy channel, two identical copies of the message are sent by the transmitter. Besides, a message will only be considered as correct if two identical copies of the same message are received consecutively.

The RDS-TMC communication protocol defines the following transmission supports:

- Transmitter information, including the number of location database being used, program identification codes or other TMC stations that overlap with the transmitter.
- Information cycle: defines the parameters of the TMC message, including information about the messages that are being transmitted for the last time in the cycle.
- Information of message shows the total number of messages in the cycle and new messages to be included.
- Message repetition
- Update message
- Delete the message
- Control codes

20. DAB - Digital Audio Broadcasting

20.1. Summary

Description	Digital Audio Broadcasting (DAB) is a one-way service which could be used to send data from a central infrastructure point to vehicles or to out-stationed parts of the infrastructure. Currently, coverage of the European landmass is not continuous, and there are relatively few vehicles provided with DAB capability.
Maturity	Mature
Availability	Available worldwide through satellite coverage
Bandwidth	128Kbps
Frecuency	Band III - 174-230MHz L Band - 1452–1492 MHz
Application to vehicular environment	Can be used for invehicle audio broadcasting
Infrastructure requirements	Receiver device.
Legal issues	Solved
Standarization	Eureka 147 DAB
Use in TeleFOT test site	Not foreseen but can be used

Table 24 – DAB

20.2. Detailed description

The Digital Audio Broadcasting (DAB) System developed within the Eureka 147 Project and standardized by the European Telecommunications Standards Institute has become a world standard. It is currently available to over 300 million people in almost 40 countries, with over 600 different services, and these numbers are increasing daily. Transmissions are in the 217.5 to 230 MHz sub-band of VHF Band III (170–230 MHz); and 1452 to 1492 MHz, at the upper end of the UHF L-Band (500–1500 MHz). [DAB]

DAB can provide in an efficient way a high quality multiservice digital broadcasting, for mobile, portable and fixed receivers using only non-directional antenna. It can operate at any frequency between 30 MHz and 3 GHz for mobile receivers and can be used in terrestrial, satellite, hybrid (satellite with complement terrestrial) and cable broadcasting.

The DAB system can be used in flexible manner, and is able to be used with different transmission speeds and allow to digitally multiplex many types of sources and channels with different encoding options for the data and the associated services.

DAB uses COFDM (Coded Orthogonal Frequency Division Multiplex) technology to convert music or speech from an analogue signal into digital (binary) code. This dramatically reduces the likelihood of transmissions being corrupted by weather conditions and other problems that can degrade reception quality. In addition, MPEG I layer 2 ('MP2') compression technologies are used to reduce the data rate required for each stream.

Broadcasts are originated from transmitters providing groups of stations on a single frequency, known as multiplexes or 'ensembles', and in a given area, several different ensembles may be able to be received. In the UK, for example, you can generally receive a BBC National multiplex, a national commercial multiplex and a local ensemble.

20.2.1. Technical Overview of the DAB System

DAB is an extremely robust transmission medium, almost completely eliminating problems with multipath, hiss and fading that can disrupt the enjoyment of analogue broadcasts. It also allows broadcasters to transmit more stations in the same bandwidth as one FM station. As well as music and speech, DAB stations can additionally carry data that can be displayed by the receiver while listening to the station.

Some of the advantages of DAB are as follows:

- Efficient use of the spectrum and power. It is achieved by the use of SFN (Single Frequency Network), a special frequency reuse technique. This allows extending the broadcast networks virtually without limit.
- Improved reception. The information transmitted is distributed in both time domain and frequency domain. This allows to remove the effects of channel distortion and attenuation from the signal received at the receiver, even when working under conditions of strong multipath (due the reflection of buildings and mountains). To achieve this, the signals are encoded and multiplexed with OFDM (Orthogonal Frequency Division Multiplexing). To protect the signal transmission errors the system uses two techniques called UEP and EEP (Unequal / Equal Error Protection).
- Quality of sound. We may achieve a quality equivalent to that of a CD thanks to the MPEG Layer II Audio (also known as MUSICAM). This system exploits the masking effect that occurs due to the psychoacoustic characteristics of human hearing, as it is unable to perceive all the sounds present in a given time, and is therefore not necessary to transmit the sounds that are not audible. This eliminates redundant information.
- Flexibility. The services can be structured and configured dynamically. For example, a radio station during a program where you can issue debate or dialogue using a low speed (with 64 or 96 kbps is sufficient), occupying a low bandwidth, while at other times may emit higher speeds stereo audio (128 or 192 kbps) and therefore more bandwidth.
- Data services. Along with the audio signal further information can be transmitted:
    - Information channel. This channel carries the configuration of the multiplex used, service information, date and time, traffic information, emergency announcements, etc.
    - Program Associated Data (PAD). It contain information directly related with the audio programs, music titles, author, text of songs in several

languages, etc. The capability of the PAD is variable (minimum of 667 bit / s with MPEG-1 or 333 bit / s with MPEG-2)

- Additional services. For instance, images and texts sent to electronic boards, including video.

It seems that the DAB-TMC inherits some of the limitations of the RDS-TMC. Although it solves many of the capacity problems, still works with location codes, suggesting that a breakthrough is just temporary, pending a final different solution. For this reason it is committed to the TPEG, which will provide references to locations instantly. RDS-TMC and DAB, collection and subsequent processing of data is much more complicated and expensive than public distribution. TPEG draw the valuable characteristics of the RDS-TMC, eliminating their major problems. (See following subsections for further information on TPEG)

20.2.1.1 Transmission coding and multiplexing

In order to transmit the data for individual services (understanding as data either audio-based or multimedia data), it must be combined into a single data stream. This process is known as multiplexing, and the resulting data stream is called the multiplex.

The frame-based DAB multiplex (shown in the following figure) comprises three different elements:

- The **Synchronization Channel** which conveys reference frequency and timing information to allow receivers to synchronize to and decode the received DAB signals.
- The **Fast Information Channel** (FIC), which carries information describing the composition of the multiplex and informs receivers how to extract and decode the information for individual services.

- The **Main Service Channel** (MSC) contains the audio frames or data packets corresponding to the different services within the multiplex. This part of the multiplex is essentially the useful payload of the DAB signal.

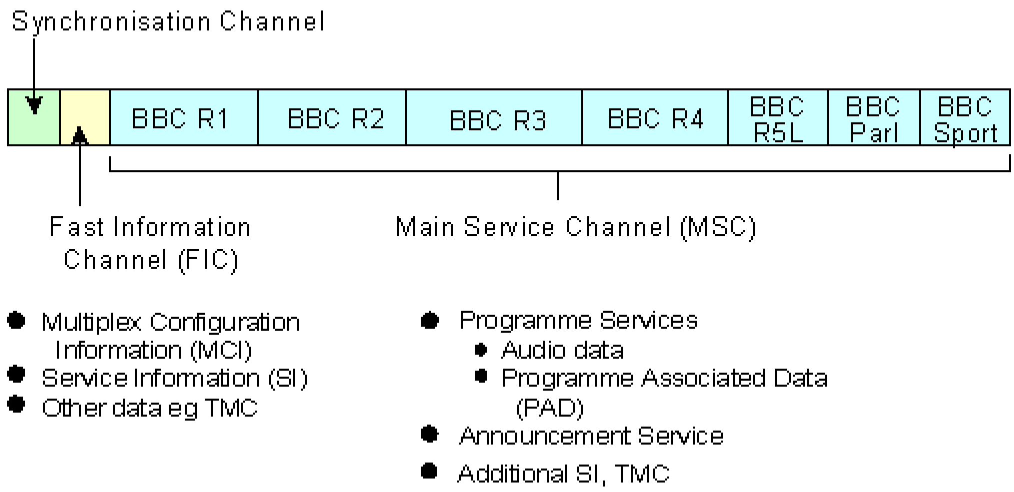

Figure 29- DAB Multiplex Frame Structure.

Both the organization and the length of a transmission frame depend on the transmission mode. FIC and MSC channels respectively are arranged in blocks (FIBS) and frames (CIF), which provide independent data packet mode transmission (see the following figure).

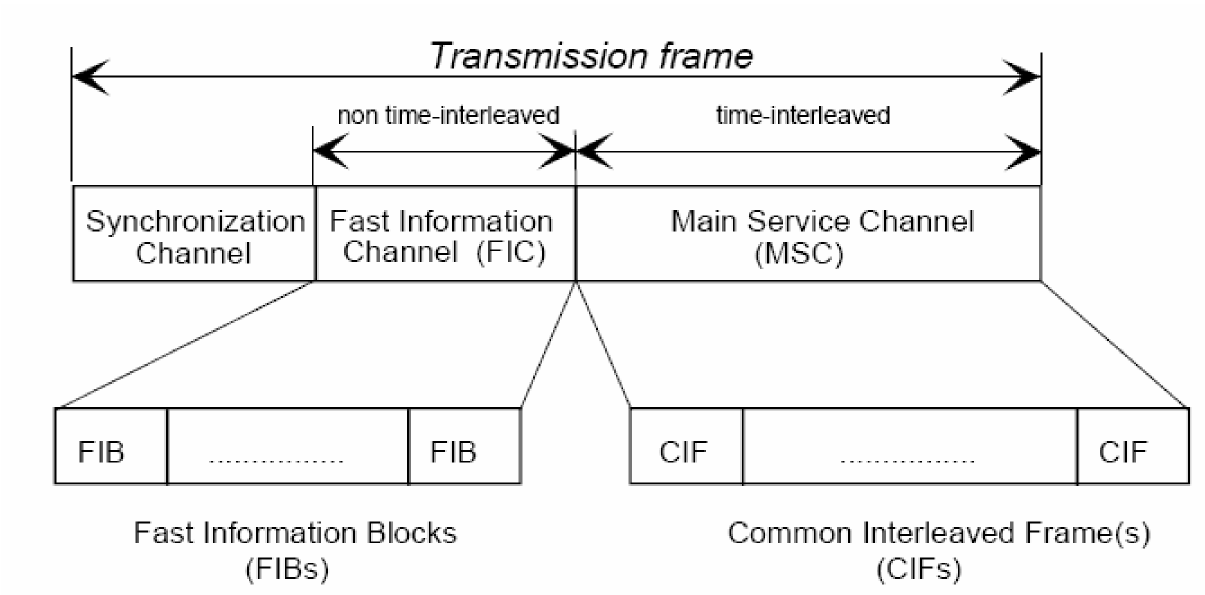

Figure 30- DAB Frame Structure.

The FIC channel is a low speed channel (4Kbps), which contains information that can be acquired quickly by the receiver. This channel includes the Multiplex Configuration Information, which describes the different channels or services (audio and/or data) that are transmitted in the main channel or MSC. Additionally, and optionally it may include Service Information - which describes the different content broadcasted (title, etc.). -

conditional access information and several low speed data services that can be transmitted on a channel named FIDC (Fast Information Data Channel) which may or may not be encrypted.

The FIC channel consists of a number of Fast Information Blocks that carry the information. Each FIB is composed by 32 bytes: 2 bytes are used for CRC error detection, the remaining 30 bytes form the FIG (Fast Information Group) in which coding for the multiplex configuration information (MCI) and other information. The FIG's are distinguished by a field "type". The FIG's type 0 are used for the MCI and service information (SI), the FIG's type 1 are used to send text labels, the FIG's type 5 are used to send data in general and FIG FIDC's type 6 are used for control systems access.

The following figure shows the FIC channel structure

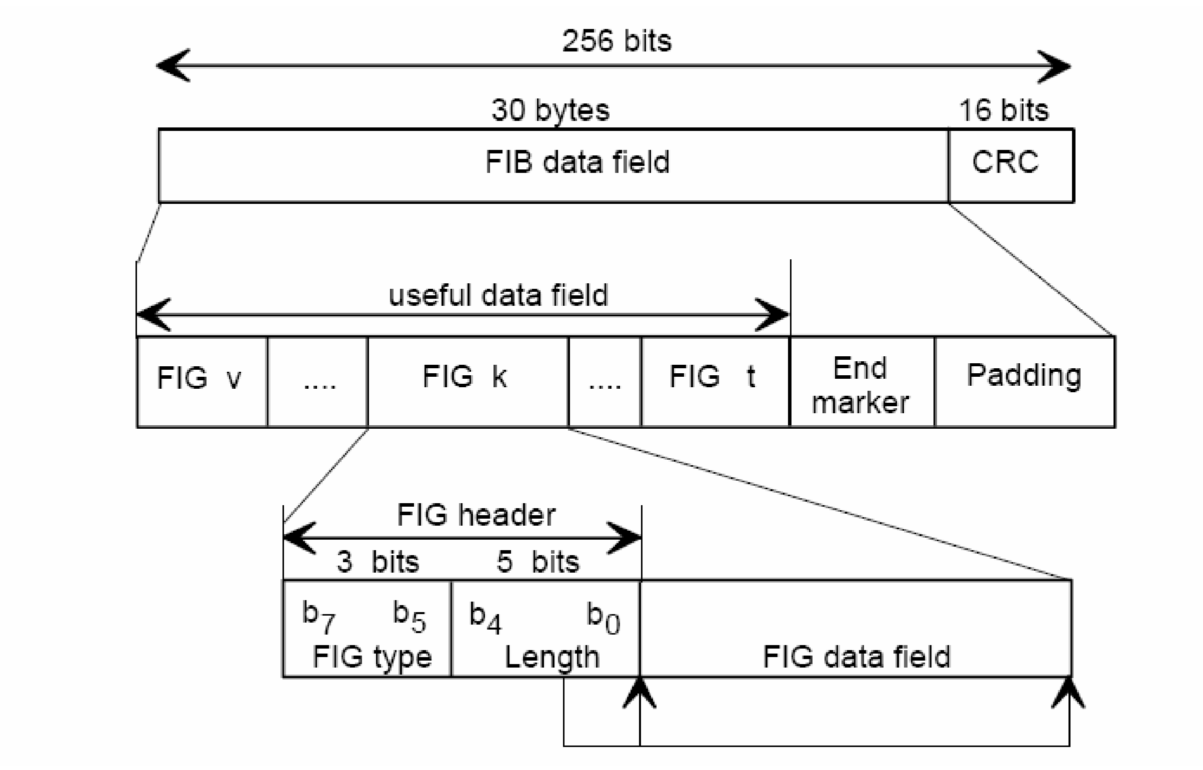

Figure 31- FIC channel structure.

The MSC channel carries the different service components (audio and data) as time-multiplexed subchannels. The MSC is composed by 55296 bits logical frames, which are 24ms long, these frames are called CIF, and provide a bit rate of 2.3 Mbps, which includes all audio and data services, as well as redundancy (convolutional coding) introduced for the correction of errors.

The MSC can be divided into a variable number of sub-channels, up to 64. In each sub-channel can carry an audio channel or a data channel, and each one of the sub-chennels conforming the MSC can have a different bit rate depending on the quality of service needed for each service.

Each subchannel contains some redundancy information (convolutional coding) used for the error correction. The error correction technique can be EEP (Equal Error Protection) or UEP (Unequal Error Protection). Different profile protection can be selected depending on the type of service transmitted. After applying the correction errors, the total bit rate 2.3 Mbps, is reduced to a useful bit rate ranging 0.6 to 1.7 Mbps, depending on the level of protection.

Convolutional forward-error-correction (FEC) channel coding, time interleaving and frequency interleaving are applied to the multiplex data to provide strong protection against bit-errors, so that the services decoded at the receiver are quasi-error-free even when the received signal is impaired.

20.2.1.2 COFDM MODULATION

DAB uses the spectrally-efficient multi-carrier digital modulation scheme, Coded Orthogonal Frequency Division Multiplexing (COFDM). Instead of a having a single digitally modulated carrier with a very high symbol rate, COFDM uses many carriers -- up to 1536, spaced at 1 kHz separation for DAB -- with each carrier independently modulated using differential quadrature phase shift keying. The multiplex data is distributed amongst all the carriers, occupying approximately 1.54 MHz of spectrum, see Fig. 3. Consequently, the symbol-rate on any individual carrier is much lower and results in a longer symbol period which affords some protection against multipath echoes where the receiver sees a signal direct from the transmitter, plus a number of delayed signals due to reflections from terrain and buildings. By deliberately repeating part of each symbol during the so-called guard-interval, COFDM provides enhanced tolerance against multipath. As long as the delay of the echo signals is less than the guard-interval, there will be a constructive benefit to reception. In fact, it allows the broadcaster to introduce the concept of the single frequency network (SFN) of transmitters, where "man-made multipath" is deliberately created by having all the transmitters in the network transmitting the same signal on the same frequency -- something that would have been impossible with existing analogue radio systems. This allows efficient use of the scare RF spectrum available -- a key advantage of the Eureka 147 DAB system.

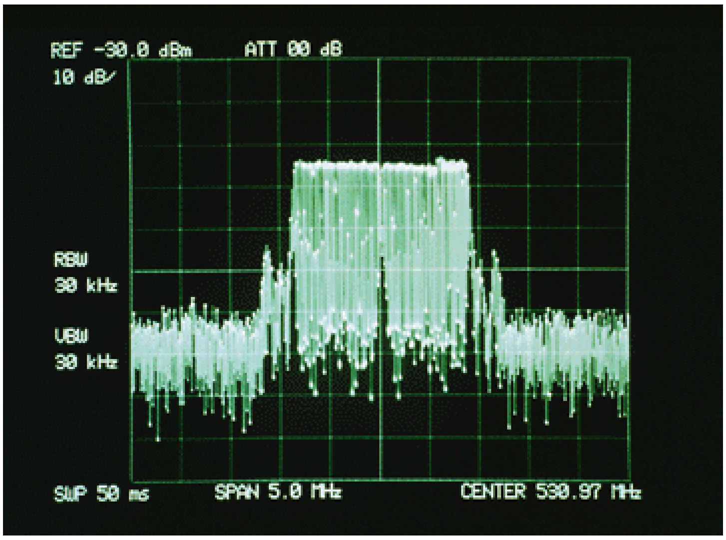

Figure 32- RF Spectrum of the DAB Signal.

20.2.2. Data transmission with DAB System

DAB is a reliable, multi-service, digital radio broadcasting system, designed specifically for robust reception by mobile, portable, and fixed receivers, using simple non-directional antennas.

The most interesting part of DAB for TeleFOT is the data transmission capabilities which are described in the following subsections.

20.2.2.1 MOT Protocol (RO – MOT: Rules of Operation for the Multimedia Object Transfer Protocol)

MOT is a transport protocol for transmission of multimedia data channels to various types of DAB receivers with multimedia capabilities. There are many possibilities for transmitting information that are incorporated in the common transport mechanism for different DAB data channels, so that access to multimedia content is unified within the OBD system.

MOT ensures interoperability between different data services and different types of applications, different receiving devices and equipment from different manufacturers. The purpose of the protocol is to transmit MOT objects of finite length from a transmitter (a content provider or services) to a receiver (a terminal).

The advantages of using the MOT protocol are:

- No restrictions apply to the content that can be transmitted.
- Segmentation and packet transmission is transparent to the user application.
- The existing MOT standard can be expanded to be compatible with the previous protocols.

The MOT protocol addresses the level of transport and not the application level, although it contains basic information about the object management and presentation of multimedia content.

The size of objects that can be transmitted using MOT is limited by the maximum body size and the size of any information that can be carried in the header is limited by the size of the field HeaderSize which can vary from 8 Kbytes to 1 Byte.

DAB transmission methods that can be used to transmit MOT objects are the package mode and the PAD (Programme Associated Data) mode.

A MOT object consists of an ordered collection of the following three parts:

Header core: The header core contains information about the size and the content of the object, so that the receiver can determine whether it has system resources to decode and present the object or not.

Header extension: The header extension includes information that supports the handling of the objects (e.g. memory handling) and provides additional information that can support an application.

Body: The body carries any kind of data, where structure and content of the data is described in the header core and the header extension.

For transportation the object is split into several segments, at least one header segment and, if present, one body segment. Each segment is mapped into one Data Group.

The header is separated from the body during transportation in order to:

- have the possibility to repeat the header several times before and during the transmission of the body (which is useful when transmitting long objects);
- send the header in advance in order to give the receiver the opportunity to "be prepared in advance" to the data that is going to be received;
- send the header unscrambled when the body is scrambled.

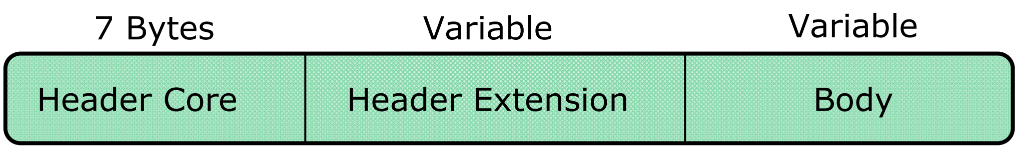

Figure 33- General MOT object structure.

20.2.2.2 TPEG Standard

The world of ITS (Intelligent Transportation Systems) has a huge variety of applications, among which a wide range of applications dedicated to broadcast information can be found. As examples the traffic information, transportation public car parks or travel times can be named. Each of these applications generate different types of information, which is distributed to the end users through the available channels. The information must therefore be adapted to the environment in question, being necessary to give a voice format, text or data (multimedia), different in each case and very specific.

An ITS service provider wanting to send information through different media carriers is required to work with multiple protocols, transmission media, databases data and applications, which means very high transmission costs.

In response to this problem, the EBU (European Broadcasting Union) established the Transport Protocol Expert Group. (TPEG) to develop a protocol for broadcasting quality information, supporting a variety of ITS applications. TPEG is also the name of this protocol.

An important consideration about the TPEG protocol is referring only to the transmission of data between a supplier and end users, that is, it is a broadcast protocol and not a protocol for exchanging data between information centers.

TPEG protocol defines the format of the information, which allows the broadcasting of this information through multiple delivery systems (DAB, Internet, radio analogue) from by generating a single message. This message is unique to each type of application (e.g. TPEG-RTM defines messages to the application of road traffic information) and is converted to appropriate format for each carrier type using an adapter layer.

On the other hand, TPEG also defines the ITS applications that will be transmitted on the TPEG information protocol. This includes broadcast information applications, among which are TPEG-RTM (Road Traffic Messages) or TPEG-PTI (Public Transport Information), and control applications, the protocol itself, as the TPEG-SNI (Service and Network Information).

A TPEG message is formed by three different parts:

- A Message Management Container Messages, which includes information about:
    - Dates and times: start time and end time of expiration of the message, time stamp of the message generation, details of times and reset the service component
    - Effect and reliability of the traffic message on the road: the severity factor, unverified information, etc.
    - cross-reference information, which allows cross-referencing each message with other messages either coming from its own application or from other TPEG applications.
- An Application Event Container, including the complete description of the event, this description is provided in a hierarchically way so that providers / decoders can choose the level of detail that they want to send / decode.
- A Location Container. TPEG technology uses a location system common to all applications; this location system is known as TPEG-Loc. This can enable the connection of different TPEG flows from its common location. Each message will

refer to a particular location. The location can be very Specific a single point in the road network or a road segment between two points, or even a more general area, often with undefined boundaries.

TPEG Protocol Characteristics

TPEG technology has been designed to be used by a wide range of applications requiring point to multipoint data transmission via potentially non reliable broadcast channels TPEG is also suitable for point to multicast applications and can be encapsulated in IP.

TPEG is especially useful for carrier independent applications. TPEG ensures the transparent transmission of data through all the carriers.

The main features of TPEG regarding multimodal transport are:

- It is unidirectional (broadcast)
- It is independent of the carrier (byte oriented). Provides a single message for each type of application.
- It has a hierarchical frame structure.
- Supports language-independent applications.
- Allows traffic and travel information (TTI) multimodal, covering all types of travel information.
- Supports a wide range of receiver / decoders, thanks to the hierarchical structure of message
  - Providers can choose the level of detail of the information they publish.
  - In addition, recipients can choose the level of information to decode.
- Maintains the highest level of compatibility with RDS-TMC.
- Use common forms of locations reference.
  - WGS 84 (latitude and longitude) and road intersections (text).
  - The main goal for that is that receivers / decoders do not need a location database to understand and use the information.

21. DVB - Digital Video Broadcasting

21.1. Summary

The following table summarizes the main characteristics of DVB

Description	Digital Video Broadcasting
Maturity	Mature
Availability	Available
Price	Available
Bandwidth	From 5 to 8 MHz
Frecuency	Band IV - 470-826 MHz

Coverage	satellite coverage
Application to vehicular environment	Can be used for invehicle video broadcasting
Infrastructure requirements	Receiver device
Legal issues	Solved
Standarization	Fully standarized -- ETSI / European Standars
Use in TeleFOT test site	Not foreseen but it can be used

Table 25 – DVB summary table

21.2. Detailed description

Digital Video Broadcasting (DVB) is a suite of internationally accepted open standards for digital television. DVB standards are maintained by the DVB Project, an international industry consortium with more than 270 members, and they are published by a Joint Technical Committee (JTC) of European Telecommunications Standards Institute (ETSI), European Committee for Electrotechnical Standardization (CENELEC) and European Broadcasting Union (EBU). Many aspects of DVB are patented, including elements of the MPEG video coding and audio coding. [DVB]

DVB systems distribute data using a variety of approaches, including:

- Satellite: DVB-S, DVB-S2 and DVB-SH
  - DVB-SMATV for distribution via SMATV
- Cable: DVB-C, DVB-C2
- Terrestrial television: DVB-T, DVB-T2
  - digital terrestrial television for handhelds: DVB-H, DVB-SH
- Microwave: using DTT (DVB-MT), the MMDS (DVB-MC), and/or MVDS standards (DVB-MS)

These standards define the physical layer and data link layer of the distribution system. Devices interact with the physical layer via a synchronous parallel interface (SPI), synchronous serial interface (SSI), or asynchronous serial interface (ASI). All data is transmitted in MPEG transport streams with some additional constraints (DVB-MPEG). A standard for temporally-compressed distribution to mobile devices (DVB-H) was published in November 2004.

These distribution systems differ mainly in the modulation schemes used and error correcting codes used, due to the different technical constraints. DVB-S (SHF) uses QPSK, 8PSK or 16-QAM. DVB-S2 uses QPSK, 8PSK, 16APSK or 32APSK, at the broadcaster's decision. QPSK and 8PSK are the only versions regularly used. DVB-C (VHF/UHF) uses QAM: 16-QAM, 32-QAM, 64-QAM, 128-QAM or 256-QAM. Lastly, DVB-T (VHF/UHF) uses 16-QAM or 64-QAM (or QPSK) in combination with COFDM and can support hierarchical modulation.

21.2.1. Overview of the System

Although the DVB-T transmission system has proven its ability to serve fixed, portable and mobile terminals, handheld terminals (defined as a light battery powered apparatus) require specific features from the transmission system serving them:

- as battery powered, the transmission system shall offer them the possibility to repeatedly power off some part of the reception chain to increase the battery usage duration;
- as targeting nomadic users, the transmission system shall ease access to the DVB-H services when receivers leave a given transmission cell and enter a new one;
- as expected to serve various situations of use (indoor and outdoor, pedestrian and inside moving vehicle), the transmission system shall offer sufficient flexibility/scalability to allow reception of DVB-H services at various speeds, while optimizing transmitter coverage;
- as services are expected to be delivered in an environment suffering high levels of man-made noise, the transmission system shall offer the means to mitigate their effects on the receiving capabilities;
- as DVB-H aims to provide a generic way to serve handheld terminals, in various part of the world, the transmission system shall offer the flexibility to be used in various transmission bands and channel bandwidths.

A full DVB-H system is defined by combining elements in the physical and link layers as well as service information.

DVB-H makes use of the following technology elements for the link layer and the physical layer:

- Link layer:
    - time-slicing in order to reduce the average power consumption of the terminal and enabling smooth and seamless frequency handover;
    - forward error correction for multiprotocol encapsulated data (MPE-FEC) for an improvement in C/N-performance and Doppler performance in mobile channels, also improving tolerance to impulse interference.
- Physical layer:
    - DVB-T (EN 300 744 [1]) with the following technical elements specifically targeting DVB-H use:
        - DVB-H signaling in the TPS-bits to enhance and speed up service discovery. Cell identifier is also carried on TPS-bits to support quicker signal scan and frequency handover on mobile receivers;

- 4K-mode for trading off mobility and SFN cell size, allowing single antenna reception in medium SFNs at very high speed, adding thus flexibility in the network design;
- in-depth symbol interleaver for the 2K and 4K-modes for further improving their robustness in mobile environment and impulse noise conditions.

To provide DVB-H services time-slicing, cell identifier and DVB-H signaling are mandatory; all other technical elements may be combined arbitrarily.

It should be mentioned that both time-slicing and MPE-FEC technology elements, as they are implemented on the link layer, do not touch the DVB-T physical layer in any way. It is also important to notice that the payload of DVB-H are IP-datagrams or other network layer datagrams encapsulated into MPE-sections.

The conceptual structure of a DVB-H receiver is depicted in figure 1. It includes a DVB-H demodulator and a DVB-H terminal. The DVB-H demodulator includes a DVB-T demodulator, a time-slicing module and a MPE-FEC module.

- The DVB-T demodulator recovers the MPEG-2 Transport Stream packets from the received DVB-T (EN 300 744 [1]) RF signal. It offers three transmission modes 8K, 4K and 2K with the corresponding Transmitter Parameter Signalling (TPS). Note that the 4K mode, the in-depth interleavers and the DVB-H signalling have been defined while elaborating the DVB-H standard.
- The time-slicing module, provided by DVB-H, aims to save receiver power consumption while enabling to perform smooth and seamless frequency handhover.
- The MPE-FEC module, provided by DVB-H, offers over the physical layer transmission, a complementary forward error correction allowing the receiver to cope with particularly difficult receiving situations.

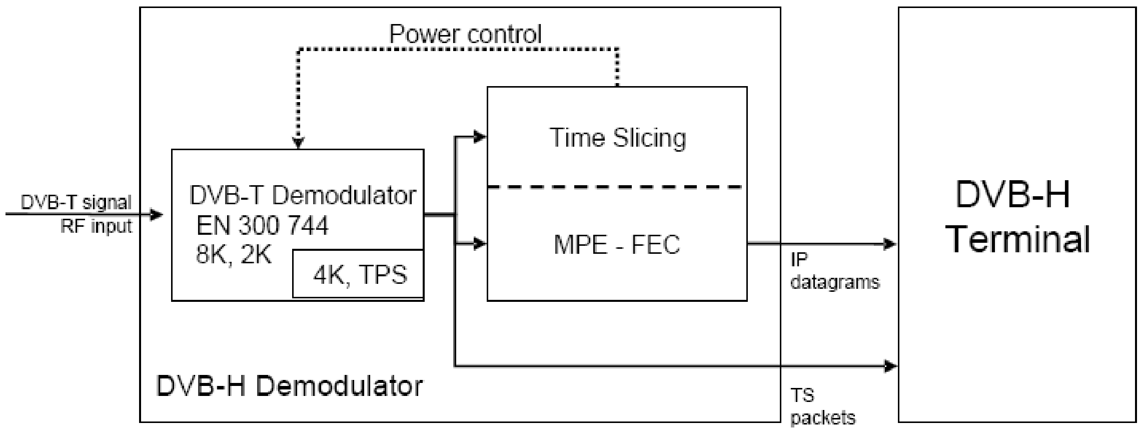

Figure 34- Conceptual structure of a DVB-H receiver.

An example of using DVB-H for transmission of IP-services is given in the following figure. In this example, both traditional MPEG-2 services and time-sliced "DVB-H services" are carried over the same multiplex. The handheld terminal decodes/uses IP-services only.

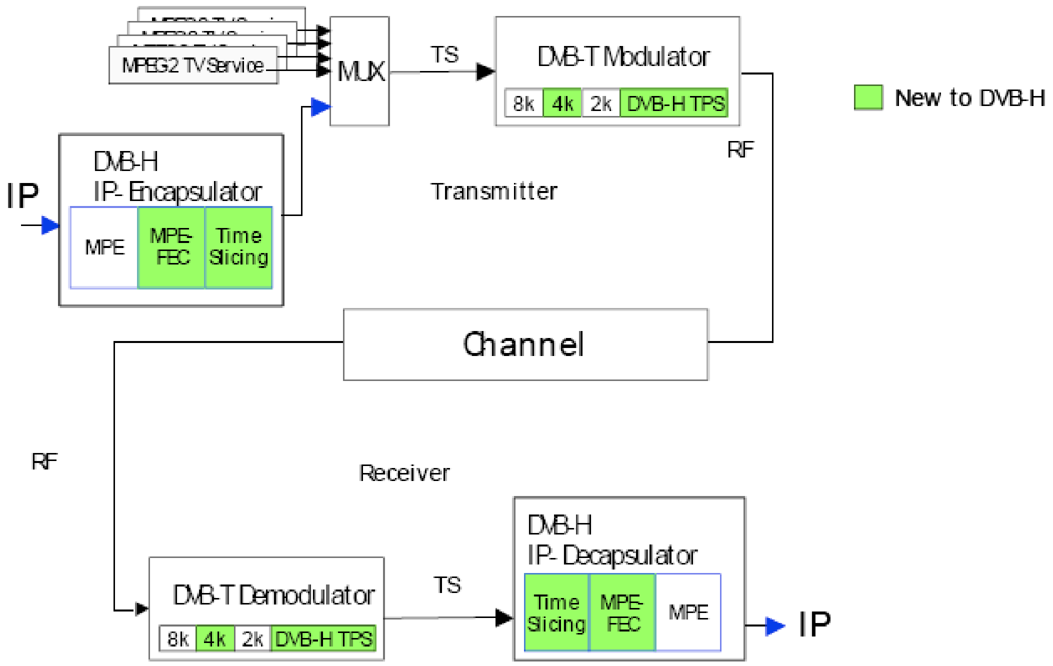

Figure 35 - conceptual description of using a DVB-H system

21.2.1.1 Time-Slicing

The objective of time-slicing is to reduce the average power consumption of the terminal and enable smooth and seamless service handover. Time-slicing consists of sending data in bursts using significantly higher instantaneous bit rate compared to the bit rate required if the data were transmitted using traditional streaming mechanisms.

To indicate to the receiver when to expect the next burst, the time (delta-t) to the beginning of the next burst is indicated within the burst. Between the bursts, data of the elementary stream is not transmitted, allowing other elementary streams to use the bandwidth otherwise allocated. Time-slicing enables a receiver to stay active only a fraction of the time, while receiving bursts of a requested service. Note that the transmitter is constantly on (i.e. the transmission of the transport stream is not interrupted).

Time-slicing also supports the possibility to use the receiver to monitor neighboring cells during the off-times (between bursts). By accomplishing the switching of the reception from one transport stream to another during an off period it is possible to accomplish a quasi-optimum handover decision as well as seamless service handover.

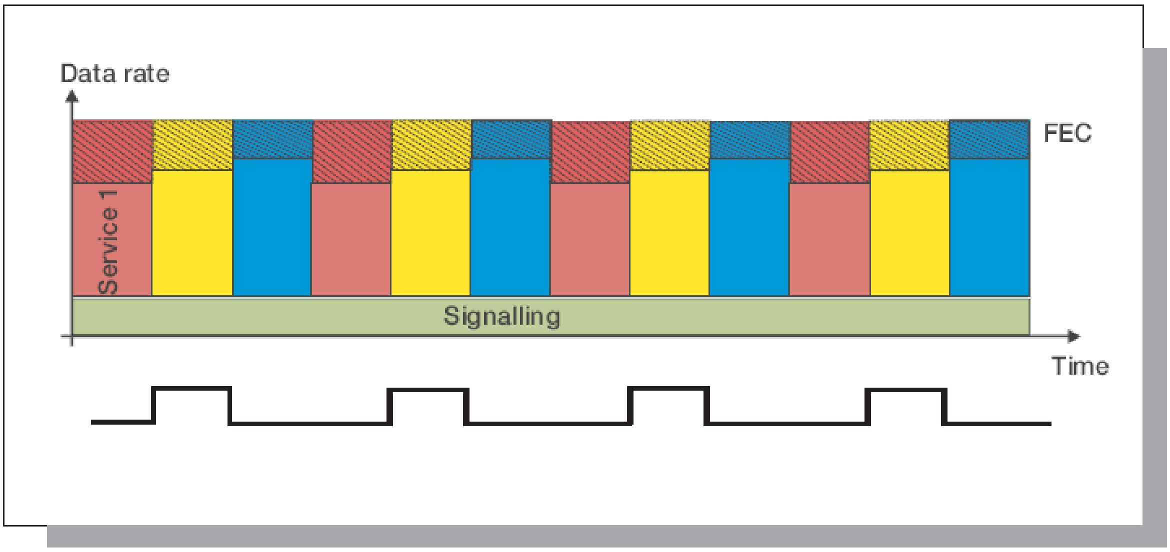

Figure 36 - Time-Slicing concept

21.2.1.2 MPE-FEC

The objective of the MPE-FEC is to improve the C/N- and Doppler performance in mobile channels and to improve the tolerance to impulse interference.

This is accomplished through the introduction of an additional level of error correction at the MPE layer. By adding parity information calculated from the datagrams and sending this parity data in separate MPE-FEC sections, error-free datagrams can be output after MPE-FEC decoding despite a very bad reception condition.

With MPE-FEC a flexible amount of the transmission capacity is allocated to parity overhead. For a given set of transmission parameters providing 25 % of parity overhead, the MPE-FEC may require about the same C/N as a receiver with antenna diversity.

The MPE-FEC overhead can be fully compensated by choosing a slightly weaker transmission code rate, while still providing far better performance than DVB-T (without MPE-FEC) for the same throughput. This MPE-FEC scheme should allow high-speed single antenna DVB-T reception using 8K/16-QAM or even 8K/64-QAM signals. In addition MPE-FEC provides good immunity to impulse interference.

The MPE-FEC, as standardized, works in such a way that MPE-FEC ignorant (but MPE capable) receivers will be able to receive the data stream in a fully backwards-compatible way.

CONCLUSIONS

This deliverable reports the activities undertaken in Task 2.6.1 (Communications Technologies) of WP 2.6 (Technology and Service Observatory). The main objective of task 2.6.1 was to provide the project with relevant and up-to-date information of the available communication technologies that could be used by the different FOTs in TeleFOT.

More concretely, this deliverable analyses the different communication technologies studied in task 2.6.1 and present them in a concise and summarised way.

The analysis is made through the different sections and in terms of their status on the market and compiled in the present document. The study of the communication technologies will include:

- Cellular communications. GSM, GPRS, EDGE, UMTS, HSDPA, HSUPA and LTE are included under this section.
- Nomad technologies. Useful for local communication in some scenarios. WiFi (IEEE 802.11), WiMAX (IEEE802.16)
- Adhoc communications. Despite they are not mature enough nowadays, Ad hoc communications will be very important for safety services deployment shortly.
- Short range communications. Needed for on board communications. NFC, Bluetooth, UWB, ZigBee
- Broadcast communications. Including RDS-TMC, DAB and DVB

The common methodology adopted by the TeleFOT Technology Observatory for the reporting suggests considering different aspects – such as description, maturity, availability, price, usability in TeleFOT, etc. - when assessing technical issues.

For each of the technologies studied, a summary description, where a table including technical information - such as Bandwidth, Frequency, Coverage, Standardization, etc. – is presented, and a detail description with a more extended description of the characteristics of each technology is presented.

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[EDGE2] The evolution of EDGE white paper: http://www.gsacom.com/index.php4

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[HSDPA1] http://www.umtsworld.com/technology/hsdpa.htm

[HSDPA 2] Understanding HSDPA and HSUPA: http://www.3g4g.co.uk/Hsdpa/

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[ADHOC1] Mobile ad-hoc networks: http://datatracker.ietf.org/wg/manet/charter/

[ADHOC2] IEEE: http://ewh.ieee.org/tc/its/

[ADHOC3] C2C-CC, www.car-to-car.org.

[DSRC] ITS forum: http://www.itsforum.gr.jp/E_index.html

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