Professional radio systems for aeronautical and public safety applications

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Abstract

Professional mobile communication systems are radio systems directly operated and/or owned by the organization that adopts them. They are targeted to business and professional users needing communication networks alternative to commercial ones. They are intended to fulfill specific requirements related to specific application fields. In general they are characterized by high levels of security, availability and resilience, guaranteeing that the communication is available even in temporary or permanent absence of coverage. Professional systems support common voice communication services with the addition of services intended to enable on field operations like group direct communications, push-to-talk, call prioritization and pre-emption, dispatcher functionality, end-to-end security, geographical function call, etc. Typical users and application areas of professional communication systems are police forces in day-to-day and emergency operations, fire and ambulance services, taxi and transport societies, port and airport operators, etc.

In the modern society, where the number and quality of services available to citizen is continually growing and improving, special attention should be addressed to applications dealing with citizen safety and infrastructure efficiency improvement. For this reason, this thesis focuses on the evolution of professional communication systems used in two environment:

- Aviation,
- Public Protection and Disaster Relief (PPDR).

The air transport can be considered as the most reliable and safe form of transport. Since traffic is expected to double

by 2025, it become of paramount importance to actualize an effective evolution of communication systems and infrastructures on which the entire avionic ecosystem relies. Principal aims are to improve pilot situation awareness, increase service capacity, provide a secure and reliable communication system to exchange air traffic management, operation control and added value passenger service high data rate data. To fulfilling these requirements, EUROCONTROL is devolopping two technologies L-Band Digital Communication System (LDACS) and AeroMACS, to be used respectively for Air/Ground and airport communications.
LDACS will operate on L-Band, where the coexistence with existing legacy systems shall be guaranteed. For this reason the issues related to electromagnetic compatibility of LDACS with other systems that operate in L-Band are investigated in this thesis. A general interference analysis is performed and an algorithm of detection and mitigation of interference generate by JTIDS on LDACS1, based on energy interference detection and signal replicas combination, is proposed. This technique is then generalized to OFDM systems affected by impulsive noise. For what concerns AeroMACS, resource allocation methodologies, based on the characteristics of propagation channel in airport areas, are presented.

At present, Professional Mobile Radios (PMRs) supply a number of services mainly based on voice and low rate data communications. However, the requirements of public security operators highlight the need to overcome the limitations posed by actual PMR standards, introducing new technologies able to support high data rate communications while maintaining all the peculiarities of professional systems.
Voice will remain an essential application in PPDR, however the availability of multimedia applications brings a number of

benefits. Data pass from being an accessory support tool to become a Mission Critical service at the same level of voice. A larger amount of data and the availability of multiple applications permit to increase the situation awareness and decision making efficiency, avoiding that corrupted, missing or late information lead to wrong decisions causing, in the worst case, even loss of lives. To this end, 3GPP, in collaboration with ETSI and TCCA, is integrating the LTE (Long Term Evolution) standard with the characteristics necessary to fulfill broadband mission critical futures. In this thesis possible solutions to implement group call services in LTE are studied, with particular attention to criticalities that can arise in terms of latencies, number of users/groups/type of services supported. A possible architectural solution is proposed and validated.

Finally, potentialities of a such broadband communication system are highlighted, proposing an efficient architecture of a Smart Public Safety platform for monitoring, forecasting and managing emergency situations, arising from environmental disasters or crimes.

This platform performs a smart and functional integration of heterogeneous components as a Smart Data Gathering and Analysis system, a novel professional communication system, wireless sensor networks and social networks. Each element not only acts as an information collector, but has autonomous capabilities to cooperate with the others in order to increase the system efficiency and to reduce the need of human interactions.

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Aeronautical Communications

Chapter 1

Aeronautical Communication Evolution Process

Today the air transport can be considered as the most reliable and the safe form of transport. Since the air traffic is expected to double by 2025 and at the same time no significant improvements have been adopted in traffic management since the last 40 years, several relevant concerns arise about the ability of the air transportation system to cope with process and procedure flexibility and real time information processing to keep up with the progress.

In particular, a special attention has to be addressed to communication systems and infrastructures on which the entire avionic ecosystem relies. To obtain an effective evolution of management systems some high level requirements can be identified: [6]:

- Pilot situation awareness shall be improved through enhanced communications with flight controllers, communications monitoring, visual look-out and navigation.

- Airport hosting capacity shall be improved and structural limits overcome.

- ATS (Air Traffic Services) shall base on new reliable communications.

- AOC (Airline Operations Control) shall be increased for efficient operations.

- Passengers and cabin communications with suitable constrains of robustness, reliability and re-reconfigurability shall be developed.

- Reliable and multi-modal transmission to the ground of safety critical information, granting that the information has been read, understood and implemented, shall be implemented.

- The convergence of protocols and interfaces in an onboard network architecture, which connects each passenger seat/crew terminal to the In-Flight Cabin server, shall be granted.

However, fulfilling these requirements implies the adoption of new technologies and procedures characterized by reliability, effectiveness, safety and security that implies complex standardization and realization procedures. Requirements listed above are strictly related to improvements needed in communications system. Currently communication system is intended to support a list of well defined services:

- Aeronautical Administrative Communication (AAC): communication used by aeronautical agencies for non-safety communications relating to business aspects of flights and transport services operating.

- Aeronautical Operational Control (AOC): communication required for the exercise of authority over the initiation, continuation, diversion or termination of flight for safety, regularity and efficiency reasons.

- Air Traffic Services (ATS): a generic term meaning flight information services, alerting, air traffic advisory, air traffic control services (e.g. area control, approach control or aerodrome control services).

- Air Passenger Communications (APC): defined as services to passengers providing them with an access to communications and entertainment similar to those that can be experienced on ground.

These services are granted in a large part by voice communications. Also some data communication links are available. They rely on digital terrestrial or satellite technologies acting in VHF or L band characterized by very slow data transmission rates (see Table 1.1). The amount and the type of information that can be exchanged is limited. Since real time exchange of large data is not feasible, effectiveness and costs of such services remain limited.

To overcome these limits, worldwide main organizations involved in air traffic management, in particular ICAO (International Civil Aviation Organization) [7], the United Nation agency in charge of coding principles and techniques of air navigation, FAA (Federal Aviation Administration) [8],

Table 1.1: Existing terrestrial and satellite data links.

Tecnology	Data Rate (kbps)
VHF Data Link 2 (VDL2)	31.5
ACARS	1.8
Iridium	9.6
Inmarsat Swift Broadband	450

the United States federal agency for civil aviation, and EUROCONTROL (European Organization for the Safety of Air Navigation) [9], the organization that manages the European air traffic, expressed their interest in developing a Future Communications Infrastructure (FCI) that integrates existing and future communications infrastructures in a system of systems.

The process of FCI developing started in 2003 when ICAO traced the evolution approach to define new functions in aeronautical communications. At a later time, in 2007, the EUROCONTROL Action Plan 17 defined the overall needs and the EU research project NEWSKY (NEtWorking the SKY) addressed the feasibility of a new aeronautical communication network based on IPv6 technologies. Moreover, EUROCONTROL and European Commission promoted two synergetic research projects: SESAR (Single European Sky ATM Research) leaded by the SESAR Joint Undertaking (SJU) [10] and SANDRA (Seamless Aeronautical Networking through integration of Data links Radios and Antennas) [11], both related to different aspects of the same topic.

SJU consortium is composed of 16 members joined under the purpose of developing a modern air traffic management system, improving effectiveness, capacity and safety of transport while reducing costs. The main activity performed by

SJU is the definition of a communication architecture for Air Traffic Management (ATM) intended to support advanced data communication services as well as specific features for 4D trajectory management, ASAS separation (Airborne Separation Assistance System), controllers' workload reduction and SWIM operations support. SWIM (System Wide Information Management) architecture is intended to change the paradigm of how information is managed in the European ATM system. Subject involved in information sharing are: Pilots, Airport Operations Centers, Airline Operations Centers, Air Navigation Service Providers (ANSPs), Meteorology Service Providers, Military Operation Centers; while the information exchanged consists mainly on aeronautical data, 4D (position and time) flight route trajectory, aerodrome operations as approaches, runways, taxiways, gate and aircraft turn-around information, past, current and future state of meteorological state, air traffic flow necessary to understand the overall air traffic and air traffic services situation, surveillance consisting in positioning information from radar, satellite navigation systems, aircraft datalinks, etc., capacity and demand information on the airspace users needs of services, access to airspace and airports and the aircraft already using it. [1]

The SANDRA project, partially funded by the Seventh Framework Program of the European Commission (FP7/2007-2013) is a consortium of 30 between industries, research organizations, universities and small and medium enterprises. It intends to define, and validate a scalable and reconfigurable Airborne Avionics Architecture that integrates new and existing communication media, providing seamless service coverage between airspace domains and aircraft classes, independently on used radio technologies. The main charac-

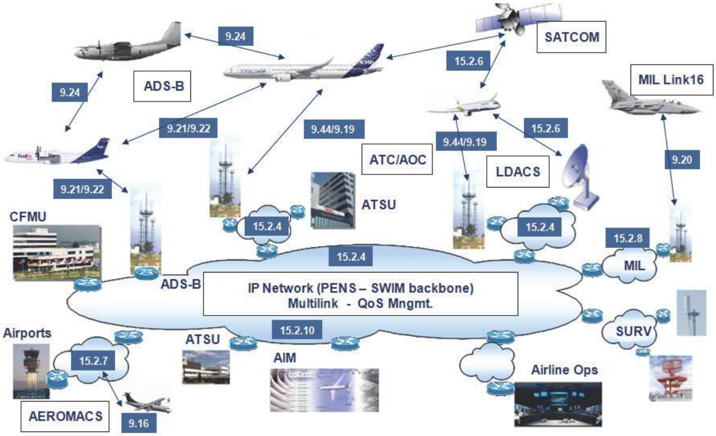

Figure 1-1: ATM SWIM Architecture [1].

teristic of this architecture is the integration of components at different levels resorting to IPv6 as unification point, in particular it will permit:

- service integration between different applications of different operational domains as: ATS, AOC/AAC, APC;
- network integration of different network and transport technologies (ACARS, ATN/OSI, ATN/IPS, IPv4, IPv6);
- radio technologies integration in Integrated Modular Radio platform;
- L-band and Ku-band antennas integration to obtain asymmetric broadband links;
- multi-domain airport integration by means of WiMAX technology.

1.1 Future communications systems

Among other thinks, SESAR defined the infrastructure for air/ground communication. It followed a multi-link approach, composed of three different subnetworks:

- a ground/air data link for continental communication in L band named L-band Digital Aeronautical Communication System (LDACS);
- a satellite system to cover oceanic airspace;
- a system to support airport operations named AeroMACS (Airport Surface Data Link).

LDACS and AeroMACS will be examined in the following chapters.

Chapter 2

LDACS Compatibility with Existing L-Band Systems

This chapter addresses the impact due to introduction of LDACS (L-Band Digital Aeronautical Communication System) in the aeronautical L-Band, a portion of spectrum already populated by some aeronautical systems, considering interference and compatibility issues that arise.

The aeronautical VHF Radiocommunication band (i.e. 118-137 MHz) is congested. Furthermore, new concepts of applications for aeronautical data-links have been developed in recent years. For these reasons, the aeronautical community is investigating the possibility to develop a new Communication Infrastructure that should provide additional aeronautical mobile communications capacity and support the communication requirements defined in the Communication and Operations Concept Requirements Document (COCR) [12] developed by ICAO. Two options for the LDACS have been

identified: LDACS1 and LDACS2. One of the two alternatives will be selected depending on the ongoing studies results.
Several legacy systems are currently allocated in the same frequency band. For this reason it is important to clarify whether coexistence in the same band is possible or not.

Figure 2-1 represents the radio spectrum allocation of different services in the L-Band.

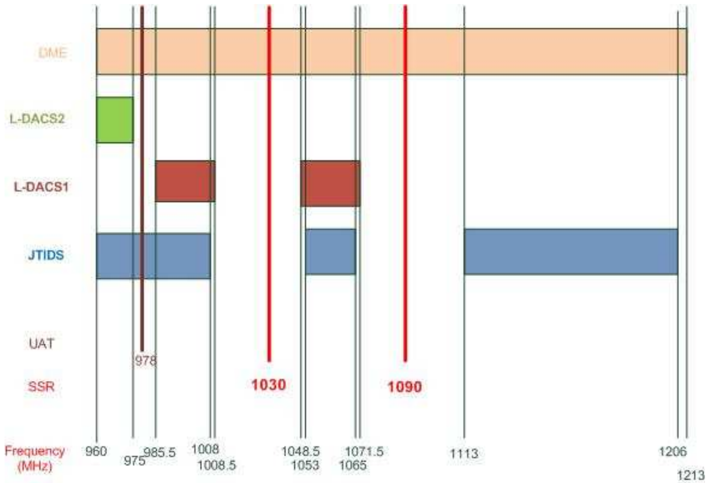

Figure 2-1: L-Band Aeronautical Systems.

This study will focus on the following aeronautical systems:

- Joint Tactical Information Distribution System (JTIDS)
- Secondary Surveillance Radar (SSR)
- Distance Measuring Equipment (DME)

2.1 Overview of involved systems

2.1.1 LDACS1

LDACS1 [13, 14] data link is intended to support ATS and AOC data communications for safety and regularity of flight in several operation scenarios as airports, TMAs (Terminal Control Area) and En-Route high-density airspaces. In accordance with FCI functionality requirements, LDACS1 is a cellular system for point-to-point or broadcast communications that should support two modes of operation for Air/Ground (A/G) and A/A (Air/Air) communications. These two modes differ in radio channel use and physical/link layer specifications. However, in the first phase of LDACS1 development only A/G communications have been considered and detailed specifications for A/A communications are not available yet. The LDACS1 operations are optimized for data link communications, but A/G mode should support also voice communications.

The LDACS1 data link can be integrated in an Aeronautical Telecommunication Network based on IP protocol suite (ATN/IPS). The LDACS1 A/G sub-system is a multi-application cellular broadband system that utilizes reservation based access control and advanced network protocols. All aircraft stations within a cell are connected to the controlling Ground Station (GS) by point-to-point links. The physical cell coverage is not directly correspondent to service coverage, but a single service can be provided in an area spanning multiple cells. As a consequence, LDACS1 supports handover procedures that are seamless, automatic, and transparent to the user. LDACS1 is designed to support operation at a maximum distance of 200 nm from the GS.

LDACS1 is based on Frequency Division Duplexing (FDD) and is designed to operate in the lower part of the L-band, [960-1164 MHz]. A first suggested solution foresees to accommodate Forward Link (FL) and Reverse Link (RL) [1] channels in the 960-1009 MHz and 1048-1164 MHz frequency ranges, respectively. However, it should be precised that this is not the only possible allocation, even if it seems one of the best, in order to avoid co-site problems between LDACS1 and 1030/1090 SSR transceivers onboard. FL/RL effective channel bandwidth is 498.05 kHz and lies on a 0.5 MHz grid. The duplex spacing between FL and RL is 63 MHz. A more recent EUROCONTROL proposal suggests to accommodate LDACS1 in four sub-bands as described in Sec. 2.2.1.

LDACS1 physical layer (PHY) is based on OFDM (Orthogonal Frequency-Division Multiplexing) technique with 64 subcarriers (with 50 active subcarriers) separated by 9.765625 kHz. Each sub-carrier is separately modulated leading to a total modulated symbol duration of Ts = 120 s. Different modulation and coding scheme are considered and the system should be able to adapt the most suitable scheme to the service and to the propagation condition for data channels. OFDM symbols are then organized in a frame hierarchical structure (multiframe of 58.32 ms duration and superframe of 240 ms duration). In particular, FL is a continuous OFDM transmission; instead RL transmission foresees a multiple access scheme based on OFDMA-TDMA bursts. The RL resources are assigned to different users on demand.

The LDACS1 specifications suggest some methods than could be used to reduce the interference on LDACS1 receiver arising from systems operating in adjacent frequency bands,

[1] FL is the link from GS to Air Station (AS) and RL is the link from AS towards GS

depending on the situations:

- Erasure Decoding
- Pulse Blanking
- Oversampling.

On the other hand, the interference arising from LDACS1 into other existing L-Band systems can be mitigated by means of TX windowing, in order to smooth the sharp phase transitions between consecutive OFDM symbols which cause out-of-band radiation.

2.1.2 LDACS2

LDACS2 [15,16] is foreseen for the same applications as LDACS1. As LDACS1, the LDACS2 A/G sub-system is a multi-application cellular system. Each cell is managed by a Ground Station (GS), and the GSs are organized in clusters and connected to an ATN. The GS provides connectivity to the aircraft within the cell up to 200 nm. The system has an efficient air-initiated cell handover mechanism founded on aircraft knowledge of cell locations and characteristics. It provides a robust link for high Quality of Service in terms of transmission latency for time-critical messages.

LDACS2 data link is based on Time Division Multiplexing (TDD): half duplex communication is realized by organizing time transmission in frame and providing separate section for RL and FL transmission. LDACS2 will operate in the lower L-Band frequency, from 960 to 975 MHz. The available spectrum shall be partitioned into a number of sub bands, each 200 kHz wide. Each of these bands shall be occupied by a Gaussian Minimum-Shift Keying (GMSK) modulated RF carrier supporting a number of TDMA time slots. To minimize the impact of LDACS2 onto other L-Band systems and vice-versa, time amplitude profile of LDACS2 transmission, power control, and pulse blanking techniques are suggested.

LDACS2 uses strong data coding for achieving the highest QoS in terms of latency by minimizing the need for retransmission. Modulated data are sent in a burst that is composed of: transmitter ramp-up and ramp-down periods, synchronization sequence, flags and guard time. The multiple access technique is TDMA: each data burst is sent in a slot; the frame is composed of FL and RL sections and each section is composed of time slots. Slot scheduling is deterministic. Frame duration is 1 second and it consists of 150 time slots of equal duration. The maximum burst duration is limited by slot duration. If an aircraft or a ground station has to transmit data that exceed the slot length, a burst can span over multiple slots and be built with unique ramp-up time and synchronization interval at the start of the initial slot, and ramp-down time and propagation guard time at the end of the last slot. Ground station transmits continuously in the uplink sections, therefore the burst transmitted in a given uplink section could be several slots long until the upper limit of the uplink section.

2.1.3 SSR

Secondary Surveillance Radar (SSR) [17–20] is a traffic management system used to monitor the controlled airspace collecting information from aircraft equipped with SSR transponders and providing these information to the ATC controllers. The SSR interrogator may be co-located with a Primary Surveillance Radar (PSR) to obtain surveillance for non-equipped targets as well. The PSR is the traditional radar that estimates the aircraft distance transmitting pulsed signal and measuring the round trip delay of the signal reflected from the metallic fuselage; since it can only detect the range and bearing of reflecting obstacles, in particular conditions it can suffer from missing or false positive detections, moreover, due to the lack of interaction with the aircraft, it is impossible to recognize the target identity. Due to the high free-space loss, PSR has to use much higher interrogation power than SSR.

The operational range of radar systems is typically about 60-100 NM for terminal radars and 150-200 NM for en-route radars. If co-located, PSR and SSR have the same range.

SSR overcomes most of PSR limitations being based on a ground interrogator and a transponder installed on the aircraft: the SSR interrogator located on the ground repetitively emits interrogation pulses at the frequency of 1030 MHz and the airborne transponder replies at 1090 MHz. The kind of information (target identity, pressure altitude, etc.) included in the reply depends on the operation mode. Standardized interrogation modes are Mode 2, Mode A, Mode C and Mode S. Mode 2 is used by military bodies only.

Mode A and Mode C permit the identification of the aircraft and the transmission of the aircraft pressure altitude, respectively. The replies are encoded in a 12 bit string that is decoded by the ground receiver. Since the identity code length is limited to 12 bits, the number of available codes is 4096. Hence, 4096 aircraft can be addressed as a maximum: aircraft addresses are assigned dynamically by the ground system during the flight, therefore, in high traffic airspaces, Mode A identity codes do not allow a unique addressing scheme for aircraft (i.e., the same identification code can be used by different aircraft). This drawback is overcome by Mode S. Mode S message utilizes a 24 bit aircraft address that result in the number of possible addresses of nearly 16.8 million, thus permitting to permanently assign an address to an aircraft and allowing the ground station to interrogate a single transponder. Ground interrogators that use Mode S are capable of tracking all aircraft in the surveillance area equipped with either Mode A/C or Mode S transponders. By means of selective addressing of aircraft, SSR Mode S can avoid overlapping of replies (Garbling) known from the other Modes. Moreover Mode S transponder supports the ADS-B (Automatic Dependent Surveillance - Broadcast) services. In Mode S, interrogations are classified in:

- All-call interrogations: they require response from all aircraft in the not locked out status (i.e. aircraft not allowed to reply to interrogations) within the radio horizon;

- Roll-call interrogation: aircraft are selectively addressed using the unique 24 bit identification code.

Depended Surveillance Broadcast (ADS-B). ADS-B is an application which can be used to determine the position of airborne, surface aircraft and surface vehicles independently of active interrogators on the ground. One technical implementation consists in the Mode S ES (Extended Squitter), which is automatically and periodically transmitted by the SSR Mode S transponder. Mode S radars can extract the ADS-B information and make it available on the ground. Appropriate receivers may receive it even without interrogation. ADS-B has the potential to optimize the use of air-space, surface surveillance, conflict management and incursion monitoring by providing additional airborne data. Moreover it reduces the visibility restrictions by providing frequently updated information to controllers and pilots. The transponder automatically and continually (i.e. about once per second) broadcasts messages containing e.g. GPS position or (for aircraft) state vector, intent or selected parameters, etc.

The service is: "automatic" because information are issued periodically without operator input, "dependent" because it relies on on-board navigation sources and "broadcast" since the information is not sent in response to an individual interrogator, but broadcast to any appropriate receiving equipment.

ADS-B operations are divided in:

- ADS-B out: data transmission capability;
- ADS-B in: data reception, elaboration and visualization capability.

Traffic Collision Avoidance System (TCAS)
TCAS is a collision avoidance system utilized to reduce the risk of mid-air collisions between aircraft. Regardless ATC, it warns pilot of the presence of other aircraft in the surveillance range. It can also issue warnings (Traffic Advisory) and advise the pilot of corrective vertical maneuvers to provide or maintain the current separation (Resolution Advisory).

2.1.4 JTIDS/MIDS

The Joint Tactical Information Distribution System/Multifunctional Information Distribution System (JTIDS/MIDS)

[21–24] is a military radio system for the distribution of information, position location and identification between different military elements, such as aircraft, ships and ground installations. It is applied in surveillance, air control, secure voice transmission, electronic warfare, mission management and identification. JTIDS is a type of Tactical Digital Information Links (TADIL). TADIL, in general, are data links used to exchange digital information to effectively manage the air battle providing timely, secure high capacity communications. They are used for data and voice transmission. MIDS is the evolution of JTIDS used in NATO implementations. Link16 (TADIL-J in U.S.) is the most recent communication system used in NATO implementations. It provides communication architecture to airborne and surface users. Its radio interface is based on JTIDS technology and on its evolution MIDS.

JTIDS operates in the frequency range 960-1215 MHz. The operative range is limited to LoS (Line-of-sight) transmissions, up to 300 nm. Also an extended range of 500 nm can be addressed. Communication beyond this distance and in NLoS (Not Line-of-sight) conditions can be possible by relaying the messages from other network users. Multiple users can access the channel by resorting to a TDMA scheme. A set of time slots for data transmission and reception is assigned to each JTIDS unit. The slot[2] is the basic unit in which users transmit and receive data. The slot assignment is not deterministic but random in order to protect data against Jamming. In addition different users can transmit on the

[2] JTIDS system subdivides the 24 hours of the day in 112,5 epochs which duration is 12.8 minutes. Each epoch is divided in 98304 slots of 7.8125 milliseconds. Time slots are grouped into three interleaved sets of 32768 slots each.

same time slot but on different frequencies. JTIDS transmissions hop among 51 different frequencies, in accord with 128 possible hopping patterns. In operative environment, the need to create separated functional groups sharing different type of functions and information is supported by the possibility to create multiple simultaneous virtual nets. This is achieved assigning to each net a different frequency hopping pattern. Transmission security is achieved by means of both frequency hopping and message encryption.

JTIDS is designed to be intrinsically resistant to jamming disturbs by means of:

- randomization of the signal with techniques as frequency hopping, spread spectrum, time hopping (jitter), cryptography, interleaving, and pulse time modulations;

- information redundancy (i.e., repetition of the data and coding techniques: Reed Solomon, parity bits);

- increased signal level and transmission range with the use of relays.

The slot is composed of a header, a synchronization message, the information data and the propagation guard time. The time slot format can vary with the message supported and the performance required (in terms of throughput and anti-jamming).

2.1.5 DME

Distance Measuring Equipment (DME) [18] is an ICAO standardized radio navigation system that allows to calculate the slant distance of the aircraft from a ground beacon by timing the propagation delay of a radio signal. It comprises an interrogator onboard the aircraft that sends an interrogation, constituted by a series of pulses pairs, to a ground transponder, that replies with an identical sequence of pulses delayed of a predefined time interval on a different frequency. The airborne equipment computes the slant distance measuring the elapsed time between the interrogation and the reception of the ground reply and provides it to the pilots. Moreover an additional Air/Air operation mode is possible in which the aircraft interrogation is direct to airborne transponder.

DME is functionally equal to TACAN (Tactical Air Navigation) distance measuring component, but while DME provides only a measure of the slant distance, TACAN can provide also azimuth information.

There are two types of DME: DME/N (narrowband) and DME/P (precise distance measuring). DME/P is part of Microwave Landing System (MLS), it is capable of providing more accurate information respect to DME/N. It has two operation modes: the Final Approach (FA) mode that supports operations in final approach and runways regions, and Initial Approach (IA) mode that supports flight operation outside approach regions and is interoperable with DME/N.

The interrogation pulse pairs are Gaussian pulses characterized by spacing, frequency of transmission and duration. The spacing existing between the two pulses defines the pulse code and, together with transmission frequency, identifies the operating channel. Pulse duration is used to distinguish DME signal from other sources. 352 different operating channels are defined, corresponding to 126 different interrogation frequencies to which correspond as many reply frequencies separated of 63 MHz from the corresponding interrogation. Channels are designated by a letter W,X,Y or

Z and a number comprised between 1 and 126. W and Z channels are reserved only to DME/P. The bands 978-1020 MHz, 1041-1083 MHz, 1094-1150 MHz and 1157-1213 MHz are defined in ICAO Annex 10 [18], while the other region of spectrum in L-Band can be used in regional implementations

2.2 Cross-interference numerical analysis

2.2.1 Analysis approach

The general purpose of this study is to determine if the receiver protection criteria requirements of each victim system stated by standards or other relevant sources are met in presence of an interfering signal, considering the mutual interference between LDACS1&2 and other L-Band systems. This study permits to judge the interference as harmful or tolerable. This is achieved by calculating the interference power level that affects the victim receiver and comparing it with the maximum tolerable power level value. The interference power level is determined by means of link budget calculations as a function of different parameters, in particular distance and frequency separation.

The procedure for the analysis that will be performed consists of three steps.

Step1. Received interference power level calculation.
The interference power level is calculated by means of a mod-

ified form of link-budget:

$$
\begin{aligned}
I(d,\Delta f) = & P_{Tx} - L_{Tx} + G_{Tx} - PL(d) + G_{Rx} - L_{Rx} + \\
& - L - OCR(\Delta f) + DC
\end{aligned}
\tag{2.1}
$$

- $I(d, \Delta f)$ is the transmission power of the interfering signal received by the victim system. It is function of spatial and frequency distance between interferer and victim system.

- P_{TX} is the transmitting power.

- G_{TX} and G_{RX} are respectively the antenna gain of the transmitter (interfering system) and of the receiver (victim system).

- L_{TX} and L_{RX} are respectively the losses at the transmitter (interfering system) and at the receiver (victim system).

- $PL(d)$ is the path loss as a function of the distance d among interferer and victim.

- $OCR(\Delta f)$ is the Off Channel Rejection factor that takes into account the capability of the victim receiver to reject the interferer signal for a given frequency offset among the two systems and is calculated as described in [25][3].

- DC takes into account the transmitter duty cycle.

[3] OCR depends on the power spectral density of the interfering signal and on the receiver IF filter frequency response.

Respect to an usual link-budget, the Off-Channel Rejection factor $OCR(\Delta f)$ term is added [26]. This term depends on the frequency separation of the interferer and the victim systems:

$$\Delta f = f_{TxU} - f_{Rx} \tag{2.2}$$

Where f_{TxU} is the interferer tuned frequency and f_{Rx} is the victim receiver tuned frequency (Hz). $OCR(\Delta f)$gives an estimate of the rejection provided by the receiver to an unwanted transmitted signal. Analytically $OCR(\Delta f)$ is given by [26]. In particular, it is possible to see that OCR is the ratio between the portion of interferer power actually "captured" by the receiver and the total interferer power.

$$OCR(\Delta f) = -10 \log \frac{\int_{+\inf}^{-\inf} P(f) \left| H(f + \Delta f) \right|^2 df}{\int_{+\inf}^{-\inf} P(f) df} \tag{2.3}$$

Where:

- $P(f)$ is the power spectral density of interfering signal (W/Hz);
- $H(f)$ is the IF frequency response of the receiver.

Its computation requires the knowledge of the power spectral density of interferer signal that depends on characteristics as modulation, power of the signal, pulse shape, bandwidth and the frequency response of the victim receiver. A low overlapping between interferer signal spectrum and receiver response leads to a large attenuation of interference power suffered by the victim system and vice versa. Power depends on path-loss and OCR and consequently on distance and frequency separation between interferer and victim equipment. Varying this two parameters, undesired signal level can be calculated and plotted.

Step 2. Maximum tolerable interference level assessment.
The maximum interference level tolerable at the victim receiver can be determined taking into account the minimum acceptable C/I ratio (also known as protection ratio) that is the minimum power ratio between the desired signal and the interferer one that guarantees acceptable performance. This value is usually indicated in the system standard as reference value. C/I is used to derive the maximum tolerable interference level at cell edge by setting the value of C equal to the receiver sensitivity, i.e. the minimum level of the desired signal that permits to the receiver to decode the signal. An eventual safe margin can be added to C, if foreseen by the standard.

Step 3. Coexistence results.
Comparing the results obtained by interference calculation and maximum tolerable interference level, several considerations can be drawn. In particular, it is possible to determine the minimum frequency and geographic separation that shall be maintained between different systems to guarantee their coexistence and correct operation.

In some cases the receiver protection criteria are not available. For L-DACS systems they have not been completely identified yet. In this case it is possible to consider a range of probable values, and, using the results coming from step 1, produce graphs of the C/I ratio varying with spatial and frequency distance. These graphs can be used for achieving evaluations on L-DACS channel allocation and frequency planning considerations. Some preliminary considerations can be achieved by hypothesizing that features of similar systems like B-AMC, WiMAX or GSM can be retained applicable to L-DACS.

Working hypotheses

Transmission power, antenna gain, cable loss, receiver sensitivity

Data required to perform link budget evaluations are derived from standard, specifications and products datasheets; all assumptions are intended to model the most general and widespread operative conditions of systems under evaluation. For the sake of brevity, these data are not reported here.

Scenarios

For the numerical analysis, several cross-interference scenarios have been detected assuming the considered systems either as victim or as interferer. Different configurations depend on the type of link considered (FL or RL) and, hence, on the location of the transmitter/receiver: on board or at ground.

Table 4.1 summarizes the scenarios involving respectively SSR and JTIDS, and DME analysis.

#	**Victim**	**Interferer**
1	FL LDACS1&2	SSR reply/1090ES/TCAS reply
2	FL LDACS1&2	JTIDS
3	SSR interrogation	RL LDACS1&2
4	TCAS reply/1090ES	RL LDACS1&2
5	FL LDACS1&2	SSR interrogation
6	FL LDACS1&2	JTIDS
7	SSR interrogation	FL LDACS1&2
8	TCAS reply/1090ES	FL LDACS1&2
9	RL LDACS1&2	SSR reply/TCAS reply/1090ES

#	Victim	Interferer
10	RL LDACS1&2	JTIDS
11	SSR reply	RL LDACS1&2
12	RL LDACS1&2	SSR interrogation
13	RL LDACS1&2	JTIDS
14	SSR reply/1090ES	FL LDACS1&2
15	FL LDACS1&2	SSR reply/1090ES/TCAS reply
16	FL LDACS1&2	TCAS interrogation
17	SSR interrogation	RL LDACS1&2
18	1090ES/TCAS reply	RL LDACS1&2
19	TCAS interrogation	RL LDACS1&2
20	FL LDACS1&2	1090 ES/TCAS Reply
21	FL LDACS1&2	TCAS Interrogation
22	FL LDACS1&2	JTIDS
23	SSR Interrogation	RL LDACS1&2
24	FL LDACS1&2	JTIDS
25	FL LDACS1&2	SSR Interrogation
26	SSR Interrogation	FL LDACS1&2
27	1090ES/TCAS reply	FL LDACS1&2
28	TCAS Interrogation	FL LDACS1&2
29	RL LDACS1&2	1090ES/TCAS Reply
30	RL LDACS1&2	TCAS Interrogation
31	RL LDACS1&2	JTIDS
32	1090ES/SSR Reply	RL LDACS1&2
33	1090ES/SSR interrogation	RL LDACS1&2
34	DME/TACAN reply	RL LDACS1&2
35	FL LDACS1&2	DME/TACAN Reply
36	DME/TACAN reply	FL LDACS1&2
37	DME/TACAN Reply	FL LDACS1&2
38	DME/TACAN interrogation	RL LDACS1&2
39	DME/TACAN Reply	RL LDACS1&2
40	DME/TACAN interrogation	FL LDACS1&2
41	FL LDACS1&2	DME/TACAN Interrogation
42	DME/TACAN Interrogation	RL LDACS1&2
43	FL LDACS1&2	DME/TACAN Interrogation
44	DME/TACAN Reply	RL LDACS1&2
45	FL LDACS1&2	DME/TACAN Interrogation
46	DME/TACAN Reply	FL LDACS1&2
47	RL LDACS1&2	DME/TACAN Interrogation
48	DME/TACAN Interrogation	RL LDACS1&2
49	RL LDACS1&2	DME/TACAN Air/Air Reply
50	FL LDACS1&2	DME/TACAN Air/Air Reply
51	DME/TACAN Air/Air Reply	FL LDACS1&2
52	DME/TACAN Air/Air Reply	RL LDACS1&2

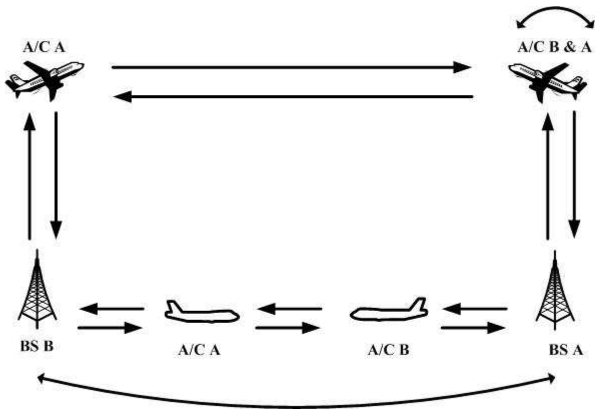

Figure 2-2: Interference paths.

Operational physical distance

The physical distance between victim receiver and interference source are defined taking into account the specific scenario. In particular it is important to detect the minimum operating distance to which the co-existence of the two systems should be guaranteed.
When victim receiver and interference source are on two airborne aircraft the minimum distance is given by the horizontal and vertical separation indicated by the flying rules.

- minimal vertical separation is fixed at 300 mt (1000 ft;
- minimal horizontal separation depends on different factors (high, visibility, type of instruments etc.) typically 1500 mt can be considered.

When victim receiver is on the ground and the interference source is on an airborne aircraft (and vice versa) the minimum distance is given by the lower distance of the airborne aircraft to the soil that is 300 mt.

Table 2.2: LDACS1 Sub-bands.

Sub-band		
B1 (FL/RL)	963.5	970.5 MHz
B2 (FL)	985.5	1008.5 MHz
B3 (RL)	1048.5	1071.5 MHz
B4 (FL/RL)	1149.5	1156.5 MHz

When victim receiver and interference transmitter are on the ground the minimum distance is almost 30 metres.
When victim receiver and interference transmitter are on the same aircraft (co-site) the minimum distance has been supposed being 15 metres.
These operational distances will be compared with the minimum physical distance at which the interference level is lower than the maximum acceptable value in order to evaluate if the coexistence is possible without limitations or not.

LDACS1/2 frequency allocation

As a first hypothesis the LDACS1&2 frequency allocation foreseen by the ongoing specifications will be considered.
For L-DACS1 four different sub-bands have been foreseen (Table 2.2). Instead for LDACS2 frequency allocation proposed is in the range: 960 975 MHz.

LDACS1/2 Pre-selection filters

Pre-selection filters (RF filters) can be considered in order to reduce the effects of interferer out-of-band energy emissions that are captured by the IF filter and can affect its capacity of interference rejection. However, pre-selection filters are strongly dependent on the radio system implementation. Assuming different pre-selection filters the receiver OCR factor changes significantly. In Figure 2-3 and 2-4 the LDACS1&2 receiver Off-Channel Rejection (OCR) factor is plotted with respect to the frequency offset between interferer transmitter and victim receiver assuming different pre-selection filters:

- not present;
- rectangular filters with band of 2,4,9 and 12 MHz and attenuation 70dB;
- the filter proposed as an alternative to the duplexer in LDACS1 specification document [16] for the realization of prototype apparatus;
- a commercial filter used by GSM [27].

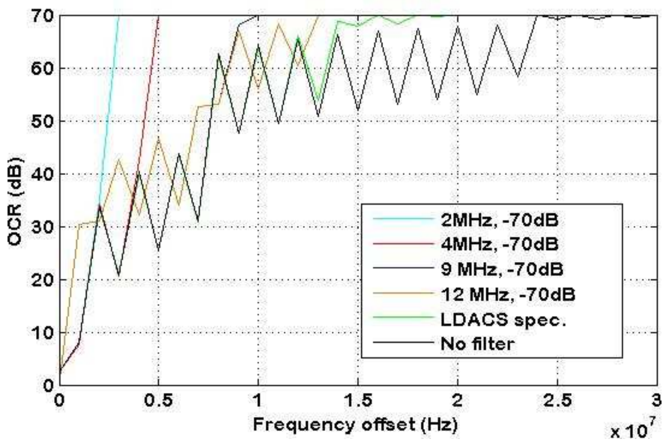

Figure 2-3: OCR factor of the LDACS1.

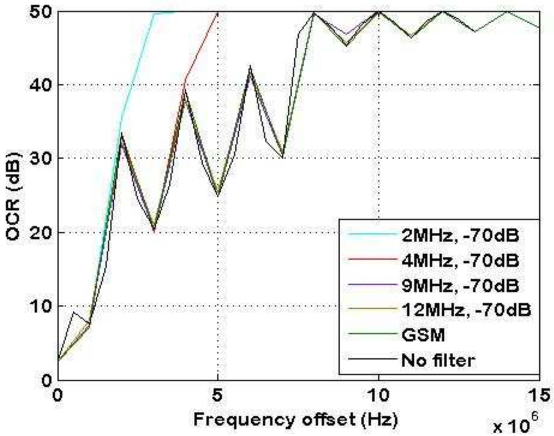

Figure 2-4: OCR factor of the LDACS2.

From Figure 2-3 2-4 it is possible to see that the effect of pre-selection filter can be very different, depending on the filter selected in the implementation. For this reason, in the following quantitative analysis the worst and the best cases will be taken into account for LDACS1/2 receivers:

- pre-selection filter not present;
- a rectangular filter with band 2MHz and attenuation 70dB.

Duty cycle

In the worst case the maximum level of interference collected by victim receiver is calculated considering the maximum possible power transmitted by the interferer. However, since the transmission is not continuous corrective term *DC* is added into the link budget formula that represents the power level reduction due to duty cycle effects.
Given a value of Duty Cycle *dc* (in percentage), the DC term is calculated as:

$$DC = 10 \cdot \log_1 0 \left(\frac{dc}{100} \right)$$

SSR modes simulation

As detailed in sec. 2.1.3, SSR system can operate with different modes:

- Mode A&C interrogation and reply;
- Mode S interrogation and reply;
- Intermode interrogation.

In addition SSR waveforms are used by 1090ES and TCAS systems.

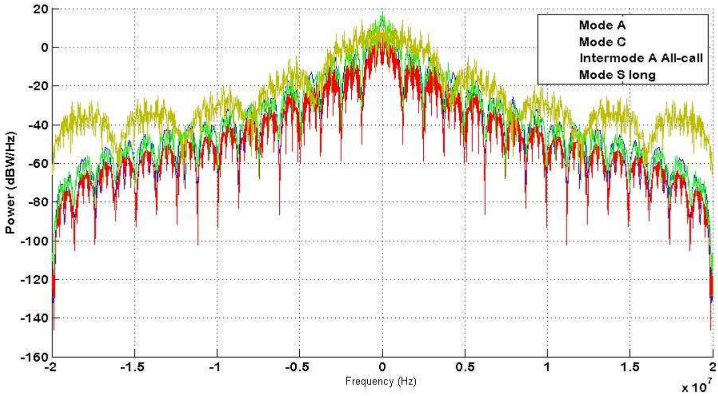

Figure 2-5: SSR interrogation signals in the frequency domain (modes A,C,S and intermode).

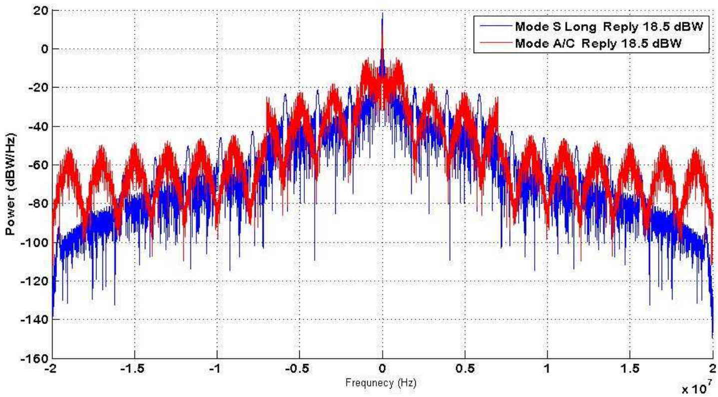

Figure 2-6: SSR reply signals in the frequency domain (modes A/C and S).

In Figure 2-5 the signal spectra of different interrogation modes are represented considering different frequency ranges. For Mode S the data message can have two different lengths (either in the interrogation or in the reply), here the "long" message is taken into account. From these figures is possible to see that Mode S interrogation has a wider main lobe and higher side lobes than mode A/C. In addition Mode S interrogation presents higher spikes.

Figure 2-6 represents different modes reply. From this figure its possible to see that the two modes reply have similar behaviour, the mode A/C main beam spectrum width is slightly greater than Mode S one (2.22 MHz vs.2 MHz) and the side lobes power is slightly higher than mode S. However Mode S spectrum shape is less linear showing marked side lobes spaced 2 MHz each other. This is due to fact the mode S reply waveform is more complex than Mode A/C reply with a data message field in which pulses representing bits are modulated PPM rather than PAM. The Mode S spectrum spikes introduce a higher interference in the victim signal as can be seen from the behaviour of the interference rejection factor (OCR) provided by LDACS1&2 receiver to Mode S and Mode A/C replies represented in Figure 2-8 and Figure 2-10. Except for few values of the frequency offset between the two systems the rejection of Mode S signal is lower that Mode A/C and it means LDACS1&2 receiver suffers higher interference when Mode S is active. The same considerations can be drawn also observing the interference rejection factor of LDACS1&2 system to Mode S and Mode A/C interrogations in Figure 2-7 and Figure 2-9.

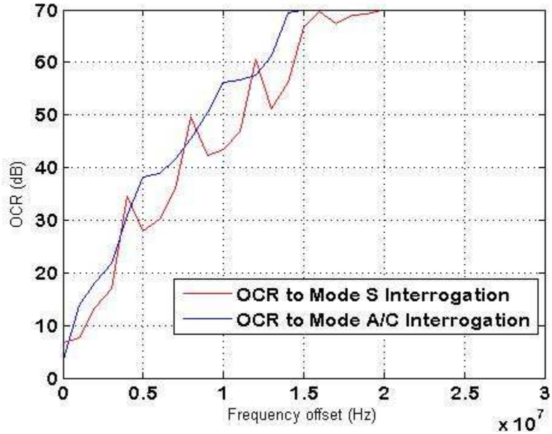

Figure 2-7: LDACS1 receiver Off-Channel Rejection Factor. SSR Interrogation Modes A/C and S.

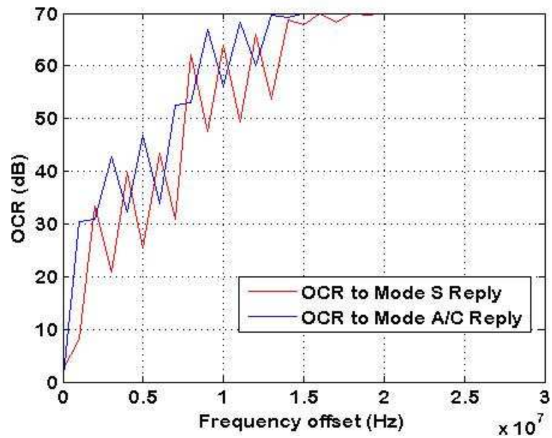

Figure 2-8: LDACS1 receiver Off-Channel Rejection Factor. SSR Reply Mode A/C and S.

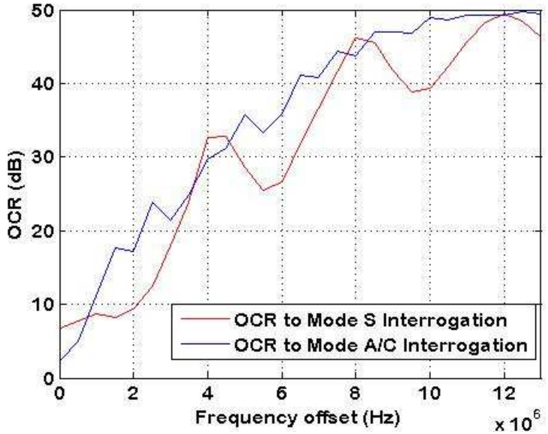

Figure 2-9: LDACS2 receiver Off-Channel Rejection Factor. SSR Interrogation Modes A/C and S.

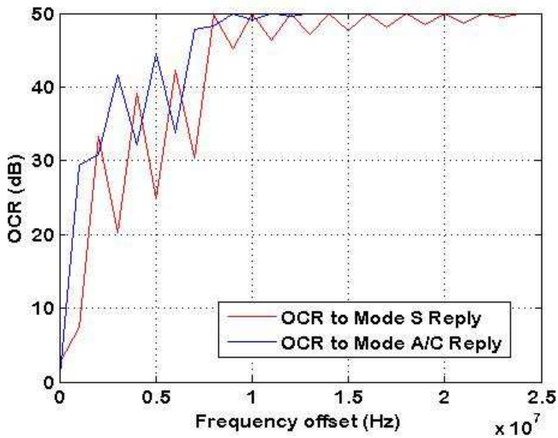

Figure 2-10: LDACS2 receiver Off-Channel Rejection Factor. SSR Reply Mode A/C and S.

This analysis shows that Mode S represents the worst case in terms of interference produced on a LDACS1&2 victim receiver. In order to simplify the following analysis without loss of generality we assume:

- when the SSR is the interference source the SSR Mode S signal will be considered in the calculation of the OCR factor term because it corresponds to the worst case.
- when the SSR is the victim receiver there is no difference between different SSR modes.

2.3 Numerical analysis results

Due to the considerable number of scenario taken into account, not all results will be reported here. Extensive analysis results will be presented only for first scenario listed in Table 4.1, while other results are outlined in Sec. 2.3.1.

Scenario 1: Airborne Aircraft to Airborne Aircraft.
Victim system: LDACS1 FL
Interferer systems: SSR reply/ 1090ES/TCAS reply.

In this case the Victim Receiver is a LDACS1 Receiver on an aircraft during flight; the Interferer Transmitter is onboard another aircraft during flight. In the following figures the interference power level affecting the victim receiver is compared with the maximum tolerable power level value, allowing detecting the minimum required spatial distance between interferer and victim stations in the considered scenario.

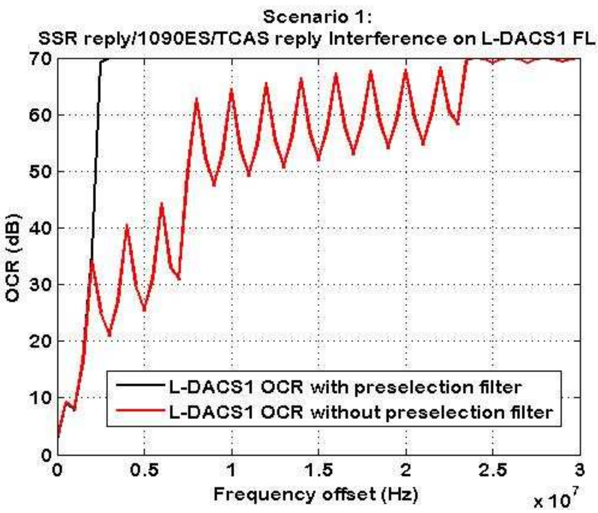

Figure 2-11: Scenario 1a: Victim Receiver OCR.

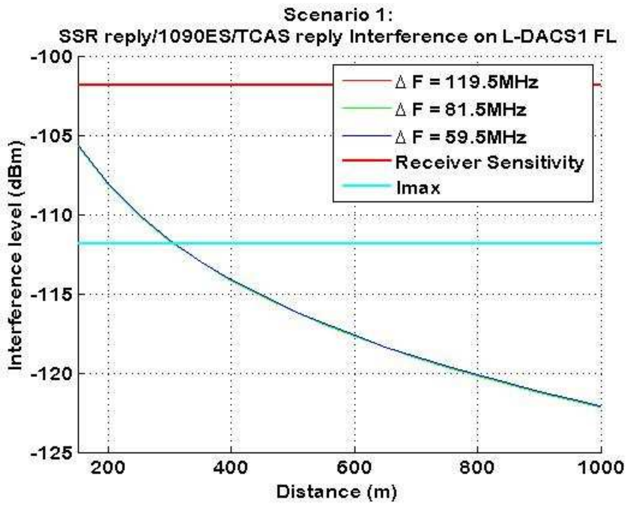

Figure 2-12: Scenario 1a: LDACS1 Frequency Allocation, with Preselection Filter.

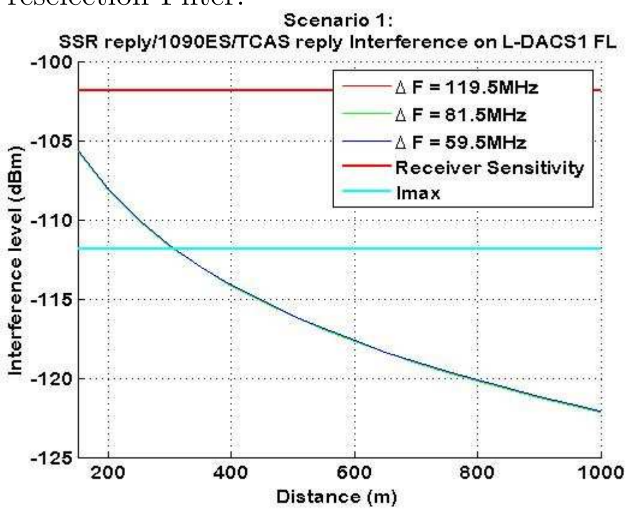

Figure 2-13: Scenario 1a: LDACS1 Frequency Allocation, without Preselection Filter.

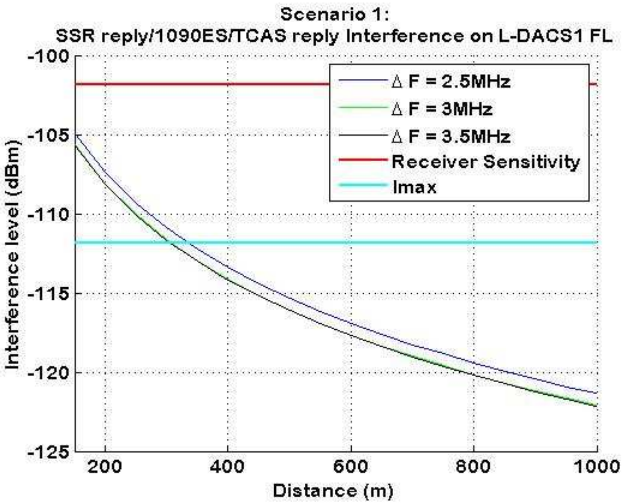

Figure 2-14: Scenario 1a: LDACS1 Free Frequency Allocation, with Preselection Filter.

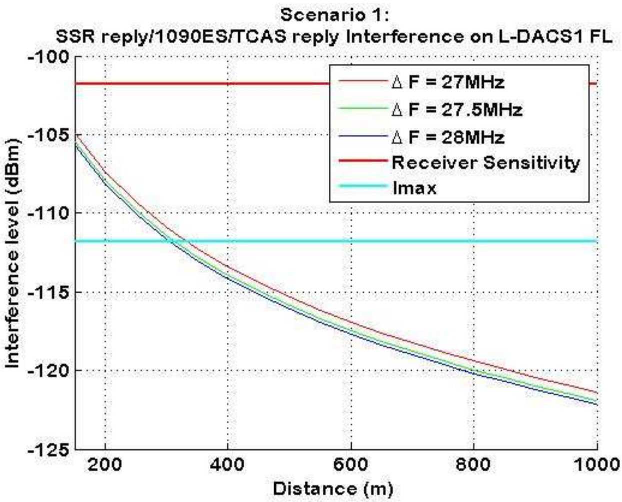

Figure 2-15: Scenario 1a: LDACS1 Free Frequency Allocation, without Preselection Filter.

From Figures 2-11, 2-12 and 2-13 it is evident that at frequency offset sufficiently high the OCR term is constant for any of the considered frequency offsets, independently from the presence of the preselection filter. In other words the preselection filter results non influential on interference level.

Another possible approach in showing the results is described in Figures 2-14 and 2-15: no LDACS1 frequency allocation hypothesis has been made, and the frequency offset has been left varying. The minimum required Frequency offset is detected, together with the minimum required spatial distance. There is no difference in results for Frequency Offsets higher than 3 MHz.

Figure 2-16 represents the minimum required spatial distance between interferer and victim stations in the considered scenario, in the various possible LDACS1 frequency allocations (B1, B2, B4).

	With Preselection Filter	Without Preselection Filter
B1 ($\Delta f = 119.5MHz$)	$d \geq 300m$	$d \geq 300m$
B2 ($\Delta f = 81.5MHz$)	$d \geq 300m$	$d \geq 300m$
B4 ($\Delta f = 59.5MHz$)	$d \geq 300m$	$d \geq 300m$
Free Frequency Allocation	$d \geq 300m$ $\Delta f = 3MHz$	$d \geq 300m$, $\Delta f = 27.5MHz$

Figure 2-16: LDACS1 Scenario 1a results.

Victim system: LDACS1 FL
Interferer systems: SSR reply/ 1090ES/TCAS reply.

Similarly,here are reported results relative to scenario that involves LDACS2.

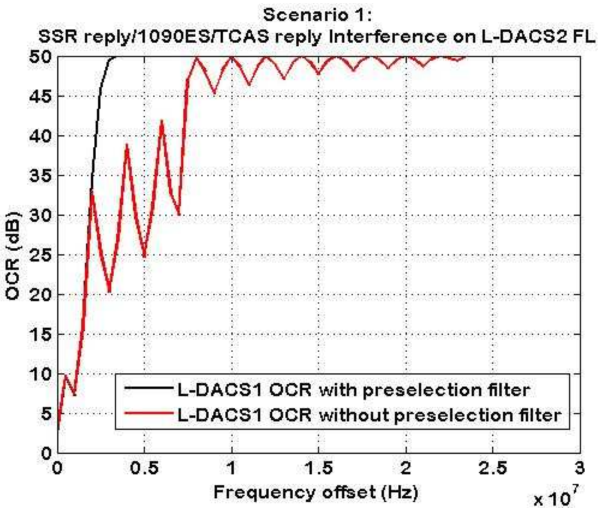

Figure 2-17: Scenario 1a: Victim Receiver OCR.

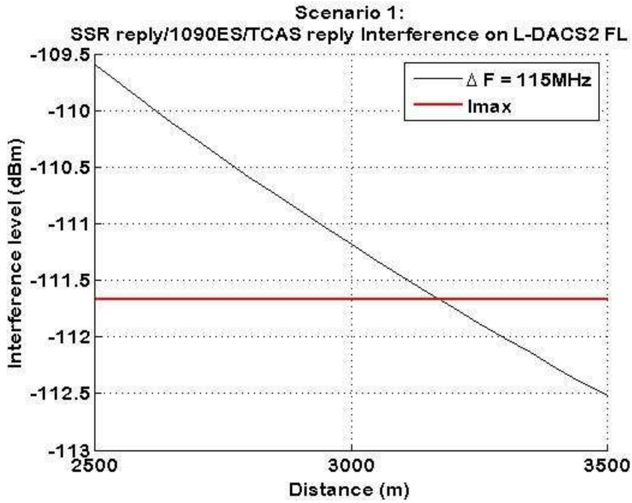

Figure 2-18: Scenario 1a: LDACS2 Frequency Allocation, with Preselection Filter.

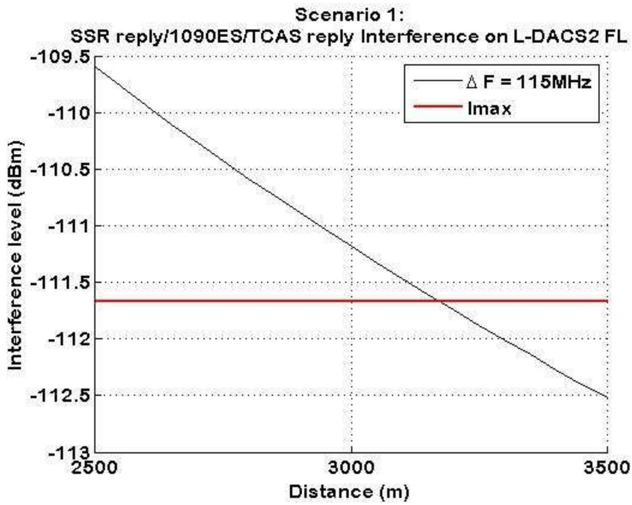

Figure 2-19: Scenario 1a: LDACS2 Frequency Allocation, without Preselection Filter.

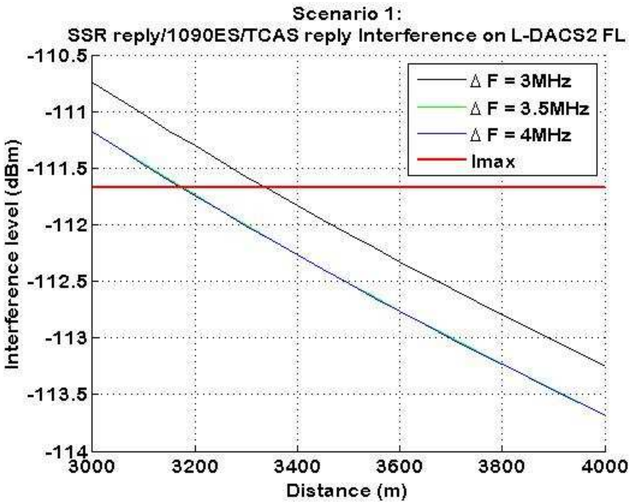

Figure 2-20: Scenario 1a: LDACS2 Free Frequency Allocation, with Preselection Filter.

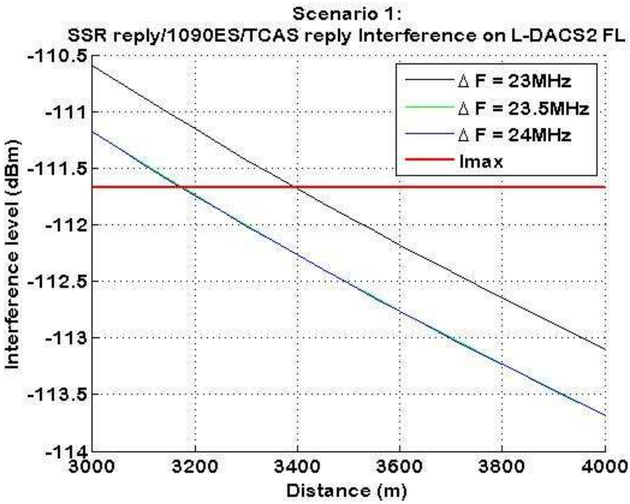

Figure 2-21: Scenario 1a: LDACS2 Free Frequency Allocation, without Preselection Filter.

	With Preselection Filter	Without Preselection Filter
L-DACS2 Frequency Allocation ($\Delta f = 115MHz$)	$d > 3180m$	$d > 3180m$
Free Frequency Allocation	$d > 3180m$, $\Delta f = 3.5MHz$	$d > 3180m$, $\Delta f = 23.5MHz$

Figure 2-22: LDACS2 Scenario 1a results.

2.3.1 Conclusions

The analysis of coexistence and compatibility of LDACS with other L-Band system is quite complex due to the high number of interference configurations that arise from each scenario that can differ each other because the high number of involved systems and of parameters variation. However, some considerations can be drawn.
Without pre-selection LDACS is totally unable to operate in the entire L-Band whatever is the distance with the interferer equipment. In general, the choose and implementation of this filter is demanded to manufacturers, system specification could recommend its use and its general characteristics. Moreover, the sub bands pointed out from EUROCONTROL for the allocation have been chosen following the indication and preferences of all nations that will be interested in LDACS deployment and in any case constitute a case more favorable respect to free frequency allocation in terms of distance that will be respected between involved equipment. For these reasons, it is reasonable to consider as most interesting the results relative to the simulation that takes into account the use of pre-selection filter and recommanded frequency allocations. Figures 2-23, 2-24, 2-25, and 2-26 summarize the scenarios in which the coexistence has been proved, under the assumptions made.

LDACS1 Interferer		
Victim system	**Compatibility**	**No Compatibility**
1090ES	In the following scenarios: -LDACS1 Airborne aircraft interferes on 1090ES airborne aircraft - LDACS1 GS interferes on 1090ES GS - Co-site	In the following scenarios: - "LDAC1 GS interferes on 1090ES airborne aircraft" - "LDACS1 GS interferes on 1090ES aircraft on the ground" - "LDACS1 Aircraft on the ground interferes on 1090ES GS"
SSR Reply	All scenarios except when LDACS1 aircraft on the ground interferes on SSR GS	When LDACS1 aircraft on the ground interferes on SSR GS
SSR Interrogation	All scenarios except when LDACS1 GS interferes on SSR aircraft on the ground	When LDACS1 GS interferes on SSR aircraft on the ground
TCAS Reply	All scenarios except when LDACS1 GS interferes on TCAS aircraft on the ground	When LDACS1 GS interferes on TCAS aircraft on the ground
TCAS Interrogation	Co-site	L-DACS1 GS interferes on TCAS aircraft on the ground
DME/TACAN	In the following scenarios: - LDACS1 Airborne aircraft interferes on DME/TACAN airborne aircraft on B1 or B3 (B3 only in case of DME Y channels) - LDACS1 GS interferes on DME/P Airborne aircraft on B1 and B3 (only in case of DME Y-channels) - LDACS1 airborne aircraft interferes on DME/TACAN GS on B1 and B2 - - - LDACS1 aircraft on the ground interferes on DME/TACAN aircraft on the ground on B1 or B3 (B3 only in case of DME Y channels) - - - LDACS1 GS interferes on DME/P Airborne aircraft in Air/Air Mode on B1 or B3 (B3 only in case of DME Y channels) - LDACS1 airborne aircraft interferes on DME/TACAN airborne aircraft in Air/Air Mode on B1 or B3 (B3 only in case of DME Y channels)	In the following scenarios: - LDACS1 Airborne aircraft interferes on DME/TACAN airborne aircraft on B2, B3 (B3 in case of DME X channels) and B4 - LDACS1 GS interferes on DME-N/TACAN airborne aircraft on B1, on any kind of DME/TACAN airborne aircraft on B2, on any kind of DME/TACAN airborne aircraft on X channels on B3, on DME/N-TACAN airborne aircraft on Y-channels on B3, on any kind of DME/TACAN airborne aircraft on B4 - LDACS1 airborne aircraft interferes on any kind of DME/TACAN GS on B3 and B4 - LDACS1 GS interferes on any kind of DME/TACAN GS on B1, B2, B3 and B4 - LDACS1 airborne aircraft interferes on any co-site DME/TACAN airborne aircraft on B1, B2, B3, and B4 - LDACS1 aircraft on the ground interferes on any kind of DME/TACAN aircraft on the ground on B2, B3 for X Channels, B4 - LDACS1 GS interferes on any kind of DME/TACAN aircraft on the ground on any band B1, B2, B3, B4 - LDACS1 aircraft on the ground interferes on GS DME/TACAN on B1, B2, B3 and B4 - LDACS1 GS interferes on Air-Air DME/N-TACAN airborne aircraft on B1, on any kind of Air-Air DME/TACAN airborne aircraft on B2, on any kind of Air-Air DME/TACAN airborne aircraft (using X-channels) on B3, on Air-Air DME/N-TACAN airborne aircraft (using Y channels) on B3, on any kind of Air-Air DME/TACAN airborne aircraft on - LDACS1 airborne aircraft interferes on any kind of Air-Air DME/TACAN airborne aircraft on B2, on any kind of Air-Air DME/TACAN airborne aircraft on B3 (X-Channels), on any kind of Air-Air DME/TACAN airborne aircraft on B4

Figure 2-23: LDACS1 as interferer. Overall scenarios results.

L-DACS1 Victim		
Interfering system	**Compatibility**	**No-Compatibility**
SSR Reply	No Scenario	All Scenarios
SSR Interrogation	No Scenario	All Scenarios
1090ES / TCAS Reply	No Scenario	All Scenarios
TCAS Interrogation	No Scenario	All Scenarios
JTIDS/MIDS	No Scenario	All Scenarios
DME/TACAN	In the following scenarios:	In the following scenarios:
	- Any kind of DME/TACAN airborne aircraft interferes on LDACS1 airborne aircraft on B1 and B2	- Any kind of DME/TACAN airborne aircraft interferes on LDACS1 airborne aircraft on B3 and B4
	- DME/P GS interferes on LDACS1 airborne aircraft on B1 or on B3 with DME/P transmitting in Y channel	- DME/N-TACAN GS interferes on LDACS1 airborne aircraft on B1, any kind of DME/TACAN GS interferes on LDACS1 airborne aircraft on B2 or B3 (B3 with DME transmitting in X channels), DME/N-TACAN GS interferes on LDACS1 airborne aircraft on B3 (B3 with DME transmitting in Y channels), any kind of DME/TACAN GS interferes on LDACS1 airborne aircraft on B4
	- DME/TACAN airborne aircraft interferes on LDACS1 GS on B1 and B2	- Any kind of DME/TACAN airborne aircraft interferes on LDACS1 GS on B3 and B4
	- Any kind of DME/TACAN airborne aircraft interferes on co-site LDACS1 airborne aircraft on B1 and B2	- Any kind of DME/TACAN GS interferes on LDACS1 GS on any Band (B1, B2, B3, B4)
	- Any kind of DME/TACAN aircraft on the ground interferes on LDACS1 aircraft on the ground on B1 and B2	- Any kind of DME/TACAN airborne aircraft interferes on co-site LDACS1 airborne aircraft on B3 and B4
	-	- Any kind of DME/TACAN aircraft on the ground interferes on LDACS1 aircraft on the ground on B3 and B4
	- Any kind of DME/TACAN aircraft on the ground interferes on LDACS1 GS on B1 and B2	- Any kind of DME/TACAN GS interferes on LDACS1 aircraft on the ground on any Band (B1, B2, B3, B4)
	- DME airborne aircraft in Air/Air Mode interferes to LDACS1 GS on B1 and B2	- Any kind of DME/TACAN aircraft on the ground interferes on LDACS1 GS on B3 and B4
	- DME airborne aircraft in Air/Air Mode interferes to LDACS1 airborne aircraft on B1 and B2	- DME airborne aircraft in Air/Air Mode interferes to LDACS1 GS on B3 and B4
	-	- DME airborne aircraft in Air/Air Mode interferes to LDACS1 airborne aircraft on B3 and B4

Figure 2-24: LDACS1 as victim. Overall scenarios results.

LDACS2 Interferer		
Victim system	**Compatibility**	**No Compatibility**
1090ES	only when an airborne aircraft LDACS2 interferes with a airborne aircraft 1090ES	In all the scenarios, except when an airborne aircraft LDACS2 interferes with a airborne aircraft 1090ES
SSR Reply	No Scenario	All Scenarios
SSR Interrogation	All scenarios except when LDACS GS interferes on SSR aircraft on the ground	when LDACS2 GS interferes on SSR aircraft on the ground
TCAS Reply	In the following scenarios: - Airborne aircraft LDACS2 to airborne aircraft TCAS - LDACS2 GS to TCAS Airborne aircraft	In the following scenarios: - Co-site - LDACS2 GS to TCAS aircraft on the ground
TCAS Interrogation	No Scenario	All Scenarios
DME/TACAN	In the following scenarios: - LDACS2 Airborne aircraft interferes on DME/P Airborne aircraft	In the following scenarios: - LDACS2 Airborne aircraft interferes on DME/N-TACAN Airborne aircraft
	-	- LDACS2 GS interferes to any kind of DME/TACAN airborne aircraft
	- LDACS2 airborne aircraft interferes on any kind of DME/TACAN GS	-
	-	- LDACS2 GS interferes to any kind of DME/TACAN GS
	-	- LDACS2 airborne aircraft interferes on any kind of DME/TACAN co-site airborne aircraft
	-	- LDACS2 aircraft on the ground interferes on any kind of DME/TACAN aircraft on the ground
	-	- LDACS2 GS interferes on any kind of DME/TACAN aircraft on the ground
	-	- LDACS2 aircraft on the ground interferes on any kind of DME/TACAN GS
	-	- LDACS2 GS interferes on any kind of Air/Air DME airborne aircraft
	- LDACS2 airborne aircraft interferes on DME-P Air/Air airborne aircraft	- LDACS2 airborne aircraft interferes on DME-N/TACAN Air/Air airborne aircraft

Figure 2-25: LDACS2 as interferer. Overall scenarios results.

L-DACS2 Victim		
Interfering system	**Compatibility**	**No Compatibility**
SSR Reply	No Scenario	All Scenarios
SSR Interrogation	No Scenario	All Scenarios
1090ES / TCAS Reply	No Scenario	All Scenarios
TCAS Interrogation	No Scenario	All Scenarios
JTIDS/MIDS	No Scenario	All Scenarios
DME/TACAN	No Scenario	All Scenarios

Figure 2-26: LDACS2 as victim. Overall scenarios results.

Chapter 3

JTIDS Interference on LDACS1

3.1 Introduction

As a consequence of results of the study depicted in Chapter 2 showing that LDACS is subject to harmful interference produced by L-Band legacy systems, we examined in depth the interference effects of Joint Tactical Information Distribution System/Multi-functional Information Distribution System (JTIDS/MIDS)[1] on LDACS-1 system.

JTIDS is a military radio technology employed as digital data link used for several purposes such as mission management, identification in surveillance and air control. Its signal results to be very robust to interference and jamming thanks to the adoption of appropriate countermeasures like frequency hopping and spread spectrum coding. On the contrary, JTIDS interference on LDACS-1 signal can be very

[1]MIDS is the NATO implementation of JTIDS.

disruptive due to its high transmission power. In addition, JTIDS frequency hopping is performed over a large range of frequencies spanning almost the whole L-Band therefore the probability of having collisions between the LDACS1 and the JTIDS signal is very high.
Impulsive noise on an OFDM system is spread over all the subcarriers by means of the DFT (Discrete Fourier Transform). This is an advantage with respect to single carrier systems until the interference power is low but becomes a disadvantage if impulsive noise becomes high. In this case OFDM is vulnerable to impulsive noise that gives rise to a strong performance loss [28], [29]. In general, without any interference mitigation the OFDM system is not able to meet the QoS requirements especially in the case of high data rate systems.
Influence of interference of JTIDS on LDACS1 was previously investigated in [30], where the LDACS1 performance in terms of Bit Error Rate (BER) is evaluated considering a pulse blanking mitigation technique and ideal interference detection. More in general impulsive interference on OFDM systems is a well-known problem in Power Line Communications (PLCs), and recently is investigated also for wireless systems. In fact the increasing number of wireless devices and systems leads to a dense spatial reuse and severe co-channel interferences that can be impulsive. Main solutions to counteract this type of interference are clipping and blanking [31–33]. In both methods impulsive noise is detected if the signal amplitude exceeds a threshold. Blanking technique sets to zero the samples where the noise has been detected and signal is removed together with noise. Clipping method truncates the amplitude of the samples where the noise is present to reduce its impact. Both are nonlinear operations

and the result is that OFDM signal is no longer orthogonal and consequent Inter-Carrier Interference (ICI) degrades the performance especially in multi-user communications systems, requiring additional countermeasures [34]. Some joint advanced schemes have been proposed. In [35] author provides an improvement of blanking nonlinearity through an iterative reduction of ICI caused by the blanking operation. In [36] and [37] impulsive noise suppression techniques in conjunction with frequency domain equalization are investigated. Alternative schemes present a noise reduction based on data detection (i.e., decision directed) and cancellation in order to estimate and remove the noise. These schemes are quite effective but present high computational cost [38] [39].

This work proposes a new scheme to detect and remove the impulsive noise based on the joint use of spectrum sensing per sample, symbols retransmission and soft combining. It presents low complexity and is suitable when the impulsive noise is often present, affecting many symbols and, hence, making interleaving techniques [40] ineffective.

In the first phase impulsive noise is detected by resorting to a spectrum sensing algorithm based on energy detector. In the context of Cognitive Radio different sensing techniques have been proposed to detect interference in [41], [42], among them the energy detector presents low computational cost and is the most useful when the signal to be detected is unknown [43]. After the interference sensing phase the proposed method foresees the joint use of symbols retransmission and blanking operation. Following an HARQ (Hybrid Automatic Repeat reQuest) principle [44] when the JTIDS interference is detected the symbol is transmitted two times: the first copy of the symbol is not discarded but stored and soft combined with its new received copy. Before the combining operation

Table 3.1: LDACS1 possible allocation bands (Bn with n=1,...,4).

B1	964 – 970 MHz
B2	985 – 1009 MHz
B3	1048 – 1072 MHz
B4	1150 – 1156 MHz

the samples affected by impulsive noise are zeroed. In that way the impulsive interference is removed without losing useful information. In addition the soft combining introduces a gain against the AWGN noise increasing the system performance. In particular two different alternatives are presented, called *full* and *partial combining*, that differs for the retransmission policy. The scheme is based on the idea that even if two consecutive copies of the same symbol are affected by impulsive noise, the samples affected by the interference are statistically independent.

3.2 System Model

In the system under consideration the LDACS1 signal is affected by the interference of a JTIDS signal operating on the same band. For the LDACS1 system we refer to the latest available specifications [45]. It is based on the FDD approach and is designed to operate in the lower part of the L-Band with a bandwidth of B_{LDACS} = 498.05 kHz. The possible allocation bands foreseen for the system, either for Forward Link (FL) and Reverse Link (RL), are reported in Tab. 3.1.

LDACS1 is an OFDM-based communication system where the available spectrum is shared between a large number of subcarriers. Data symbols are efficiently modulated on these subcarriers by resorting to the use of the Inverse DFT (IDFT) in transmission and the DFT at the receiving end. According to [45], we consider an OFDM system with a total number of available subcarriers, N, equal to 64 where only $N_u = 50$ are used to carry information. The bit stream is first mapped into complex QPSK symbols v_k $k = 0, \cdots, N_u$. After a serial to parallel (S/P) conversion, the block of N_u symbols $\mathbf{v} = [v_0, \cdots, v_{N_u-1}]$ is transformed into a block of N symbols $\mathbf{c} = [c_0, \cdots, c_{N-1}]$ inserting $(N - N_u)$ zeros in correspondence of the first and the last symbols.

The i-th transmitted OFDM symbol results to be

$$s_i(n) = \sum_{k=0}^{N-1} c_k \cdot e^{j\frac{2\pi}{N}nk} \quad \text{for} \quad \begin{cases} n = 0, ..., N-1 \\ i = -\infty, \cdots, \infty \end{cases} . \qquad (3.1)$$

Then, a cyclic prefix of length N_g equal to 11 samples is inserted by repeating the final part of the OFDM symbol, $s_i(n)$, resulting in the transmitted signal $\hat{s}_i(n)$ with duration T_s equal to 120 μs. The base-band signal is up-converted to the radio frequency and transmitted through the channel. We consider a frequency selective multipath fading channel with discrete impulse response $h_i(n)$ shorter than the cyclic prefix, so that, by removing the cyclic prefix at the receiving end we avoid ISI. By using the cyclic prefix, the convolution between the transmitted signal and $h_i(n)$ is a cyclic convolution, hence, the down-converted i-th received symbol can be defined as:

$$\hat{r}_i(n) = \hat{s}_i(n) \otimes h_i(n) + n_i(n) + u_i(n) \qquad (3.2)$$

where $n_i(n)$ is white Gaussian noise term, with zero mean and variance $N_0/2$ introduced by the communication channel and $u_i(n)$ represents the impulsive noise due to the JTIDS signal as described in the following. The useful received signal $r_i(n)$ is obtained by removing the cyclic prefix.

Then the signal $r_i(n)$ is processed by the DFT block. The DFT output for the k-th subcarrier, before equalization, is

$$\begin{aligned} Z_i(k) &= \frac{1}{N}\sum_{n=0}^{N-1} r_i(n) \cdot e^{-j\frac{2\pi}{N}nk} \\ &= S_i(k) \cdot H_i(k) + N_i(k) + U_i(k) \end{aligned} \tag{3.3}$$

where $S_i(k)$, $H_i(k)$, $N_i(k)$ and $U_i(k)$ are the samples of the transmitted symbol, channel impulse response, AWGN and JTIDS interference contribution, respectively, in the frequency domain, on the k-th subcarrier of the i-th symbol. The channel coefficients can be written as

$$H_i(k) = \alpha_i(k)e^{\phi_i(k)} \tag{3.4}$$

where $\alpha_i(k)$ and $\phi_i(k)$ are the attenuation and the phase rotation of the channel impulse response at the k-th subcarrier of the i-th symbol, respectively. We assume a fading channel whose amplitude varies with a Rice distribution, with Ricean factor equal to $K = 10$ dB and normalized power $\Omega = 1$.

The signal is equalized by the means of a Zero Forcing operation assuming perfect knowledge of the channel response.

JTIDS is a TDMA (Time Division Multiple Access) system operating in the frequency range $[960 - 1215]$ MHz [21]. Each time slot is composed by a sequence of N_p pulses of length $T_p = 6.4$ μs and spaced of $T_{int} = 6.6$ μs: each pulse contains 5 bits multiplied for a spreading sequence of 32 chips MSK (Minimum Shift Keying) modulated. The chip duration is 200 ns with a consequent bandwidth of 5 MHz. Then the total bandwidth is filtered and reduced to 3 MHz. Each pulse is transmitted over a different carrier frequency selected among $N_{ch} = 51$ possible values according to a specific hopping pattern. In particular the N_{ch} frequencies are spaced of 3 MHz from 969 MHz to 1206 MHz. JTIDS is a military system and the hopping sequences represent a classified information, for this reason in our model we consider randomly generated sequences where each frequency has the same probability to be selected. JTIDS has a bandwidth that is significantly higher than that of LDACS1, hence, when JTIDS transmits on frequencies near to the LDACS1 central frequency the entire LDACS1 signal is affected by the interference. One or two hopping frequencies can interfere with LDACS signal depending on its exact frequency allocation. We can conclude that JTIDS interference on LDACS1 signal is not deterministic but it depends on several parameters. In particular JTIDS transmission results to be intermittent in time and also in frequency (due to frequency hopping) and the probability that a JTIDS pulse interferers with LDACS1 signal can be expressed as:

$$P_p = \frac{q}{N_{ch}} \eta \frac{N_p * (T_p + T_{int})}{T_{slot}} \tag{3.5}$$

It depends on:

- the probability that JTIDS is transmitting on the hopping frequencies that are near to the LDACS1 central frequency that is $\frac{q}{N_{ch}}$ with $q = 1 or 2$. For simplicity here we do not consider multi-frequency JTIDS networks (i.e., simultaneous transmission on more frequencies following orthogonal hopping patterns, and, hence, $q > 2$);

- the probability that the JTIDS time slot is active that is $\frac{N_p(T_p+T_{int})}{T_{slot}}$, where:
  - $N_p = [72, 258, 444]$ is the number of pulses in a JTIDS time slot (the standard value is $N_p = 258$);
  - $T_{slot} = 7,8125$ ms is the overall JTIDS time slot duration, which includes an inactive period where the transmitter is in idle mode;

- the probability that the JTIDS time slot is effectively used that corresponds time slot duty factor (TSDF), that is the percentage of time slots occupied in a JTIDS frame. It depends on the number of users actives on the LDACS1 area (i.e. that can interferer): for a single user the maximum value[2] is 50% that can arrive to 100% for a higher number of users.

In order to have the same sampling frequency of the LDACS signal, the JTIDS pulse is decimated and then the two signals are summed at sample level. This decimation operation extends the pulse duration and reduces the amplitude, making the JTIDS signal more difficult to detect. Hence, when the power of the interfering signal is high, the decoding is difficult for all the samples carried by the OFDM symbol.

[2]This analysis considers the maximum values of the TSDF because they represent the worst case from an interference point of view.

3.3 Interference Sensing and Mitigation

This section describes the proposed sensing and interference method. In particular two different alternatives are presented. In the first one (named *full combining*) we suppose that all the symbols are transmitted twice and the interference sensing is performed on the signal resulting form combination of the two replicas. The second option (named *partial combining*), considers a scheme where only the OFDM symbols corrupted by the interference are retransmitted and, hence, the sensing is performed on the signal replicas before combining. In the first case it is possible to obtain a more reliable interference detection at the expense of a reduction of the transmission capacity. In the second case, interference detection is more affected by the useful signal fluctuations but it permits to save transmission capacity. In addition when JTIDS interference is present the probability that even the successive retransmissions of the OFDM symbols can be affected by interference is very high. In a system with an HARQ mechanism this should lead to a high number of retransmission and delays. Thanks to the joint use of blanking and retransmission this problem could be overcome.

The first operation of the proposed schemes is the interference detection that is used to determine which samples are corrupted. This is done by adapting a well known spectrum sensing scheme: the energy detector. This detector is well-known in the Cognitive Radio networks where a secondary user (SU) looks for spectrum holes that are not used by the primary licensed user (PU). Free spectrum portion can be used by the SU for transmission. Energy detector scheme is blind and does not require the knowledge of the interfering system features[3], in addition it presents very low complexity. In general it computes the energy of the samples received during a specified sensing period and compares it with a threshold which depends on the power of the noise at the receiver. If the energy is over the threshold, it is assumed that another system is transmitting. The detector performance depends on the threshold choice and the accuracy is proportional to the duration of the sensing period, which can be performed only for a limited time: if the number of collected samples is sufficient, it is possible to reach any desired performance even under low Signal to Noise Ratio (SNR) conditions. Usually during the sensing period the receiver exclusively listens the spectrum and does not receive any communication.

We propose a modified energy detector. The spectrum sensing is not performed on a dedicated interval but together with communication. In addition the goal is to know which samples are affected by the JTIDS interference instead of detecting spectrum holes. By adapting the sensing concept, it is possible to exploit a sliding window to compute the mean energy of successive portions of the signal. If the value exceeds a certain threshold, we assume the interference affects that portion of the signal. A significant difference with the traditional spectrum sensing is represented by the impact of the sensing period on the performance: increasing the window length does not leads to an improvement, since the energy of the pulse is spread on a longer interval making more difficult to discriminate which samples are affected. The window length depends on the duration of the interfering signal that cannot be exactly known because the pulse is filtered by the receiver, however a rough estimation of the JTIDS signal du-

[3] JTIDS is a military system and many features of the system are not known.

ration can be calculated as $L = T_p \cdot f_s = 4$ samples, where f_s is the sampling frequency. We adopt a windowing size, W, equal to $L + 1$ samples.

In order to get refined results, the sliding window analyses every sample of the received symbol: each value returning from the windowing operation is referred to a specific sample, since the windows are partially overlapped. The n-th sample of the i-th symbol in output from the sliding window is

$$w(n) = \sum_{j=-\frac{W-1}{2}}^{+\frac{W-1}{2}} a_j \cdot \|r_i(n-j)\|^2 \tag{3.6}$$

where the a_j terms are the window weight and are selected in order to give more weight to the central sample.

Fig.3-1 shows an example of the windowing effects. It is possible to note that windowing reduces significantly the fluctuations of the signal thus permits to better discriminate the interference selecting an optimal threshold value. In fact, with windowing and using Threshold1 value, it is possible to detect all the samples affected by interference. Instead, observing directly the received signal an optimum threshold value is more difficult to be selected: with Threshold1 some missing detections and many false alarms occur. Increasing the threshold (Threshold2) permits to reduce the false alarms but increases the missing detections.

The window output obviously depends also on LDACS1 signal, that is based on an OFDM transmission and, hence, it is characterized by a high peak to average ratio. This can increase the probability of false alarm. To counteract this aspect, and taking into account the length of the JTIDS signal, this probability can be reduced by observing M consec-

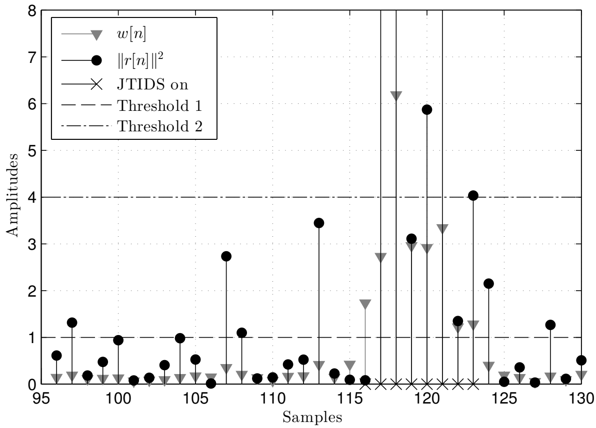

Figure 3-1: Windowing effects on spectrum sensing.

utive window outputs: the outputs $w[n]$ are compared with a threshold, if at least $M = L - 1$ consecutive outputs are over the threshold we assume the interference is present on correspondent received samples.

After sensing operation the interference is removed by means a blanking operation. Blanking simply puts to zero the received samples where the interference has been detected and lets unchanged the other samples.

The difference among the two proposed alternatives is that:

- in *full combining* all the symbols are transmitted twice and hence the sensing is performed on the difference of the two signal's replicas;
- in *partial combining* the sensing is performed on the received signal and the sensing decision is used to select which symbols must be retransmitted.

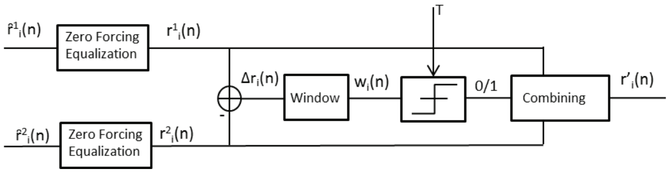

Figure 3-2: Full combining scheme.

3.3.1 Full combining scheme

In *full combining* scheme two replicas of all the symbols are always available. This can be exploited in order to improve the interference detection performance.

Full combining scheme is pictured in Fig.3-2

To detect exactly which samples are affected by interference the proposed method performs a sample by sample difference between the two copies of the symbol that has been previously equalized to remove the channel effect. In that way the dependence on the LDACS1 signal fluctuation is removed. Assuming a perfect channel zero-forcing equalization, the difference on the n-th sample of the i-th symbol is

$$\Delta r_i(n) \doteq r_i^1(n) - r_i^2(n). \tag{3.7}$$

Since each signal r_i^l, with $l = 1, 2$, is

$$r_i^l(n) = s(n) + \frac{u_i^l(n)}{\alpha_i^l} + \frac{n_i^l(n)}{\alpha_i^l}. \tag{3.8}$$

The difference Δr_i can be expressed as

$$\Delta r_i(n) = \frac{u_i^1(n)}{\alpha_i^1} - \frac{u_i^2(n)}{\alpha_i^2} + \frac{n_i^1(n)}{\alpha_i^1} - \frac{n_i^2(n)}{\alpha_i^2} \tag{3.9}$$

where:

- $r_i^l(n)$ is the n-th received sample of the i-th symbol of the l-th transmission after equalization
- $u_i^l(n)$ is the JTIDS interference on n-th sample of the i-th symbol of the l-th transmission
- $n_i^l(n)$ is the AWG noise on n-th sample of the i-th symbol of the l-th transmission
- α_i^l is the Ricean fading term on the i-th symbol of the l-th transmission, assumed constant on an OFDM symbol.

In Eq.s 3.8 and 3.9 the fading coefficients appears in the denominators because the signal is equalized. The difference $\Delta r_i(n)$ depends on the JTIDS interference and the noise power: the LDACS1 signal fluctuation does not affect the detection procedure.

The windowing operation is performed along the difference signal samples $\Delta r_i(n)$ and the result is compared with a threshold, T. The threshold comparison output is 1 if the sample is above the threshold or 0 otherwise. Finally the combining block checks if windowed signal keeps above the threshold for at least $M = L - 1$ consecutive samples (i.e. it receives at least M consecutive ones). In that case it assumes the interference is present in those samples. At this step, it is possible to know which samples are affected by the interference but it is needed to determine if the interference is introduced by the first, $r_i^1(n)$, or the second $r_i^2(n)$ copy of the received signal. Hence the combining block performs a comparison of the two signals before combining. Indicating as $\overline{n}_q$, with $q = 0, 1, ..., Q - 1$ the samples affected by the

Table 3.2: Analysed cases.

$CASE$	q	$TSDF$	N_p	P_p
$Standard$	1	50%	258	$4.2E-3$
$Worst$	2	100%	444	$2.3E-2$

interference, for each $\overline{n}_q$ the values of $r_i^1(\overline{n}_q)$ and $r_i^2(\overline{n}_q)$ are compared: the maximum is blanked while the minimum is doubled. The resulting signals are summed together, and the final i-th symbol can be expressed as

$$r_i'(n) = \begin{cases} r_i^1(n) + r_i^2(n) & \text{if } n \neq \overline{n}_q, \ \ \forall \overline{n}_q \\ 2 \cdot r_i^1(n) & \text{if } n = \overline{n}_q, r_i^1(\overline{n}_q) < r_i^2(\overline{n}_q) \\ 2 \cdot r_i^2(n) & \text{if } n = \overline{n}_q, r_i^1(\overline{n}_q) > r_i^2(\overline{n}_q). \end{cases} \tag{3.10}$$

Demodulation and decision are performed on r_i' signal.

3.3.2 Partial combining scheme

In the *partial combining* scheme interference detection is applied on the received signal in order to detect which symbols are affected by interference and thus requiring a new transmission of these symbols. *Partial combining* is pictured in Fig.3-3.

In this case the windowing operation is performed along the symbol $r_i^1(n)$ and the result is compared with a suitable threshold. When the windowed signal keeps above the threshold for at least $M = L - 1$ samples, the interference is assumed to be present in those samples and in the relative symbols. Following an HARQ mechanism the receiver requires a new transmission of the corrupted symbols that

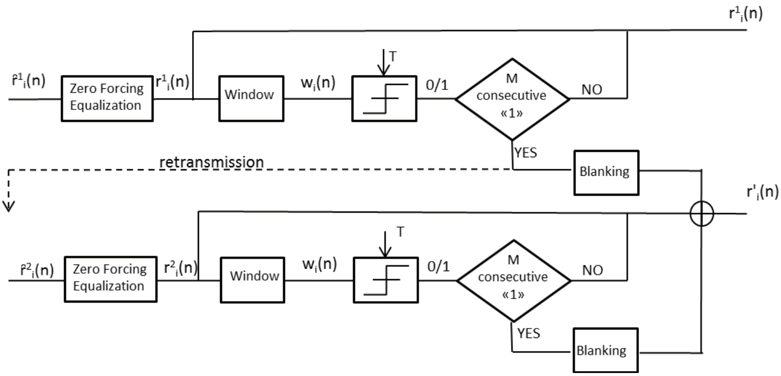

Figure 3-3: Partial combining scheme.

are not discarded but stored after removing the interference by the means of a blanking operation. Indicating as $\overline{n}_q$, with $q = 0, 1, ..., Q-1$ the samples affected by the interference, the resulting i-th symbol becomes

$$\overline{r}_i^1(n) = \begin{cases} r_i^1(n) & \text{if } n \neq \overline{n}_q \\ 0 & \text{if } n = \overline{n}_q \end{cases} . \tag{3.11}$$

In presence of JTIDS interference the probability that even the second transmission is affected by the interference is very high, for this reason sensing is performed also on the second received signal, $r_i^2(n)$ and the eventual interference is removed by the means of a blanking operation like in (3.11), obtaining $\overline{r}_i^2(n)$. Finally, the two copies of the i-th symbol are softly combined

$$r_i'(n) = \overline{r}_i^1(n) + \overline{r}_i^2(n). \tag{3.12}$$

Demodulation and decision on corrupted symbols are taken on r_i' rather then r_i^1.

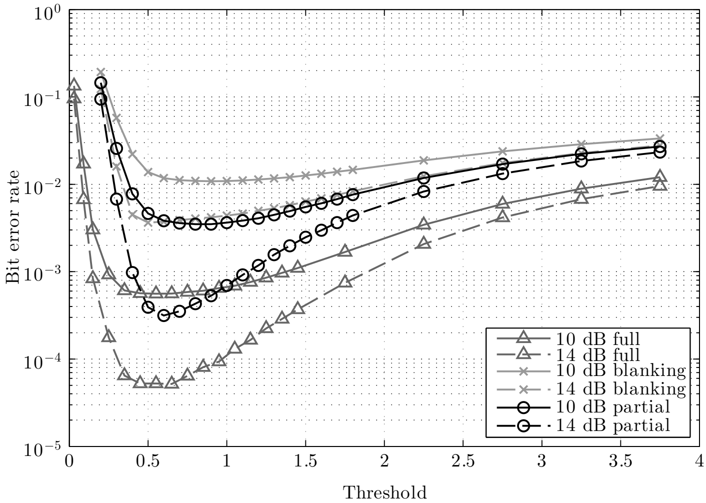

Figure 3-4: BER as a function of the threshold for different SNR (SIR=0 dB in the Worst Case).

3.4 Numerical Results

In order to study the effectiveness of the proposed mitigation method, the performance of our system has been simulated and compared with that of a classical blanking method and with the system that does not adopt any interference mitigation technique. For blanking scheme the same windowing interference detection method presented in Sec. 3.3 has been considered. JTIDS interference has been characterized by considering two profiles, named *standard* (STD) and *worst* (WORST), reported in Tab. 3.2. From this table we can see that the probability of interference between the two systems is quite high, especially for the WORST case.

Since the proposed method performance depends on the threshold value, T, an optimization process has been carried

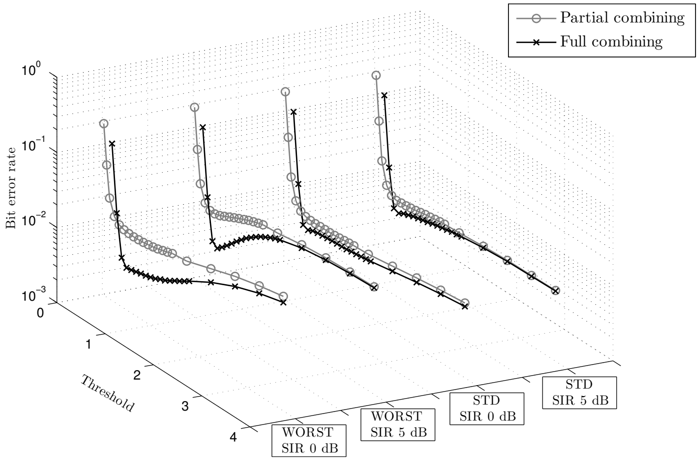

Figure 3-5: BER as a function of the threshold for different SIR and interference profiles (SNR=10 dB).

out. Fig.s 3-4 and 3-5 show the Bit Error Rate (BER) as a function of the threshold value for different system configuration. In particular the optimum threshold behavior has been evaluated through computer simulations, either for the proposed and the blanking methods for different Signal to Noise Ratio (SNR) (Fig. 3-4) and for different interference parameters (Fig. 3-5).

These figures highlight the existence of an optimum value of T for which we have the lowest BER for all the methods. We can note that the optimum threshold value does not change significantly when the SNR or the interference configuration change even if cancellation benefits are more evident for higher SNR values and when the interference is more frequent (in Fig. 3-5 only the proposed methods curves are reported for a better readability, but the same simulations have been carried out also for the blanking method, leading to similar results). In addition Fig. 3-5 shows that the performance of the proposed methods does not change significantly when the SIR changes. This is because when the SIR is high the interference is harder to detect but it has a minor impact. On the other hand, when the SIR is low the detection and, hence, the recombination become more reliable. It is important to underline that the JTIDS power is very high if compared with LDACS1 transmission power and low SIR values can often occur. Simulations with SIR values lower of 0 dB have been carrier out showing performance very close to the case of SIR=0 dB, for this reason the results are not reported here. In general, observing the threshold optimization figures we can observe that a slope close to zero is visible around the optimum T value and we can deduce that a value of T close to the optimum does not affect notably the receiver performance. The optimum T value determination has to be performed only for specific target applications and service scenarios. The following results were obtained for an optimum threshold value, different for each method, which was determined by computer simulation, as outlined above.

In *partial combining* scheme the threshold choice affects also the system throughput because it determines the number of retransmissions as shown in Fig. 3-6. From this figure it is possible to see that the maximum useful throughput (i.e., the amount of symbols correctly received on a given time period) is achieved for threshold values that are almost the same of those for which we have minimum BER. We can note that for low threshold values (up to the value for which the maximum throughput is reached) the results are independent of the SIR value. This is because all the interference is detected for both values of SIR. When the threshold increases, there is a range of values for which having lower SIR leads to a

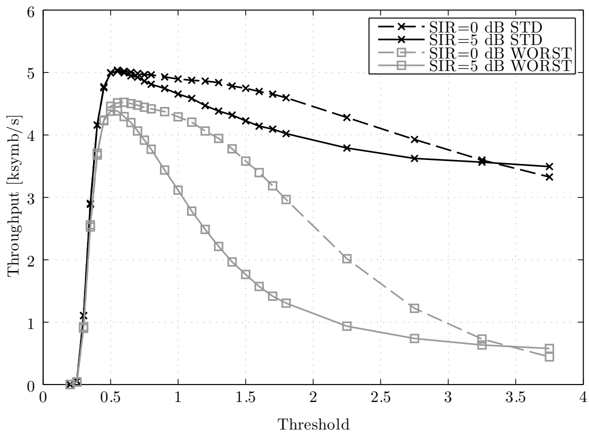

Figure 3-6: Throughput as a function of the threshold for different SIR and interference profiles (SNR=10 dB).

higher interference detection probability and, hence, a higher interference mitigation. Conversely there is a following range of values for which missing detection probability increases, thus having higher SIR leads to a residual interference that has less detrimental effects.

The interference detection and mitigation capabilities of the proposed methods are evident also by observing BER performance. Fig. 3-7 represents the BER as a function of SNR. The performance of the proposed methods is compared with that of the classical blanking method and the case without mitigation and without interference. Standard and Worst cases are considered. It is evident that for both the proposed methods the interference effects are almost completely removed, in fact the partial combining method permits to approach the performance of the case without interference, and the full combining scheme presents almost a 3 dB gain

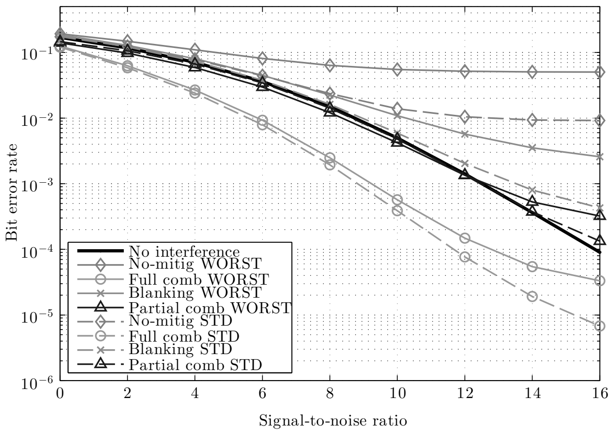

Figure 3-7: BER comparison as a function of SNR when SIR=0 dB, worst and standard cases.

thanks to the diversity combining. Only for high SNR and high interference (worst case) the BER results to be slightly affected by the residual interference. The proposed methods present in all the cases a significant gain if compared with the case without mitigation, that is heavily deteriorated, and the blanking method. In particular, for high SNR values blanking operation reaches more rapidly a performance floor due to the fact that the signal is removed together with the interference and ICI is introduced by the nonlinearities. It is more evident in the worst case.

In order to evaluate the effect of retransmission on the throughput we have to take into account that the symbol retransmission leads to a reduction of the transmission capacity, on the other end, the useful received throughput increases thanks to the reduced error rate. Fig. 3-8 presents the final useful received throughput in terms of the number of correct OFDM symbols received, evaluated taking into account the transmission capacity loss due to retransmission. It is evident that despite the retransmissions the *full combining* scheme presents an advantage also in terms of throughput for low SNR, while the *partial combining* scheme results to be always the best choice for high SNR especially when the interference is high (worst case).

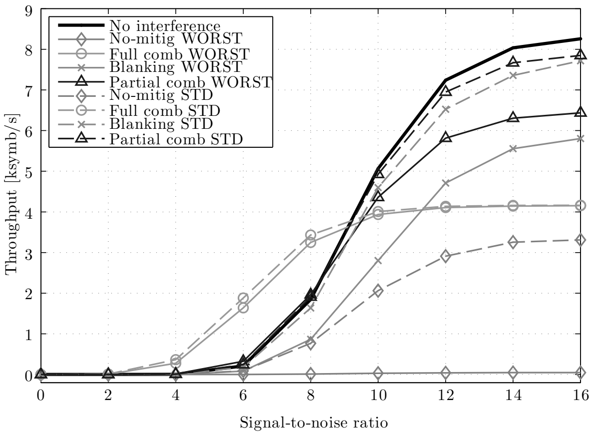

Figure 3-8: Throughput of the proposed and blanking methods as a function of SNR when $SIR = 0$ dB.

Chapter 4

Impulsive Noise Detection and Mitigation

Orthogonal Frequency Division Multiplexing (OFDM) is a transmission technique widely adopted in recent years in high data rate communication systems. It is a multi-carrier modulation technique, with orthogonal sub-carriers, that exhibits several advantages over the single carrier modulation approaches, such as a higher bandwidth efficiency and the capability of mitigating frequency-dependent distortion across the channel band with a consequent simplified equalization in multipath fading environments.

For these reasons OFDM has been selected as the air-interface of many wireless standards such as Third Generation Partnership Project (3GPP) Long Term Evolution (LTE/LTE-Advanced), Wireless Metropolitan Area Networks (IEEE 802.16), Wireless Local Area Network (IEEE 802.11), Digital audio/video broadcasting (DAB/DVB) and wired systems such as power line communications (PLCs).

Performance of OFDM systems can be affected not only by thermal noise but also by Impulsive Noise (IN). The problem of IN in OFDM transmission is well-known in PLCs [46, 47], while only recently it has been investigated for wireless systems where the increasing number of wireless devices and systems leads to a dense spatial reuse and a severe co-channel interference that can be impulsive. In addition, IN can be generated by other sources like lightning discharges or man-made noise.

In an OFDM system the IN is spread over all the subcarriers due to the Discrete Fourier Transform (DFT) operation performed at the receiver side. This is an advantage compared to the single carrier systems in case of low interference power, but it becomes a disadvantage if IN is powerful, as may happen in PLC and dense cellular networks. In this case, OFDM has low immunity to impulsive noise, and performance is heavily affected [28,29]: without any interference mitigation, the OFDM system is not able to meet the QoS requirements especially in high data rate systems.

The most known solutions to counteract the impulsive interference are clipping and blanking [31,33]. In both methods IN is detected when the signal amplitude exceeds a certain threshold. On one hand, blanking sets to zero the samples where the noise has been detected, removing the signal together with noise, on the other hand clipping truncates the amplitude of the samples to a certain value, where the noise is present, to reduce its impact.

Both methods operate with nonlinear operations, making the OFDM signal subcarriers no more orthogonal, thus resulting in an increased inter-carrier interference (ICI). This degrades the performance especially in multi-user communication systems, requiring additional ICI cancellation schemes.

In [35] the authors provide an improvement of blanking non-linearity through an iterative reduction of ICI caused by blanking operation. In [36, 37] IN suppression techniques jointly with frequency domain equalization are investigated. Alternative schemes show a noise reduction based on data detection (i.e., decision directed) and cancellation in order to estimate and remove the noise. These schemes are quite effective but present high computational costs [38,39]. In [48], the authors propose a decision direct approach for mitigating the impulsive noise in xDSL environments.

This work proposes a new method to detect and remove the impulsive noise based on spectrum sensing and symbols retransmission. The method is a generalization of that proposed in the Chapter 3, considering as victim a generic OFDM system and as interferer a gaussian impulsive noise.

As in Chapter 3 the impulsive noise is detected by resorting to a spectrum sensing algorithm based on energy detector and a combination of symbols retransmission and blanking operation after the interference detection is performed.

Similarly to previous partial combining scheme, according to the soft symbol combining principle [44], in order to increase the performance in continuous ARQ (Automatic Repeat reQuest) based communication systems, the symbol is transmitted two times whenever the impulsive interference is detected: the first replica of the symbol is stored and soft combined with the second replica. Before the soft combining operation, the samples affected by IN are zeroed on each copy, in order to remove the impulsive interference without losing useful information. An additional gain is introduced by the soft combining against AWGN noise. The scheme is based on the idea that, even if two consecutive replicas of the same packet are affected by IN, the samples affected by the interference can be considered as statistically independent.

4.1 System Model

In OFDM communication systems the available spectrum is divided in several subcarriers. Data symbols are efficiently modulated on the subcarriers by resorting to the use of the inverse DFT (IDFT) in transmission and the DFT at the receiving end.

We focus here on an OFDM-based communication system with K subcarriers. The information bit streams are coded with the concatenation of an outer Reed-Solomon (RS) code and an inner variable-rate convolutional code in packets of 2500 bits, and then mapped into a base-band symbol c_k with $k = 0...N_{\mathrm{FFT}} - 1$ using QPSK modulation scheme, where $K = N_{\mathrm{FFT}}$. The information sequence results to be:

$$s(n) = \sum_{k=0}^{K-1} c_k \cdot e^{j\frac{2\pi}{K}nk} \quad n = 0, ..., K-1. \tag{4.1}$$

A cyclic prefix with length N_g symbols is considered by repeating the final part of the OFDM symbol, $s(n)$, resulting in the transmitted signal $\hat{s}(n)$. The base-band signal is up-converted to the radio frequency and transmitted through the channel. The channel is modelled here as a frequency selective multipath fading channel following a Rayleigh distribution, with a discrete impulse response $h(n)$ shorter than the cyclic prefix, which allows the cancellation of ISI effect at the receiver side. Hence, the received signal can be defined as

$$\hat{r}(n) = \hat{s}(n) \otimes h(n) + v(n) + u(n) \tag{4.2}$$

where $v(n)$ $\mathcal{N}(0, \sigma_v^2)$ represents the Additive White Gaussian Noise (AWGN), while $u(n)$ denotes the IN. The useful received signal $r(n)$ is obtained by removing the cyclic prefix.

Some statistical or empirical models have been proposed in the literature to model the IN, but they refer to specific environments and interference sources [48–50]. Even if the characterization of impulsive noise is quite difficult, two features are in common among all the impulsive interference sources: the energy is concentrated in short time periods and its power is much higher than the background noise. For this reason we model the IN as a Gaussian pulse with duration D, such that

$$D \ll T_s \tag{4.3}$$

where T_s is the OFDM symbol duration. In addition, the probability that the IN affects a certain OFDM symbol is P_i (i.e., in average one symbol every $\frac{1}{P_i}$ is affected by the interference). The position of the interference within the OFDM symbol is considered random with uniform distribution. The signal $r(n)$ is processed by the DFT block at the receiver side. The DFT output for the k-th subcarrier, before equalization, is

$$\begin{aligned} Z(k) &= \frac{1}{K}\sum_{n=0}^{K-1} r(n) \cdot e^{-j\frac{2\pi}{K}nk} = \\ &= S(k) \cdot H(k) + V(k) + U(k) \end{aligned} \tag{4.4}$$

where $S(k)$, $H(k)$, $N(k)$ and $U(k)$ are the samples in the frequency domain of the transmitted signal, channel impulse response, AWGN and IN contribution on the k-th subcarrier, respectively. The signal is equalized by exploiting a Zero Forcing equalization algorithm by dividing each subcarrier signal $Z(k)$ by the channel response $H(k)$[1].

[1] We assume to have a perfect knowledge of the channel response for equalization purposes.

4.2 Proposed interference mitigation method

This section presents the proposed method to counteract the effects of an impulsive interference on an OFDM signal. In particular, it is composed of a first stage of IN detection, and then a new mitigation strategy based on symbols retransmission.

4.2.1 Interference Detection

Interference detection is needed to determine which samples and, hence, which symbols, are corrupted by interference in order to put in act suitable countermeasures. In particular, we consider here a spectrum sensing approach based on the energy detector. This method is blind, hence it does not require any knowledge of the interfering system features, and can be implemented by resorting to a low complexity algorithm.

In general, spectrum sensing is used to determine if a signal is present or not on a given area. To this aim, the energy detector estimates the energy of the samples received during a dedicated period (i.e., the communication is avoided in this interval) and compares it with a certain threshold, whose value is set depending on the background noise power at the receiver. If the energy is over the threshold, another system is assumed to be present and transmitting, i.e., the interference is present.

The detection accuracy is proportional to the duration of the sensing period: if the number of collected samples is sufficient, it is possible to reach any desired performance even in low Signal to Noise Ratio (SNR) regime. A modified energy detector is proposed in order to know which samples of the received signal are affected by the interference. Hence, the sensing is not performed on a clear interval but on the whole received signal.

By adapting the sensing concept, we exploit a sliding window for estimating the average energy of a portion of the signal. If the value exceeds a certain threshold, we assume that interference affects that portion of the signal.

A significant difference with the traditional spectrum sensing is represented by the impact on the performance of the sensing period: an increasing in the window length does not necessarily lead to an improvement, because the energy of the pulse is spread on a longer interval making more difficult to discriminate which samples are affected. The choice of the window length, W, is shown in Sec. 4.4.

In order to get refined results, the sliding window is partially overlapped, each value returning from the windowing operation is referred to a specific sample. The n-th sample of the signal in output from the sliding window is computed as

$$w(n) = \sum_{i=-\frac{W-1}{2}}^{+\frac{W-1}{2}} a_i \cdot \|r(n-i)\|^2. \tag{4.5}$$

The a_i terms are the window weights, which have been chosen as described in Sec.4.4.

The interference detection phase produces as output the value "1" if the interference is present on a given sample, "0" otherwise.

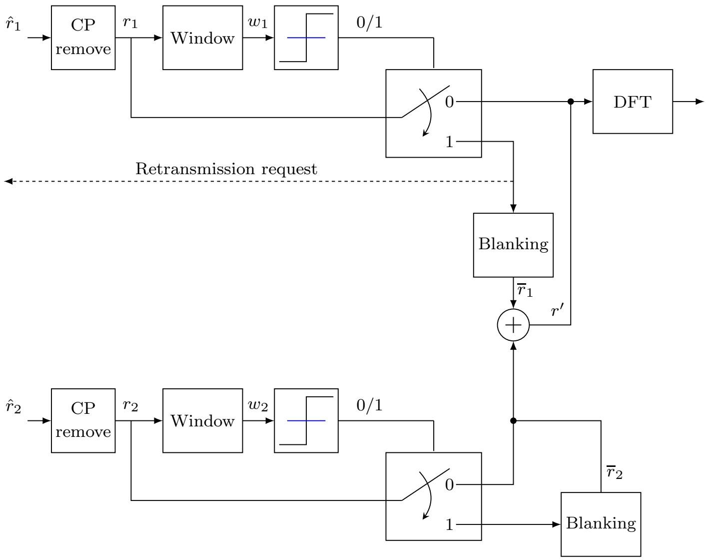

Figure 4-1: Block diagram of the proposed method in the OFDM receiver.

4.2.2 Interference Mitigation

The sensing decision is used to request a new transmission of the symbols where the interference has been detected during the previous sensing phase. The receiver does not discard the corrupted symbols, which are stored after removing the interference by means of a blanking operation. Then, they are softly combined with the new transmitted replica. Blanking puts to zero the received samples where the interference has been detected, while leaves unchanged the other samples. Indicating with n_i, where $i = 0, 1, ..., I - 1$, the samples affected by the interference on the first received signal $r_1(n)$, the resulting signal after blanking operation becomes:

$$\bar{r}_1(n) = \begin{cases} r_1(n) & \text{if} \quad n \neq n_i \\ 0 & \text{if} \quad n = n_i \end{cases}, \qquad \forall n_i. \tag{4.6}$$

Even the second transmission of the symbol, $r_2(n)$, may be affected by interference; for this reason, the sensing procedure and the blanking operation as in (4.6) are repeated also for the second replica of the symbol, obtaining the signal $\bar{r}_2(n)$. Finally the two copies of the symbols are softly combined, giving

$$r'(n) = \bar{r}_1(n) + \bar{r}_2(n). \tag{4.7}$$

Demodulation and decision on corrupted symbols are performed on $r'(n)$. A block diagram of the proposed method is presented in Fig.4-1.

4.3 Theoretical Analysis

In this section a theoretical analysis of the proposed method is presented. We focus here on an AWGN channel[2] and on an impulsive noise that has a Gaussian distribution. Thus, the total noise $v'(n) = v(n) + u(n)$ affecting the signal has a Gaussian distribution when interference is present. The impulsive noise is spread on the whole OFDM symbol by means of the DFT operation, hence, denoting with d the number of samples affected by the interference the bit error

[2]The results can be easily extended to a Rayleigh fading channel by averaging the performance with the Rayleigh distribution.

probability can be expressed as

$$P_e(d) = Q\left(\sqrt{\frac{S}{\sigma_v^2 + \frac{d}{K}\sigma_i^2}}\right) \tag{4.8}$$

where $Q(\cdot)$ is the Q-function, S is the useful signal power and σ_i represents the power of each interference sample.

To remove the dependence on d, it is necessary to find its probability density function (pdf), $f_d(x)$. This can be calculated by averaging all the possible interference configurations within a generic OFDM symbol with the related probability of occurrence. To this aim, the characteristics of the impulsive interference must be taken into account. As stated before, interference has fixed time duration equal to D, and P_i is the probability that an OFDM symbol is affected by the interference. Hence, to derive $f_d(x)$ we consider a time axis divided in slots of duration D, and the probability that a slot is affected by the interference is defined by p, that can be derived by P_i (i.e., $P_i = 1 - (1-p)^N$ where N is the number of slots in the OFDM symbol defined in what follows).

In the most general case, the interference timing is different from the timing of the OFDM signal and the symbol length K (i.e., the number of samples of an OFDM symbol in the time domain after the CP removal) is not a multiple of D. Hence, at the beginning and at the end of the OFDM symbol the interference can affect less than D samples (i.e., part of the interference is on the next symbol or on the CP that has been removed). To take into account this aspect we denote with t the time offset between the OFDM symbol and the slots timing. It is straightforward to note that t may assume values in the interval $[0; D-1]$; when $t = 0$ the interference

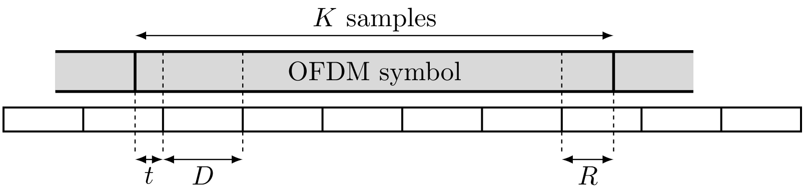

Figure 4-2: Interference and symbol timing comparison.

and signal timings are aligned[3].

Hence, the impulsive interference can affects an initial portion of signal of duration t, a certain number of central slots of duration D, and/or a final portion of signal with duration R, as depicted in Fig. 4-2.

For any given value of K, t and D, the duration of the final slot is

$$R = K - t - \lfloor (K - t)/D \rfloor D \tag{4.9}$$

where $\lfloor s \rfloor$ denotes the greatest integer minor than s, and the total number of slots in an OFDM symbol is

$$N = \lfloor (K - t)/D \rfloor + 2. \tag{4.10}$$

The interference can affect m slots in the OFDM symbol, with $m = 1, \cdots, N$. In addition, it produces a less detrimental effect on the performance if it is on the first or the last slot because in this cases the duration is shorter than D. For this reason, we have to distinguish among four main cases:

- l=1, the interference is on the first t samples but not on the last R;

[3]In general, even if the interference and signal timings are aligned at the beginning of an OFDM symbol, in the next one they will not be synchronized unless K and CP are multiples of D.

- l=2, the interference is on the last R samples but not on the first t;

- l=3, the interference is neither on the first t samples nor on the last R;

- l=4, the interference is both on the first t samples or on the last R.

The probability density function of d, $f_d(x)$, can be derived as the sum of the probabilities, $P_{m,t,l}$, to have $d = x$ for all the values of m, t, and l, that is

$$f_d(x) = Pr(d = x) = \sum_{m,t,l\,:\,d_{m,t,l}=x} P_{m,t,l} \quad , \quad \text{for } x = 0, 1, \cdots, K. \tag{4.11}$$

The probabilities $P_{m,t,l}$ are derived in the Sec. 4.5. Following (4.8) and (4.11), the mean bit error probability of the system without any interference mitigation, P_e^I, can be expressed as:

$$\begin{aligned} P_e^I &= E_x\left[Q\left(\sqrt{\frac{S}{\sigma_v^2 + \frac{x}{K}\sigma_i^2}}\right)\right] \\ &= \sum_{x=0}^{K} Q\left(\sqrt{\frac{S}{\sigma_v^2 + \frac{x}{K}\sigma_i^2}}\right) f_d(x) \end{aligned} \tag{4.12}$$

Let us consider now the proposed retransmission and combining technique. To derive a closed form expression of the error probability we consider an ideal sensing phase. We are interested in validating the main element of the proposed method that is the joint use of blanking and retransmission. Each sample corrupted by the interference is blanked, and combined with the corresponding sample of an OFDM symbol replica. Sensing and blanking are applied on the second replica too, and retransmission occurs only if at least one sample of the OFDM symbol is corrupted by interference. It is straightforward to note that with ideal sensing the performance does not depend on the SIR. The retransmitted OFDM symbols have twice the SNR of the others, but each sample affected by interference in any replica leads to an increase of the power of the noise equal to $\frac{\sigma_v^2}{K}$.

However, this is true only if the same sample is not interfered in both the symbol replicas. In this case, the signal is just blanked and the effect on the performance is more detrimental and complex. On the other hand, the probability that a sample is affected by interference in both replicas is equal to p^2, which means that for small values of p can be considered negligible. Hence, we give here only a lower bound of the mean bit error probability, assuming the absence of the aforementioned condition. In this case, the mean bit error probability of the mitigated system, P_e^M, can be expressed as:

$$P_e^M = Q\left(\sqrt{2\frac{S}{\sigma_v^2}}\right) f_d(0) + \\ + \sum_{x_1=1}^{K}\sum_{x_2=0}^{K} Q\left(\sqrt{\frac{S}{\sigma_v^2\left(1+\frac{x_1+x_2}{K}\right)}}\right) f_d(x_1) f_d(x_2) \qquad (4.13)$$

that is the sum of the error probability when interference is not present in the first replica and the re-transmission does not occur, and of the error probability of all the possible combinations of interference on the first and/or on the second replica. In the next section, the theoretical results will be validated through simulations.

4.4 Numerical Results

In order to demonstrate the effectiveness of the proposed mitigation method, we provide here some numerical results.

First, we validate the theoretical analysis provided in Sec. 4.3. To this end, Fig. 4-3 presents the bit error probability of the proposed system obtained by means of theoretical analysis and computer simulations.

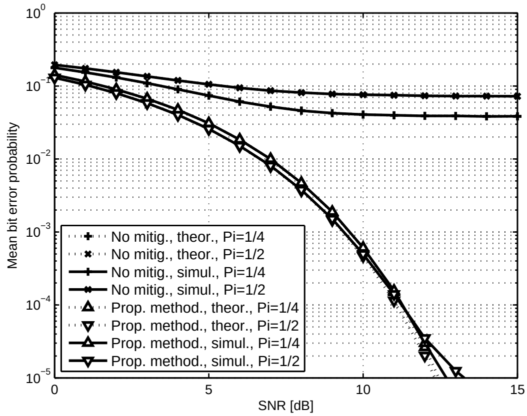

Figure 4-3: Bit Error Probability of the proposed method when $D = 4$.Theoretical analysis vs simulations.

The curves are compared with those of a system without interference mitigation for different interference probabilities P_i, when $D = 4$. It is possible to see that there is a good match between the two curves that slightly diverge at high Signal to Noise Ratio (SNR). In facts, as explained before, the theoretical analysis does not consider the case in which the same samples of both the replicas are affected by the interference, hence, is a lower bound of the performance. To derive these results we assumed an AWGN channel and omitted the channel coding gain.

The following system performance has been derived by means of computer simulations, taking into account also the multipath Rayleigh channel, the coding gain and the sensing phase. The obtained performance is compared with that of a classical blanking method and that of a system that does not adopt any interference mitigation technique. For what concerns the blanking technique, the same interference windowing detection method presented in Sec. 4.2 has been considered.

The parameters assumed in deriving our numerical results are:

- Number of subcarriers $K = 64$;
- Variable length of impulse noise, $D = 4, 8$ samples;
- Variable probability of interference P_i ;
- QPSK modulation scheme;
- Variable Signal to Interference Ratio (SIR) ;
- The ITU-R pedestrian channel model A [51] whose parameters are reported in Tab. 4.1

A cyclic prefix consisting of $K/4$ samples has been assumed to counteract the multipath effects.

The sensing phase is based on a windowing operation, hence we have verified the dependence of the system performance on the window parameters (i.e, length and weights) even if a deep investigation of different window shapes is out of the scope of the paper. The Fig. 4-4 presents the behaviour of the Bit Error Rate (BER) as a function of the window

Table 4.1: Channel Parameters.

Path	Delay (ns)	Averaged Power (dB)
1	0	0
2	110	-9.7
3	190	-19.2
4	410	-22.8

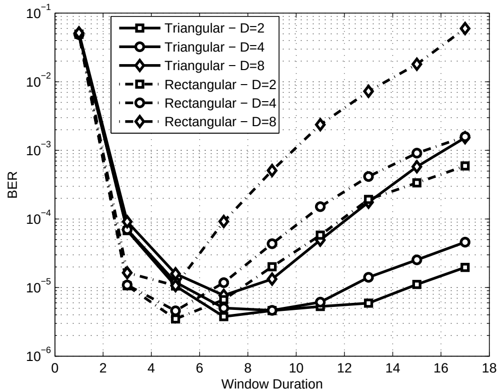

Figure 4-4: BER for different windows weights and lengths ($SIR = 0dB$, $P_i = 1/4$, and $D = 2, 4, 8$).

length, using different window weights (i.e., rectangular and triangular shapes with normalized coefficients) when the interference duration varies. From this figure we can see that the choice of the window weights has not a deep impact on the system performance if the right length of the window is chosen. We selected the triangular window with $W = 7$ to derive the following results because it presents a slope close to zero around the optimum value of W, which leads a value of the window length close to the optimum to not affect notably the performance.

Since the performance of the proposed method depends on the threshold value, T, an optimization process has been carried out: Figs. 4-5 and 4-6 show the BER as a function of the threshold value for different system configurations. In particular, the optimum threshold behaviour has been evaluated for the proposed method and the blanking alternative for different SNR values and for different interference parameters (P_i,D, SIR). These figures highlight the existence of an optimum value of T allowing the lowest BER. It is interesting to notice that such optimum value does not changes significantly with the SNR (Fig. 4-5) and interference parameters. Furthermore, from Fig. 4-5 it is evident that, in general, a slope close to zero is evident around the optimum T value. Hence, we can conclude that a value of T close to the optimum does not affect notably the receiver performance and, therefore, the optimization of T could be performed only for specific target applications and service scenarios.

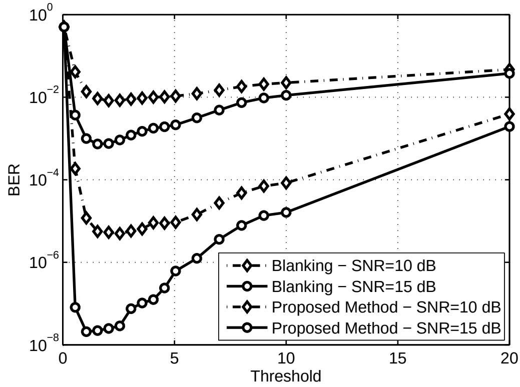

Figure 4-5: BER of the proposed method and blanking as a function of the threshold for different SNR values ($SIR = 0dB$, $D = 4$ OFDM samples, $P_i = 1/4$

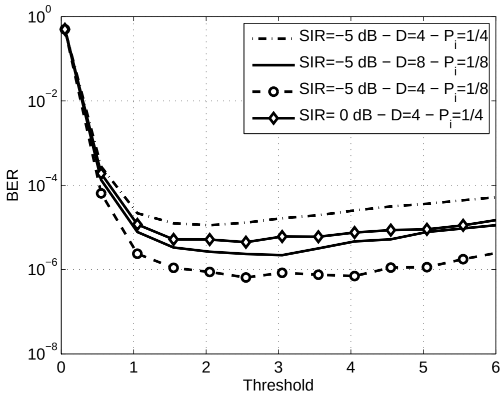

Figure 4-6: BER of the proposed method as a function of threshold for different interference configurations ($SNR = 10dB$).

The following results were obtained with the optimum threshold, as previously described. Fig. 4-6 shows that the performance of the proposed method does not change significantly when the SIR changes because the interference is correctly detected and removed when its power is sufficiently higher than the useful signal power.

The BER performance as a function of SNR is depicted in Figs. 4-7 and 4-8. In particular, in Fig. 4-7, the performance of the proposed method is compared with that of the classical blanking method and the cases without mitigation and without interference. It is possible to see that the proposed method allows to approach the performance of the case without interference. It presents a significant gain compared to the case without mitigation and the blanking method. In particular for high SNR values, blanking operation presents a degradation of the performance due to the fact that the signal is removed together with the IN, and ICI is introduced by the non-linearities, while the proposed method overcomes these problems. In addition, it presents a gain due to the symbol combining. Fig. 4-8 shows the BER when the interference parameters change. A good behaviour is evident in the figure under all the operational conditions.

In order to evaluate the effect of retransmission on the transmitting rate we can assume to have an ideal interference detection (i.e., the probability of false alarm and missing detection equal to 0). Consequently, the probability of having a retransmission is equal to the probability of having interference on a symbol, i.e., P_i. It means that the transmitted capacity is reduced of $100/(\frac{1}{P_i}+1)\%$. However, the useful received throughput (i.e., the amount of packets correctly received on a given time period) increases thanks to the reduced error rate, as shown in Fig. 4-9.

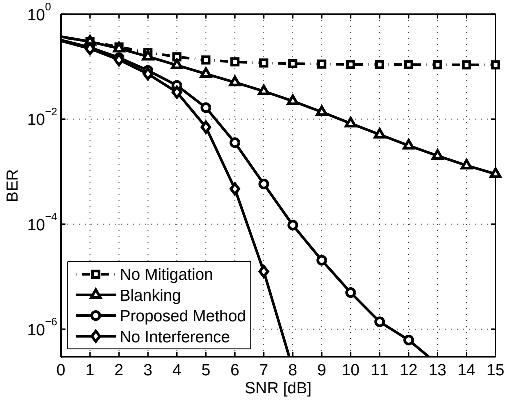

Figure 4-7: BER comparison as a function of SNR when SIR=0dB, $P_i = 1/4$ and $D = 4$ OFDM samples.

In addition, the proposed method reduces the delay due to retransmission mechanism of classical ARQ schemes, which require a retransmission every time the symbol is incorrectly received due the presence of the interference. Indeed, even the successive retransmissions can be affected by interference, and hence, many attempts could be needed to correctly recover the information with a consequent long delay. This problem is overcome by the proposed method where the interference is removed by each copy sample by sample.

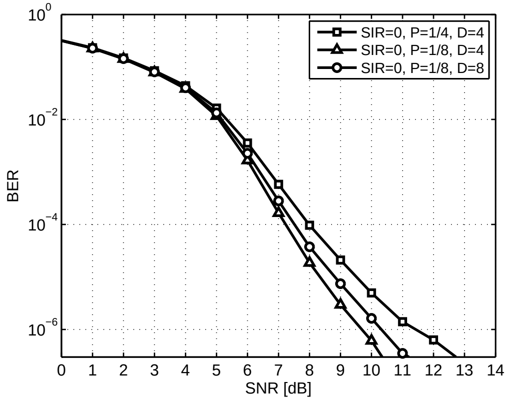

Figure 4-8: BER as a function of the SNR for different interference configuration.

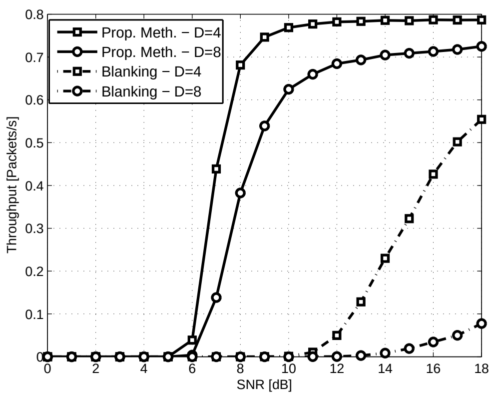

Figure 4-9: Throughput of the proposed and blanking methods as a function of SNR when $SIR = 0dB$, $P_i = 1/4$ and $D = 4, 8$ OFDM samples

4.5 Appendix

The interference on a given OFDM symbol depends on the number of samples, d, affecting the symbol. In order to achieve a closed form of the error probability it is needed to remove the dependence on d in (4.13) by averaging its pdf that must be derived.

As stated before we have

- t is the portion of slot at the begin of the OFDM symbol;
- R is the portion of slot at the end of the OFDM symbol;
- N is the number of slots on the OFDM symbol;
- m is the number of slots affected by the interference.

For each possible interference configuration we can detect the number of samples affected by the interference d and the probability of occurrence of this configuration, P. We assume that the presence of the interference in different slots is uncorrelated.

In particular we have

- $m = 0$, represents the event of "absence of interference". This happens with probability $P = (1-p)^N$, and the samples affected by interference are $d = 0$.
- $m = 1$ happens with probability $P = Np(1-p)^{N-1}$. However, the number of corrupted samples depends on where the interference takes place, that is
    (I) $d = t$, when interference is present in the first slot. This occurs with probability $P = p(1-p)^{N-1}$

(II) $d = R$, when interference is present in the last slot. This occurs with probability equal to case (I), $P = p(1-p)^{N-1}$

(III) $d = D$, when interference is present in a slot different from the first and the last. This may happen in $N-2$ cases, hence the related probability is $P = (N-2)p(1-p)^{N-1}$

- $m = 2, \cdots, N-2$, represents the general case of an OFDM symbol affected by m interference slots, which has probability $P = \binom{N}{m} p^m (1-p)^{N-m}$. In particular, the $\binom{N}{m}$ configurations can be differentiated in four cases, as follows

  (I) interference is present in the first slot and in $m-1$ central slots, hence

  $$d = t + (m-1)D. \tag{4.14}$$

  This happens with $\binom{N-2}{m-1}$ different interference configurations, so the related probability is

  $$P = \binom{N-2}{m-1} p^m (1-p)^{N-m}. \tag{4.15}$$

  (II) interference is present in the last slot and in $m-1$ central slots, hence

  $$d = (m-1)D + R. \tag{4.16}$$

  $$P = P_{m,I} = \binom{N-2}{m-1} p^m (1-p)^{N-m}. \tag{4.17}$$

  (III) interference is present in the first, in the last and in $m-2$ central slots. This may occur in $\binom{N-2}{m-2}$ cases, which leads to

$$d = t + (m-2)D + R \tag{4.18}$$

$$P = \binom{N-2}{m-2} p^m (1-p)^{N-m}. \tag{4.19}$$

(IV) interference is present only in central slots, that is

$$d = mD \tag{4.20}$$

$$P = \left(\binom{N}{m} - 2\binom{N-2}{m-1} - \binom{N-2}{m-2} \right) p^m (1-p)^{N-m} \tag{4.21}$$

- $m = N-1$, happens with probability $P = Np^{N-1}(1-p)$ and can be seen as a dual case of $m = 1$. In particular,

  (I) interference may be present in the whole symbol with the exception of the first slot, which leads to

$$d = K - t = (N-2)D + R \tag{4.22}$$

$$P = p^{N-1}(1-p) \tag{4.23}$$

  (II) interference may be present in the whole symbol with the exception of the last slot, and hence

$$d = K - R = t + (N-2)D \tag{4.24}$$

$$P = p^{N-1}(1-p) \tag{4.25}$$

  (III) interference is present in the whole symbol with the exception of a central slot

$$d = K - D \tag{4.26}$$

$$P = (N-2)p^{N-1}(1-p) \tag{4.27}$$

- $m = N$, happens with probability $P = p^N$, the samples affected by interference are $d = K$.

The aforementioned analysis can be generalized. First we introduces an index l that represents the following cases

- l=1, the interference is on the first t samples but not on the last R;

- l=2, the interference is on the last R samples but not on the first t;

- l=3, the interference is neither on the first t samples nor on the last R;

- l=4, the interference is either on the first t samples or on the last R.

We extend the domain of the Binomial function as

$$\begin{bmatrix} A \\ B \end{bmatrix} = \begin{cases} \begin{pmatrix} A \\ B \end{pmatrix} & \text{if} \quad 0 \le A \le B \\ \\ 0 & \text{otherwise} \end{cases} . \tag{4.28}$$

Hence, for a given value of t, we have

$$
\begin{aligned}
d_{m,t,1} &= t + (m-1)D & (4.29)\\
d_{m,t,2} &= (m-1)D + R & (4.30)\\
d_{m,t,3} &= t + (m-2)D + R & (4.31)\\
d_{m,t,4} &= mD & (4.32)
\end{aligned}
$$

and the related probabilities are

$$
\begin{aligned}
P_{m,t,1} &= \begin{bmatrix} N-2 \\ m-1 \end{bmatrix} \frac{p^m(1-p)^{N-m}}{D} & (4.33)\\
P_{m,t,2} &= \begin{bmatrix} N-2 \\ m-1 \end{bmatrix} \frac{p^m(1-p)^{N-m}}{D} & (4.34)\\
P_{m,t,3} &= \begin{bmatrix} N-2 \\ m-2 \end{bmatrix} \frac{p^m(1-p)^{N-m}}{D} & (4.35)\\
P_{m,t,4} &= \left(\begin{bmatrix} N \\ m \end{bmatrix} - 2 \begin{bmatrix} N-2 \\ m-1 \end{bmatrix} - \begin{bmatrix} N-2 \\ m-2 \end{bmatrix} \right) \frac{p^m(1-p)^{N-m}}{D} & (4.36)
\end{aligned}
$$

for $m = 0, 1, \cdots, N$ and $t = 0, 1, \cdots, D-1$.

Now it is possible to express the pdf of the number of samples affected by interference, d, taking in account that certain value of d can be the outcome of different interference configurations. Hence, the pdf of d can be expressed as

$$
f_d(x) = Pr(d = x) = \sum_{m,t,l \,:\, n_{m,t,l} = x} P_{m,t,l} \quad , \quad \text{for } x = 0, 1, \cdots, K \tag{4.37}
$$

Chapter 5

AeroMACS

As introduced in Chapter I, Eurocontrol has recently identified an IEEE 802.16e/Mobile WiMAX [52] based system, named AeroMACS (Aeronautical Mobile Airport Communication System) as the solution to fulfil the foreseen communication needs within the airport area. AeroMACS will operate in the Microwave Landing System (MLS) band: 5091-5150 MHz. Despite the fact that the IEEE 802.16e standard has been selected as the most suitable technology for the Future Communication Infrastructure (FCI) [53], until now there are not available definitive results in the literature to validate the effectiveness of the AeroMACS solution. In particular, specific problems as synchronization between the aircraft and control station, frequency offset compensation and impairments due to propagation channel conditions still require a deep investigation. For what concerns the characterization of the channel on the airport surface an interesting outcome arisen from recent studies [54–56] is that some features as fading severity, wide delay spread and tap correlation, are always applicable. Among them, the exploitation of tap correlation to improve the capacity of an Orthogonal Frequency Division Multiplexing (OFDM) scheme seems to be promising [57–60] and it allows to have more accurate information on the communication channel also for fast varying environments where channel information quickly become outdated [61]. Moreover, in the case of an Orthogonal Frequency Division Multiple Access (OFDMA) scheme this characteristic can be exploited by dynamically assigning resources to the users in order to increase the system capacity. The subcarriers allocation in OFDMA systems exploiting channel tap correlation is a new topic and only a limited number of publications dealing with it are available in the literature. Among them, in [62], [63], [64] the authors present a power and subcarriers allocation method based on the Tap Correlation Information (TCI). The main idea devised in [62], [63], [64] is a sub-optimal solution dependent on the assumption that the subcarriers are not exclusively allocated to users (i.e., subcarriers sharing factors are introduced). This assumption allows to reduce the objective function to a convex optimization problem lowering the computational complexity. In addition to this, in [62] and [64] an alternative sub-optimal method based on an objective function approximation and the waterfilling approach is presented. In both cases complexity grows up quickly when the number of users and subcarriers increases. In order to relax this drawback here is presented a heuristic characterized by reduced complexity with respect to different alternatives that assigns sub-channels to users with a procedure similar to that adopted for frequency allocation by exploiting the multi-user diversity in MIMO (Multiple Input Multiple Output) systems [65]. The users correlation matrices are considered here in order to derive the mean channel gains on all subcarriers, then each subcarrier is assigned only to the user having the highest channel gain on it. It is important to emphasize that a specific feature of the proposed method is that the correlation matrix knowledge can be replaced with the observation of the mean square gain of the channel on the subcarriers without significant performance degradation even in the case of fast fading conditions (i.e., frame duration longer than the channel coherence time). The performance of the proposed method will be evaluated in terms of overall system capacity for different Signal to Noise Ratio (SNR) values and compared with that achieved by the uniform subcarrier allocation and optimal method [62], [63], [64] in order to highlight its better behaviour.

5.1 System Model

The system under consideration, as specified by the IEEE 802.16e standard [52], is OFDMA based and operates in the MLS band according to the draft version of the AeroMACS specifications [66]. OFDMA provides a flexible multi-user access scheme based on OFDM transmission technique. OFDM allows simultaneous transmissions of several data flows dividing the available bandwidth into smaller bands, named subcarriers. The OFDMA scheme assigns disjunctive sets of subcarriers to users. The subcarriers orthogonality ensures to avoid interference among sub-channels. Efficient modulation schemes can be also adopted to transmit data information on these subcarriers by resorting to the use of the Inverse Discrete Fourier Transform (IDFT) in transmission and the Discrete Fourier Transform (DFT) at the receiving end. It is important to note that the method proposed is suitable for both uplink and downlink communications. However, for the sake of simplicity, we focus here on the uplink communications only, by assuming a number of K subcarriers and U users, e.g., aircraft (AR) or service vehicles, that can simultaneously communicate with the Control Station [1] (CS), by using separate sets of subcarriers. We assume a Time Division Duplexing (TDD) structure because it is compliant with the preliminary draft version of the AeroMACS system specifications and suitable to handle data traffic such as new IP based multi-rate and multi-QoS services. In particular, we focus here on a Mobile WiMAX frame composed by 30 OFDM symbols for downlink and 18 OFDM symbols for the uplink and assume the same propagation channel conditions for both uplink and downlink communications. Moreover, the channel response estimated on an uplink frame is used by the CS to allocate subcarriers to users in the next frame, hence, introducing a latency of one frame[2]. With reference to this, it is important to point out that, even assuming ideal channel impulse response estimation at the receiving side, the one-frame latency introduced by the channel estimation algorithm makes its use critical in the case of fast fading channels as discussed later in Sec.5.3. In our analysis we mainly assume that the communication channel is in NLoS (No Line of Sight) condition and affected by Rayleigh fading (i.e., small scale fading) considered as a special case of the Weibull distribution. However, also LoS conditions and Rice fading distribution have been considered to take into ac-

[1] The CS is the control tower or the company control center depending on the service type: ATC-Air Traffic Control or AOC-Airline Operational Control services.

[2] In some cases, the latency degree can be higher, especially in systems adopting a closed-loop estimation method (i.e., using the downlink channel estimation provided by the AR).

count high mobility scenarios [3]. Weibull represents in general the most suitable solution to characterize the fading affecting airport surface communications [56] due to its flexibility to be properly adapted to different environments. However, the assumption of Rayleigh fading is compliant with [56] where it is stated that this special realization of the Weibull distribution (obtained by setting the shape parameter equal to 2, [67]) often recurs in actual airports in NLoS propagation conditions. Moreover, we point out that the main scope of this considerations is that of proposing a method based on the exploitation of the tap correlation (and the consequent slow variation of the channel) whose advantages are always evident independently from the fading distribution even if with a different relevance.

According to our assumptions, we consider that the communication channel for the u-th user at the n-th time instant, $c_{u,n}$, can be modeled as a discrete time finite impulse response (FIR) filter with L taps given by:

$$c_{u,n} = \sum_{l=0}^{L-1} h_{u,n}[l]\delta(n - \tau_u[l]). \tag{5.1}$$

By assuming the channel conditions constant over the observation interval (i.e., over an OFDM symbol time), we omit hereafter the dependence on the time for terms $h_{u,n}[l]$, hence considering $h_{u,n}[l] = h_u[l]$. Moreover, we have that the terms $h_u[l]]$ result to be Zero Mean Circularly Symmetric Complex Gaussian random variables (ZMCSCG) [68] i.e.:

$$h_u[l] \sim CN(0, \delta_u^2[l]), \quad l = 0, ..., (L-1). \tag{5.2}$$

[3] AR moves with high speed in runways where LoS propagation conditions occur.

In (5.2), the terms $\sigma_u^2[l]$ denote the variance of each tap coefficient resulting by the Power Delay Profile (PDP) with unit norm for the u-th user. From previous studies in [54] and [55], it results that the airport channel is affected by tap correlation so that the channel covariance matrix contains extra-diagonal entries. In particular, the tap covariance matrix for the u-th user results to be:

$$\mathbf{C_v}^{(u)} = E[\mathbf{h}_u \cdot \mathbf{h}_u^H]. \tag{5.3}$$

with h_u the vector of length L, whose elements are the L FIR coefficients $h_u[l]$ and the (n, m) entry given by:

$$\{\mathbf{C}_\mathbf{v}^{(u)}\}_{n,m} = E[h_u[n]h_u^*[m]] = \sigma_u[n]\sigma_u[m] \cdot \rho_u[n, m] \tag{5.4}$$

with $[\bullet]^H$ in [69] and $(\bullet)^*$ in (5.4) denoting the Hermitian operator and the complex conjugate operator, respectively. We can note that being hu an L-dimensional vector formed by complex coefficients given by (5.2), the extra-diagonal elements of the channel covariance matrix for the u-th user, $\mathbf{C_v}^{(u)}$, result to be complex too. Since standard deviations σ_u are real numbers, the correlation coefficients $\rho_u[n, m]$ are complex numbers satisfying the following constraints:

1. $\rho_u[n, n] = 1$
2. $|\rho_u[n, m]| \leq 1$
3. $\rho_u[n, m] = \rho_u^*[m, n]$

The values $|\rho_u[n, m]|$ are randomly generated assigning higher probability to the values in the range $[0.2 - 0.5]$ in order to take into account the characteristics of the measured data as reported in [55]. The vector $\mathbf{H_u}$, whose elements are the channel frequency response at the sub-channel frequency values, can be defined as:

$$\mathbf{H}_u = \mathrm{DFT}\{\mathbf{h}_u\} \tag{5.5}$$

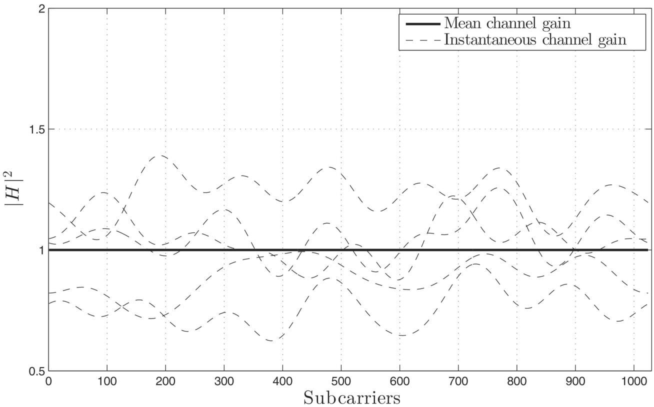

Figure 5-1: Channel behavior for the uncorrelated scattering (US) case.

Were DFT($\mathbf{x}$) denotes the Discrete Fourier Transform of the vector $\mathbf{x}$. Hence, the channel frequency response for the u-th user at the k-th subcarrier results to be:

$$H_u[k] = \mathbf{w}_k^T \cdot \mathbf{h}_u \qquad \text{for} \quad k = 0, ..., (K-1) \tag{5.6}$$

with $\mathbf{w_k}^T = [1, e^{-j2\pi\frac{k}{K}}, ...e^{-j2\pi k(\frac{L-1}{K})}]$ and X^T denoting the transpose of matrix X. From above, it follows that the mean square value of the channel gain of the k-th subcarrier results to be [62]:

$$\gamma_u[k] = E[|H_u[k]|^2] = E[H_u[k] \cdot H_u[k]^H] = E[\mathbf{w}_k^T \mathbf{h}_u \mathbf{h}_u^H \mathbf{w}_k^*]$$
$$= \mathbf{w}_k^T E[\mathbf{h}_u \mathbf{h}_u^H] \mathbf{w}_k^* = \mathbf{w}_k^T \mathbf{C}_{\mathbf{v}}^{(u)} \mathbf{w}_k^* \tag{5.7}$$

From (5.7), it is possible to note that if the channel covariance matrix ${\mathbf{C}_{\mathbf{v}}}^{(u)}$ is a diagonal matrix (i.e., the channel is not affected by tap correlation) the $\gamma_u[k]$ terms are all equal to one, as shown in Figure 5-1 the solid line refers to the mean square value of each subcarrier, while the dashed lines refer to independent realizations of the channel. Conversely, when correlation between taps is present, the $\gamma_u[k]$ terms are no longer unitary, and their values depend on the channel covariance matrix ${\mathbf{C}_{\mathbf{v}}}^{(u)}$ (the covariance matrix used in deriving the results shown in Figure 5-2 was randomly generated with non-zero extra-diagonal elements and is the same for all the channel realizations).

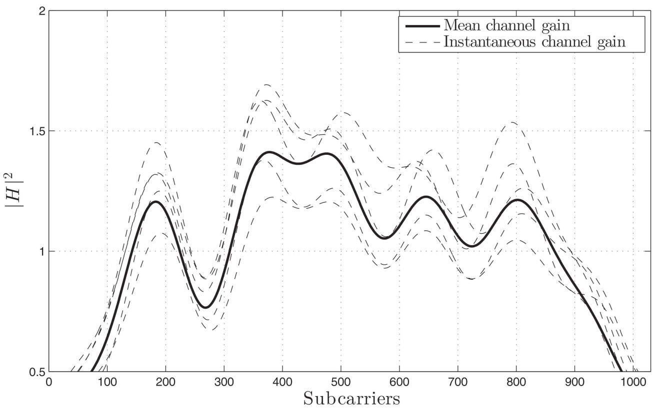

Figure 5-2: Channel behavior for the correlated scattering (CS) case.

Moreover, we define Ω_u as:

$$\Omega_u = \frac{1}{K}\sum_{k=0}^{K-1}\gamma_u[k] = \sum_{l=0}^{L-1}\sigma_u^2[l] = 1 \tag{5.8}$$

From (5.8) it is straightforward to note that u represents the energy of the PDP for user u and if it happens that some subcarriers are attenuated ($\gamma[k] < 1$), consequently others have to be necessarily amplified ($\gamma[k] > 1$). Assuming to use a cyclic prefix longer than the channel delay spread, the received baseband signal for the u-th user on the k-th subcarrier results to be:

$$y_u[k] = x_u[k]H_u[k] + n[k] \tag{5.9}$$

where $n[k]\tilde{C}N(0,\sigma_n^2)$ denotes the additive white Gaussian noise (AWGN) contribution and $x_u[k]$ denotes the symbol transmitted on the kth subcarrier by the u-th user. In particular $x_u[k]$ has to satisfy the following constraint:

$$\text{if} \quad x_u[k] \neq 0 \quad \text{then} \quad x_v[k] = 0 \quad \text{for} \quad v \neq u \tag{5.10}$$

because only one user can transmit on each subcarrier. Finally, assuming all transmitted symbols having unitary energy, we have

$$\|x_u[k]\| = 1 \qquad \forall u, k \tag{5.11}$$

5.2 Subcarrier allocation method

The subcarrier allocation method devised here makes use of the tap correlation property of the channel model highlighted in [56] in order to improve capacity of the communication system and lower the overhead due to the updating rate of the channel parameters estimation. Previous works dealing with resource allocation methods based on channel tap correlation [62, 64] have focused on subcarrier and power allocation in OFDMA systems, by assuming the knowledge of TCI at the transmitting end in order to maximize the channel capacity. In the multi-user case, since the subcarrier exclusive allocation condition makes the optimization problem analytically intractable, subcarrier sharing factors have been proposed in the channel capacity definition [62, 63] in order to relax the constraint (5.10) and, hence, formulate the optimization problem in a more convenient form. In particular, once power allocation coefficients and subcarrier sharing factors are obtained, a hard decision is performed by assigning the kth subcarrier to the user with the highest sharing factor at that subcarrier. This approach allows an improvement on performance, but the search for the optimal value becomes too heavy in a computational complexity point of view, when the number of users and available subcarriers is high. From the results presented in [62], [63] it appears that the main contribution to the performance improvement is due to the adopted subcarrier allocation criterion rather than the power allocation. As a consequence, we lower the implementation complexity by resorting to a suboptimal method that focuses only on optimal subcarriers scheduling among users. As shown before, the channel affected by tap correlation is characterized by different $\gamma_u[k]$ values, depending $\mathbf{C_v}^{(u)}$. In particular, due to multi-user diversity (i.e., the instantaneous channel conditions for different users are almost mutually independent), the resulting users channels will be characterized by different channel covariance matrices. This analysis suggests the following subcarriers scheduling criterion: since the same subcarrier K has different mean square values $\gamma_u[k]$ for each AR (i.e., for each u with u =0,........,U-1), it can be allocated to the AR that presents the maximum value[4]. The set of operations to be executed in sequence according to the proposed algorithm is provided in procedure 1, shown in the following:

$Procedure 1 - Proposed Algorithm$

K: Number of OFDM subcarriers
U: Number of ARs
$\gamma_{u,k}$: Mean gain of the kth subcarrier of the uth AR
C_u: Covariance matrix of uth AR
w_k: DFT vector set
S: scheduling matrix with dimension UxK, with:

$$s[u,k] = \begin{cases} 1 & \text{if subcarrier } k \text{ is assigned to user } u \\ 0 & \text{otherwise} \end{cases}$$

Mean gain computation:
for $k = 0$ to $K - 1$ **do**
  for $u = 0$ to $U - 1$ **do**
    $\gamma_{u,k} = w_k^T \cdot C_u \cdot w_k^*$
  end for
end for
Subcarrier Allocation
for $k = 0$ to $K - 1$ **do**

[4]The proposed subcarrier allocation criterion is based on the fact that the mean square values of the channel gains $\gamma_u[k]$ haven different values for each subcarrier. With a different fading distribution (i.e., Weibull) but with the same PDP and correlation matrix, the $\gamma_u[k]$ do not change: the particular fading distribution affects only the amplitude of the instantaneous fading fluctuations around the mean value. For this reason similar results can be obtained with Weibull distribution.

for $u = 0$ to $U - 1$ **do**
  if $\arg\max_u\{\gamma_{u,k}\} = u$ **then**
    S[u,k]=1
  else
    S[u,k]=0
  end if
end for
end for

The algorithm outlined in procedure 1 is very simple. It takes as input the channels covariance matrices and returns the matrix S, whose elements indicate the subcarriers assignment to the users (i.e., the element $S[u,k]$ is 1 if subcarrier k-th is assigned to the uth user, or 0 otherwise). The computational complexity is low and increases linearly with the number of users and subcarriers. However, it can be easily noted that procedure 1 requires the estimation of the channel covariance matrix. This is usually a hard issue [5] that needs significant computational effort. In order to skip this drawback, the proposed algorithm replaces the estimation of the channel covariance matrix $C_v^{(u)}$ with the knowledge of terms $\gamma_u[k]$ as outlined below. By considering an appropriate time interval where the channel conditions can be considered stationary, it is possible to approximate the $\gamma_u[k]$ terms with the average of a sufficiently long sequence of channel gain samples as:

$$\gamma_u[k] = E[|\tilde{H}_u[k]|^2] \cong \frac{1}{M}\sum_{m=t_0}^{t_0+M}\left|\tilde{H}_u^{(m)}[k]\right|^2 \tag{5.12}$$

[5] Discussion on approaches to achieve the estimation of covariance matrix is out of the scope of this work.

where $\tilde{H}_u^{(t)}[k]$ represents the channel response estimation on the kth subcarrier performed at time i. In Section5.3 the performance of the proposed algorithm is evaluated and compared with that of different alternatives on the basis of the system capacity defined as:

$$C_{sys} \doteq E[C_{ist}] \tag{5.13}$$

where E[•] represents the mathematical expectation of the instantaneous capacity defined as:

$$C_{ist} = \sum_{k=0}^{K-1} \sum_{u=0}^{U-1} S[u,k] \frac{1}{K} \log_2 \left(1 + \mathrm{SNR}_u \cdot |H_u[k]|^2\right) \tag{5.14}$$

where $H_u[k]$ is channel frequency response on the k-th subcarrier allocated to the u-th AR by the algorithm and SNR is considered equal to $||x_u[k]||/\sigma_n^2$. The random variable $|H_u[k]|^2$ is exponentially distributed (i.e., the marginal probability density function is chi-square distributed with two degrees of freedom, [62]) with mean given by $w_k^T C_v^{(u)} w_k^*$. Finally, in Section5.3 the statement is validated that whenever the coherence time of the communication channel is lower than the channel gains estimation latency (i.e., fast aircraft or relaxed estimation constraints) and the tap correlation property is applicable, it is better to allocate subcarriers on the basis of (5.12) rather than using knowledge of terms $\tilde{H}_u^{(m)}[k]$.

5.3 Numerical Results

This section deals with the performance evaluation of the proposed subcarrier allocation algorithm defined by procedure 1, derived by resorting to computer simulations. In particular, we focus on the following AeroMACS system:

1. RF carrier f_c=5GHz;
2. bandwidth of 10 MHz;
3. K=1024 subcarriers;
4. subcarriers frequency spacing Δf=10,94 kHz;
5. QPSK modulation;
6. NLOS area in airport with maximum delay spread equal to 714 ns [55] and Rayleigh fading distribution for low mobile speed (3km/h);
7. LOS area in medium airport with maximum delay spread equal to 178 ns [55] and Rice fading distribution for high mobile speed (125 km/h);
8. normalized PDPs proposed in [55];

We started our analysis by comparing the performance achieved by the proposed algorithm with that of the optimal power allocation method outlined in [62], [63]. Limited to this case only, we have considered a reduced system with 64 subcarriers and 4 users, in order to prevent convergence times that are too long for the method proposed in [62], [63] . The obtained results are shown in Figure 5-3. It is important to note that, despite a reduced implementation complexity, this figure highlights that the proposed method achieves performance close to the optimal approach [62], [63].

In Table 5.1 the proposed subcarrier allocation algorithm is compared with the uniform allocation method. In deriving

Table 5.1: System Capacity Comparisons of the Proposed Algorithm with the Uniform Allocation Assuming Covariance Matrix Knowledge at the Transmitter.

SNR [dB]	U, Uniform Allocation [Mb/s]	C, Uniform Allocation [Mb/s]	C, Proposed Allocation [Mb/s]
4	14.641	14.092	16.434
6	18.388	18.010	20.689
8	22.914	22.425	25.401
10	27.339	27.190	30.376
12	33.003	32.262	35.675
14	38.256	37.819	41.378
16	43.991	43.464	47.159
18	50.247	49.320	53.120
20	56.102	55.373	59.224

**Note*: U - uncorrelated fading, C - correlated fading.

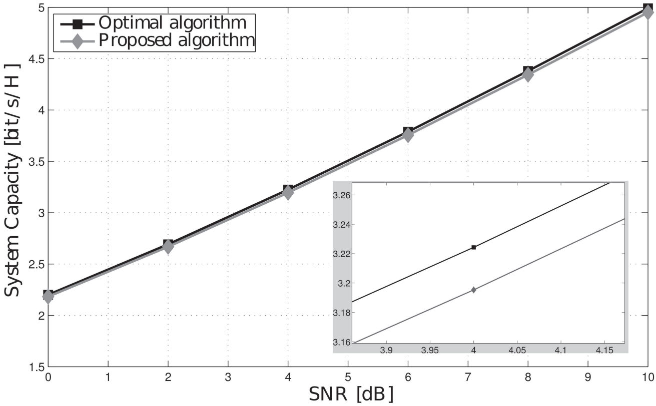

Figure 5-3: System capacity comparison of proposed algorithm with optimal solution assuming covariance matrix knowledge at the transmitter.

the results of Table I we have considered eight ARs and the cases of correlated (C) or uncorrelated (U) fading channel in NLOS conditions. In particular, the case of U-fading channel has been considered mainly for comparison purposes since the uniform allocation approach is the best solution under this condition. For the sake of completeness results in terms of bit error rate (BER) are provided in Figure 5-4. The good behaviour of the proposed approach is again evident in this figure. As for the advantages of replacing the correlation matrix estimation by the knowledge of the $\gamma_u[k]]$ terms according to (5.12), we can note that by considering a time interval on which the propagation channel can be assumed as stationary [6] the accuracy of the estimation of the $\gamma_u[k]]$ terms can be

[6]Simulation results have been derived by considering a time interval for which the stationary property of the communication channel is applicable.

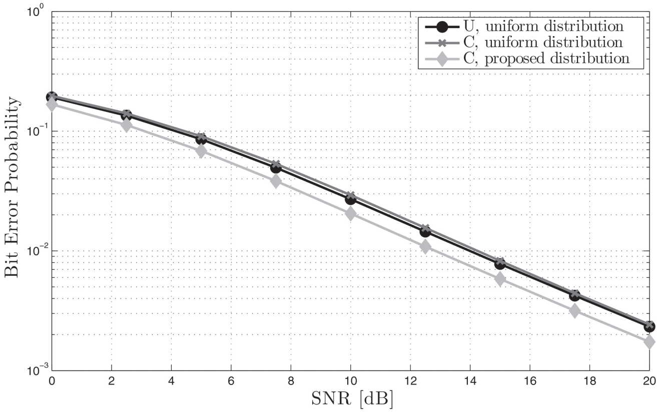

Figure 5-4: BER performance.

improved by taking the average of all the available channel gains samples $\tilde{H}_u^{(i)}[k]$ until the current one (i-th) as

$$\tilde{\gamma}_u^{(i)}[k] \cong \frac{1}{i} \sum_{j=1}^{i} \left| \tilde{H}_u^{(j)}[k] \right|^2 \tag{5.15}$$

Moreover, whenever it is necessary to lower the memory use amount, we can resort to the use of a cumulative averaging approach given by

$$\tilde{\gamma}_u^{(i)}[k] = \frac{(i-1) \cdot \tilde{\gamma}_u^{(i-1)}[k] + \tilde{H}_u^{(i)}[k]}{i} \tag{5.16}$$

without compromising the estimation accuracy. Equations (5.15) and (5.16) have been derived under the assumption that the statistical characteristics of the channel do not change over the considered interval time. Usually, over longer time intervals the stationarity property of the channel is no longer applicable so that the cumulative average $\tilde{\gamma}_u^{(i)}[k]$ has to be reset periodically.

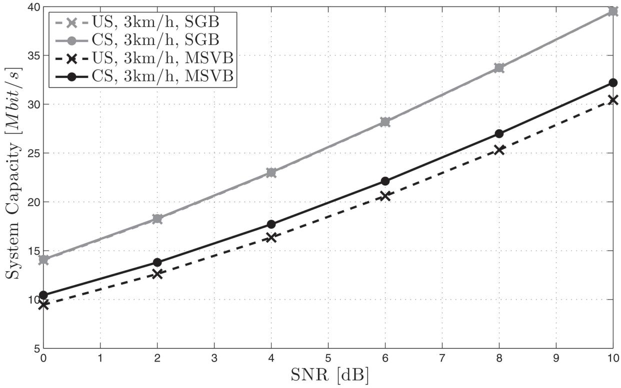

Figure 5-5: System capacity comparisons for AR speed of 3 km/h (NLOS).

Figure 5-5 and Figure 5-6 show the performance of the proposed subcarrier allocation algorithm based on the estimation of the subcarrier mean square value, through the cumulative average method (indicated as MSVB, mean square value based) considering either correlated or uncorrelated channel taps. 5-5 and 5-6 refer to NLOS and LOS propagation conditions, respectively. These results have been compared with those obtained by the proposed subcarrier allocation algorithm based on the instantaneous subcarrier channel gain values [7] (denoted as SGB, subcarrier gain based).

In both cases we have assumed the subcarrier channel gains evaluated during the previous frame. Usually, subcarrier allocation schemes exploit instantaneous subcarrier

[7]Each subcarrier is allocated to the AR that presents the maximum subcarrier channel gain $\tilde{H}_u^{(i)}[k]$ instead of the maximum mean square value $\gamma_u[k]$ as assumed in the proposed method.

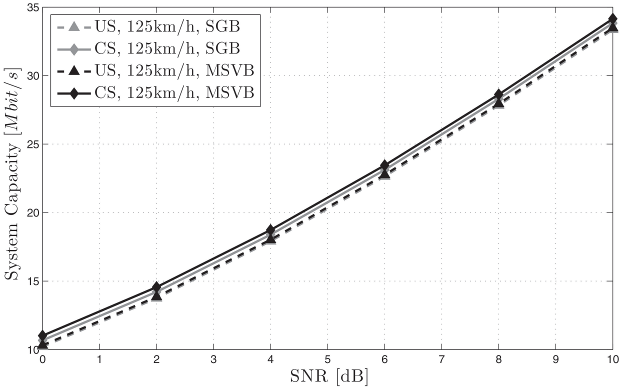

Figure 5-6: System capacity comparison for AR speed of 125 km/h (LOS).

channel gain values to adaptively assign the subcarriers to the users. This requirement cannot be fulfilled in practical applications where usually the subcarrier channel gains are estimated on a given frame and used at the next frame, i.e., at least with a latency of one frame. This constraint strongly degrades the performance whenever the channel coherence time is lower than the latency related to the channel gains estimation. In this case the channel estimation quickly becomes outdated and, as a consequence, the subcarriers allocation scheme is no longer optimal. This drawback can be overcome if correlation is present because the mean channel gain of each subcarrier remains constant over a long time interval and it can be used for subcarrier allocation instead of instantaneous values. In 5-5 it is possible to note that under low user mobility conditions (i.e., 3 km/h) the allocation algorithm performs better using instantaneous subcarrier gain values (SGB) achieving the same performance in the cases of correlated and uncorrelated taps. Conversely, for fast ARs (5-6) the system behaviour in the two cases (i.e., correlated and uncorrelated taps) is different, since for the correlated taps case the mean square values (MSVB) approach achieves better performance than the SGB alternative. Hence, from 5-5 and 5-6 we can state that for the uncorrelated channel taps case the SGB approach results in the best solution, while for the correlated channel taps case the MSVB approach becomes the most suitable solution. In particular, on the basis of the achieved results, we can stress that the higher the relative speed between the two terminals (or equivalently the latency related to the estimation of the channel gains with respect to the channel coherence time), the greater the improvement of performance will be. In addition SGB alternative requires the knowledge of instantaneous channel gains, hence, the performance is more dependent on the channel estimation accuracy. [8]

Moreover, in the case of tap correlation it is possible to lower the overhead due to the channel parameters estimation rate. As an example, Figure 5-7 shows the performance of the proposed algorithm when the channel parameter estimation is updated with a relaxed frequency. In particular, this figure shows the case of an updating rate of one time every five frames for both SGB and MSVB approaches.[9] From Figure 5-7, it is possible to highlight the better behavior of the MSVB approach with respect to the SGB alternative.

[8] Channel estimation errors have not been considered in deriving the simulation results. However, errors could have a deeper impact on SGB method performance rather than on MSVB where an average value is considered.

[9] Roughly speaking, the computational cost of the channel estimation is reduced of about 80%.

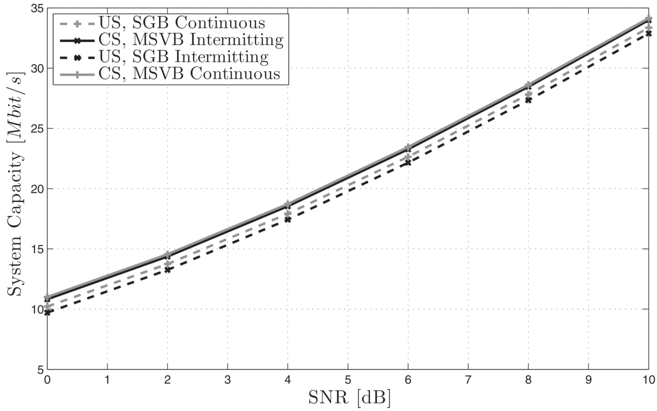

Figure 5-7: System capacity comparison for AR speed of 125 km/h (LOS), assuming channel estimation performed with reduced frequency.

5.4 Conclusions

Here, is proposed an efficient subcarriers allocation method in order to improve the communication capacity in new generation ATM systems. In particular, the approach devised here has been based on a smart use of a specific characteristic (i.e., tap correlation) of the communication channel. It has been shown that the use of long term channel statistics allows a significant gain for the system capacity and permits less use of the control channel, reducing the system overhead. In order to relax the computational complexity with respect to similar approaches previously proposed in the literature [62] [64], an efficient heuristic has been proposed here. The results presented, have highlighted that the proposed method allows performance close to the optimum one with a significant reduction in the implementation complexity, hence making it an attractive solution for the use in future AeroMACS systems.

Part II

Professional Mobile radio

Chapter 6

TETRA Broadband

The requirements of public security operators highlight the need to overcome the limitations posed by actual PMR standards introducing new technologies able to support high data rate communications while maintaining all the peculiarities of professional systems.

Voice will remain an essential application in PPDR, however the availability of multimedia applications brings a number of benefits. Data pass from being an accessory support tool to become a Mission Critical service at the same level of voice. A larger amount of data and the availability of multiple applications permit to increase the situation awareness and decision making efficiency, avoiding that corrupted, missing or late information result in wrong decisions that, in the worst case, can cause loss of life [70, 71]. To respond to these needs an evolution of legacy Professional Mobile Radio system has to be foreseen and performed. This system, namely TETRA Broadband (TETRA-BB) takes advantages from achievement of commercial market in Long Term Evolution (LTE) development, adding the necessary added values

for use in professional environments.
In this Chapter an overview of the TETRA-BB development process is performed, then issues related to deployment of new communications network are analysed. In particular a possible deployment path of TETRA-BB network, starting from the sharing of resource with commercial operators networks to building of fully dedicated professional networks, is illustrated. A critical analysis of spectrum resources availability is also performed.

6.1 TRETRA-BB standard evolution process

Globally, the voice and data communications for mission critical applications are distributed through different technologies such as TETRA, TETRAPOL, P25 and GSM-R. The increasing need to be able to support the services of the traditional professional networks with advanced applications that require broadband communication technologies has led governments and organizations that deal with critical communications to take an interest in the development of broadband data services.

In USA, the National Public Safety Telecommunications Council (NPSTC) together with other organizations such as the Association of Public-Safety Communications Officials (APCO) and the National Emergency Number Association (NENA) has highlighted the need to have a national interoperable standard for next generation of public safety networks with broadband characteristics.

In June 2009, NPSTC has indicated LTE as the technology chosen for the evolution and contacted 3GPP, the body in charge of the development of the LTE standard, to discuss changes to be made to the standard in order to meet the needs of public safety applications. The hope of NPSTC was to integrate the greatest number of features in future versions of the LTE standard making it fully responsive to the needs of PRM and at the same time attractive for business and consumer applications.

In Europe, the main organizations dealing with Critical Communications initiated a similar process. The TETRA and Critical Communications Association (TCCA) is an association that takes care of collecting the opinions of authorities and national governments, application providers, manufacturers, operators and end-users of TETRA, to derive from them requirements and operational needs in the use of these technologies. Within TCCA, the Critical Broadband Communications Group was created with the purpose of driving the development and adoption at global level of standards and common solutions for mission and business critical applications. CCBG collaborates also with the TCCA Spectrum Group and PSC-Europe to find an appropriate portion of harmonized spectrum to be used for these applications. In particular, the purpose of CCBG is to collect requests, needs, and requirements proposed by subjects directly involved in the use, production and management of these technologies and synthesize them into requirements to produce a system that is able to allow users to ubiquitously and continuously access to information systems, Internet and Intranet through a broadband connection, in the same way that access to the services provided narrowband voice and data by means of TETRA, TETRA, P25, etc. In [72] TCCA expressed the intention to drive the development of broadband technology solutions for mission and business critical operations. After reviewing various technologies, it has identified LTE as the system characterized by better prospects and therefore has expressed its intent to collaborate with 3GPP to include the required functionality in the standard. TCCA has also expressed its intention to cooperate with ETSI to create a common and interoperable solution required by the user community Critical Communications.

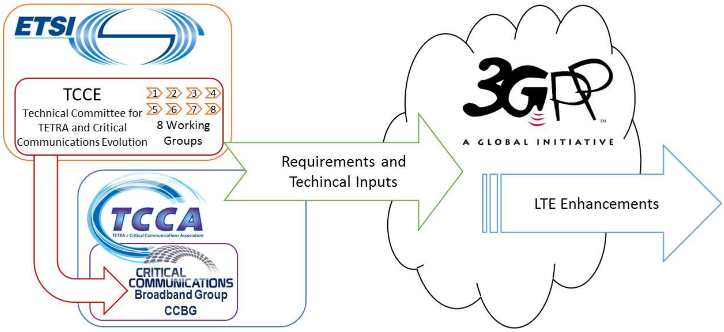

Figure 6-1: Standardization bodies relationship.

ETSI (European Telecommunications Standard Institute), as promulgator of the TETRA standard, is directly involved in the evolution process. Inside ETSI, the Technical Committee (TC) TETRA and Critical Communications Evolution (TCCE) has the task of:

- Providing standard for secure voice and data communications whose development process is directly driven by the needs of users.

- Collecting and specifying the requirements from major stakeholders such as emergency services, government, military, transportation, industrial organizations and operators of Public Access Mobile Radio (PAMR).

- Maintaining and developing the existing TETRA standard.
- Coordinating European and international standardization organisms.

TC-TCCE is composed of 8 working group. It maintains an own representation into the CCBG of TCCA to contribute to development of requirements, use cases and architecture of mission critical communications that can results in a unified standard. Moreover, TC-TCCE participates in the CEPT/ECC project team PT 49 to find at European level additional spectrum resources for PPDR services. Requirements and technical recommendations issued by NPSTC, TCCA and ETSI are supplied to 3GPPG as input for the development of LTE standard able to support requirements typical of PMR and at the same time maintaining its appeal for commercial applications. This synergic collaboration is aimed to obtain the best results in this evolution process.

6.2 Ongoing steps in LTE standard

Mission critical communications are characterized by more stringent requirements in terms of resilience, reliability, security and availability than the commercial counterpart. At Present, LTE standard is not fully compliant to professional needs. The primary criticality concerns generic support to voice calls over IP (Internet Protocol). LTE has been developed mainly for fully IP data communications, as a consequence it does not include the circuit switching domain on which rely voice calls and short messaging in 2G/3G technologies.

At present, there are various technological options for voice call on LTE, that include Circuit Switched Fallback (CSFB), Volga and One Voice and Voice over LTE (VoLTE). CSFB consists in transferring users involved in voice calls for the duration of the call on a Circuit Switched domain provided by a pre-existing radio technology such as UMTS and GSM. This technology has two many drawbacks: LTE and legacy networks shall to simultaneously co-existence and call setup time is very high significantly degrading the user experience. This solution can be adopted in the short term on the commercial networks, but due to its poor performance, it is not feasible for PMR networks. A solution in the medium/long term is represented by Voice over LTE via Generic Access Network (Volga), defined by the Volga forum [73]. Volga is based on the concept of connecting the existing Mobile Switching Centers to the LTE network via a gateway. In this case, because communication does not fallback to legacy networks, setup times are lower than the CSFB. The most advanced solution consists in the support of voice calls through IP Multimedia Subsystem (IMS). IMS is an IP-based architecture of connectivity and control to managing multimedia services over IP standardized by 3GPP. One Voice, an association of leading operators and manufacturers in the field of telecommunications, in 2009 defined a 3GPP compliant basic profile, which contains the network of terminal features to support the IMS basic voice services. The organization of operators GSMA (Global Systems for Mobile Communications Association) has extended the work done by One Voice, producing a more advanced profile called VoLTE (Voice over LTE) published in [74]. At present, VoLTE is getting greater interest from the industry, but it is not the unique solution in a context of fragmented approaches to the problem. Main

VoLTE features consist of:

- set-uping of the transmission path between the terminal and IMS;
- providing security features for user authentication;
- providing the core functionality for the establishment and termination of the call (via SIP, Session Initiation Protocol);
- support to call forwarding, caller ID presentation and restriction, call-waiting and multiparty conference.

However, VoLTE is not able to support primary functions of mission critical communications, i.e. group communication, direct mode communication, push-to-talk management.
A Group Communication Service is a fast and efficient mechanism for the distribution of the same content to multiple users in a controlled manner [5]. The service shall support data, voice and video communications and the participation of an user to multiple groups at the same time. Group creation, affiliation and detach shall to be dynamic and group definition can be done with restrictions on a geographical basis. Group communication shall support "Push-to-talk", the ability to make a call and start talking without waiting for the response from the receiver. Requirements associated with this service are the ability of the caller to be advised if no user has received the call, or to be notified that a user joined to the chat session, and the ability to receive a warning if the call is coming from the CS domain.
Currently, 3GPP specifications for 2G/3G networks include the definition of "Push to talk over Cellular (PoC)" technology introduced for the first time by the Open Mobile Alliance (OMA). The PoC has been designed to support ad-hoc

and group chat-like communications for both home users and business applications, as well as to enable individual users to dynamically create ad-hoc groups for voice communications and chats. PoC, however, is not able to satisfy the requirements of performance and interoperability required by mission critical communications. To enabling this service an appropriate modification of the IMS framework is needed, especially to maintain low the setup time of the call that constitutes a critical aspect in contexts PMR.
Another basic functionality regards the possibility of establishing direct calls between users regardless even outside the network coverage. This implies the need to provide the user device with the capacity of local discovery of other devices without the network management control and with the ability to establish the call.

To address these features, 3GPP started a number of activities performed by dedicated study and work items [75]:

- Group Communications System Enablers for LTE (GCSE_LTE) is an active Work Item Description (WID) in 3GPP SA1 (System Architecture Working Group 1) that addresses the LTE standard support of voice/video/data group communications. Its output will be integrated in Release 12 of the standard.

- Proximity-based Service (ProSe) is a WID interested in Direct Mode e device-to-device communications. It defines User Equipment services to discovery, direct communications and relay capabilities that will be included in Release 12.

- EPC-less E-UTRAN Operation for Public Safety is a Feasibility Study preparatory to a Work Item that addresses the network resilience topic. Network operations have to be guaranteed also when the connection between base stations and the core network is not available. Consequently it shall be possible to setup a local connection between base stations to extend connectivity in a geographic area. This feature is expected to be integrated in the Release 13 of the standard planned for 2016.

- Public Safety Broadband High Power User Equipment for Band 14 for Region 2 defined requirements of compatibility and coexistence of LTE in 700 MHz Band. It has been completed in Release 11.

- Study on LTE Device to Device Proximity Services - Radio Aspects (FS_LTE_D2D_Prox) is a feasibility study item on radio aspects related to proximity services. It will be completed in Release 12.

- Study on Group Communication for LTE (FS_LTE_GC) evaluates the ability of LTE of meet PPDR requirements for group call communications. It will be completed in Release 12.

Table 6.1 summarizes the main documents either Technical Reports and Technical Specifications related to each study/work item. Most of major updates are expected to be published within the upcoming Release 12 of the standard. However the 3GPP interest in PPDR communications is present also in Release 11 (frozen in 2012) where an high power level class of terminals was added to US PPDR users. Also in future Release 13 and 14 further enhancements about Mission Critical Push-to-talk (MCPTT) and air-ground-air (AGA) are expected.

Table 6.1: 3GPP relevant documents.

Group Communications System Enablers for LTE	TS 22.468, TR 23.768, TR 23.468 , TR 33.888
Proximity-based Service	TR 22.803, TR 23.703, TS 23.303, TR 33.833
Public Safety Broadband High Power User Equipment for Band 14 for Region 2	TR 36.843
Study on Group Communication for LTE	TR 36.868

6.3 PMR Network Deployment Solutions over LTE

Despite the big work still needed in standard developing and harmonization, the migration of PMR network from current technologies to LTE is expected in a near future; costs and timing for the deployment of LTE dedicated network are not negligible. The simplest way for the initial diffusion of high data rate services for PMR network based on LTE technologies is the use of already deployed commercial networks. PMR operators can partially or completely rely on already active commercial networks for providing broadband services to their users. LTE networks are composed by several modules and equipment, which can be collected in different logical domains:

- the services domain, managing the contents and the services to be provided to users, injecting the traffic in the network through a Service delivery platform;

- the Evolved Packet Core network, mainly responsible of the control functions;
- the radio access network, composed by the eNodeB;
- the users equipment domain.

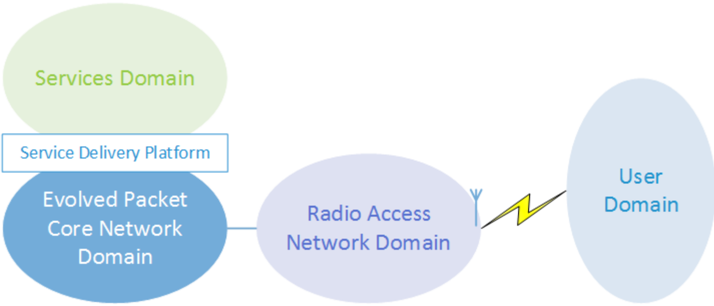

Figure 6-2: LTE Networ domains.

Professional and commercial operators may partially or completely share one or more domains of their networks, in order to guarantee the services needed by different kinds of users. For PMR communications, some domains, as the services domain, shall be dedicated, due the specific services to be provided, while others, as the core network and the radio access network, can be shared between the professional and the public networks, as represented in Figure 6-3. In an initial phase the PMR network can completely relies on a commercial network. In this case the authority that manage the TETRA-BB network acts as an Enhanced Service Provider and the quality of service assurance, as well as the reliability and the availability of the network, are fully committed to the contract with the operator. It is important to note that in this case

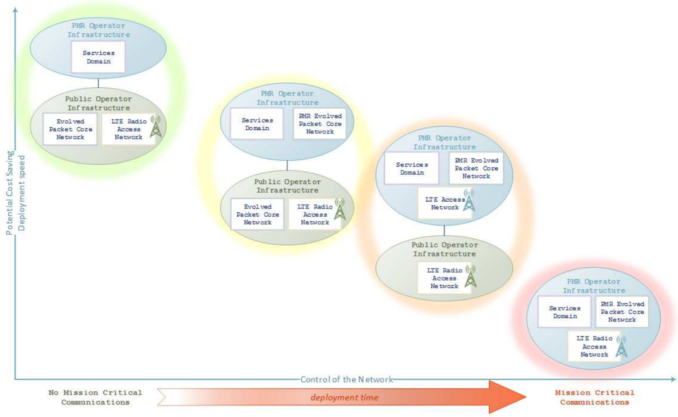

Figure 6-3: Network Deployment Evolution.

it is risky to use such network for Mission Critical communication [76]. 3GPP provides recommendations and technical specifications for the sharing of network devices and modules between operators [77], commonly used in commercial networks [78]. Resort to services provided by Mobile Network Operator (MNO) is the simplest option, and probably also the cheapest one, with the strong drawback of not having the direct control of the network and of the QoS policies. All the sensitive characteristics of PMR services are demanded to the negotiation of an appropriate Service Level Agreements (SLAs) with the operator. Moreover in particular critical situation, commercial networks are usually congested, overloaded and exposed to failures, with unpredictable effects on high priority emergency communication. The network evolution is by the way oriented to getting the professional network gradually more independent from commercial networks, with a step-by-step approach. In the second and the third configurations represented in the figure, the PMR operator acts, from a network architectural point of view, as a Mobile Virtual Network Operator (MVNO), in Core Network Sharing or Full RAN Sharing configuration respectively. The two networks partially share their EPC can follow the Gateway Core Network (GWCN) specifications of 3GPP [77]. In the third configuration the professional network expands to the eUTRAN, with a partial coverage of the territory. The current radio coverage of European PPDR networks is nationwide very high, usually over 98 per cent of the landmass [79]. The needs of a deep diffused network on the area of interest is a requirement representing one of the major obstacle on the upgrade of current PMR networks over new and more efficient communications technologies. LTE allows the sharing of part of the eUTRAN access network between operators, following the Multiple Operator Core Network (MOCN) 3GPP specification. The last presented configuration represents an autonomous LTE PMR network, with full radio coverage of the territory. In this case, the PMR operator is the owner of the eUTRAN network but some form of passive sharing, such as mast sharing or site sharing, can be anyway implemented for reducing the costs and the environment impacts of the new network deployment. The experience in network sharing for commercial network shows a strong saving in both Capital Expenditure (CAPEX) for the network deployment and Operating Expenditure (OPEX) for the network maintenance, in the order of roughly 20-30% [80] [81]. The cost saving obtained by Telcos joining forces with rivals in their markets to build, run, and maintain mobile networks, can be replicated by professional operators during the deployment of a new generation PPDR network. Of course, the costs saving is earned partially sacrificing some of the controls that a standalone operator has over its network, which shall be safeguarded through severe SLA contracts. Considering both the appeal of costs reduction and the strategic interests, some forms of sharing can be anyway recommended not only in the initial coverage-driven roll-outs of the PPDR network, but also as solution for the distribution of professional services in areas where there are seldom or temporarily needed, as low population density or rural areas, where network sharing results more convenient for both CAPEX and OPEX [81].

6.3.1 Sharing of network apparatus

LTE networks are full-IP and this advantages and makes easy the chance of successfully share physical equipment between two logical networks. For this reason the possibility of share part of the core and backhaul network between operators has been taken into account not only for purpose of cost saving but also for increasing the network resiliency, for providing a fast recovery mechanism in case of network failure or for managing the peak traffic problems, which is one of the most common issue on PPDR networks.
The network sharing model between TETRA-BB and a commercial operator is similar from an architectural point of view to the sharing of the network between civil operators.
There are five possible levels of network sharing between operators, with three possible sharing modes: roaming, sharing of passive component, sharing of active component.

Ran Sharing has to be adopted in second and third configurations. 3GPP defines two possible architecture for Ran Sharing:

- MOCN (Multiple Operator Core Network)
- GWCN (Gateway Core Network)

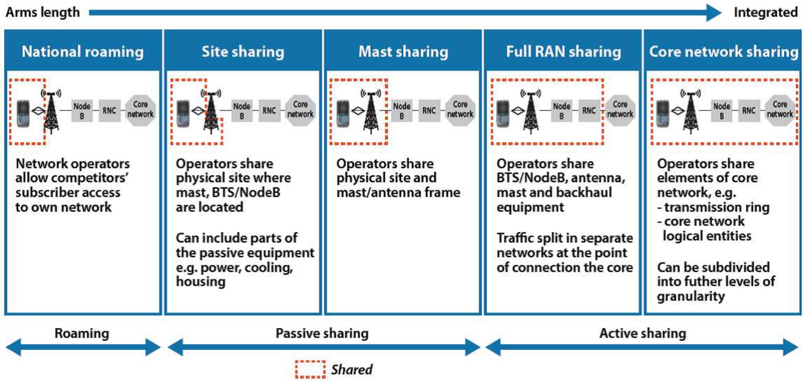

Figure 6-4: 3GPP Network Sharing Models.

In the first case operators share only the radio access network while in the second case they also share the Serving Gateway (S-GW) and the Mobility Management Entity (MME) of the Evolved Packet Core too.

Virtual Operator Configuration

First configuration shown in Figure 6-4 corresponds to *Core Network Sharing*. TETRA-BB management can be compared to *Enhanced Service Provider (ESP)* defined by 3GPP. In the commercial networks a mobile virtual network operator is a company that offers mobile services without owning the license for the radio spectrum, using a part of the infrastructure of a mobile operator (MNO) with whom he has commercial agreements.

Although this definition is not directly applicable to TETRA-BB network, the architectural solutions are similar. The operator that provides the access network is called MHO (Mobile Host Operator). For the hosted network are provided

Sharing mode	Sharing level	Shared MHO elements	TETRA-BB network domains	Configuration (from Figure)	Mission Critical communications support
Active Sharing	Core Network Sharing	E-UTRAN, EPC	Services, Terminals	Config. 1	No Mission Critical
	Full RAN Sharing	E-UTRAN EPC (partial)	Services, Terminals, EPC (partial)	Config. 2	
		E-UTRAN	Services, Terminals, EPC	Config. 3	
			Services, Terminals, EPC, E-UTRAN (partial coverage)		Mission Critical
Roaming	Roaming	Out coverage	Services, Terminals, EPC, E-UTRAN (partial coverage)	Config. 3	
Passive Sharing	Mast Sharing	antenna e supports	Services, Terminals, EPC, E-UTRAN	Config. 4	
	Site Sharing	eNB sites	Services, Terminals, EPC, E-UTRAN		

Figure 6-5: TETRA-BB and commercial networks configurations.

two modes:

- Full MVNO (Mobile Virtual Network Operator)
- ESP-MVNO (Enhanced Service Provider)

In the Full MVNO solution, the TETRA-BB has its own core network, and relies on the MHO for the access network, establishing a roaming relationship with the MNO; users are treated as guests under the MNO network coverage. An ESP, however, does not have its own switching network, or any part of the Core Network, but only the infrastructure to VAS (Value Added Services) services provisioning. Not owning the Core Network, the SIM card must be issued by MHO, which has to supply the basic connectivity too.

The TETRA-BB network working as ESP can provide IP telephony and other advanced services, making use of the transmission capacity of the evolved LTE network support.

The relationship with the MHO is very tight, because the users of the network TETRA-BB are, to the effects of LTE networks, users MHO, since they use its SIM. Thus to ensure that an adequate service experience to PMR users, commercial arrangements very well defined, through policies of quality of service (QoS) constraints, have to be defined.

The 3GPP standard also includes two other models of virtual operator:

- SP (service Provider) MVNO model;
- Reseller MVNO model.

In both cases the virtual operator is responsible only of handle sales offers and supply user support. Such agreements may cover only commercial networks and can not be used for the purpose of achieving PMR networks.

GWCN (Gateway Core Network) Configuration

Second configuration in Figure 6-4, in addition to the eNodeB, the commercial and the TETRA-BB network share the MME (Mobility Management Entity) and possibly other components of the core network EPC (Evolved Packet Core).

MOCN (Multiple Operator Core Network) Configuration

MOCN mode shown in third configuration in Figure 6-4 foresees the virtualization of two or more radio networks through the same hardware.

[77] foresees two sharing mode:

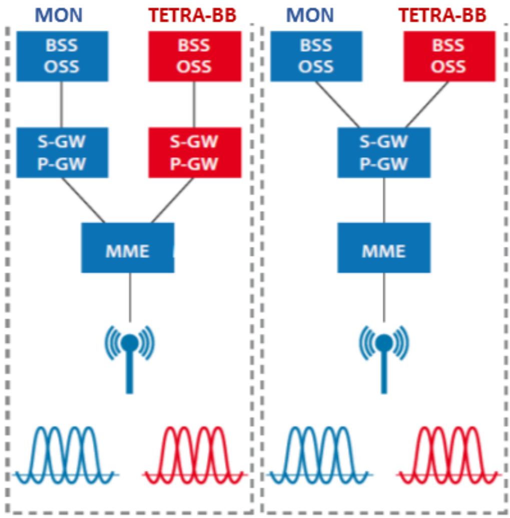

Figure 6-6: GWCN configuration.

- Dedicated Spectrum:networks share only the hardware infrastructure that transmits different carriers dedicate to each operator (MORAN configuration).

- Shared Spectrum: networks share the same frequency resources.

MOCN and MORAN architectures permits to an eNodeB to connect to different EPCs. MME, SGW and PGW are distinct for commercial and TETRA-BB operator. Where TETRA-BB network coverage is not available, PMR users can be served through commercial network.

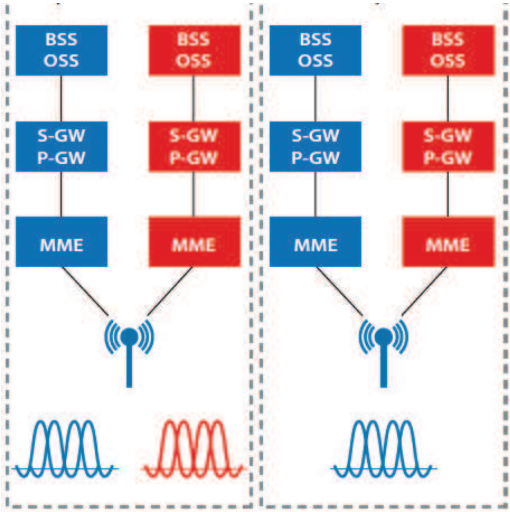

Figure 6-7: MOCN configuration.

Autonomous TETRA-BB Network and Passive Sharing

Site Sharing could be used even when the TETRA-BB network will completely deployed to reduce costs of network deployment and management.

6.4 Radio Spectrum Evaluations

Suitable radio spectrum resource availability is a major issue for an effective deployment of mission critical communications. Traditional professional communications, as TETRA and Tetrapol in Europe e APCO P25 in North America take place in reserved portions of spectrum that are located in the 380-400 MHz or 150-174 and 412-521 MHz bands. These bands are intended exclusively for narrowband voice or low bit rate data communications (few tens of kbps). Also TETRA wideband extension that is TETRA Enhanced Data System (TEDS), which is able to reach at most speed of the order of hundreds of kbps. To support broadband communications additional resources shall however be located. In July 2007, the USA Federal Communications Commission assigned a portion of 700 MHz band, released by TV analog broadcast, to safety communications [82] with the aim of establishing a nationwide interoperable communication network. FCC designated the lower portion of the spectrum (i.e. 763-768/793-798 MHz) for broadband communication and the upper half (i.e. 769-775/799-805 MHz) for narrowband communications. Later, in 2012, the reserved radio spectrum has been extended to 2x10 MHz in band 758 MHz-768 MHz and 788 MHz-798 MHz.

In Europe, PPDR community expressed the need to identify a portion of spectrum below 1 GHz to be harmonized [83]. A study conducted by CEPT-ECC with the participation of the EU Councils Law Enforcement Working Party (LEWP) [84], concluded that the minimum spectrum needed for broadband data services is 2x10 MHz, in accord with decision taken in USA. These resources do not include voice services, Air Ground Air, Direct Mode Operation and adhoc networks that could be allocated in additional spectrum portions.

Additional studies are ongoing within the CEPT-ECC Working Group FM 49 [85] with the aim of finding suitable solutions for radio spectrum harmonization. At large, the harmonization concept implies the identification of an unique spectrum range, a tuning range, within which each administration can select the portion of spectrum that has to be used locally. In facts, not all frequencies can be available in every country, but terminals are intended to operate in whole range. This solution presents numerous advantages in terms of increased interoperability and cooperation between different organization or countries, and leads to higher volume of business, that results in the possibility of obtain wider economies of scale for terminals and infrastructures and equipment availability. Harmonized spectrum makes more difficult the emergence of competing technologies. Moreover, having a dedicated spectrum permits to fulfil critical operation requirements of resilience, availability, coverage and security.

Locating a common portion of spectrum is subject to practical constrains due to coexistence with other systems and electromagnetic compatibility other than the need to mediate different interests (public and private operators, final users, etc.) and countries politics that reserve resource following local needs and contexts. The use of frequency below 1 GHz is preferred due to the better signal propagation conditions respect to higher frequencies, that implies better coverage both in outdoor and indoor environments, and permits to fulfil critical communications performance requirements. Currently, two frequency bands have been designated as possible candidates: 700 MHz (IMT band 694 790 MHz) and 400 MHz (subranges 410-430 MHz and 450-470 MH). The former solution is that preferred by the majority of FM49 members. It has the advantage of allowing the deploying of the network both as a dedicated and as commercial network or a combination of theme. Inside FM49 there are different opinions about the beneficial of using an harmonized exclusive band for PPDR: some countries take the view that will be more changes to allocate spectrum at national level if there is not competition with commercial operators. Other administration, instead, believe that dedicated spectrum will create a niche market that negatively impacts on economies of scale and interoperability. The 400 MHz band has the advantage of good propagation characteristics, permitting to reduce the number of sites necessary to provide the radio coverage. On the other hand, in some countries it is not possible to find enough spectrum resources in the 400 MHz band, where only 2x5 MHz bands may be available. This leads to hypothesize a future scenario where critical communications will be allocated in the 700 MHz band, using the 400 MHz band in some countries as complementary, taking advantage from a flexible solution with the drawback of a possible fragmentation. To make this possible and to respond to market needs it is indispensable that manufactures will produce terminals able to operate in channels allocated in wide range of frequencies . The results of this study will be published in the ECC 199B Report and a final decision is expected to be ratified in an ITU-R Resolution of the ITU-R World Radio Conference in 2015, as consequence of agenda items 1.2 and 1.3 that deals with use of 694-790 MHz band for mobile services and PPDR needs.

Special attention has to be paid to complementary applications, as they require additional spectral resources, some hypothesis can be drawn [86]. For instance, ad-hoc networks can be used in case of unexpected big event to increase the capacity of the congested network, or where the network infrastructure is not available. The required spectrum can be obtained installing additional temporary base stations or repeaters or in the 4940-4990 MHz band, used by PPDR in the ITU Region 2 and 3. Another complementary application is Air-Ground-Air communications, mainly used for a monitoring video streaming from a camera mounted on an helicopter or an Unmanned Aerial Vehicle (UAV). In this case the use of external frequency band is demanded to national decisions subject to specific regulatory and technical conditions. Finally DMO communications, allowing terminals to communicate directly in reduced spaces without the intervention of the network infrastructure, can be allocated in the same frequency band of the network (e.g. 700 MHz) or in other spaces to be identified in commercial IMT bands.

To reserve a portion of spectrum to PPDR, in other hand, produces a cost comparable with the value that the spectrum would have if allocated to other applications as commercial mobile communications. As an example, in [87] this cost has been estimated tacking as reference auctions of 800 MHz band took place in European countries. By comparing it with an estimate reduction of cost for disaster managing, due to the use of broadband PPDR communication, [87] demonstrates the effectiveness of the deployment of a dedicated radio spectrum for these applications.

Chapter 7

Group Calls in TETRA-BB

Group Call (GC) represents one of the most important and indispensable service of PMR networks. It enables an efficient management of the rescue teams and permits to send commands to all the PSS operators in a disaster area and share information. The communications addressed to more than one user can be distributed with a Point-to-Point (P2P) approach, using one unicast transmission for each involved user, or with Point-to-Multipoint (P2M) flows. In LTE the distribution of multicast communications is demanded to the evolved Multimedia Broadcast Multicast Service (eMBMS), introduced in the 3GPP standard starting from Release 10 [2]. It has been mainly developed for multicast and broadcast distribution of multimedia data, such as video streaming provided by external sources that act as service providers. For this reason only a downlink channel is considered.

Providing mission critical services in a public safety LTE network involves a proper network architecture solution in

order to achieve and maintain the required performance and reliability levels. Hence, it is easy to foresee that several research efforts and further releases of the LTE standard will be needed before this task will be fully accomplished.
For the above reasons, we discuss current approach to GCs management in a public safety LTE based network, analysing recommendations provided by 3GPP [4, 5]. Then two efficient architectural solutions are proposed, analysing how the network shall be configured and improved, introducing new hardware and software elements performing specific tasks related to multicast distribution of mission critical services. Compliance of these configurations to the PPDR needs are then assessed in Chapter 8.

7.1 Functionalities description

Group calls is the service that most characterizes professional radios, constituting the main source of traffic in today networks [76]. It permits to manage voice communication between multiple users in a flexible, dynamic and efficient way. In an operational scenario, most of the communication will take place through group calls. Moreover, part of the communications that appear to the user as individual calls, can actually be treated by the system as group calls, where members of the group are, in addition to the parties, communication control and monitoring centers or apparatus intended to record and automatically archive communications on the network.

Group calls are PTT (Push-to-Talk) thus performed in half-duplex mode, hence only one user at a time is authorized to talk [5]. During a group call, the network shall allow the terminal to:

- ask the call floor by pressing the (virtual or physic) PTT button;
- notify the end of the broadcast with the release of the PTT button.

Group calls are inherently "direct set-up signalling"; users belonging to the group, in fact, are not advised of the incoming call and do not have to accept it.

The TETRA-BB system shall provide necessary services for group calls management, such as:

- distribution of data and voice calls to pre-configured static groups;
- management of dynamic groups;
- management of call priority to allow the connection of the group even when the network is loaded;
- management of groups created on the basis of users location.

In order to provide this functionality, the network has to be able to manage groups by means of dedicated reporting mechanisms identifying the location (geographical or logical in the network) of the users belonging to a group, however a such feature is not natively supported by LTE.

Requirements that have to be satisfied by TETRA-BB network can be deduced by operative conditions of professional system [84] [5]. Figure 7-1 summarizes main performance requirement for group calls.

A relevant characteristic of voice call in professional system beside the very low call establishment time (up to 300 ms) is a reduced latency of data transmission that permits

Performance	Requirement	Unicast	Multicast
Packet Delay	<150ms	OK	OK for the required number of active groups
PELR (Packet Error Loss Rate)	$<10^{-2}$	ok until 210/270 users. Performance requirements are not fulfilled	ok 120/125 groups.
Number of active groups	>36 (suggested >75)	Not influential, the limit is the total number of users	OK
Number of users per group	>500	No	Not influential, the limit is the total number of groups
Number of active users	>2000	No	Not influential, the limit is the total number of groups

Figure 7-1: Group calls performance requirements.

to obtain a great user experience. During a voice communication, delay experienced by packets shall be less than 300 ms. The average number of users per group is about 50 units. The network, however, shall to support up to 500 fully functional terminals for half-duplex voice call and short data service messages. The number of members of a group can reach 2000 users, above that the network can choose to forbid users from joining the group. [5, 88]

Native solution to support group calls in LTE is the use of eMBMS (evolved Multimedia Broadcast Multicast Services), the LTE built-in technology that permits to efficiently delivery contents to groups of users [2, 89, 90].

7.2 Multicast distribution in LTE

The distribution of broadcast and multicast flows in LTE is performed by Multimedia Broadcast Multicast Service (MBMS), introduced by 3GPP from Release 6. Afterwards

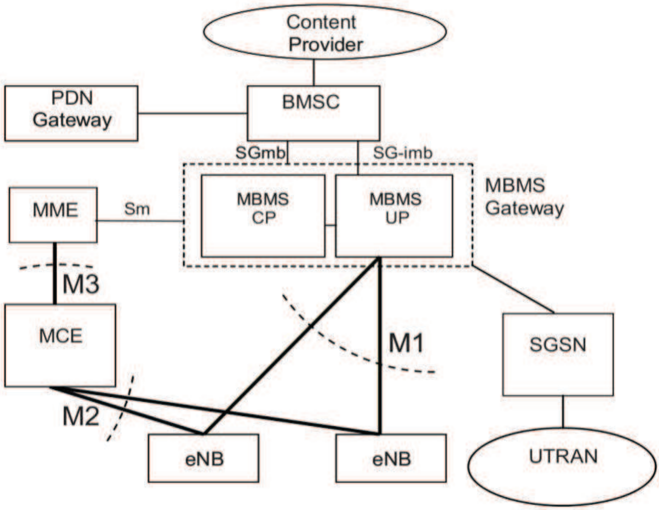

Figure 7-2: MBMS architecture.

new features have been implemented and from Release 10 the framework take the name evolved MBMS (eMBMS) [2].

From a structural point of view, the multicast and broadcast traffic can be distributed in two ways:

- Point-to-Point (P2P): the multicast stream is transmitted to the each EU through a different unicast traffic. As a consequence the same content is repeated several times depending on the number of receiving terminals with an high occupancy of resources;

- Point-to-Multipoint (P2M): an unique data flow is transmitted which is received by multiple users at the same time.

Transmission mode is chosen by EPC.

7.2.1 MBMS architecture

An *MBMS Service Area* is a geographic area where MBMS services are transmitted. In MBMS Service Area, multiple eNodeB can be enabled to transmit the same content in a synchronous and overlapping manner. This type of transmission is called *Multicast Broadcast Single Frequency Network (MBSFN)* and the involved geographic area is the MBSFN Synchronization Area. SFN transmission differ from Single Cell (SC-eMBMS) where each eNodeB sends it multicast flow independently from others. Within a Single Frequency Network can be distinguished one or more MBSFN Areas, composed of two or more eNodeB that transmit the same stream multicast or broadcast synchronously. An eNodeB can participate to different overlapping MBSFNs. In a MBSFN, similarly to a multipath propagation, a terminal receives several replicas of the same signal, each characterized by a certain delay. This approach permits to improve reception of UEs that camp on the cell edges.

MBMS content provider is usually external to LTE core network. *Broadcast-Multicast Service Center (BMSC)* acts as interface between Core Network and content provider and is in charge of traffic scheduling and flows synchronization within MBMS areas. *MBMS-Gateway (MBMS-GW)* is involved in Access Stratum (AS) and Non Access Stratum (NAS) tasks, e.g. control signalling (start, modification and termination of MBMS sessions) and MBMS flows addressing to eNodeBs involved in the communications. *Multicell Coordination Entity (MCE)* allocates MBMS flows into the transmission frame and receives from MME information about start, modification and termination of MBMS sessions. MBSFN data are encapsulated in MCH (Multicast Channel) transport channel, which at the physical layer is mapped into Physical Multicast Channel (PMCH). At physical level, a subframe in a frame can not be used for both multicast and unicast transmission. Moreover, the standard explicitly prohibits the allocation of multicast traffic within certain subframe, in particular:

- Subframes 0, 4, 5, 9 in FDD frames;

- Subframe 0, 1, 2, 5, 6 in TDD frames.

7.2.2 Radio Access Protocol and signaling

To support MBSFN the LTE standard has extended MAC layer functionalities introducing a new transport channel, Multicast Channel (MCH), and two logical channels Multicast Traffic Channel (MTCH) and Multicast Control Channel (MCCH).

MTCH transports MBMS data, while MCCH conveys information necessary to the reception of this service, including the allocation of the subframe and the modulation used for each MCH.

As shown in Figure 7-3, one MCCH channel and one or more MTCH channels are mapped within a MCH channel which is sent to the destination through the PMCH (Physical Multicast Channel) channel. In particular, there is only one signaling channel MCCH for MBMS area.

7.2.3 MBMS session start

Current LTE standard does not define UE session joining-/leaving procedures. Signaling consists only in messages exchange for session start and stop management. Figure 7-4

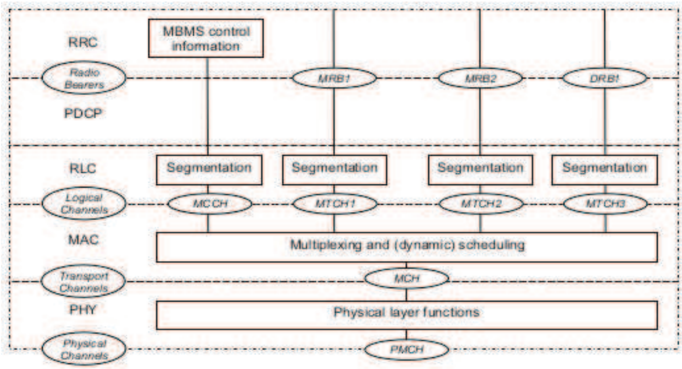

Figure 7-3: MTCH, MCCH e MCH channels.

illustrates the signalling messages used for the MBMS session establishment. The BM-SC initiates a session by sending to MBMS-GW the MBMS Session Start Request message containing indication about QoS, TMGI (Temporary Mobile Group Identifier), service area and service duration. The message is propagated to the MCE module that carries out the IP multicast address allocation and configuration of radio resources. Finally, the Session Start Request message and the MBMS Scheduling Information message are sent to the eNodeBs belonging to MBMS area. When the setup is complete, the EU belonging to the MBMS area are advised of the presence of a new service through the MCCH, MBMS and MBSFN Area Configuration Change Notification.

7.2.4 MBMS scheduling

Since MBMS data can be transmitted by multiple eNodeBs at the same time using the same format and physical resources, eNodeB can not perform a dynamic resource allocation as

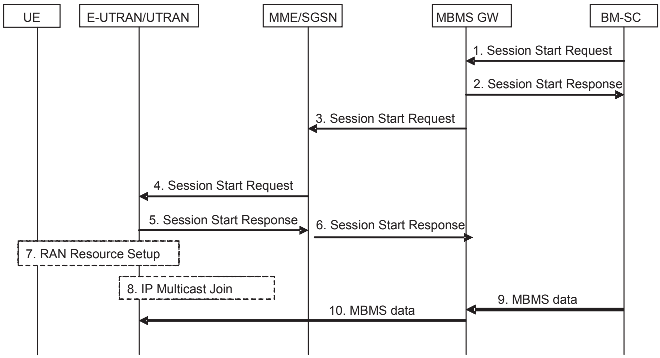

Figure 7-4: MBMS session start procedure [2]

in unicast transmissions; on the contrary, the operation of scheduling is performed exclusively by MCE and sent to the terminals as part of the information present in the MCCH channel.

The MBMS service scheduling is done on a time basis; SIB13 (System Information Block 13) messages, transported within the PBCH channel (Physical Broadcast Channel), indicate which part of the LTE frame contains the MCCH channels.

LTE standard foresees a periodic allocation of resources. It is configured by two parameters: the MCH Scheduling Period (MSP) and the Common Subframe Allocation (CSA). For MBMS services quasi-static scheduling is used with the purpose of reducing the activation times of the EU, with a consequent battery saving.

The MCH channels belonging to a certain MBMS area occupy a set of subframes that is called Common Subframe Allocation (CSA). As shown in Figure 7-5, the CSA has a periodic behavior and within it MBMS services are mapped. Each MCH is transmitted following the scheme defined in

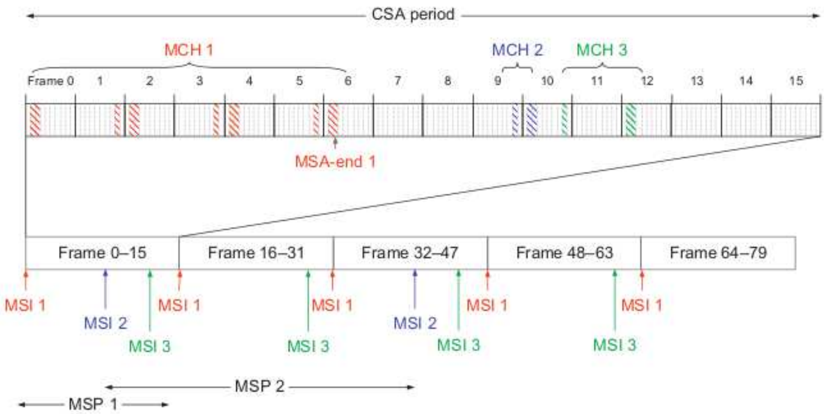

Figure 7-5: MBMS scheduling [3]

MCH Subframe Allocation (MSA). Also MSA is periodic. At the beginning of each new MSP a MCH Scheduling Period (MSI) is sent, which defines what subframes are used to a certain MTCH. Figure 7-5 highlights another feature of MBMS transmissions: transmission of MCH channel is consequential, i.e. the subframes used by MCH n-th are sent before subframes used by MCH n + 1-th. Control information carried by MCCH channels is transmitted at predetermined time intervals and can only be changed during the MCCH Modification Period. Between two MCCH Modification Period, MCCH channel is repeated several times (MCCH Notification Period) in order to facilitate the entry of a EU within area MBMS. Defining m as the modification period and SFN as the System Frame Number, the control information can be updated only when $SFN\ mod\ m = 0$. During the Modification Period, updated MCCH information is transmitted through the PDCCH channel using the message Downlink Control Information (DCI) Format 1.

7.2.5 Content synchronization

Since in a Single Frequency Network the multicast service is transmitted in different cells at the same time, eNodeBs shall to be synchronized in time. The MCE module has the task of coordinating the eNodeB belonging to the same area MBMS through the *MCCH update time* parameter. The synchronization procedure is performed through the SYNC protocol as described in [91]. To each MBMS bearer is associated an instance of the SYNC protocol containing a timestamp which indicates at each eNodeB the start of the MBMS packets transmission. The time stamp is based on a common reference time and must take account of the delivery of the packages to the various eNodeBs. The MBMS data stream is

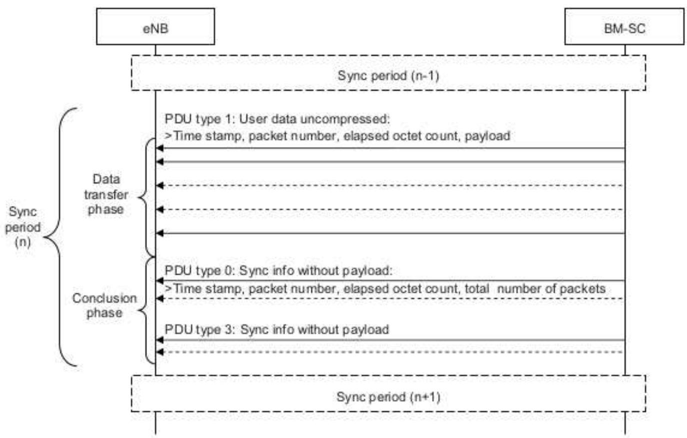

Figure 7-6: Synch Protocol Procedures [2].

divided into a number of synchronization sequences (Sync period) from the BM-SC; within each synchronization sequence are transmitted three types of SYNC PDUs containing data or information relating to sync period. Referring to Figure 7-6, the user data is transferred via PDU type 1, which contains the timestamp, the sequence number of the packet (reset for each new sequence SYNC) and the payload. On the other hand, the SYNC PDUs type 0 and 3 carry only information sent by the BM-SC.

7.2.6 Counting procedure

The counting procedure has been introduced starting from Release 10 with the aim of providing eNodeBs with a tool to know the number of EUs interested in a MBMS service. Based on the number of responses received, the eNodeB can decide whether to provide the flow of data through a multicast or unicast transmission. Currently there are some limitations regarding the procedure of counting: the first is that only the terminals compliant to Release 10 may respond to a request for counting. The second is that only the EUs that are not in the idle state may receive requests for counting.

7.3 Architectural solutions for group call implementation

LTE enhancements for PPDR (Public Protection and Disaster Relief) have been introduced by 3GPP in three main documents, i.e TS 22.468 [5], TS 23.468 [92] and TR 23.768 [4]. In particular TS 22.468 introduces the functional requirements of the Group Communication System Enabler (GCSE). GCSE is the module in charge of managing all the phases of a group call (Group Communication Service) for a set of users. Even though group call in TETRA contexts correspond only

to Push to Talk (PTT) services, in LTE Group Communication Service can be understood in a broadest sense and is also capable of supporting video and data streams.
The Multipoint Service [4] is the service created by GCSE and used to distribute group communication to the EUs belonging to a GCSE group. It should be noted that if group calls include different media streams (e.g.. voice, video, data), a different multicast session has to be allocated for each of them. The reason for this lies in the fact that each data stream may present different constraints on quality of service (QoS). An EU can decide whether to receive all the flows of the group call or only a part. Finally, a user can participate in multiple Group Communications simultaneously.
Some of the tasks performed by GCSE are:

- creation, modification and termination of group calls;
- management of users belonging to a group call;
- management of Group Communication with Geographical Scope;
- management of priority and pre-emption between users belonging to the same GCSE group;
- continuity of service maintenance even in the event of mobility between different cells;
- management of requests for group association and transmission from user;
- choosing of an efficient mechanism for group calls distribution (multicast or unicast).

TR 23.768 [4] presents several solutions to transmission of group calls problem in an LTE network. In general, the approach is to adopt, when possible, a unique multicast transmission in downlink and multiple unicast transmissions in uplink. The main differences between the proposed solutions concern the way in which the group call is established and managed. Figure 7-7 shows a general diagram of the compo-

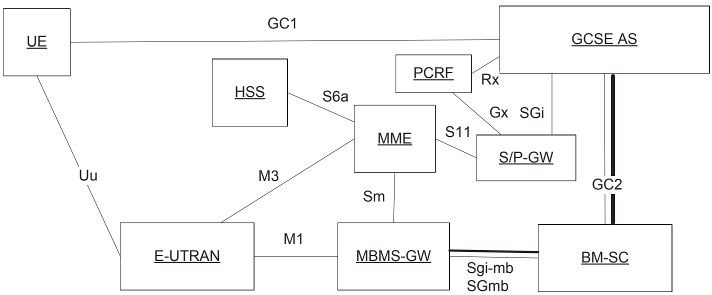

Figure 7-7: GCSE Architecture [4].

nents involved in a group call; compared to the classical LTE architecture, new interfaces linking the GCSE, the PCRF, the BM-SC and S/P-GW have been introduced. The guidelines that are common to the solutions proposed in [4] are:

- adoption of the MBMS framework for the distribution of the content provided by GCSE Application Server (AS);
- use of GC2 interface by the GCSE AS for the setup of the group call;
- use of GC1 interface by the EU for the association request to a group GCSE and, more generally, for the exchange of control messages with the GCSE;

- uplink traffic carried out only in unicast;
- the Multipoint Service implemented using the eMBMS framework described in [4].

All solutions presented in [4] are not alternative each other, but are extensions of the same basic solution based on the use of eMBMS framework and differing in the approaches used for the distribution of real-time multicast voice streams over an LTE network.

Main common characteristics are:

- use of Evolved Multimedia Broadcast Multicast Service for downlink multicast distribution;
- use of unicast transmission in uplink;
- possibility of combining multicast with unicast communications for the nodes that are not covered by the eMBMS service (nodes at cell edge or that experience bad channel conditions).

Figure 7-8 outlines the main features of the solutions proposed by the 3GPP in [4]. A first classification of the proposed solutions concerns the area of the network that will handle group calls and eMBMS flows management:

- Solutions 1, 2 and 5 provide a controlled management of the network by the Core Network.
- In solution 3 the management is entrusted to the EU-TRAN, directly through the eNodeB.
- Solutions 4 and 6 are based on high-level protocols similar to the signalling architectures in IMS (IP Multimedia Subsystem).

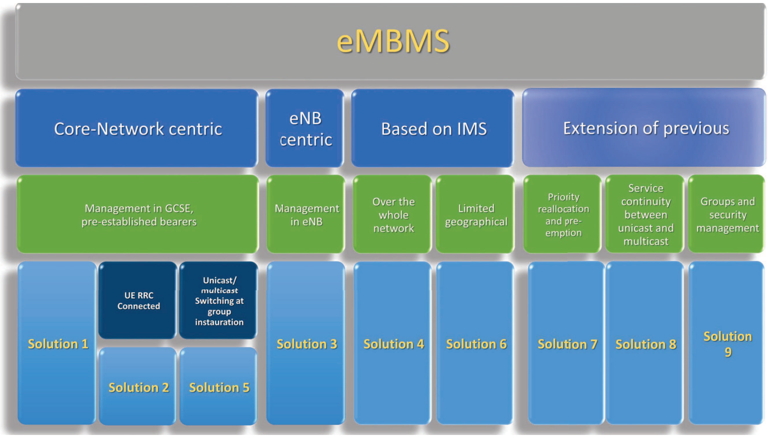

Figure 7-8: 3GPP solutions [4].

- The additional solutions 7 and 8 are not self-consistent, but they add functionalities about management of call priority and service continuity.

- The solution 9 addresses groups management, suggesting to allocate the tasks related to groups creation and management to application layer. This leads to the deactivation of the security procedures of eMBMS, moving any encryption to application layer; GCSE-AS is responsible for ensuring that users removed from a group are no longer able to receive multicast data.

3GPP does not recommend a specific solution, however, it is clear that solutions based on high-level protocols and IMS architectures are currently less mature than the others. Since solution 3 demands the management of group calls and eMBMS flows to eNodeB, the LTE network has to be autonomous from the point of view of radio coverage. As described in Section 6.3, TETRA-BB network from the early stages of its deployment will likely evolve over time in a modular way, relying on commercial networks to ensure services and adequate radio coverage. The EUTRAN represents the most peripheral network portion and the eNodeBs will be the components longer shared between the TETRA-BB and commercial networks. Since from a security point of view it is not recommended that the control of group communication is carried out by decentralized elements shared with commercial networks, it appears advantageous to focus on solutions that commit the management of the group communications to the core network.

The first two solutions shown in [4] are based on the assumption of eMBMS bearers reservation to speed up the establishment of communication. The second solution foresees that the terminals are always in RRC_Connected state so that the network is constantly aware of their connection to one or more multicast streams. This solution allows to improve the performance of latency in call establishment and allows the network to optimize the distribution of group communications among the various users in the cell. A group call is initiated by the GCSEs through a request to the BM-SC. The MBMS session information (such as TMGI) is sent periodically on MCCH channel. In order to receive them, users must be in RRC_Connected state. Moreover, if EUs are in the RRC_Connected state, GCSE can operate accurate counting procedures to decide the most convenient transmission mode (multicast or unicast). The main drawback in this case is a considerable drain of the devices battery. Therefore, for professional use of the LTE network, where the latencies pose very strict constraints, the second solution can be more advantageous compared to the first, also taking into account that some types of terminals (for example those vehicular) have no special requirements for energy savings.

The GCSE acts inside the network as an Application Server (AS) that, besides delivering the group calls, shall handle all the signalling coming from the UEs (in uplink) and towards the LTE Core Network (in downlink). In solutions 1 and 2 of [4], GCSE also manages the switching between multicast/unicast transmission for the EUs performing handover in a cell not covered by the eMBMS service or not experiencing good propagation conditions.
Figure 7-9 shows the messages that are exchanged between EU, Core Network and GCSE in order to maintain the service continuity in Make-Before-Break mode.

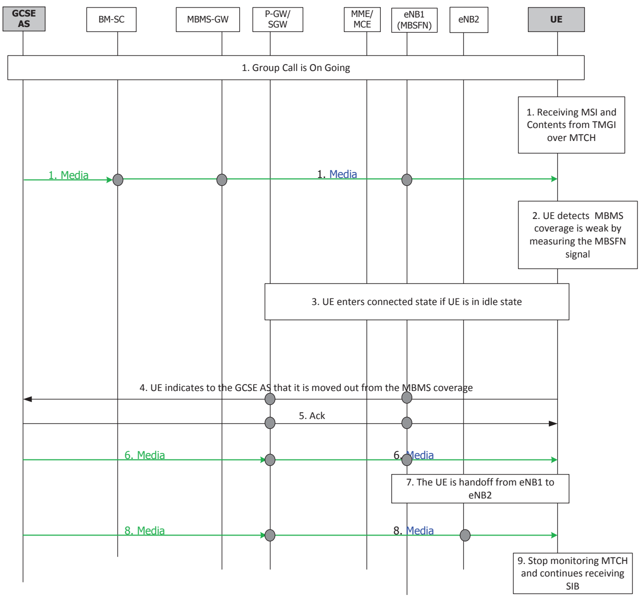

Figure 7-9: Make before Break Transmission Mode Switching [5].

In solution 3, eNodeBs belonging to the MBSFN choose, without the intervention of the Core Network, which downlink transmission mode to adopt and whether to transmit the group call within the cell (solution eNodeB centric). In an initial phase, when no EU requires the activation of a Group Call, the eNodeBs transmit only information regarding the eMBMS bearers present within the cell and trigger the transmission of one or more MBMS flows only in response to a EU request. This solution reduces the workload of GCSE AS leaving to each eNodeB the decision about mode of transmission to be adopted independently from others eNodeBs belonging to the MBMS area.

Figure 7-10 shows the procedure followed when an user join a Group Call. The EU sends a registration message to the GCSE (point 9 in Figure), which sends as response the TMGI (the identifier of the MBMS flow) and through the PCRF establishes a unicast communication. The user starts to receive the data stream and simultaneously listens to MCCH channel; if the TMGI is supported inside this cell, the terminal begins to receive the MBMS stream and sends a message to GCSE with the request to terminate the unicast communication.
The aim of this procedure is to minimize the access time of an EU to a Group Communication. However it can be noted that: MCCH channel is sent periodically with a time interval (*repetition period*) that can vary from 320ms to 2560ms [93]; this means that a user can wait for a time greater than 300 ms (the maximum required by PPDR applications) before receiving a group call. In addition, the initialization of a MBMS bearer involves considerable exchange of messages between the Core Network and the eNodeB. As a result, the setup phase and the beginning of the transmission of multimedia

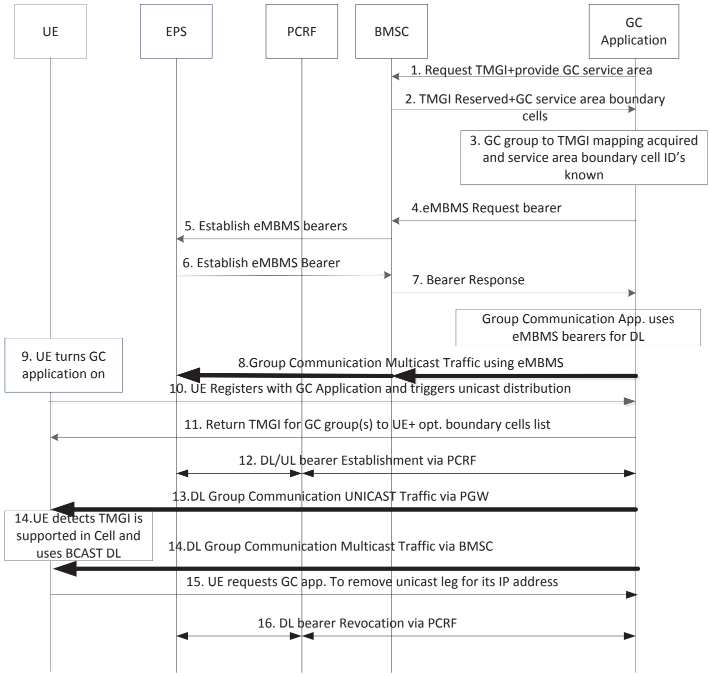

Figure 7-10: UE Group Call Joining Procedure [5].

content can be subject to considerable delays.
Solution 5 represents a possible variation of the previous towards the goal to accelerate the establishment of a group call. It proposes to initiate the call via multiple unicast flows in the downlink, leaving the possibility for individual terminals to switch to the reception of the multicast stream as soon as it is active and available.

In view of this consideration it is legitimate to suppose that the most suitable network architecture that will be realized will follow the indications outlined in solution 2 and it is schematized in Figure 7-11 it is based on pre-established

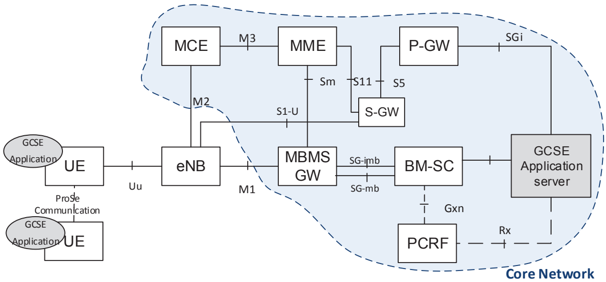

Figure 7-11: Network Architecture for proposed solutions 1,2 and 5.

eMBMS bearers (a characteristic inherited from the solution 1), with the aim to speed up the set-up of the communications group. This means that the identifier TMGI (Temporary Mobile Group Identity) of the group communications is pre-assigned by the BM-SC (Broadcast Multicast Service Center), as well as the QoS characteristics of the bearers. Nodes that are located at the cell edge or in areas affected by disadvantageous radio channel, where multicast stream can be received with an adequate quality, can take advantage of simultaneous unicast communications that replicate the MBMS flows.

7.4 The proposed solution

In accordance with the indications of the 3GPP, we outline here a promising solution for GCs in the future LTE-based PMR networks and support our vision by providing suitable performance evaluations and comparisons. A more extensive evaluation of architecture performance is presented in Chapter 8.

The envisaged solution is based on eMBMS framework and GCSE as previously described, and assumes a centralized approach for the call management (i.e., performed by the EPC) that leads to benefits in terms of coverage and reliability.
In particular, we will discuss here two possible approaches. The first is fully based on multicast transmissions, named *static eMBMS activation*, where the unicast mode is considered only as backup solution when the multicast transmission is not available, e.g., due to bad propagation conditions or because the UE moves out of a MBMS area. Unicast mode is activated only to provide service continuity when the UE detects an increasing packet loss.

A second, more advanced solution, named *dynamic eMBMS activation*, is considered in order to increase the system flexibility and efficiency. *Dynamic eMBMS activation* manages group communications over both unicast and multicast transmissions. The idea is based on the ability to perform a counting procedure for the GCSE, with the aim to transmit downlink media contents through multiple unicast bearers until a certain number of users requires the same GC service, hence to activate a multicast bearer. The advantage is that the GC is never broadcast in a cell with no active group members, and, hence, the spectrum efficiency is maximized.

For both solutions, the network architecture is represented in Fig. 7-12, where the GCSE acts as an *Application Server* (GCSE AS) providing the functionality for the management of GCs. In addition, the GCSE uses the existing LTE standardized interfaces in order to communicate with the EPC and selects the proper transmission scheme for the delivery of GCs. Finally, multicast bearers are considered for downlink communications whenever possible. On the other hand,

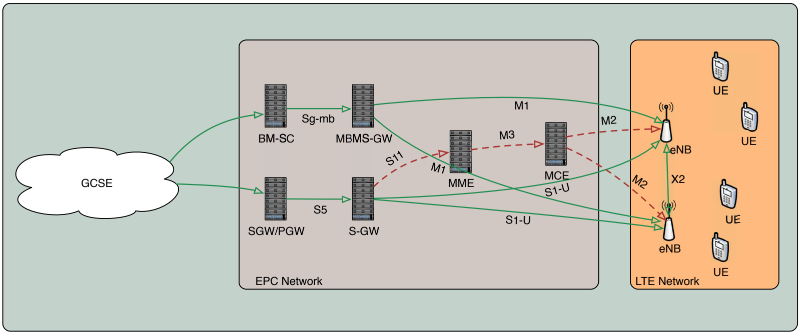

Figure 7-12: System view of PMR network.

uplink traffic is always sent via unicast bearers. GCSE provides a multicast to unicast switching mechanism depending on the selected approach. Hence, the GCSE is connected to both the PGW (for the distribution of unicast traffic) and to the BM-SC (for the multicast traffic) through the SGi and GC2 interfaces, respectively.

Static eMBMS activation

In the *static eMBMS activation* solution, we assume that the GCs are always managed by resorting to multicast transmission except when it is not available. This approach exploits the basic structure of the eMBMS, for which it is needed to verify the satisfaction of the PSS requirements. From Fig. 7-13 it is possible to see that this is true for the expected requirements in terms of number of GCs supported for each type of service [84].
However, the most critical requirement is represented by the GC set-up time that must be lower than 300 ms . This depends on the time requested to establish the eMBMS bearer

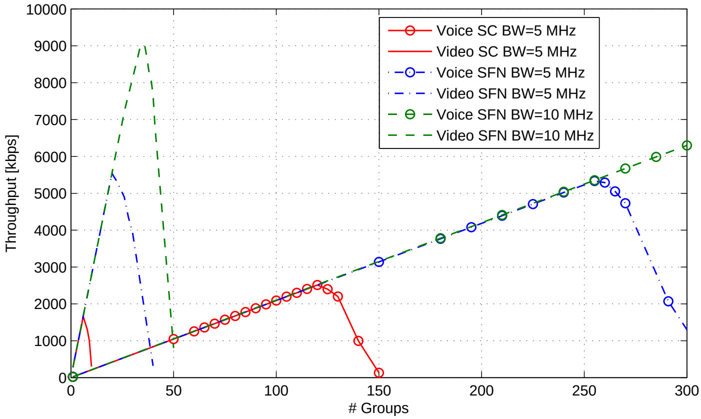

Figure 7-13: Throughput for different types of services in SC and SFN eMBMS.

and to start-up the call. Two options have been analysed:

- *Pre-established eMBMS Bearer.* The network establishes the eMBMS bearer over preconfigured MBMS areas before the GC starts. This implies that the BM-SC pre-establishes in advance all the information related to the GC, such as the *Temporary Multicast Group Identifier* (TMGI), QoS class and the eNBs belonging to the MBMS area. In particular, the GCSE AS requests the creation of an eMBMS bearer to the BM-SC by means of the PCRF interface, which is in charge of the exchange of the information related to the eMBMS session. As soon as a UE requests a GC, the downlink traffic is transmitted using one of the pre-established eMBMS bearer. This solution provides a fast total set-up time for the GC service that depends only on the time to start-up the call. 3GPP estimates that it is nearly 220-250 ms [94], which is consistent with the requirement for the PMR voice communication.

- *Dynamic bearer setup at group call start-up.* In this case the eMBMS bearer is established only when needed. It means that in addition to the 220-250 ms for the call set-up, also the delay for the downlink bearer establishment shall be considered for evaluating the user experience. The additional latency is assessed on the order of 115 ms [94], taking into account 10 ms for radio interface delay, 5 ms for network interface delay ad 5 ms of request processing delay. This additional latency is not negligible and the total delay experienced by users, for call set-up and downlink bearer set-up, also exceeds 300 ms.

Fig. 7-14 shows the delay contributions composing the overall end-to-end latency at call start-up, for both options (A more extended analysis of delays are performed in Sec. 8.1).

Even if dynamic bearer set-up is more flexible and permits an optimization of the user resources, pre-established bearer is the option selected in our solution, because it permits to satisfy the GC set-up time requirement.
However, it could be not sufficient because when the MBMS service is mapped on the eMBMS bearer and on the logical MTCH channel, it has to be multiplexed with the control information (MCCH) and then sent on the physical MCH to be transmitted. In the current LTE standard, the MCCH can be updated with a minimum period of 5.12 s (called *MCCH modification period*), it means that the GC should wait also this time before to be transmitted exceeding the set-up de-

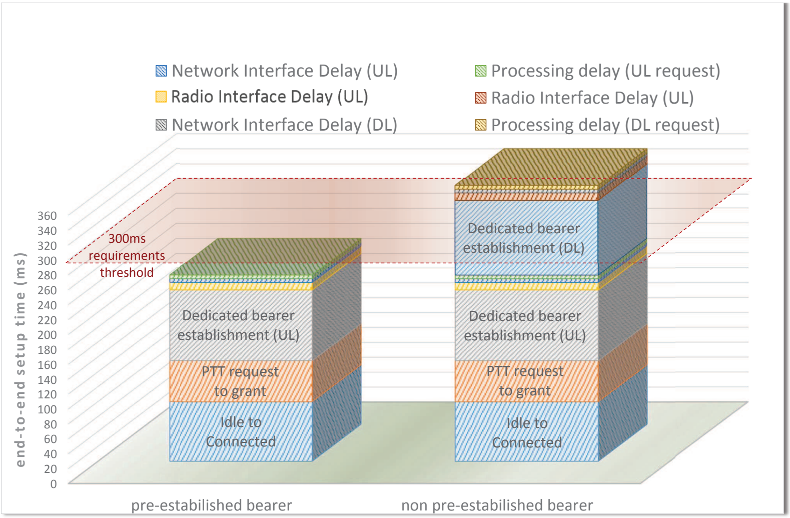

Figure 7-14: End-to-end setup latency for the two considered case with and without pre-established downlink bearer.

lay requirement. Hence, we propose also a shorter MCCH modification period that should be about 50 ms (once every 5 LTE frames).

Dynamic eMBMS activation

In this case the use of unicast and multicast is more flexible and also the GC set-up time requirement can be relaxed. Indeed in this solution UEs are always served through unicast transmissions at the communication start-up, and the use of pre-established unicast bearer shall be considered in order to accomplish the required end-to-end call set-up time. On the other end the activation of eMBMS bearer can be dynamic, since users are already receiving the service and will be able

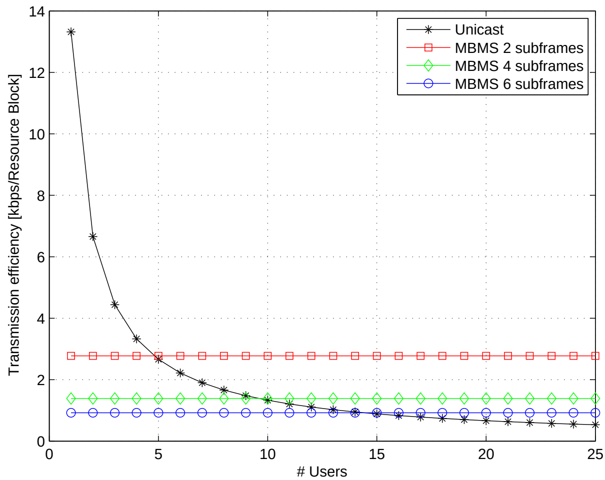

Figure 7-15: Spectral efficiency evaluations for dynamic eMBMS activation.

to switch on multicast reception as soon as eMBMS has been completely activated, as a consequence the MCCH modification period can be neglected and unmodified. The number of users associated to a GC is provided on a per-cell basis by the PGW by means of the *User Location Information* (ULI) procedure. Consequently, the GCSE selects the most efficient transmission scheme. An alternative to the ULI procedure is represented by the UE that sends a message to the GCSE whenever moves from one eNB to another one. This solution implies a higher data transmission over the core network, but at the same time allows the UE to switch to IDLE mode in order to save battery life delivering high burst of data in short period of time.

In the proposed *Dynamic eMBMS activation* solution the

GCSE exploits the information regarding the UEs associated to a GC service, to detect the most efficient solutions. In particular, the total amount of resources requested by the UEs if they would be served with unicast mode is evaluated and compared with the resources reserved for the multicast mode. As an example Fig. 7-15 shows the spectral efficiency in terms of throughput of a group member, normalized to the resources allocated to the multimedia GC service when the number of group(s) members increases. We can see that the multicast solution is independent on the number of group members, the resource usage depends only on the number of subframes allocated to the multicast service. Conversely, the unicast spectral efficiency decreases with the number of served UEs. With the proposed *Dynamic eMBMS activation* procedure the system is able to switch from unicast to multicast when the number of UEs overcomes a given value represented by the intersection of the curves, hence the spectral efficiency is always the maximum value (i.e., the envelope of the unicast and multicast curves).

Chapter 8

LTE Group Calls Performance Evaluation

Network architecture presented in previous chapter introduces remarkable innovations respect to current LTE network configuration, therefore it becomes important to assess its performance, its compliance with the group calls requirements in professional networks, its advantages and possible weaknesses. In this chapter will be investigated the ability of LTE network to maintain low the call establishment time, that is a fundamental feature in emergency situations, and support an adequate number of groups and users per groups in the main operational scenarios. To validate the architecture proposed in Sec. 7.4, we compare transmission of multimedia group call traffic both on unicast and multicast bearers that can pre-established and not. Additional evaluation condition are the RRC status of the caller and receiver users and use of Discontinuous Reception mode. This analysis is useful both to evaluate performance of LTE as it is and to identify possible constrains or improvement that should to carried out to the standard.

8.1 Time latencies

A major feature that distinguishes PMR networks from commercial ones is the very low call establishment required time. As described in [94], three different latency requirements have to be fulfilled:

- **End to end setup time:** defined as the amount of time between when an EU sends a request to start a group communication and when it can begin to transmit voice or data. It shall be less than or equal to 300 ms. This requirement is valid under the assumption that the caller can start to communicate without waiting for a response from the others members of the group.

- **Time for joining an ongoing group communication:** defined as the time between when a node requires to participate in a communication, and the time when it actually receives data.

- **End to end average distribution delay:** time between data are sent and the time when they are received. 3GPP recommends a limit of 150ms for this delay.

8.1.1 Latencies for group communications distributed through unicast bearers

End to End setup time

3GPP defines as *End to End setup time* the latency for connection establishment between a group member and the GCES AS. The maximum allowed delay is 300ms, and 3GPP assumes that is not necessary to wait a feedback from group members before starting a communication.

Call initiated by UE in RRC_IDLE Status. This section analyses the latency for the call setup of a terminal in RRC_IDLE status, which previously joined the group at application layer.

In LTE, a terminal can be in two different states, RRC_CONNECTED and RRC_IDLE.

- RRC_CONNECTED: in this state an RRC context is established. Since terminal and network know the parameters necessary for the communication, data can be transmitted and received by user.

- RRC_IDLE: there is no RRC context in the radio-access network. The terminal does not belong to a specific cell and no data transmission is possible.

The exit from the ILDE status is directly triggered by the need of transmitting data (Figure 8-1). As long as the terminal is in IDLE status, no resources are allocated to it. In order to be able transmitting the UE shall first synchronize himself to the network, then send the request to be considered CONNECTED to the RRC. The connection process requires between 50 and 80 ms to let the terminal pass in the RRC_CONNECTED status.

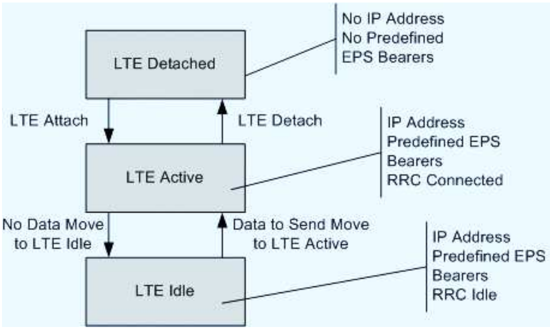

Figure 8-1: RRC states transition diagram.

Once connected the terminal sends a "PTT floor Request" and waits the corresponding "clear to talk" carried by the "PTT floor Grant" response message (see Figure 8-2).

This messages exchange is not directly dependent from the radio access used technology (either LTE or TETRA) and the corresponding latency is therefore hard to be evaluated. 3GPP assumes that about 55ms are needed to accomplish the transmission request. The second step is the establishment of two LTE radio bearer, one for the uplink and the other for the downlink. The downlink bearer is supposed to be pre-established: every terminal of a group will have a radio bearer associated to the communication, avoiding the introduction of further delay. The uplink unicast radio bearer, between the source UE and the GCSE, is instead activated on the run; this operation requires 115ms, considering 10ms delay at the radio interface and 5ms processing delay at the network interface.

The total end-to-end setup time is therefore about 250ms.

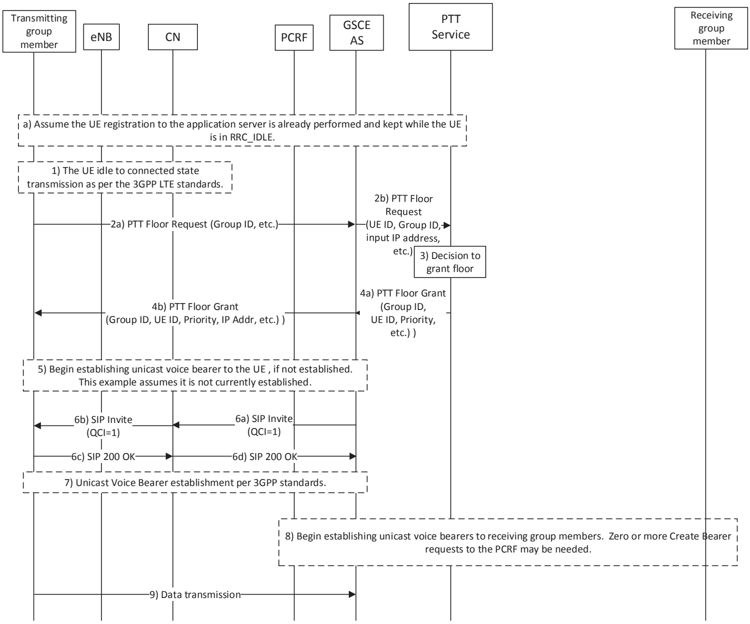

Figure 8-2: GC establishment by UE in RRC_IDLE mode sequence diagram.

The requirement of having an end-to-end setup time lower then 300ms is therefore verified, even in the case the transmitting terminal starts the communication from a RRC_IDLE Status (see Figure 8-3).

UE receiving the call The end to end setup time, as defined in the Technical Report 36.868, is not considering any feedback from receiving terminals; the considered delay ends when the transmitter is able to start the transmission to the

Azione	Tempo (ms)	Commenti
RRC_IDLE → RRC_CONNECTED	80	Ref. Sec. 16.2 of TR 36.912
Time for the PTT channel acquisition.	55	The PTT channel acquisition delay may be variable and dependent on network implementation. 3GPP estimates an average value of 55ms.
Uplink bearer initialization	115	The dedicated bearer for VoIP is assumed to be established using IMS. Time needed for the establishment of the bearer used foc communications from UE towards GCSE. It can be estimated in 115ms and comprises 10ms for air interface delay, 5ms for network interface and 5ms for computation time.
End to end setup time	250	

Figure 8-3: End to end setup time, terminal in RRC_IDLE mode.

GCSE, even if no other users are listening the communication.

It is anyway interesting to evaluate the additional latency experienced form receiving users.
In case the receiving UE is in RRC_CONNECTED status, the additional delay for the communication set-up just correspond to the delay for data packet distribution.
On the contrary, if the receiving UE is in RRC_IDLE status, the delay experienced form the user shall take into account the time for the communication paging (up to 320ms) and for changing the status from IDLE to CONNECTED (about 80ms), resulting in a total additional delay of 400ms.

The receiving UE can moreover operate in DRX (Discontinuous Reception) mode, for power saving purpose, and a further delay for the periodic data distribution have to be considered. In case the terminal have no data to be sent or received for a time longer then T1, it goes in "Short DRX" status and maintain the radio interface active just during a portion of the radio frame. If no data are received, after a second T2 timer, the UE goes in a "Long DRX" state, characterized by less frequent wake-up time listening the channel (see Figure 8-4). In case no data are again received after a

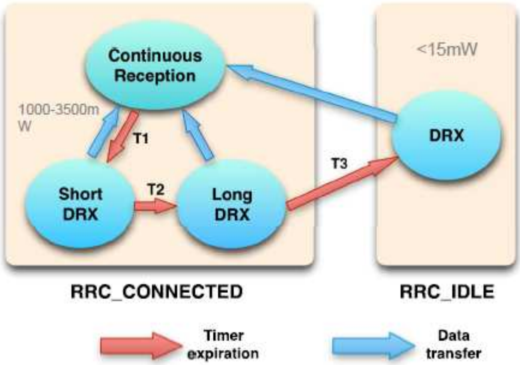

Figure 8-4: Terminal transition from active to DRX state.

T3 timer, the terminal turns the receiver off, maximizing the power save and going in IDLE status (with a power consumption lower then 15mW). The three described timers are set up by the network, and not from the UE, and may be optimized at network level. UEs have to be aware of the presence of data to be received in order to ask the change of their status from IDLE to CONNECTED; an additional latency in data reception, due to the less frequent listening of the channel when operating in DRX mode, shall be therefore considered.

The minimum Short DRX Cycle is 2 Subframe (2ms), while the maximum one is 640 subframes (640ms). The Long DRX Cycle is between 2,56 and 10s. Even if the shorter timers are used, the implementation of DRX mode may introduce additional latencies between 2ms and 2,56s. It is therefore mandatory to avoid the use of DRX technology for system to be use in PMR frameworks, where usually the terminal are characterized by large battery and low latencies have more importance then power saving.

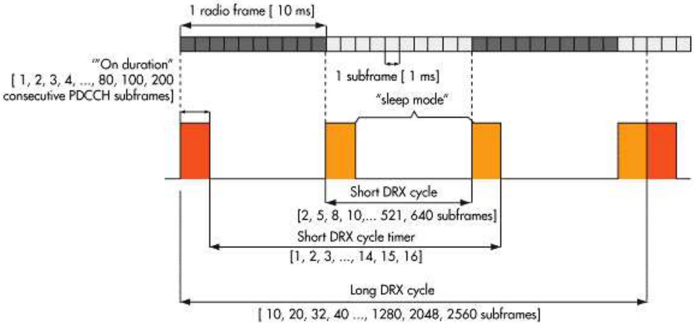

Figure 8-5: DRX cycles duration.

Pre-established bearer use. A further user-experience optimization can be achieved by the use of pre-established downlink bearers.

3GPP requirements about latencies for calls set-up just take into account the time needed for the uplink activation and are not considering additional latencies for bearer initialization from the GCSE to the listening users; even if there are no requirements constrains, the delay for the downlink initialization can have a strong impact on user experience, and all possible optimization have to be considered. In case pre-established downlink bearer are used, the network establishes the eMBMS bearer over preconfigured MBMS areas before the GC starts. This implies that the BM-SC pre-establishes in advance all the information related to the GC, such as the Temporary Multicast Group Identifier (TMGI), QoS class and the eNBs belonging to the MBMS area. In particular, the GCSE AS requests the creation of an eMBMS bearer to the BM-SC by means of the PCRF interface, which is in charge of the exchange of the information related to the eMBMS session. As soon as a UE requests a GC, the downlink traffic is transmitted using one of the pre-established eMBMS bearer. This solution provides a fast total set-up time for the GC service that depends only on the time to start-up the call, which has been previously estimated in 220-250 ms. In case the bearer are not pre-established, they are activated only when needed. This means that in addition to the 220-250 ms for the call setup, also the delay for the downlink bearer establishment shall be considered in order to evaluate the user experience. The additional latency is assessed on the order of 115 ms [15], taking into account 10 ms for radio interface delay, 5 ms for network interface delay ad 5 ms of request processing delay. This additional latency is not negligible and the total delay experienced by users, for call set-up and downlink bearer set-up, also exceeds 300 ms.

Figure 8-6 shows the delay contributions composing the overall end-to-end latency at call start-up, for both options. The total delays are compared with the latency requirement of 300ms for the uplink (red threshold in the picture), in order to have a metric to evaluate the two solutions.

Active call joining time (DL unicast distribution)

The time needed for a terminal to join an active call distributed in unicast is related to the time for the downlink bearer activation. The user is considered as previously registered to the logical group at application layer and the terminal ready for the secure communication instauration. First of all the terminal shall obtain the network connection, passing to the RRC_CONNECTED status (see Figure 8-7); this transaction request about 50-80ms. Then it proceeds with

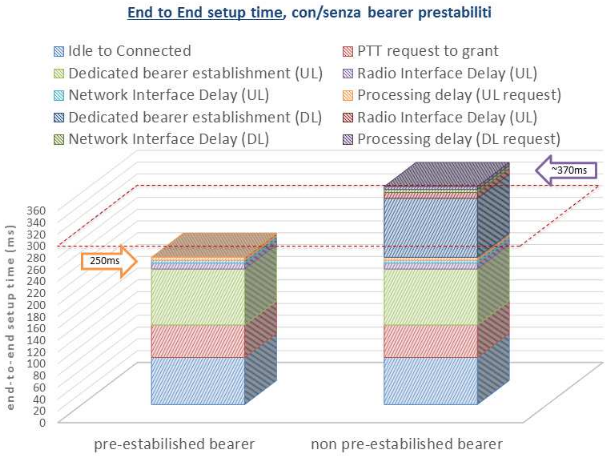

Figure 8-6: Group Call reception latency evaluation with and without pre-established bearers.

a Service Attach Request and waits for the Service Attach Response. This message exchange is not directly related to the used radio access technology (LTE or TETRA as well), it is therefore difficult to exactly evaluate the correspondent delay; 3GPP assumes 55ms by default. After this request an unicast bearer for the downlink shall be activated; this operation requires 115ms, as evaluated for the uplink. The overall delay for the call join is about 250ms.

This delay, considering a terminal already registered to the group at application layer and ready for a secure call establishment, is therefore lower than the requested 300ms (see Figure 8-8).

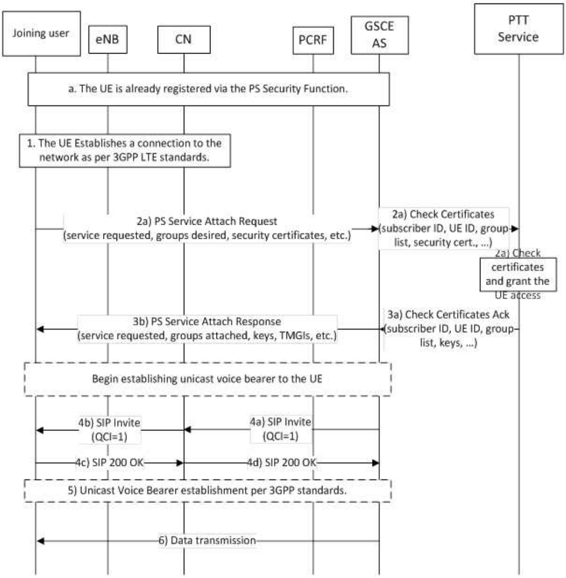

Figure 8-7: Group call joining procedure, considering terminal in RRC_IDLE state.

Action	Time (ms)	Comments
RRC_IDLE → RRC_CONNECTED	<80	Ref. Sec. 16.2 of TR 36.912.
Time for the PTT channel acquisition	55	The PTT channel acquisition delay may be variable and dependent on network implementation. 3GPP estimates an average value of 55ms.
Uplink bearer initialization	115	The dedicated bearer for VoIP is assumed to be established using IMS. Time needed for establishment of the bearers used for the communications from UEs towards GCSE. It can be estimated in 115ms and comprises 10ms for air interface delay, 5ms for network interface and 5ms for computation time.
Call joining time	250	

Figure 8-8: Active call joining delay computation.

End to end delay (uplink unicast, downlink unicast)

The end to end delay can be decomposed in three main segments: two over the eUTRAN (one inside the transmitter cell and one inside the receiver cell) and one over the Core Network. 3GPP consider delays for the three segments as shown

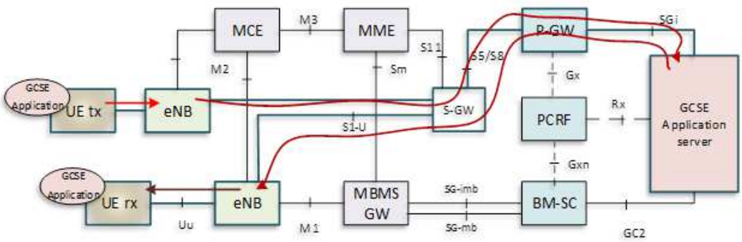

Figure 8-9: User data stream in unicast.

in Figure 8-10: The obtained end to end delay is therefore

Action	Time (ms)	Comments
UE in TX → eNodeB	10	See Annex B.2 of TR 36.912.
eNB→SGW→PGW→GCSE→eNB	20	Delay experienced within Core Network is variable and depends on network implementation, topology, cell position, etc. 3GPP assumes a delay of 20 ms.
eNodeB → UE in RX	10	See Annex B.2 of TR 36.912.
End to end delay	40	

Figure 8-10: End-to-end transmission delay computation - Unicast distribution

just 40ms, well lower than the requested threshold of 150ms.

8.1.2 Latencies for group communications distributed through multicast bearers

Pre-established bearer use

Latencies calculation for the case of downlink MBMS distribution takes into account the use of pre-established multicast bearer (MRB). Every group is managed by the GCSE AS, which requests to the BM-SC the assignment of an ID (TMGI), previously defined for the corresponding MRB. Every UE downloads the USD (User Service Description) of the service and establishes a unicast bearer with QCI = 5 for the signalling; at application layer the GCSE AS advices the GSCE about its capacity of receiving MBMS flows in the area where is located. The group call is activated in the same way as a PoC (Push to talk over Cellular) session. The UE starts the reception of MBMS flows just after the reading of SIB2, SIB13 and SIB15 over the MCCH channel. When the user pushes the PTT button, the UE enters in RRC_CONNECTED status and send a floor request to the GCSE AS, which transmits the talker identity to other UEs listening on the correspondent bearer. In reception the UEs under the MBMS coverage, listen the MSI (MBMS Scheduling Information) information over the MCH at the beginning of each MSP (MCH Scheduling Period). In current 3GPP standard the minimum MSP periodicity is 80ms; using this value the average delay is expected to be 40ms (and the maximum 80ms). At the same time the GCSE AS send a floor grant message to the transmitting terminal; the delay for the floor grant elaboration and sending depends of the GCSE implementation, but can be considered negligible considering that every UE sends its Talker ID together with the first transmitted data packet (in the same MSP); on the contrary a further delay corresponding to MSP periodicity have to be added. At this time the node is able to send its media to the GCSE, which will forward them over the MRB to other call participants.

End to End setup time

Considering the 3GPP definition of end to end setup time, involving just the uplink, all the consideration done for the unicast case are valid also for the MBMS distribution (since this second just involves the downlink). The end to end delay can therefore be considered about 250ms, which is lower than the 300ms threshold required.

Active call joining time (DL multicast distribution)

When a UE joins an active call it has been previously registered at application layer and it already knows the call TMGI; the terminal just verifies if it is distributed in MBMS in the area where it is located. In this case the UE acquires the scheduling information about the TMGI and starts receiving data over the MTCH channel.
If the UE, before joining the call, already was monitoring the SIB13 and maintaining updated the MCCH and MTCH configuration for the TMGI of interest, the latency for call joining is just related to the MCH periodicity (80ms) and to the MSI acquisition for the service (i.e. 5ms). In this case the average delay is just 45ms (85ms in the worst case).
On the contrary, if the UE was not maintaining the information a longer delay is required. In particular 10 ms are needed for the acquisition of the MCCH on the SIB13, 160ms

are needed (in average) for waiting next MCCH (supposing the minimum periodicity of 320ms), 10 ms are needed for the MCCH and MTCH acquisition and for the configuration of the TGMI as well. The overall additional delay in this case is 180ms (and the maximum 340ms).

If multiple MCCH are foreseen, they are characterized by an offset of 100ms, adding therefore 50ms of average delay.

It is clear that in this second case it is very difficult to obtain a delay lower than 300ms, since on average it will be 225ms (275ms in case of multiple MCCH) and can increase up to 525ms.

Activity	Mean Latency	Maximum Latency	Mean Delay	Maximum Delay
Call acquiring (MCH periodicity: 80ms)	40 ms	80 ms		
MSI Acquisition	5 ms			
Joining time (channel configuration already acquired)			**45 ms**	**85 ms**
SIB13 configuration acquisition on MCCH	10 ms			
Wait for MCCH , periodicity of 320ms (minimum possible)	160 ms	320 ms		
MCCH, MTCH and TGMI configuration acquisition	10 ms		180 ms	340 ms
Joining time (channel configuration not already acquired)			**225 ms**	**425 ms**
Additionatl Delay in case of multiple di MCCH			50 ms	50 ms
Joining time (channel configuration to be acquired e multiple MCCH)			**275 ms**	**475 ms**

Figure 8-11: Call joining delay computation - Multicast distribution.

End to end delay (UL unicast, DL multicast)

The end to end delay can be decomposed in four main segments: two over the eUTRAN (one inside the transmitter cell and one inside the receiver cell), one over the Core Network in unicast and one over the Core Network from BM-SC to the eNodeB in multicast downlink.

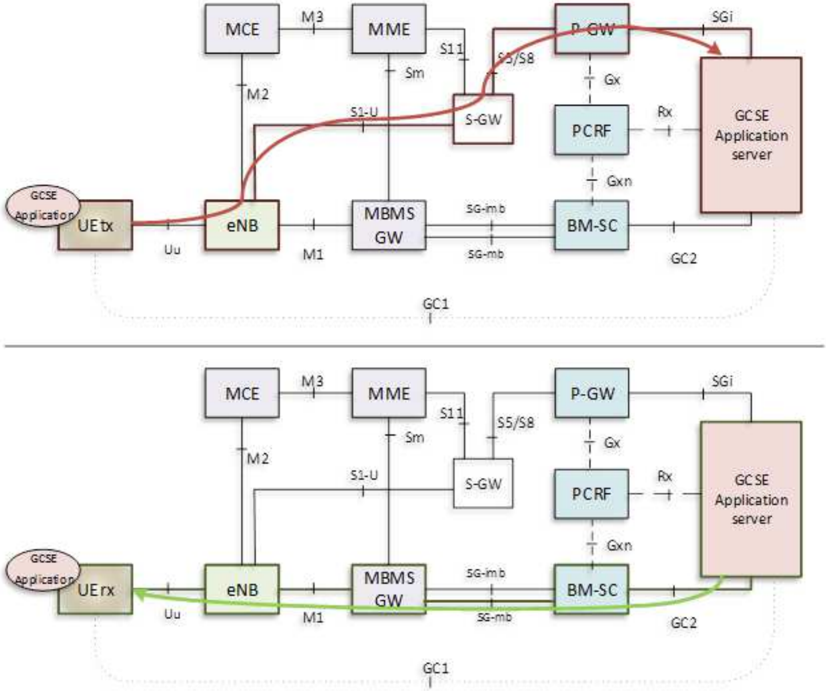

Figure 8-12: User data distribution for group call in uplink (red arrow) and in downlink (green arrow).

To obtain an evaluation of this delay , 80 ms needed for the MSI (MBMS Scheduling Information) reading over the MCH at the beginning of each MSP (MCH Scheduling Period) shall be added to the latencies of the uplink and downlink already calculated.
3GPP considers delays shown in Figure 8-13 for the four network segments.

The bi-directional total delay is therefore 160ms; this delay is higher than the one requested by 3GPP.

Action	Time (ms)	Comments
Talker UE → eNodeB:	10	See Annex B.2 of TR 36.912
eNB → SGW/PGW → GCSE → BM-SC	20	Delay within Core Network is variable and dependent on network implementation. 3GPP assumes 20ms.
BM-SC → eNodeB	40	SYNC sequence of MSP/2 = 40ms assumed
eNodeB → Receiver UE	10	See Annex B.2 of TR 36.912
Receiving and reading MSI time	80	Equal to MSP (MCH Scheduling Period), assumed 80ms
End to end delay	160	

Figure 8-13: End-to-end delay computation - Multicast distribution.

8.1.3 Default parameters

The latency previously described are calculated considering following hypotheses:

- The receiver terminal knows all the System Information (e.g. SIB13).
- If the terminal acts in DRX mode (for energy saving) an additional latency for data reception shall be considered. The DRX mode is, by the way, not compatible with the latency requirements of 3GPP and ETSI.
- The shorter possible repetition period of the MCCH is used (320ms). Larger admitted values are 640ms, 1.28s e 2.56s (all too large to respect the requirements).
- The shorter possible MSP (MCH Scheduling Period) periodicity is used (80ms). Larger admitted values are between 80ms and 10,24s.
- The application layer messages exchange, as "PTT floor Request", "Clear to talk" and "PTT floor Grant", is not directly related with the radio access network technology. For these latencies the reference values provided by 3GPP have been considered.

8.2 TETRA-BB support to multimedia Group Call validation

As clarified in Chapter 7, group calls will be carried out using the LTE eMBMS framework. This section provides an analysis of benefit and limits of this implementation choice in scenarios characterized by various types of traffic, availability of resources, number of the users within the cell and MBMS areas dimensions. The goal of the simulations is to verify that the TETRA-BB system requirements are met in main operative PMR scenarios.

8.2.1 Macro scenarios

TETRA-BB will operate in scenarios similar to those described in [84]:

1. Ordinary operations.

2. Planned public events.

3. Unplanned public events - emergencies.

4. Disasters.

The first scenario is related to routine day-to-day tasks performed by operators. It is characterized by a reduced number of operators acting on their area of expertise involved in calls or data exchanges of low entity. The scenarios 2 and 3 instead refer to situations in which exceptional events occur that involve a large number of operators, converging in a certain geographical area and generating heterogeneous traffic also of relevant entity. Finally, the last scenario refers to a situation of emergency, for example, produced by a natural disaster, in which the operation of the network could be reduced due to the lack of infrastructure in a particular area or because a failure or unavailability of network facilities. All of these scenarios can be related to generic network use cases.

Use case 1: group call employment

The call may involve a dispatcher and multiple operators on the field. The list of users who have access to the group and their areas of expertise are preconfigured in the system.
The enabled user equipment that enter or exit from the operating area automatically join or leave the group. When a user initiates a call on his terminal, the system sets up the call. The user receives a confirmation of the establishment and in a time less than or equal to 300 ms can start talking. At the same time other users are informed about call start. The system efficiently allocates resources on the cells involved. Each user hears the voice of the user who is talking but not the voice of the other users. Each user displays the identity of the speaker. When another user makes a PTT request, the talking user may be interrupted depending on system configuration.
An authorized user can join the call, even if it has already been started, by selecting the group on its terminal.
When the user terminates the active call on its device, other users are notified that the transmission ceased. After a short period of time since the last user has spoken and nobody else has made a request for transmission, the group call is cancelled and the resources released.

Use case 2: "See what I see"

This use case corresponds to a scenario in which a user shares multimedia contents (video or images) with the members of a predefined group formed by dispatcher and/or operators.
For example, an operator which detects an emergency can communicate it to the group via a voice communication, and then it can start the transmission of multimedia traffic towards the control center. At this point the dispatcher shares

the video with other users starting a video group call. When the emergency ends, the dispatcher terminates the video sharing with the members of the group, and the caller interrupts the transmission of video, maintaining active only the audio call.

Use case 3: Multi-agency harmonized operations This use case illustrates the situation in which different agencies (e.g. police forces and rescue teams) should co-operate in the same geographical area.
If applications used by different operators are compatible, the dispatcher can create miscellaneous groups by merging preexisting groups; at the end of the emergency dispatcher releases the composite group and users can continue communicating within the native group.

8.2.2 Simulations overview

Simulations scenarios for the analysis and validation of the eMBMS ability to support of group communications can be defined following the macro scenarios analysis proposed in previous section.

Simulations are carried out with the aim of highlighting performance and limits concerning the use of the eMBMS framework for the distribution of group calls. Since TETRA-BB system shall be able to handle services characterized by different transmission rates, a resource allocation procedure that does not cause inefficiencies to group call users is required. For this purpose, a first phase of the simulation is aimed to define a series of scenarios that analyse the limits of the framework MBMS in terms of data rate. In particular, the maximum rate that can be experienced by a user involved in the reception of broadcast and multicast flows in a Single Frequency Network is evaluated. The simulation activity then proceeds with the analysis on the number of group calls that the TETRA-BB network is able to support. Simulations are performed taking into account users experiencing different propagation conditions and considering different system channel bandwidths. Furthermore, since the eMBMS framework can only use some subframe of the LTE frame, the proposed scenarios analyse the rate varying the number of subframes allocated for the transmission of multicast and broadcast streams and the number of active groups within the MBMS area.

8.2.3 Services and traffic models

To analyse the performance of TETRA-BB access network, three main services are taken into consideration: voice group calls, video group calls, high definition video group calls. The characteristics of the three services are described in detail in the following.

The simulation campaign takes into account the group data distribution in both unicast (one flow per service per user) and multicast (a service flow for each group) mode. In case of unicast distribution each service directed to an EU is associated with a flow; the limit of the network is represented by the number of simultaneously active services for all users. In the case of multicast distribution, on the contrary, the number of streams transmitted on the network corresponds to the number of services multiplied by the number of groups involved in the service, independently from the number of users constituting the groups.

The approach followed in simulations provides an analysis of system performance as function of the number of active flows on the network, unicast and multicast, without explicitly tying the number of streams to the number of users or the number of groups. In the case of unicast distribution, for example, 10 calls directed to an user employ the same amount of resource of 10 calls to different users under the hypothesis of equal channel conditions. In the case of multicast, vice versa, the number of users that receive the call has no effect on performance. For this reason, in results illustration we will speak generically of data flows, defined as services transmitted by eNodeB.

Characteristics of the services, described in the following, have been obtained by analysing the table LEWP-RCEG (Law Enforcement Working Party - Radio Communication Expert Group) [84] of TCCA, and the documents of the 3GPP, describing the services supported by the civil and professional LTE network(e.g.TS 26.114).

Group Voice Call

The service that most characterizes PMR systems.
Its main characteristics are:

- End to end setup time less than or equal to 300 ms (under the hypothesis that the caller does not need to wait for a response from the members of the group before he can start communicate). Simulations do not take into account the latency due to call setup, evaluated analytically in the previous section.

- Low distribution latency, less than 150 ms.

- Large number of users, at least 2000 supported in a particular area, with at least 36 active groups at the same time (recommended up to 75 active groups). The same call can be distributed to a large number of people, up to 500 per group.

- Tolerance to packet loss. For VoIP voice calls LTE standard defines QCI 1 service class characterized by PELR (Packet Error and Loss Rate) of 10^{-2}.

In simulations a media stream with data rate of 16kbps, comparable with the data rate used in VoLTE (Voice over LTE) systems, has been taken into account.

Conversational Video

Conversational video extends traditional PMR group call service with the simultaneous sending of video data.
Group calls are half-duplex communications; a single user transmits its voice stream in push-to-talk mode, while the video is selected by the operations center, which chooses whether to send its own clip or video data received from one of the other users, by redistributing it to all.
Voice and video are multiplexed together in downlink from operative center in an unique flow.
Main characteristics of this service are:

- Latency requirements inherited from voice service, even if for video communications these constrains can be relaxed.

- Moderate data rate; 3GPP Technical Specification TS 26.114 provides some examples of services for video telephony (referring in particular to two-way video calls + voice). The services considered in the document resort to video transmission based on AVC (Advanced Video Coding) with data rate comprised between 192 kbps and 384 kbps. In the simulations we assumed a data rate of 256 kbps.

- Moderate number of users per cell; the reference scenario consists of two groups with up to 10 users (typically 6 users).

High definition video

In TS 26.114, 3GPP considers the possibility to use high-quality video, based on protocols HEVC (High Efficiency Video Coding) for multimedia call with data rate ranging between 500 kbps and 790 kbps.
This data rate is sufficient for the distribution of asynchronous high-definition video, captured by fixed or semi-mobile cameras and sent from the control center to a small number of users. In the simulated system a reference media stream with rate of 768 kbps has been taken into account.

Simulated metrics

- Dropped Packet: the ratio between the number of sent and received packets. It includes packets that exceed transmission queues and that will never be sent.
- Packed delay.
- Average flow throughput; as long as the network traffic does not exceed the total capacity, the throughput corresponds to the service transmission flow rate (payload + overhead).
- Overall group communications throughput (NET throughput).

These metrics shall to satisfy requirements introduced in section 7.1 and resumed in Figure 8-14

Performance	Requirement	Unicast	Multicast
Packet Delay	<150ms	OK	OK for the required number of active groups
PELR (Packet Error Loss Rate)	$<10^{-2}$	ok until 210/270 users. Performance requirements are not fulfilled	ok 120/125 groups.
Number of active groups	>36 (suggested >75)	Not influential, the limit is the total number of users	OK
Number of users per group	>500	No	Not influential, the limit is the total number of groups
Number of active users	>2000	No	Not influential, the limit is the total number of groups

Figure 8-14: Group calls performance requirements.

8.3 Theoretical network capacity analysis

The ability of the network to support multimedia services, including high data rate services and a large number of group calls simultaneously active, is strongly dependent, among other things, from the network capacity in terms of bandwidth available for users. To evaluate the expected network capacity it is useful to analyse the structure of the radio frame used in LTE reference architecture (LTE-FDD). An LTE frame has a duration of 10 ms and it comprises of 10 subframes of 1 ms, up to 6 usable for MBMS. A subframe is composed of 2 Time Slot of 0.5 ms in time and a number of subcarriers in frequency variable with the channel bandwidth. Each time slot has the same duration of a Resource Block (RB) that is composed by 6 or 7 OFDM symbols (in case of extended or normal cyclic prefix). In frequency the number of RB per symbol is 25 or 50 corresponding to a channel bandwidth of 5 or 10 MHz.

The number of bits that can be transmitted during a 1 ms TTI (Transmitting Time Interval) depends on modulation and channel coding used. 3GPP identifies 27 possible combination of modulation and coding, named TBS (Transport block size) related to adopted MCS (Modulation and Coding Scheme).

The modulation and coding scheme that can be used depends on the quality of the signal perceived by the EU. Figures 8-17 and 8-18 show the trend of the Block Error Rate obtained with various combinations of MCS respect to signal to noise ratio variation at the receiver.

The choose of the modulation and coding scheme to be used depends on the quality of the signal perceived by the EU. Figure 8-19 and 8-20 show the trend of the Throughput respect to SNR obtained with various combinations of MCS.

MCS Index I_{MCS}	Modulation Order Q_m	TBS Index I_{TBS}
0	2	0
1	2	1
2	2	2
3	2	3
4	2	4
5	2	5
6	2	6
7	2	7
8	2	8
9	2	9
10	4	9
11	4	10
12	4	11
13	4	12
14	4	13
15	4	14
16	4	15
17	6	15
18	6	16
19	6	17
20	6	18
21	6	19
22	6	20
23	6	21
24	6	22
25	6	23
26	6	24
27	6	25
28	6	26
29	2	reserved
30	4	
31	6	

Figure 8-15: MCS/TBS association. TS 36.213

I_{TBS}	N_{PRB}					
	1	**...**	**25**	**...**	**50**	**...**
0	16		680		1384	
1	24		904		1800	
2	32		1096		2216	
3	40		1416		2856	
4	56		1800		3624	
5	72		2216		4392	
6	328		2600		5160	
7	104		3112		6200	
8	120		3496		6968	
9	136		4008		7992	
10	144		4392		8760	
11	176		4968		9912	
12	208		5736		11448	
13	224		6456		12960	
14	256		7224		14112	
15	280		7736		15264	
16	328		7992		16416	
17	336		9144		18336	
18	376		9912		19848	
19	408		10680		21384	
20	440		11448		22920	
21	488		12576		25456	
22	520		13536		27376	
23	552		14112		28336	
24	584		15264		30576	
25	616		15840		31704	
26	712		18336		36696	

Figure 8-16: Number of bits per subframe transported on 1, 25 (5MHz) or 50 (10MHz) Resource Blocks . TS 36.213

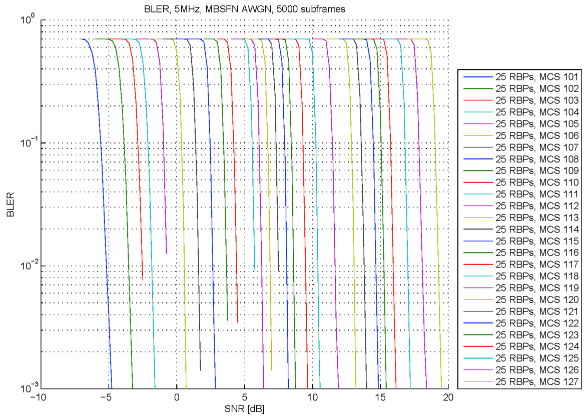

Figure 8-17: Block Error Rate for different MCS. Channel bandwidth = 5 MHz.

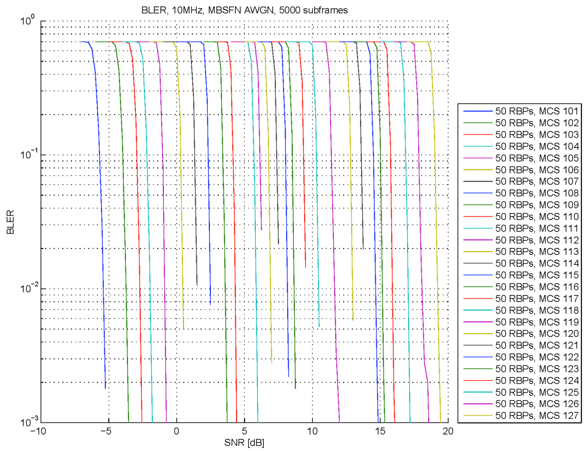

Figure 8-18: Block Error Rate for different MCS. Channel bandwidth = 10 MHz.

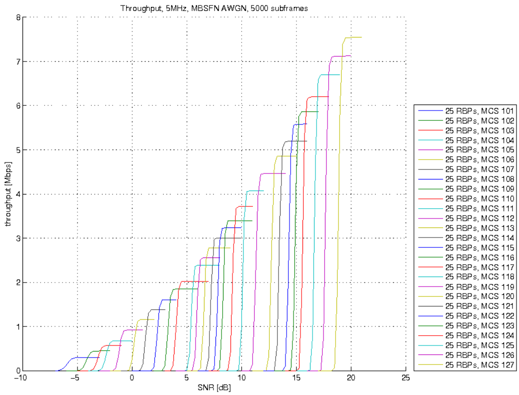

Figure 8-19: Throughput for different MCS. Channel bandwidth = 5 MHz.

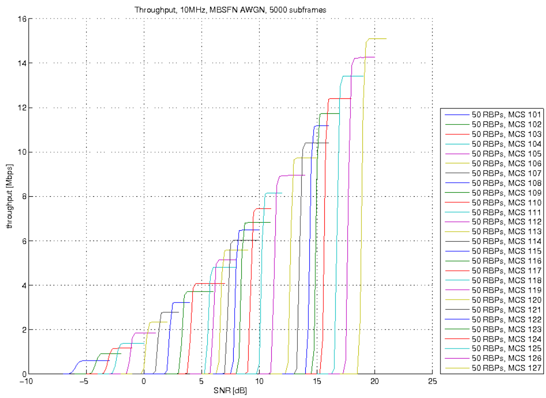

Figure 8-20: Throughput for different MCS. Channel bandwidth = 10 MHz.

In unicast transmissions the MCS is different for each user, whereas in multicast communications it is tuned on reception condition of the most disadvantaged user.

Once fixed modulation and coding, the maximum achievable rate in a 10 MHz system is approximately twice that at 5 MHz. However, increasing the bandwidth from 5 to 10 MHz, causes the doubling of contribution of AWGN noise; as a consequence if transmitted power does no change the SNR perceived by users decreases of 3dB. SNR reduction implies the need to use more robust modulation and coding schemes, characterized by lower spectral efficiency, which reduces the advantage of the greater availability of bandwidth (up to annul it in some cases).

SNR (dB)	CQI	MCS Index I_{MCS}	Modulation	Modulation Q_m Order	Code Rate (x1024)	I_{TBS}	NET Throughput 25 RB (5MHz)	NET Throughput 50 RB (10MHz)	Throughput Gain (10MHz vs 5MHz)
<-5	1,2	0	QPSK	2	78-120	0	680	1384	
-5		1				1	904	1800	0,00%
-4	3	2			193	2	1096	2216	0,00%
-2,5		3				3	1416	2856	9,85%
-2		4				4	1800	3624	
-1	4	5			308	5	2216	4392	
0,5		6				6	2600	5160	28,74%
1,5	5	7			449	7	3112	6200	
2,5		8				8	3496	6968	
3,5	6	9			602	9	4008	7992	60,87%
4		10	16QAM	4		9	4008	7992	39,33%
6		11				10	4392	8760	13,24%
6,5	7	12			378	11	4968	9912	
7		13				12	5736	11448	
7,5	8	14			490	13	6456	12960	
8		15				14	7224	14112	
9	9	16			616	15	7736	15264	66,93%
9,5		17	64QAM	6		15	7736	15264	
10,5		18				16	7992	16416	
12		19				17	9144	18336	
13	10	20			466	18	9912	19848	
14		21				19	10680	21384	
14,5		22				20	11448	22920	
15,5	11	23			597	21	12576	25456	
16		24				22	13536	27376	
17	12	25			666	23	14112	28336	
18		26				24	15264	30576	
19	13	27			772	25	15840	31704	
>20	14,15	28			873-948	26	18336	36696	

MULTICAST: minor advantages in bandwidth increase

UNICAST: greater advantages, but no proportional to bandwidth increase

Figure 8-21: SNR-CQI association and throughput vs bandwidth increase.

Table 8-21 shows data rates achievable either with channel bandwidth of 5 and 10 MHz. Last column shows the throughput increment due to bandwidth doubling. It can be noted that in correspondence of low SNR the detrimental effect of the noise increase is more evident.

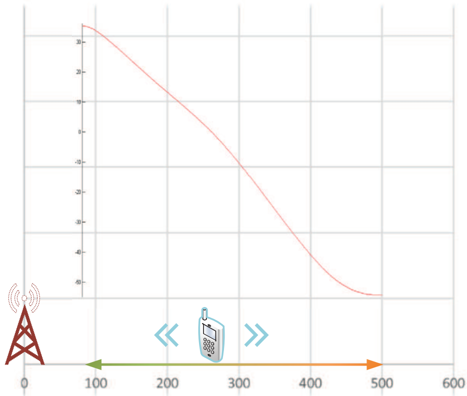

Figure 8-22: SNR vs UE distance from eNodeB.

Figure 8-22 shows how SNR changes with the EU distance from eNodeB, in the region of space considered for the simulations. Users are distributed through the simulation space in a uniform manner excluding extreme position at cell border or in the eNB near proximity.

Figure 8-23 shows an example of users distribution within a cell represented by the red circle. In simulations users have been located in the space delimited by green and orange circles.

Since for multicast transmission the MCS is chosen according to reception conditions of the most disadvantaged user (highlight in red in figure), there is an high probability of having low SNR and hence low data rate MCS, as a

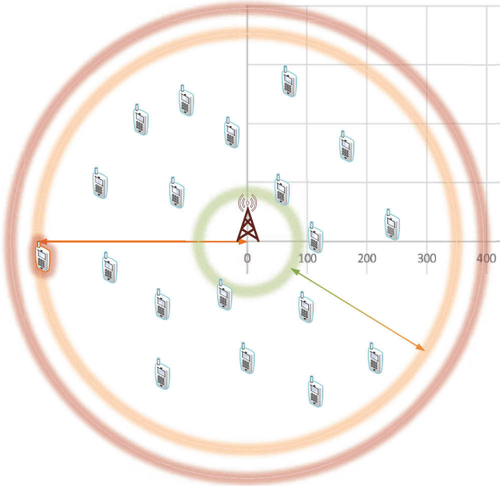

Figure 8-23: Example of UEs distribution in a cell.

consequence it is possible to hypothesize a throughput gain of about 10% (see Figure 8-21). On the other hand, when transmission is carried by unicast communications the SNR of each user is taken into account. If users are distributed within the cell in a uniform manner the system will experiment an average SNR equal to that of users that are in the central part of the cell coverage area, as a consequence the expected throughput increment can be estimate to approach the 30%.

8.4 Simulation results

8.4.1 Single cell performance

Conversational video distributed via unicast flows

The simulated conversational video service is characterized by a datarate at application layer of 256 kbps; in case of unicast distribution each user receives a different video flow independently from other users are receiving the same data. The number of active groups or the number users composing the groups is irrelevant. Once introduced the overhead due to user data encapsulation in the transport and network layer packets, the data rate is 277.3 kbps.

As long as the network is able to support the traffic load, the user throughput is equal to the communication rate. The number of communications supported by the network is 18 and 24 respectively for a bandwidth of 5 and 10 MHz.

Increasing the number of transmitted video flows the percentage of dropped packets increases exceeding the limit imposed by requirements and reducing the network throughput. Figure 8-25 shows the evolution of network throughput respect to active video flows. The net throughput reaches its maximum (i.e. 5 Mbps and 7 Mbps), and then maintains values near it due to the increase of the number of dropped packets .

As anticipated in Section 8.3, the channel bandwidth doubling does not correspond to an equivalent network capacity gain. The packed distribution delay remains moderate. Extending the constrain imposed to voice call to video call (i.e. delay less or equal than 150 ms), it is possible to note that the system would be able to support a number of communications greater than those imposed by packet loss constrain.

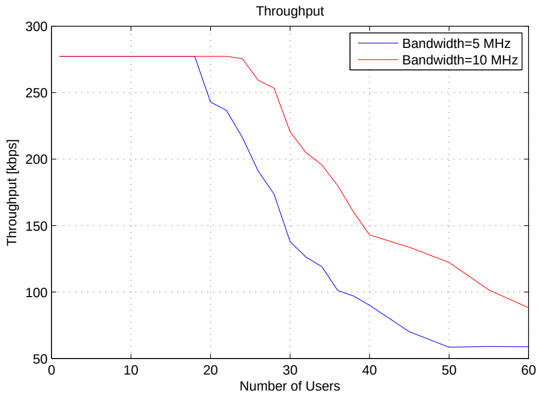

Figure 8-24: Throughput - Conversational Video - Unicast - Single cell.

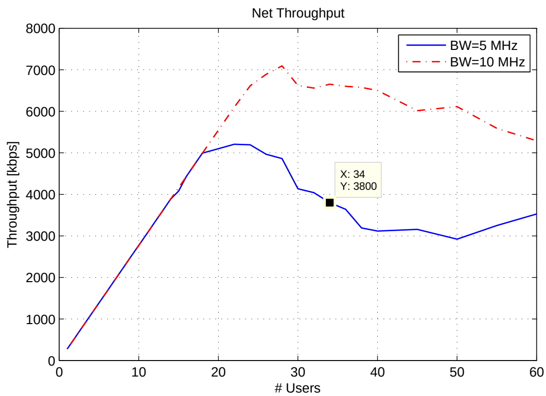

Figure 8-25: Network Throughput - Conversational Video - Unicast - Single cell.

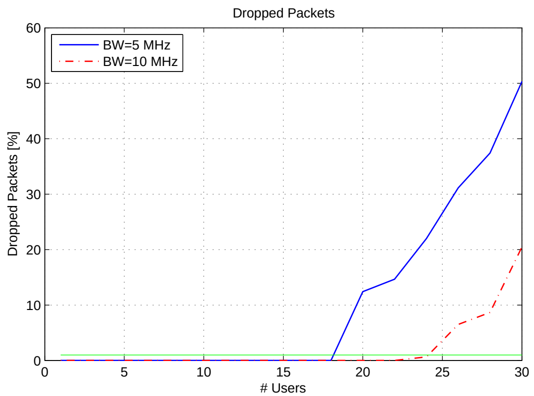

Figure 8-26: Dropped Packets - Conversational Video - Unicast - Single cell.

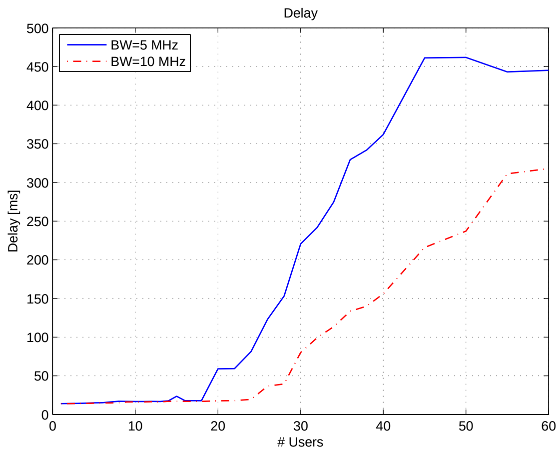

Figure 8-27: Delay - Conversational Video - Unicast - Single cell.

Conversational video distributed via multicast flows

The number of MBMS flows corresponds to actives groups independently from the number of users composing the groups.

The network throughput linearly increases with the number of active groups. Reserving 6 slot of 10 in the LTE frame to MBMS transmission, the system is able to support up to 6 active groups. Above this number, the network runs out resources and loses packets. At the same time a throughput reduction occurs.

As long as there resources are available, the delay experienced by the users is very low and is not a limiting factor for the number of video calls simultaneously supported.

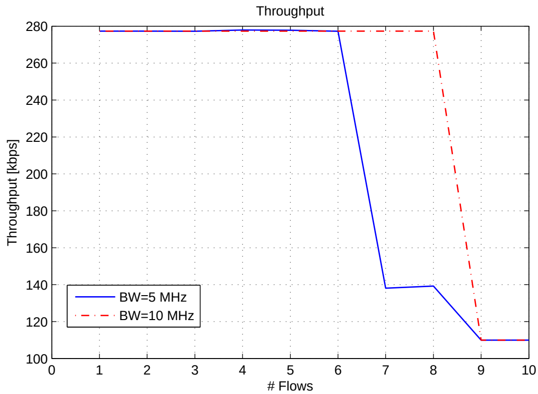

Figure 8-28: Throughput - Conversational Video - Multicast - Single cell.

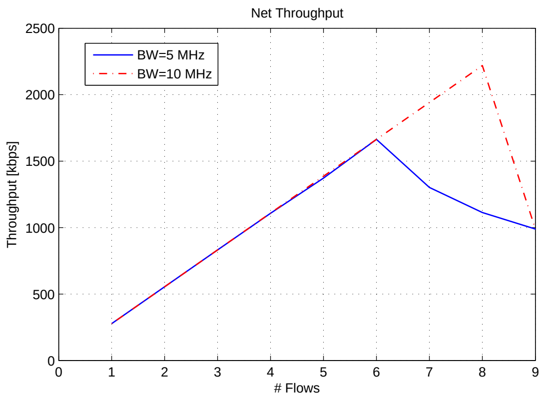

Figure 8-29: Network Throughput - Conversational Video - Multicast - Single cell.

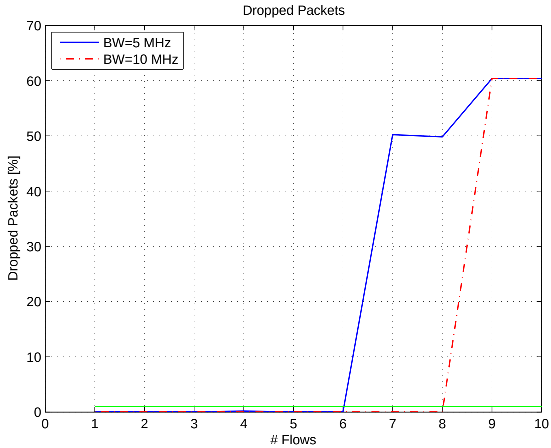

Figure 8-30: Dropped Packets - Conversational Video - Multicast - Single cell.

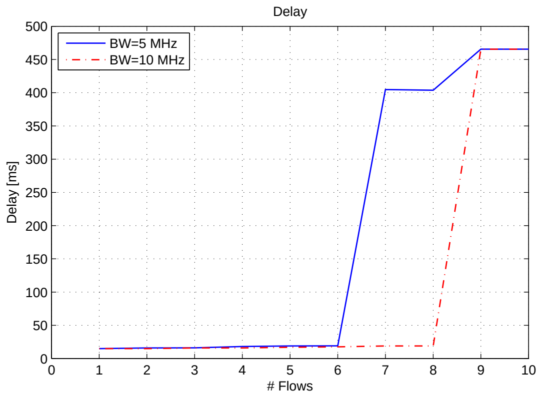

Figure 8-31: Delay - Conversational Video - Multicast - Single cell.

HD video distributed via unicast flows

Following graphics show performance of distribution of high definition video streams at 768 kbps based on HEVC (High Efficiency Video Coding) through unicast bearers. Once added the overhead due to IP and LTE signalling, the service rate reaches 860 kbps. For HD video distribution only a channel bandwidth of 10 MHz is considered. The user throughput remains constant and equal to service data rate until 5 users, then packet loss occurs and throughput decreases. In this case, the more stringent limit is imposed by packet loss ratio, applying the same limits imposed to audio transmission (i.e. $PER \leq 10^{-2}$), the system is able to support until 5 unicast flows.

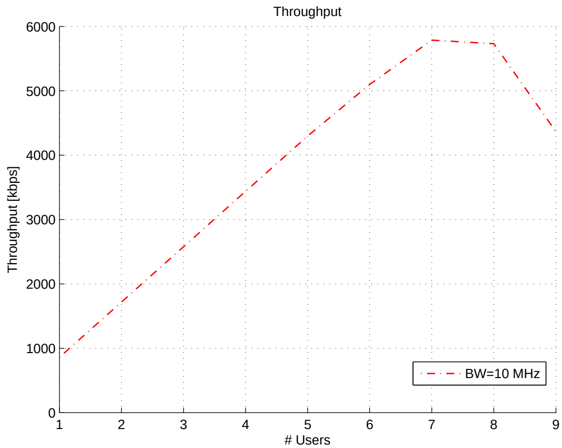

Figure 8-32: Throughput - HD Video - Unicast - Single cell.

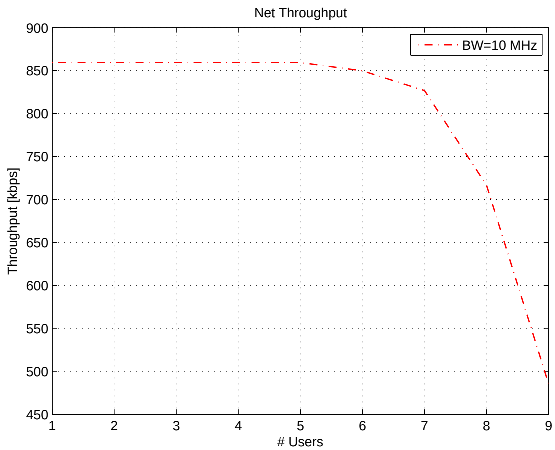

Figure 8-33: Network Throughput - HD Video - Unicast - Single cell.

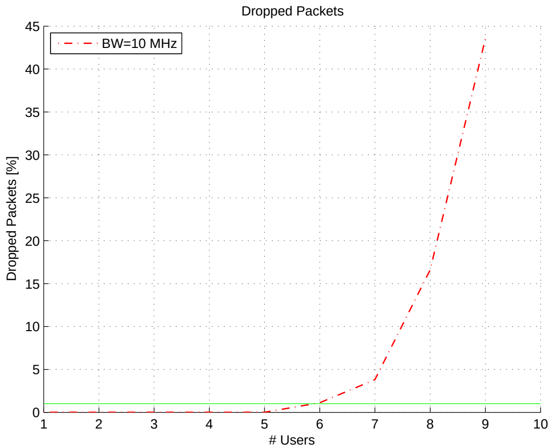

Figure 8-34: Dropped Packets - HD Video - Unicast - Single cell.

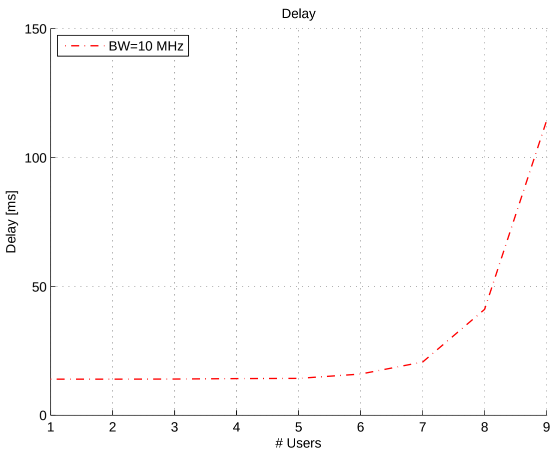

Figure 8-35: Delay - HD Video - Unicast - Single cell.

HD video distributed via multicast flows

In case of multicast transmission, each flow corresponds to a group. Performance are independent from the number of users that compose them, but are limited by the user of the group that experience worst channel conditions. Until 4 active groups no packet loss occurs, user throughput remains equal to service rate and network throughput increases. Beyond 4 active groups packet losses increase, earlier moderately then more considerably. In correspondence of 5 groups throughput starts to decrease, delay augments even assuming moderate values.

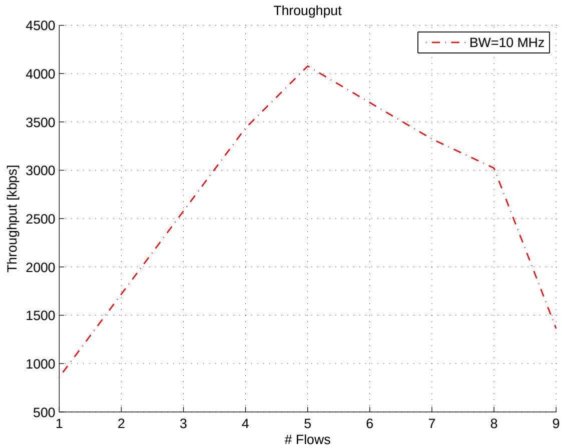

Figure 8-36: Throughput - HD Video - Multicast - Single cell.

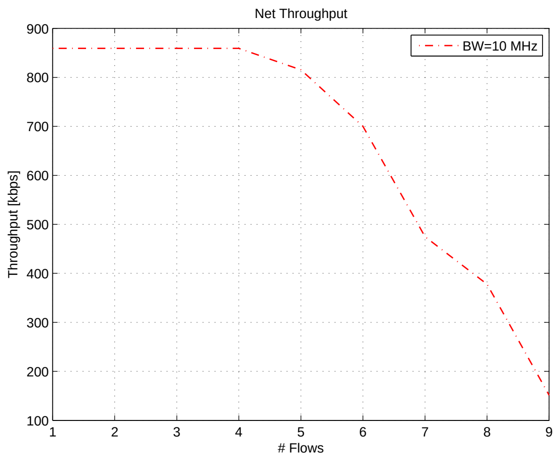

Figure 8-37: Network Throughput - HD Video - Multicast - Single cell.

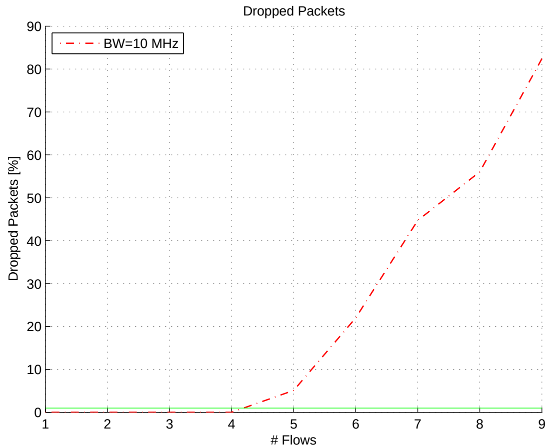

Figure 8-38: Dropped Packets - HD Video - Multicast - Single cell.

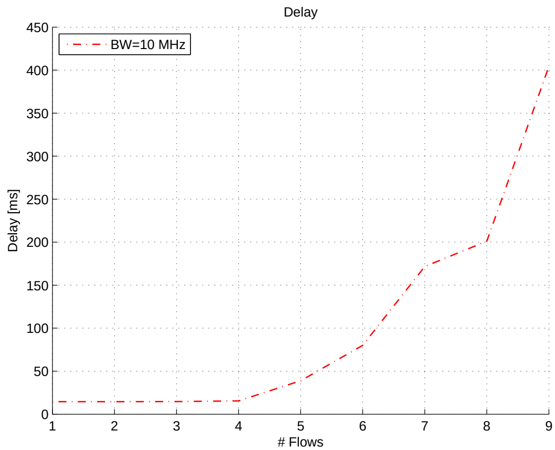

Figure 8-39: Delay - HD Video - Multicast - Single cell.

Voice call distributed via unicast flows

Group voice call service requires the management of an elevated number of users, until 2000, and at least 36 active groups (recommended 75). The number of users per group can reach the 500 units. Low distribution latency ($< 150ms$) and packet error rate ($< 10^{-2)}$) shall to be guaranteed. In case of voice group call distributed in unicast, the number of lost or dropped packets represents the limit of the system. The number of lost or dropped packets exceeds the 1% ($PER = 10^{-2}$) when the number of involved users exceeds either 210 or 270 respectively in correspondence of a channel bandwidth of 5 or 10 MHZ. These values are considerable lower than that required. In the other hand, distribution latencies are always low always below required value.

The VoIP codec considered for this service produced a data flow of 16 kbps at application layer. Adding IP and LTE overhead, the gross communication data rate reaches 21 kbps.

The user throughput remains equal to the gross data rate until relevant packet loss occurs, that is 210-270 users for channel bandwidth of 5-10 MHz.

Network throughput linearly increases with number of users, obtaining a maximum rate of 4.5-5.8 Mbps for bandwidth of 5-10 MHz.

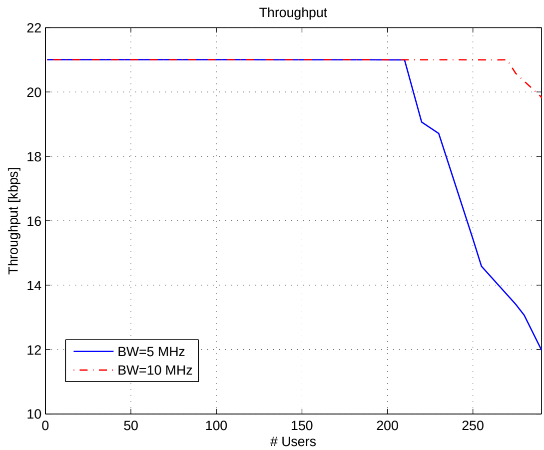

Figure 8-40: Throughput - Voice - Unicast - Single cell.

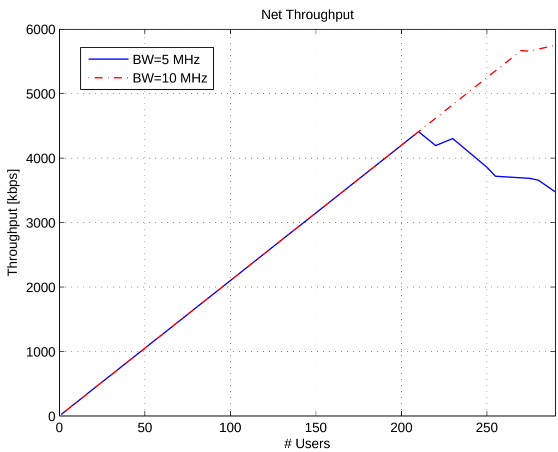

Figure 8-41: Network Throughput - Voice - Unicast - Single cell.

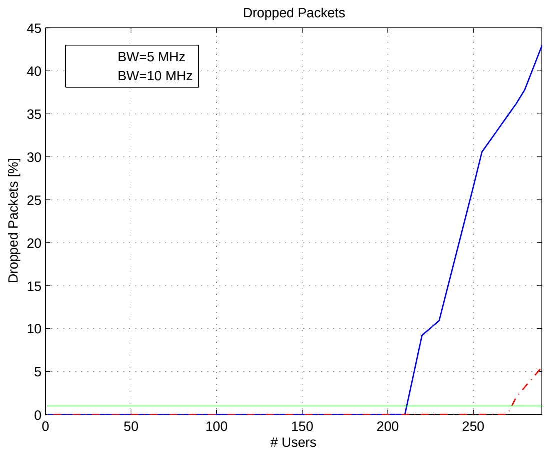

Figure 8-42: Dropped Packets - Voice - Unicast - Single cell.

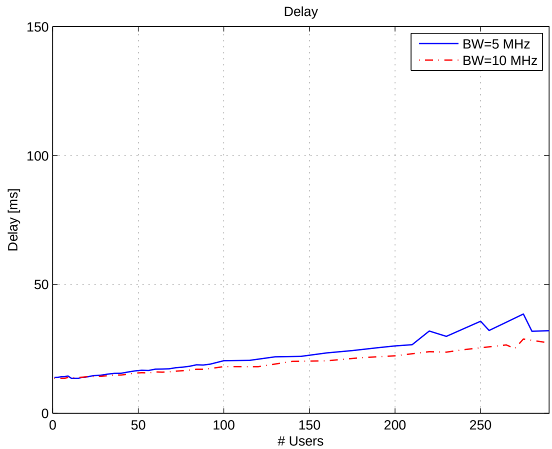

Figure 8-43: Delay - Voice - Unicast - Single cell.

Voice call distributed via multicast flows

Multicast distribution of group calls has the advantage of being independent from number of receiving users and being limited only by the number of simultaneous active groups.

Also in this case the limits to communication are imposed by the number of lost or dropped packets, that exceeds the limit in correspondence of 120-125 respectively for a channel bandwidth of 5 and 10 MHz. System performance respects requirements supporting 75 active groups irrespective of number of users per group. As long as no packet loss occurs, distribution latency is below the required limit, user throughput is constant with the increasing of number of user and the overall network throughput reaches 2.5. Mbps.

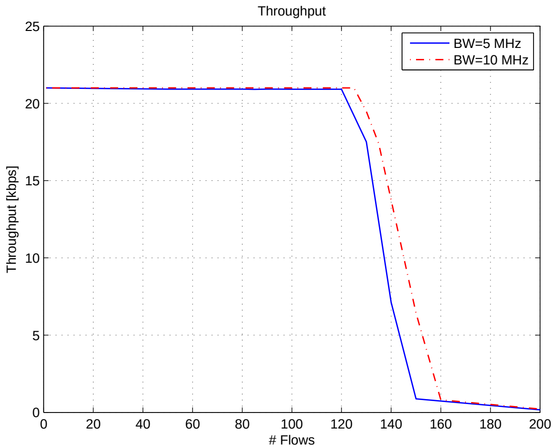

Figure 8-44: Throughput - Voice - Multicast - Single cell.

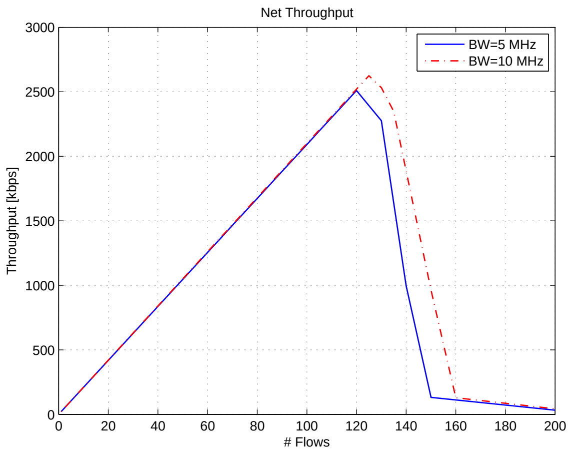

Figure 8-45: Network Throughput -Voice - Multicast - Single cell.

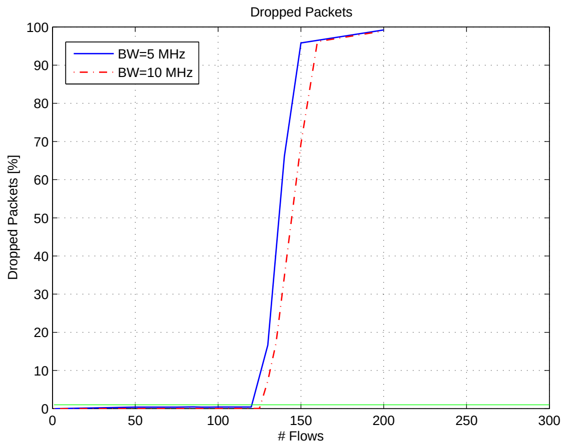

Figure 8-46: Dropped Packets -Voice - Multicast - Single cell.

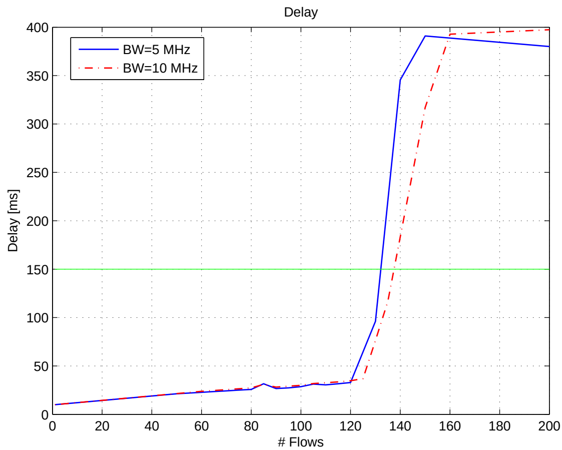

Figure 8-47: Delay - Voice - Unicast - Single cell.

8.4.2 Single Frequency Network

The signal received by a user in a Single Frequency Network (SFN) is the sum of different contributions coming from multiple eNodeBs differing in channel conditions and distance. The scenario taken into account in our simulation consists in a SFN composed of LTE cells arranged along an hexagonal grid with an inter-distance between eNodeBs of 500 m. Without loss of generality we took into account 4 cells as illustrated in Figure 8-48.

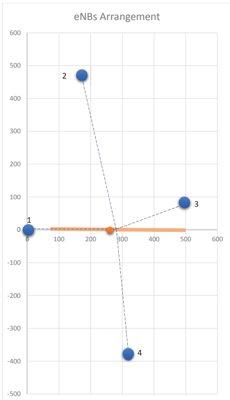

Figure 8-48: Voice performance on unicast and multicast distribution.

Figure 8-49 shows the SNR perceived by users both in case of single cell or SFN transmission. It is possible to note that, in the scenario under evaluation, the higher SNR values available in the SFN improve signal reception conditions that permit to resort to higher MCS schemes. In particular, hypothesizing to pass from a CQI 4 to a CQI 7, the throughput increses from 2.5 Mbps to 5.5 Mbps in 5 MHz networks and from 3 Mbps to about 7 Mbps in 10 MHz networks (see

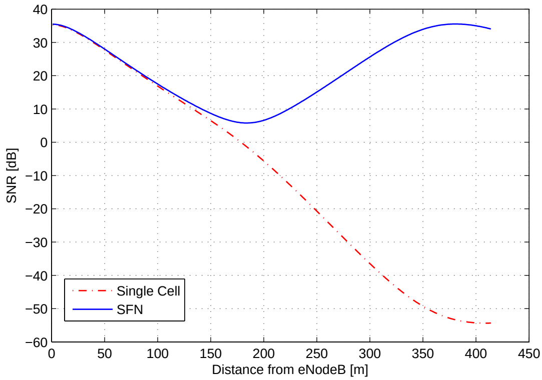

Figure 8-49: SNR trend in the SFN.

Figures 8-19 and 8-20). The quality of the signal received by the most disadvantaged user in the SFN is comparable with that of an average user in a single cell. Similarly to the unicast single-cell case, we can see a performance increment due to the bandwidth doubling; this improvement is more significant in SFN networks where the throughput gain (due to bandwidth doubling) is about of 40% (7Mbps vs 5Mbps) respect to single cell where the throughput gain is about of 20% (3Mbps vs 2,5Mbps).

This theoretical analysis is validated by following simulation results that have been obtained considering a SFN composed of 4 cells.

Conversational video

While in single cell case packet loss start to be relevant in correspondence of 6-10 groups (for 5-10 MHz bandwidth), now it is possible to maintain 20-26 active calls before performance degrades. User throughput remains constant and equal to gross datarate until MBMS resource saturation, allowing the overall throughput to reach 5.5-7.5 Mbps.

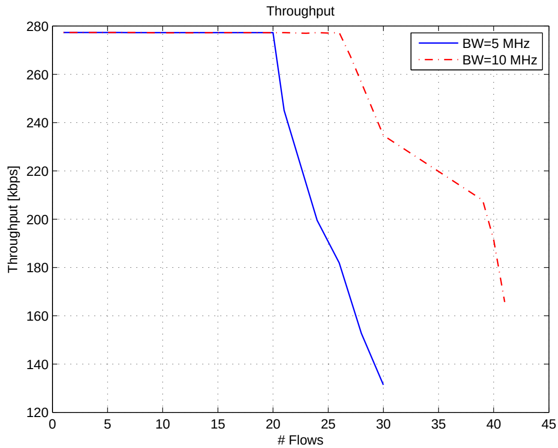

Figure 8-50: Throughput - Conversational Video - SFN.

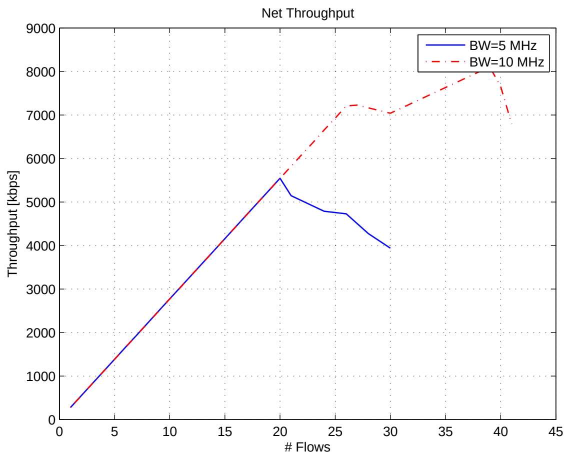

Figure 8-51: Network Throughput - Conversational Video - SFN.

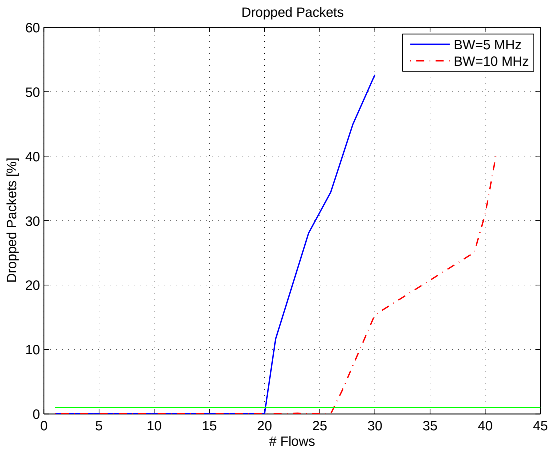

Figure 8-52: Dropped Packets -Conversational Video - SFN.

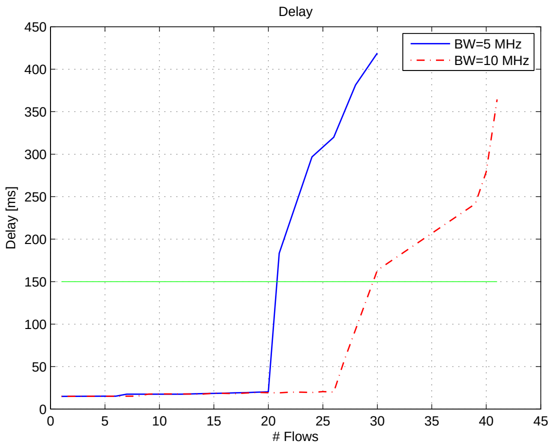

Figure 8-53: Delay - Conversational Video -SFN.

Latency is not a stringent requirement for conversational video service, however it assumes low values and considerably increases only when the system saturates.

HD video

The HD video transmission through SFN is supported until 6-8 groups of users in 5-10 MHz networks respectively. The maximum network throughput is 5-6.9 Mbps. Even if 3GPP does not define particular latency requirements for HD video, it is possible to note that the transmission delay is lower than 25 ms until the network becomes congested. The number of supported active groups is tied to packet drop requirements.

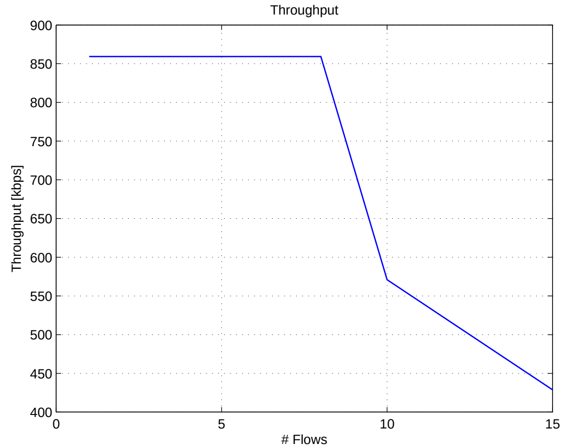

Figure 8-54: Throughput - HD Video - SFN.

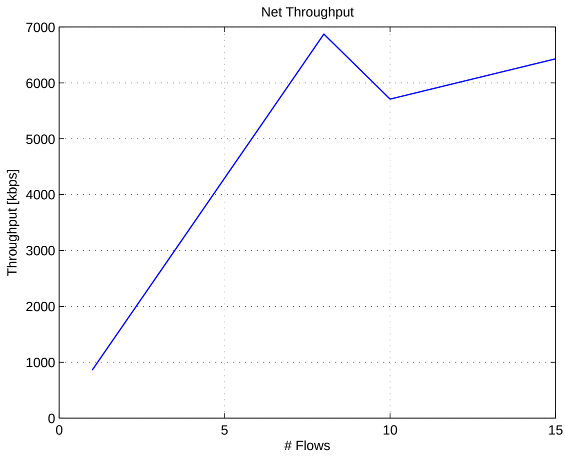

Figure 8-55: Network Throughput -HD Video - SFN.

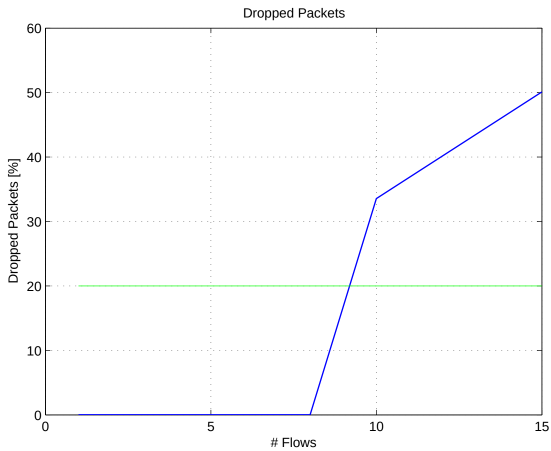

Figure 8-56: Dropped Packets -HD Video - SFN.

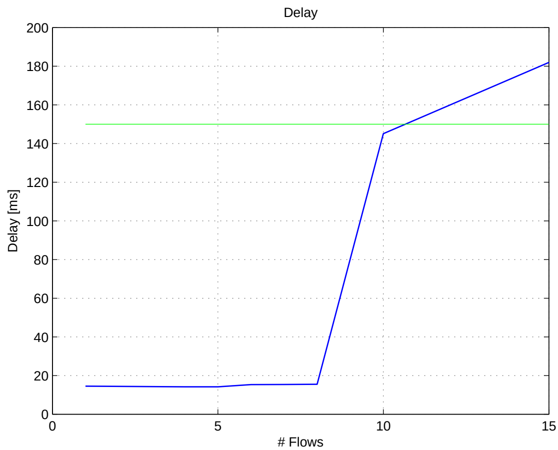

Figure 8-57: Delay - HD Video -SFN.

Voice call

These simulations consider the transmission of a multimedia flow characterized by an application rate of 16 kbps, that results in a physical rate of 21 kbps after overhead addition. Results show that the system can support up to 260-340 active groups (5-10 MHz) irrespective of number of users per group. The overall network throughput is 5.5-7.3 Mbps in the two considered cases. Above 260-340 groups the number of dropped packets increases and prevents the communication. Voice service is particularly sensible to packets distribution latency but in this case it is less influential respect to packet drop requirement.

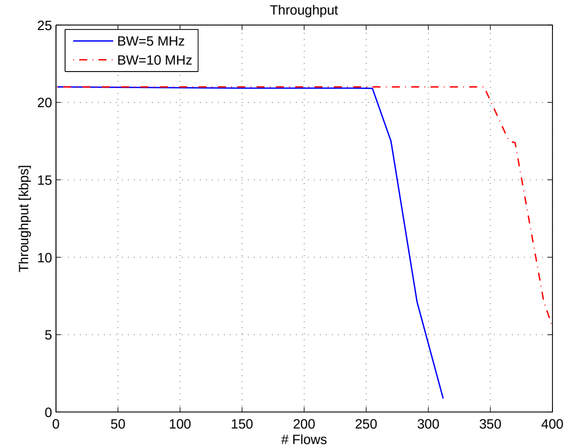

Figure 8-58: Throughput -Voice - SFN.

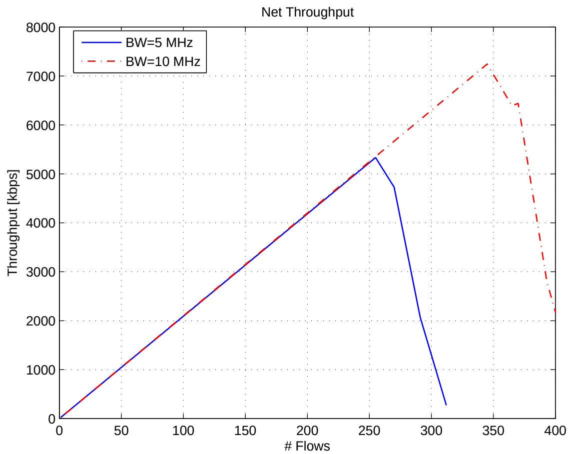

Figure 8-59: Network Throughput -Voice - SFN.

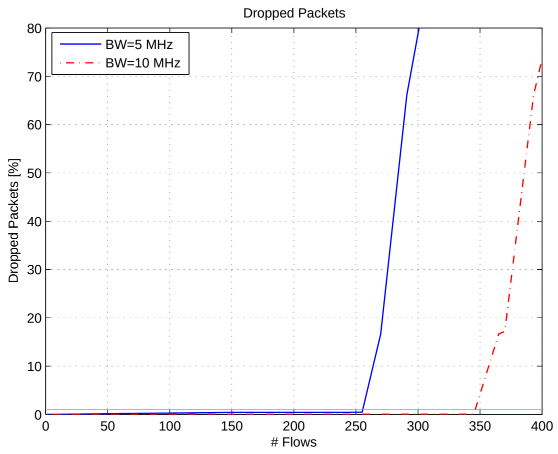

Figure 8-60: Dropped Packets -Voice - SFN.

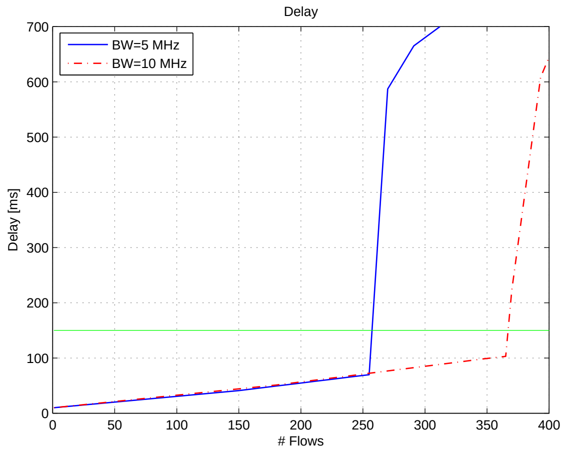

Figure 8-61: Delay - Voice -SFN.

8.4.3 Dynamic eMBMS activation assessment

In this section the dynamic eMBMS activation performance is evaluated considering a particular solution that foresees that, for downlink transmission, a multicast bearer is established and maintained by the network for each type of service. This bearers are shared by all communications that have the same characteristics of QoS at their beginning until a dedicated bearer is established, then the service is transferred. This solution should permit to reduce the call setup time using reduced quantity of resources. The limits of this solution consist in finding of a trade off between eMBMS resources allocated to setup and dedicated bearers.

Simulations show the system behaviour when multimedia flows are distributed either on a single or multiple bearers. Without loss of generality we took into account the conversational video service.

The network can transmit 13 video flows on a unique bearer and 8 video flows on multiple bearers. Indeed the number of supported calls is less or equal to 8 since this is the maximum number of communications that would be supported if all resources were available, but some resources are used by start up bearer, therefore this number is not actually reachable. Moreover, if more of 8 calls would be contemporary begun, exceeding ones will be dropped, since no resources would be available. Transmission delay is the most evident advantage of this solution. The limit of the network in this case is imposed by dropped packets.

This solution has some drawbacks. It reduces the system flexibility and introduces security issues. All communications, indeed, are receiver by all users and the discrimination has to be performed by application that discards calls belonging to other groups.

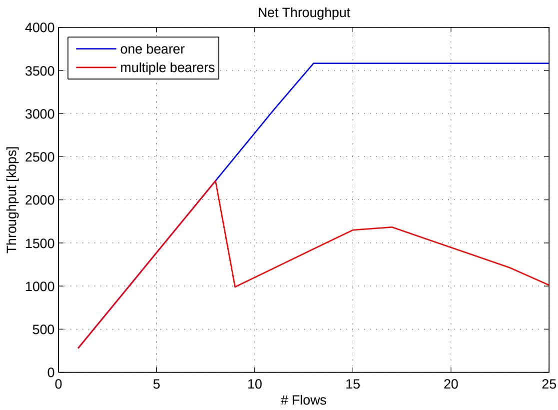

Figure 8-62: Throughput -Single multicast starting bearer.

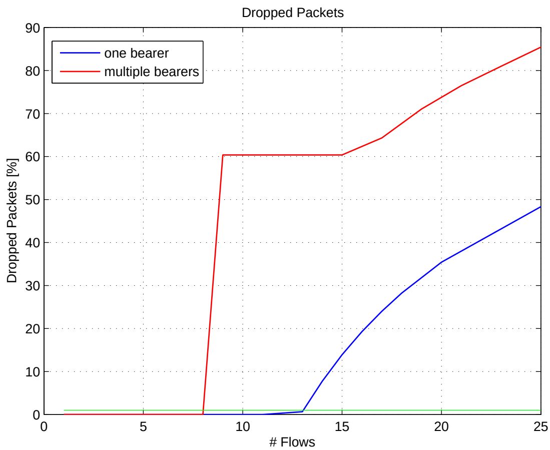

Figure 8-63: Dropped Packets -Single multicast starting bearer.

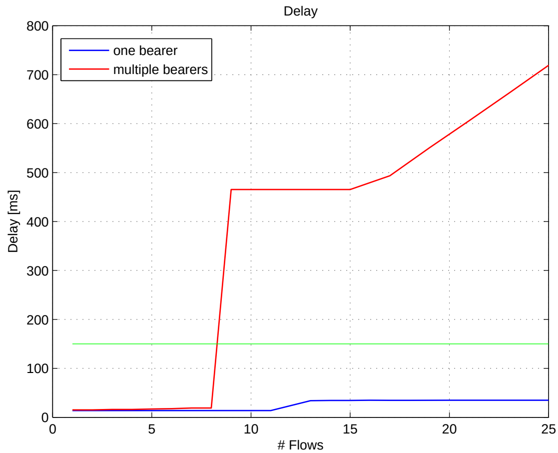

Figure 8-64: Delay -Single multicast starting bearer.

8.4.4 Conclusions

The two main requirements related to high quality video distribution are:

- 8 active users at the same time: this requirement is not respected in unicast transmission and it is irrelevant in multicast.

- 2 active groups at the same time: this requirement is totally fulfilled in multicast and partially in unicast partially, where there is no distinction between members of different groups.

Figures 8-65 and 8-66 summarize results obtained for voice and video group call distribution on unicast and multicast bearers.

The analysis conducted in this chapter confirms that the architectural choices advanced in Sec. 7.4 permit to obtain good results for the distribution of group call services. Unicast transmission in uplink permits to maintain set-up, joining and distribution delays as low as possible even in case of the caller terminal is in RRC_IDLE state. Quality and flexibility of transmission respect requirements imposed by the operative scenarios.
Unicast transmission in downlink is possible if the number of users and group is not elevated. Multicast transmission in downlink permits to distribute all types of traffic to a large number of groups irrespective of number of users that compose them. Even if results have been obtained for traffic composed of the same type of media in each simulation, indicative results for scenarios that foresee heterogeneous kind of media can be obtained opportunely combining them taking into account the ratio between the transmission rates of

different services and adding corrective factors due to different transmission overheads.
eMBMS have to be pre-established to obtain acceptable delay of call establishment. Dynamic eMBMS activation solution permits to save resources, but the delay experienced by receiver can be reduced activating first single unicast flows that switch on a multicast bearer (activated during the initial phases of the communication) when the number of user grows.
Resorting to Single Frequency Network transmission gives remarkable advantages in terms of number of group calls that can be provided at the same time(see Figures 8-67, 8-68, 8-69). Benefits derive from signal-to-noise ratio improvement due to reception of multiple replicas of the signal transmitted by different eNBs, especially of the users that experience worst channel conditions and influence the choose of modulation and coding scheme to be used in the SFN.

However, in order to respect the requirements due to mission critical operation some limits have to be respected. Receiving terminal should not enter in discontinuous reception, because too high timers. This implies the adoption of terminal with high energy capacity or, in alternative, the possibility to resort to a mobile gateway mounted, for example, on a vehicle, that communicates with mobile equipment through lower energy technologies, e.g. Wi-Fi.
Moreover, to permit more flexible resource allocation procedures the Multicast channel Modification Period shall be reduced.

Performance	Requirement	Unicast	Multicast
Packet delay	<150ms	OK	OK
PELR (Packet Error Loss Rate)	$<10^{-2}$	ok up to 210/270 users requirement on number of users non satisfied	ok up to 120/125 groups
Number of active groups	>36 (suggested >75)	Not influential, the limit is the total number of users	OK
Number of users per group	>500	No	Not influential, the limit is the total number of groups
Number of active users	>2000	No	Not influential, the limit is the total number of groups

Figure 8-65: Voice performance on unicast and multicast distribution.

Performance	Requirement	Unicast	Multicast
Packets delay	-	OK	OK for required number of active groups
PELR (Packet Error Loss Rate)	$<10^{-2}$	The number of required active users can not be supported on 5MHz	OK
Number of active groups	>2	Not influential, the limiti s the total number of users	OK
Number of users per group	>10	Only 10MHz bandwidth networks	Not influential, the limit is the total number of groups
Number of active users	>20 (2 groups of 10 users)	Only 10MHz bandwidth networks	Not influential, the limit is the total number of groups

Figure 8-66: Conversational video performance on unicast and multicast distribution.

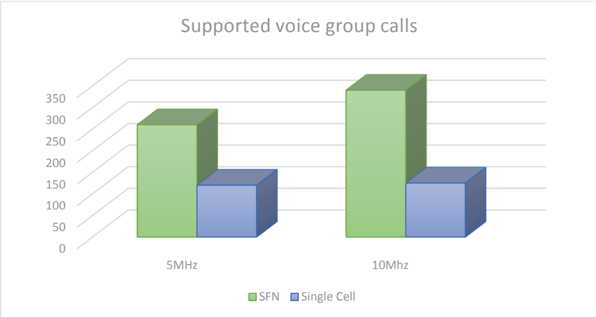

Figure 8-67: Voice performance comparison between single cell and SFN distribution.

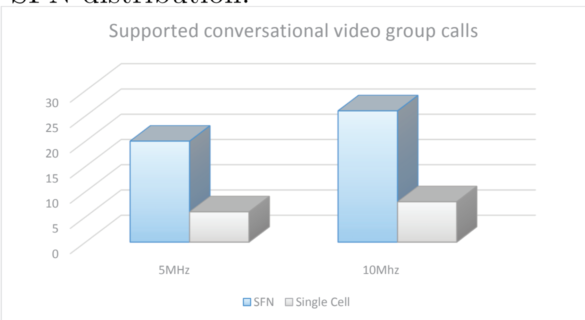

Figure 8-68: Conversational video performance comparison between single cell and SFN distribution.

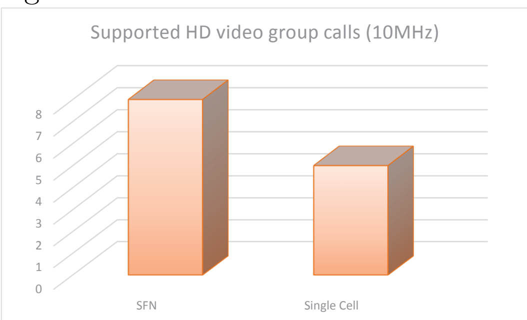

Figure 8-69: HD video performance comparison between single cell and SFN distribution.

Chapter 9

Smart Management Platform for Public Safety

In the near future, quality of life of billions of people will depend on the capability of cities of saving energy, reducing harmful emissions, improving the living conditions and increasing citizens security. Today, cities host approximately half of the world population and it is expected that in 2019 more people will live in cities than in rural areas. This introduces new challenges regarding services and infrastructures. Indeed, resources (like water, energy, healthcare, education, ...) are limited and often poorly managed. Two other important aspects concern the pollution that can lead to respiratory disorder and other diseases, and the global warming that is leading to intense flooding, violent storm, forest fires. For these reasons, the interest of worldwide authorities, manufactures and services suppliers is toward the possibility of introducing Information and Communication Technology (ICT) intelligent solutions in the urban ecosystem, to contribute to the creation of a "smart city" together with human, capital and infrastructure investments. What makes a city "smart" is the combined use of software systems, network infrastructures and users devices. Communication and collaboration technologies are key elements of a smart city. In particular, a smart city can be seen as a set of smart infrastructures and services [95] as:

- **City Management**, improving quality of offered services by reducing costs, introducing transparency and citizen participation. It has to give to the citizen an appropriate, precise and clear information awareness about the correct use of the resources made available by the community and show the possibilities deriving from a direct involvement in civil life;

- **Education**, allowing the use of technology to increase access, improving the quality and experience, and reduce costs.

- **Healthcare**, allowing more accurate and rapid diagnosis and responses to emergency. Storage and communication platforms permit data to be stored and shared for diagnosis and research.

- **Transportation**, improving services, as traffic management in congested areas, and at the same time reducing costs. The goal is to reduce traffic congestion while encouraging the use of public transportation;

- **Environment**, improving the environmental conditions, limiting the sources of pollution, increasing the green areas, providing widespread health services, etc.;

- **Economy**, promoting innovation, productivity, exportation, etc.;

- **Public safety**, improving the capability of monitoring critical environments, forecasting disasters due to both natural causes or terrorist attacks. It uses real-time information to respond rapidly to emergencies and threats.

Among these smart systems, this chapter aims to identify an efficient architecture of a smart public safety platform. Indeed, with the huge increasing of cities population density public safety operators need to respond more quickly to emergency situations. It has made essential the need of effective and smart emergency management solutions. New and advanced systems and platforms have to be identified in order to guarantee the safety and well-being of citizens, properly protect services and critical infrastructures. Smart solutions for public safety involves to accomplish complex tasks related to:

- **Monitoring and Forecasting**. Adopting specific monitoring activities and analysis of results arising from the monitoring campaigns to prevent natural or man-made disasters and crimes;

- **Planning**. Preparing action plans to be adopted in case of disaster;

- **Emergency responding**. Management and coordination of the operations of the first responders, that follow a natural disaster to limit damages and restore security;

- **Recovering**. Handling post-emergency activities with the aim of coordinating, designing, and verifying the restoration works for a rapid return to normal life conditions.

In order to fulfil all the required tasks providing a complete emergency management framework, this work proposes a system made by different components that interoperate among them. Main components are a new generation Professional Mobile Radio system, Wireless Sensor Networks and Social Networks.

In recent years all the above components have been widely discussed in the literature due to the increasing fear of possible disruptive events and, hence, to the increasing interest in the emergency management. However, to the best of our knowledge, the definition of a platform that integrated all of them is novel.

In particular, several papers and projects have proposed specific solutions for communications in emergency situations. In all of them, data exchange is always considered of paramount importance both to coordinate first responders operations and collect information from the field in order to enable suitable actions [96–99]. Security issues concerning MANET in emergency scenarios have been considered in [100] where the concept of mission-specific certificate to manage public keys is introduced.

Likewise, several proposals are available from the literature for what concerns the environmental monitoring and data gathering. The aim is to collect data from different sensing networks such as satellite and sensors mounted on terrestrial, aerial or naval stations. In particular many research efforts have been devoted to the sensor networks optimization and realization of efficient sensors for specific application in emergency situations [101–103]. More complete systems integrate data from remote sensing with spatial information in order to build a geographic information platform for public services. These timely and geo-referenced information is processed and provided to the operators involved in dealing with natural disasters, environmental protection, agriculture, health, transportation, sustainable development, civil protection, tourism, etc. In particular, earth observation is the main subject of the research projects related to the European Programme Copernicus [104]. The objective is to supply information and solutions for timely response and planning of appropriate risk mitigation solutions in areas highly vulnerable to natural disasters. The application scenarios span breaking of dams, storms, floods, earthquakes and landslides.

In [105] an Event Driven Architecture for information services for monitoring public areas and infrastructures is presented. The system is mainly based in the notifications produced by the data observed by the sensors that are correlated, generating alerts that are distributed to the public safety operators and the citizens via broadcast services. Capability of direct communications among Machines (M2M communications) without human intervention is a field of great interest [106, 107] for sensor networks. Indeed M2M communication is considered as key technology enabling the smart city paradigm.

Data collected by sensors need to be aggregated and elaborated in order to generate alarms and, hence, to trigger consequent actions. In [108, 109] two examples of fusion of collected data to generate alerts are presented.

Sensors can be used also to support rescue operations or emergency management policies within the risk area [110].

Another component of the proposed system is represented by the information sharing among public safety operators and/or citizens. In [111] it is proposed an open virtual collaboration environment that facilitates collaborative work in a virtual space made by a dynamic website and 3D space for meetings. The citizens are usually destination points of alerts and warnings as in [105, 112] where emergency-related information are distributed by means of multi-channel communication links. Recently, also the active citizens participation has being considered. In particular, the pervasive diffusion of social networks can be exploited as a new information source [113–115]. An interesting research field is the participated sensing represented by the integration of data coming from social networks with sensed data [116, 117].

Despite the above intense research efforts, efficient and intelligent solutions compliant with the smart city paradigm and requirements still need to be fully defined. Smart city can use advanced ICT infrastructure and analysis methods to enhance and coordinate the information flow between first responders, public authorities (transport, utilities, hospital etc.) and citizens.

In this work we present an innovative framework that incorporates many of the key aspects of a smart emergency management system in an unique platform. It integrates different components (i.e., monitoring, computing, and communication) that provide heterogeneous sensed data, participated sensing, efficient communications, data gathering and analysis. The proposed framework aims to provide a more accurate, responsive and autonomous system for an evolved "smart city" environment.

9.1 System outline

Intelligent monitoring methods need of specific information acquisition, which is typically related with multimedia contents and often cannot be provided by the sole sensors networks. The novel system architecture outlined in this work, seems to be a promising solution to enable a smart management of the Public Safety. The proposed platform aims to functionally integrate four different components: a new generation Professional Mobile Radio (PMR) system able to allow broadband data communications among the first responders and the Authorities, suitable Wireless Sensor Networks (WSNs), Social Network tools and a Data gathering and analysis system able to collect and elaborate heterogeneous information coming form different sources. Each element acts not only as an information collector/distributor, but can also cooperate with the others in the realization of a smart framework. This makes the system more effective and autonomous, hence, reducing (or even avoiding) the need of human interactions.

The WSNs are the essential element in any system devoted to the emergency monitoring and prevision and, in general, to gather information needed for a smart environment management. The integration of heterogeneous WSNs permits to monitor different physical parameters and the trend of ongoing processes (e.g., water level of a river, rainfalls, soil movements). The collected data are then elaborated in a suitable way thus generating alarms and providing useful information to a more complex system able to take decisions. The alarms are forwarded to Public Safety agents to promptly intervene with ad-hoc tools and assets. The PPDR (Public Protection Disaster Recovery) operators on the field shall be equipped with advanced communication and acquisition devices, so that they can take pictures and videos in order to provide the Operative Center (OC) with a real knowledge about the current situation and the possible risks. In this way, the OC can coordinate all the assets on the field and identify the best procedures to preserve the citizen security and safety.

Social Networks represent novel tools, which allow the convergence of civil users activity and professional security monitoring by authorities, permitting the OC to exploit the large variety of information coming from the citizens. The pervasive diffusion of smartphones and the availability of broadband connection make the citizens a primary source of real-time information even in areas that are still not reached by sensors [118]. This allows the rescue missions to be more effective and ubiquitous. Moreover, the proposed framework intends to exploit the Social Networks also to broadcast useful information to the citizens, texting users groups to alert about possible dangers. The added value of Social Networks is the wide accessibility obtained through the diffusion of smart mobile devices, making prompt information available to users anytime and everywhere.

The proposed system represents a smart solution for a disaster management for authorities and citizen, covering all the phases of an emergency: monitoring and forecasting, planning, response and recovery.

The main goal of the prevention activity is to identify the possible risks for a specified area, monitoring data provided by sensors networks as well as from Social Networks. After the identification of possible risks, for a particular area, the system informs the population about the hazard with asynchronous informative campaign or quick alert messages, trying to moderate the effect of the possible evolutions from risk to emergency and alerts authorities. Social Networks offer a great opportunity for establishing a bi-directional communication line with the population; two are the main differences with the classical mass media information system:

- users provide real-time feedbacks to the authority about the situation;

- users can be reached with push messages on mobile devices.

In case the emergency cannot be avoided, the proposed framework provides authorities with an efficient communication system for emergency managing, with the availability of the Professional Mobile Radio (PMR) broadband network.

The proposed framework, described in details in the next section, needs many interfaces to communicate with the information sources, which would lead to a technology-dependent context. In order to increase the architecture flexibility, suitable *adapters* have been foreseen to make the framework technology-independent, so that data can be processed regardless the exploited technologies.

9.2 System architecture

The proposed framework is characterized by a flexible and scalable structure that permits its use in several emergency scenarios to preserve the citizens safety and the environment care.

Fig. 9-1 represents the general architecture of the framework, whose basic components are:

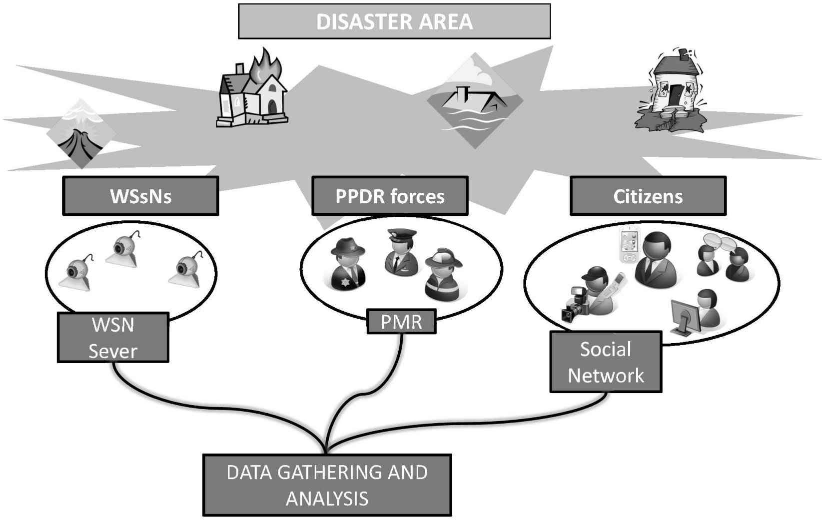

Figure 9-1: General framework architecture.

- Professional Broadband Communication system.
- Data Gathering and Analysis system.
- Wireless Sensors Networks.
- Social Networks.

9.2.1 Professional broadband communication system

The Professional Broadband Communication system allows the OC to remotely coordinate the operations of all subjects involved in the emergency as well as to gather and broadcast information from/to professional operators. However, the communication system supporting information flows between OC and PPDR operators has to provide secure and reliable communication modes and specific capabilities not envisaged in public networks. Therefore, the proposed platform has planned a PMR communication system in order to support services like trunked and Direct Mode Operation (DMO) communications, preemption, group call, dispatcher function, encryption, quick call setup [119]. Usually these functions are not provided by commercial mobile systems, but are essential to support emergency management operations. In addition, these networks are characterized by a high resilience and can be temporally deployed. This is important in disaster areas where commercial communication systems can be not available or have been destroyed. One of the most widespread PMR technology is TETRA (TErrestrial Trunked Radio) an open standard defined by ETSI (European Telecommunications Standards Institute) operating in [400 - 800] MHz bands in order to guarantee large coverage with few Base Stations [119]. However, TETRA together with its evolution named TETRA2 or TEDS (TETRA Enhanced Data Service) is characterized by severe limits in data transmission bandwidth (19.8 kbps and 500 kbps respectively). In order to build a system that can facilitate and make more efficient, intelligent and effective the on field operations it is mandatory to adopt a novel technology. The smart platform makes available a lot of heterogeneous information that can be very useful to the PPDR operators. Hence, the communication system must be able to support all services of a typical PMR and maintaining a complete interoperability, with additional broadband data transmission capability. This permits to increase communication potentialities with a set of advanced services, for example the

PPDR operators can receive/transmit videos in real-time of disaster scenes from/to the OC, in order to plane the intervention in a more precise manner. Video Information are quite important in order to get disaster awareness, e.g., to know if people are leaving the area and evacuation routes are ready and available, properly plan actions and rapidly respond to emergency situations. Therefore, the communication system has to be able to deliver video information in a secure and reliable mode. Moreover, the operators can be provided with advanced tools like interactive maps or ad-hoc applications.

Currently, the ETSI started the standardization process for the development of the third stage of TETRA system, known as TETRA BroadBand (TETRA-BB) o TETRA3. It will be oriented toward integration of services and performance required by the PMR users on the 3GPP LTE (Long Term Evolution) technology, that represents the most advanced commercial mobile system.

LTE networks allow high data rate services and flexibility with limited costs. It is characterized by a simple all-IP system architecture and flexible air interface with low latency (10ms) and enhanced performance and efficiency. The theoretical transmission rates are equal to 326 Mbps in downlink e 86.4 Mbps in UL. Furthermore, the LTE system offers great flexibility to support a wide variety of deployment scenarios and operators needs. In particular, we propose here a solution TETRA communications are carried by an LTE network used as access network. A PMR user with a TETRA client application running on an LTE terminal, can communicate with other TETRA users over an LTE network, while using all of the TETRA services. This TETRA/LTE hybrid solutions combines the best of the two technologies to provide broadband overlay services to existing TETRA networks. However, the critical nature of much of the first responders communications places high demands on the LTE radio access system. The network must be able to manage different and dynamic traffic bearers with varying priorities and requirements. This can be achieved thanks to the end-to-end Quality of Service (QoS) framework defined in LTE that permits to manage QoS class identity (QCI) and access retention priority (ARP). Conversely, to enable critical communications services there are currently two key open issues (Sec.9.4) to be addressed within the LTE standards:

- Group Communications
- Direct Communications (proximity-based services).

In addition LTE coverage, resilience and security must be carefully evaluated in proposing enhanced solutions.

9.2.2 Data gathering and analysis system

This system permits to collect and integrate all the information sources in a single overview of the City status and of the event. It provides a wide and complete view of the situation. Heterogeneous information are correlated, detecting links that are even not apparent, thus providing a smart support to the identification of problems.

As depicted in Fig. 9-2, the Data Gathering and Analysis system consists of five elements:

- Adapters: interfaces for the interaction with Social Networks, PMR networks and WSNs.
- Data Gathering subsystem: software managing the information collection.

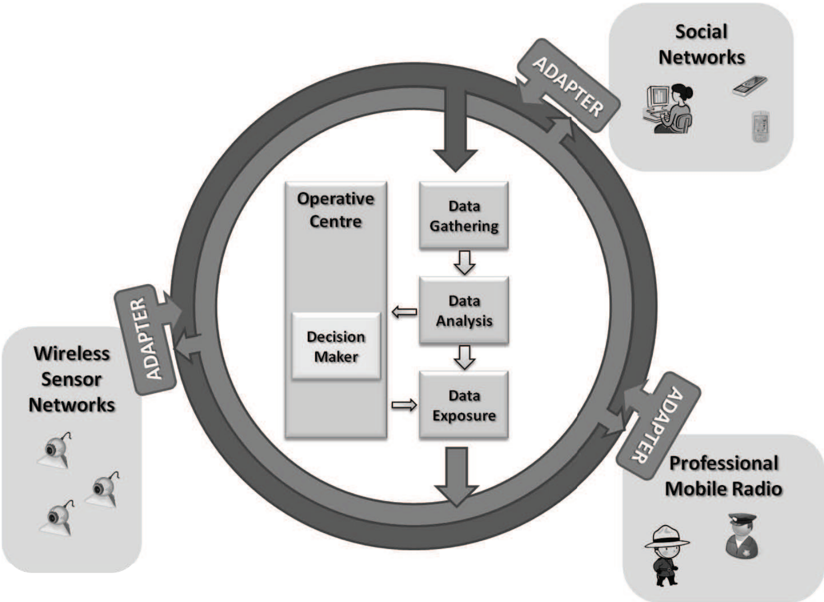

Figure 9-2: Data Gathering and Analysis system.

- Data Analysis subsystem: databases and algorithms for the data processing.
- OC: core of the structure.
- Data Exposure subsystem: exploited to represent the results of the elaboration in a human-readable format.

Adapters

The adapters act as bidirectional interfaces between the specific used technology (e.g., web, blogs, smart phone apps, WSN, Social Networks) and the Gathering and Exposure subsystems. Each protocol, procedure and data structure has to be converted in a technology-independent format and made available to the system. The aim of each adapter is also a pre-filtering of information according to a programmable logic defined by the system manager (Decision Maker). In particular, not all the information coming as inputs into the Platform has the same relevance and reliability and, on the other hand, not all the information can be distributed to all the possible final users (PPDR operators, Authorities, Citizens...). For this reason the Decision Maker will actuate policies to handle either input and output information.

The most relevant adapters include:

- mobile device adapter, which interacts with the information on mobile devices and made available on a voluntary base;

- specific Social Network adapter, like Facebook, Twitter, Skype, MSN, etc., by exploiting web services and API. This adapter has to gather and filter relevant information (e.g., position, pictures, videos, comments) from the large amount of data available on these platforms.

- Public Authority systems adapter, to manage the informations coming from official channels.

- WSN adapter, assisted by a WSN server, due to the data complexity and non-homogeneity coming from the environment sensing.

The solution envisaged here is to translate incoming data in an application independent data interchange format, like XML [120] or JSON [121]. Both have the advantage of representing data in text format that can be read from human people or serialized to be used from a machine. The data are structured: in XML the structure is defined in the DOM while in JSON it is intrinsically defined by its basic components array and records. Both have the ability of including binary information. However, the array-objects JSON format respects the DOM based format of XML. It reduces the data redundancy and makes the first more efficient in terms of data parsing and transfer [122].

Data Gathering subsystem

The Data Gathering subsystem interacts with the adapters to collect environment measures, alarms and multimedia data. Furthermore, the subsystem is in charge to integrate the data in order to improve the knowledge of the surrounding environment. To achieve this goal, semantic models (ontology-based) are exploited to represent entity and relationship among them. The logic of this subsystem is directly controlled by Decision Maker by setting configuration parameters and semantic rules suitable to the specific application.

Data Analysis subsystem

The information coming from heterogeneous sources must be stored and analysed in real-time to identify the actions to be done for the prevention or the management of the emergency. This subsystem provides functionalities of *data-warehousing* and *data-mining* operating on current data, in order to identify immediate actions, and on historical data, in order to identify medium/long term policies. Emergency situations are detected by means a statistical analysis of the events. The prevision and analysis models vary depending on the specific application scenario (e.g., floods, fire, landslides, etc.). The

Data Analysis subsystem has mainly in charge to store and elaborate the information with the aim of:

- providing the Decision Maker with notifications, alarms and suggestions to support the decisions;
- identifying homogeneous groups of people to forward the same informative stream;
- identifying non-trivial relation in data (data mining);
- detecting users viewpoint relative to a certain event (opinion mining).

Advanced techniques of machine learning are exploited to identify non-trivial relation in data. Identification techniques are used to understand and represent particular events, like dominant and recurring terms, sentence semantic clustering and ordinary used pattern.

Operative Center

The OC is logically situated at the core of the framework, collecting the outputs of data analysis subsystem and elaborating them before their exposition to authorities operator or citizen. This element represents also the contact point where the autonomous component of the framework interacts with the humans. The notifications and alarms of the data analysis subsystem are critically elaborated by a Decision Maker component, which is in charge to assign to the received data the right weight and to estimate the risk of the emergency to be managed. The Decision Maker also filters the data to be presented to human operators and represent therefore the interface between the automatic system and the operators.

The human activity assumes an active role inside the OC. It has to be highlighted that all the operative decisions, actively impacting to the security of citizen, are validated by operators before their execution. Parts of the received information are also directly exposed to the citizen, with the right compromise between informing the population and avoiding panic and false alarm.

Data Exposure subsystem

This subsystem aims to provide users groups with a representation of the data coming from the system. This is accomplished by considering:

- information content;
- message type (e.g., alert, info);
- multimedia contents;
- users group;
- restrictions;
- broadcast mode, that can be *channel* (unicast, multicast) or *media* (web, sms, Social Network platform).

The data representation processed by the Data Exposure subsystem is sent through the adapters, which work according to the policy defined by the Decision Maker.

9.2.3 Wireless sensor networks

Sensors represent a key element for an efficient monitoring and a smart management of environment [123]. They provide an updated view of the events of interest on the field.

In particular, we are interested in Wireless Sensor Networks, that are more suitable for large areas monitoring instead of wired networks. This subsystem is composed of several heterogeneous WS sub-Networks (WSsNs) made of many sensors. The underlying idea is to consider a WSsN as a single cluster, and all the clusters interact in a distributed manner and then concentrate all the information toward a central point. The collected data are aggregated and made available to the smart platform through standard channels (like web services).

In particular we consider a tiered structure as depicted in Fig. 9-3:

- *Sensor Nodes*: are optimized for data acquisition (i.e., they require minimal maintenance and have low power consumption) using on-board transducers; they communicate with Gateways using short-range communications, either directly or through Routers.

- *Routers*: these elements provide high level connection among the nodes, forming the network infrastructure and extending the WSN range; Routers have cardinal importance in the WSNs and represent critical elements since they determine the network geometry and extension [124].

- *Gateway*: is a specific node with additional functions, interacting with the routers infrastructure and with the Gateway of other clusters. It represents the coordinator of the WSN. The Gateway is in charge to connect the WSN with the outside and to receive programming commands, like alarm thresholds and sampling interval values. It processes, stores, and periodically sends the field data to the application server using long-range communication channels. We consider that each Cluster (i.e., Gateway) can be connected to each other and even to the Data Gathering and Analysis system, according to the Internet of Things (IoT) paradigm, mainly in an autonomous mode [125]. This is of paramount importance to facilitate the coordination and collaboration of geographically sparse WSNs enabling a seamless and autonomous heterogeneous data exchange with the aim of improving the capabilities of capturing the physical status of the environment.

- *WSN Application Server*: it is in charge to convey all the data gathered by many heterogeneous WSNs and make them available to the framework. The WSN Application Server implementation can follow the *cloud* approach, by distributing both data and application, leading to a complete abstraction of the WSN. This server performs a space-temporal analysis of the data coming from the WSNs and through suitable models provides a general description of the environment conditions and the ongoing processes.

9.2.4 Social networks

Social Networks represent an innovative tool for the information gathering and broadcasting [126]. The proposed system takes advantage from the use of a specific platform, that allows the data collecting and the distribution of suitable contents according to the Decision Maker policy. The proposed platform has to validate, filter and correlate the data with the information coming from the official channels by exploit-

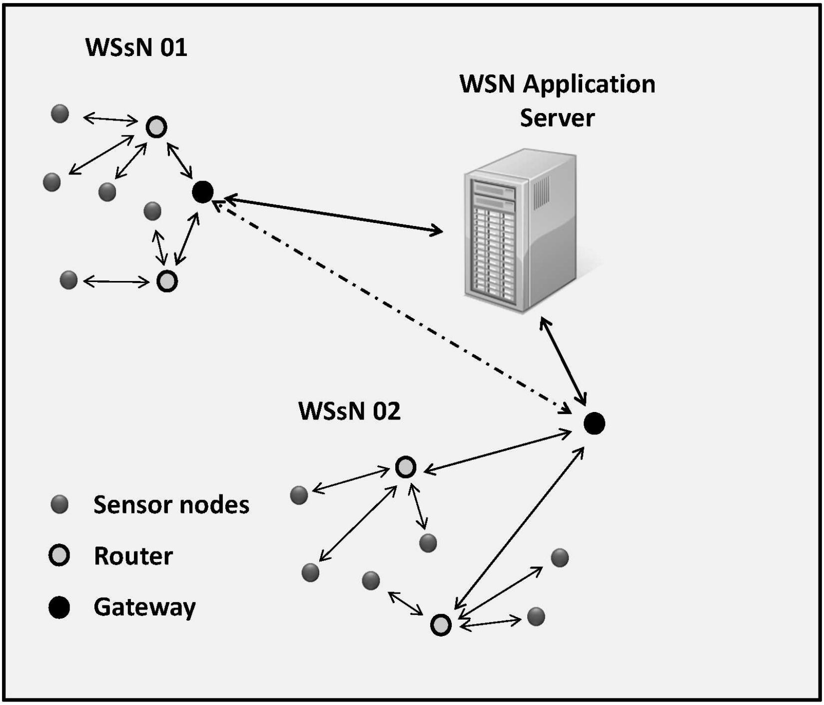

Figure 9-3: WSNs tiered structure.

ing algorithms and rules dynamically programmed. The interaction with the other elements of the framework allows to define efficient and pervasive communication methods toward the users.

The use of Social Networks from public Authorities and organization is surely increased in these years, but information is often fragmented and not easy to be accessed by the population. Social Networks are currently mostly used in unidirectional way for information campaign to the citizen, rather than for obtaining feedbacks and advices from the users. The proposed framework takes advantage by a strong integration with Social Networks, considered not only as publication method but also as information sources. The new on-line media are in fact used for several activities:

- inform citizen with prevention campaigns;
- obtain feedbacks form population and create a bidirectional communication channel;
- reach users with quick advices and short messages in case of emergency, on PC or smartphones;
- collect data analysing information and media spontaneously generated by the population with their post, tweets and statuses;
- integrate the data obtained from the WSNs, using users as *social sensors* widely distributed on the territory.

The wide diffusion of devices with always-on connection, such as smartphones and tablets, especially in the cities, allows the active participation of the population on discussion about topics of public interest. Managing this large amount of data in efficient way is the key for obtaining concrete benefits form the use of this new communication channels. The proposed platform is able to smartly monitor the users activity on Social Networks, identifying anomalies in the produced data flow and informing the OC about that. Half of the Social Networks users, access them also from mobile phone, increasing the rapidity of information diffusion between peoples involved in calamity and disasters. In many cases news are on-line disseminated by civil users quicker than by official portals. Social Networks can also be source of not exact information, the system is therefore able to filter and validate the data, verifying the actual diffusion and coherence within the Social Networks themselves and the other data obtained from WSNs. It is very important for Authorities to correctly tune the right compromise between accuracy of news and rapidity of alarms, before the use of information derived from Social Networks for official advices or intervention.

The effectiveness of information analysis is related with the capacity of filtering the large amount of data produced by users, basing on Keywords, Time, Localization, shared links and multimedia data. It has been demonstrated that the vocabulary used in crisis situations is characterized by a low variance: *tweets* and *statuses* tend to describe an event in similar ways, with reduction of entropy in the Social Network activity [127]. The system is able to continuously monitor the public data produced by the users, and to identify when a large part of the citizen focuses their attention on particular key topics reducing the average variance of the spontaneously updated content. This approach has been successfully followed in studies for disease diffusion analysis and epidemic intelligence, e.g., during the Swine Flu pandemic [128] [129], and can by generalized to various different cases for automatic emergency prevention and monitoring in a smart city environment. In the proposed framework social networks can be exploited in two different manners. The first, called *active*, implies the creation by the Authorities of suitable and official channels of communication in which citizens should act in a proactive manner. Citizens can follow or join Facebook, Twitter, Google+, etc., institutional accounts to post their textual or multimedia warnings, questions and observations or to interact with other people and obtain news and indications in real time. The second solution acts in a *controlled* mode and consists of a software tool that can be used to monitor not institutional resources. To reduce its charge of work this tool can be activated when the alert status is already started to obtain more precise information on ongoing events. For example, the tool could integrate a Facebook application doing research on the social network for specific contents. Indeed, Facebook provides ad-hoc Application Programming Interface (API) namely Graph API, that permits the access to Facebook data represented by the Graph. Through Graph API it is possible to search in the Facebook data (the Graph) different types of objects like public posts, people, events, groups, places, pictures that correspond to specific keywords. Searches can be related and limited to a geographic area. However, accessing user data introduce privacy issues. Hence, the use of Graph API is subordinate to a security mechanism that enables the access to data from an application only after obtaining a Access Token. Not all information are freely accessible. Each user can set different privacy polices and let open only some information or even none. So with basic permissions only public information are accessible. The information is produced in JSON format, that simplifies the Adapter implementation.

9.3 Application scenarios

One of the main characteristics of the proposed system is the flexibility. It can be applied to different scenarios, which are strongly dependent on context where the platform is used. Various types of emergencies and events can occur depending on local territorial topology, population density and type of environment, e.g., urban, sub-urban, industrial or rural. The platform subsystems must be suitably designed for the specific event/emergency to prevent or manage. In particular for what concerns the type of monitoring sensors and the used data analysis models. However, a general operating way of the proposed framework can be identified, as described in the following. Some details are reported for a specific use case as an example. In particular we refer to the hydrogeological risks (floods and landslides) that represent the major geological hazards whose frequency of occurrence is the highest including heartquake and volcanic eruption [130].

Each element of the system assumes a peculiar role in different phases of the emergency evolution, as represented in Fig. 9-4. The rows show the current status, which can be: *ordinary* when no emergency is present or foreseen; *pre-alter* in case something indicates the chance of an emergency occurrence; *alarm* if the possibility of upcoming disaster is concrete; *emergency* as soon as serious occurrence happens and demands immediate action and *restoring* when emergency is finished. The arrows show the activities of each of the three main elements of the proposed framework during an emergency evolution indicating the main types of exchanged information.

- *Ordinary.* A background monitoring activity is always on, collecting and analysing data from sensor networks, social networks and surveillance cameras deployed in the Smart City environment. In an ordinary situation, the Social Networks are used also for disseminating informative campaign and suggesting the correct procedures to adopt in case of emergency, with the sake of preparing the citizens. Sensor networks are placed in the territory to monitor specific physical parameters, that in the considered example can be the level of the water in a river or in a dam, the movement of the land, weather sensors, etc. Social Networks are analysed, for

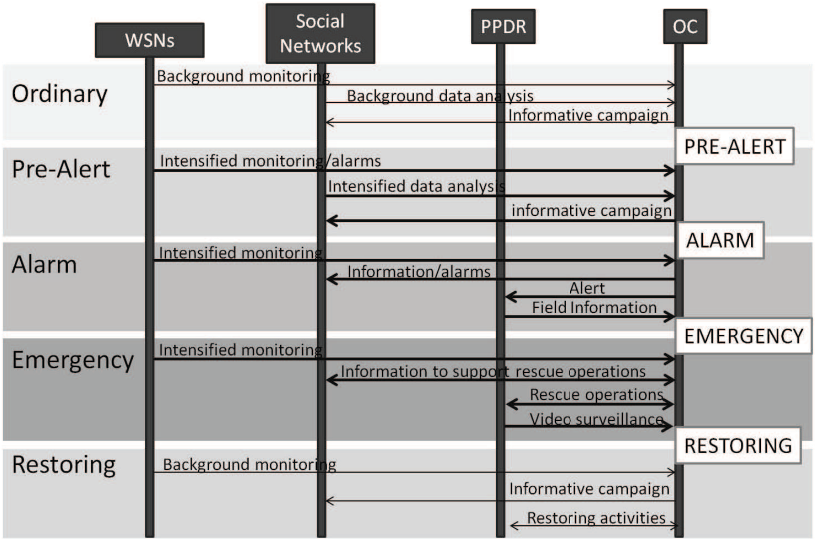

Figure 9-4: Phases of the emergency and involved System elements

example, by observing the number of occurrences of specific keywords (ex. flood, landslide,etc.) on a given time interval and in a restricted geographic area, associated to a peak in web traffic that denote the sudden focus of public on a specific object.

- *Pre-alert.* Anomalies in the social network data flow, and/or alarms from the analysis of the collected data, increase the alert in the OC, allowing the Authorities to actuate planned actions to prevent or limit problems that may arise. In addition in the *pre-alert* phase the OC requires to the WSNs and SNs an intensification of the monitoring activity and of the informative campaign toward population.

For example an exceptional torrential rain falls can attire the attention of the people that is reflected in their *tweets* and *statuses*. In addition to the most popular social networks, also weather-dedicated web-forum are analysed, where people keep up-to-date about weather information and data from their own home weather stations. It is very important to note that people involved in the weather event can be in areas where physical sensors are not present, providing therefore very useful information to the platform. The Data Analysis subsystem identifies the anomalies in the users information flow and cross check data with those received by WSNs. Data gathered from physical sensors are elaborated following specific models. As an example, rainfall thresholds determined by historical data can be used to verify if the condition that can determine a river flooding or a landslide is reached. Alarms can be generated directly by the WSNs that perform a pre-processing of the data. WSNs receive by the OC the alert threshold values for the measured parameters.

- *Alarm.* In case of a concrete chance of disaster PPDR forces are activated and a bidirectional information exchange between the Authorities and the population is open. Videos and area images become very important, moreover PPDR operators can reach areas not covered by fixed camera and transmit video data to Authorities using mobile advanced PMR devices. Monitoring still continues, but particular relevance is given to the countermeasures that can be put in place to limit the damages based on the forecasting models. The Data Analysis subsystem uses these models to determine the evolution of the situation and the most critical events that can happen. These models take into account studies on physical and social processes together with the territory conformation. They are used to determine the expected disasters that can be by the environment but also by the disruption of artificial structures. For example suitable models for floods and landslides, applied to a specific city layout, can estimate the effects on buildings, streets, cars etc. to detect the most critical areas.

- *Emergency.* When disaster occurs PPDR operators are involved in rescue operations. Important information can be derived from the experiences of citizen they share on the social media. It may help for psychological support/recovery, to identify and locate people in risk situations. Many Social Networks allow users to insert their position in the posted status. The localization can be very accurate if obtained by a GPS-equipped smartphone, but also rough positions autonomously calculated by system may represent vital information in some situations. The importance of the population positions knowledge is evident analysing the emergency managing of recent disasters, such as the Haiti earthquake. Volunteers of Boston Tufts University developed an open source software, Ushahidi [131], representing the location of injured people of over an on-line map, updated through the advisory received by phone calls, text messages and official press release. Collecting the position information directly from Social Networks the proposed system can easily be used also for an effective graphical representation of advices over a map. PPDR operators can exchange multimedia data, thus increasing the

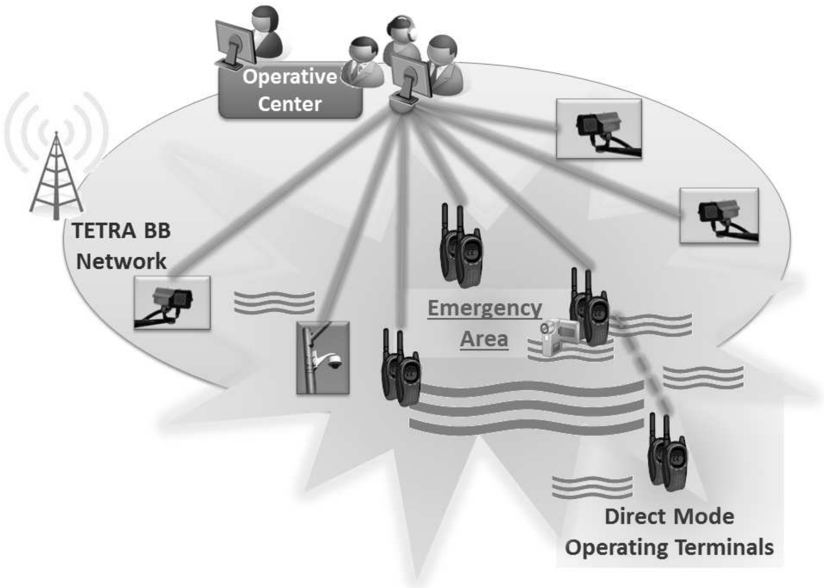

Figure 9-5: Use of the *TETRA BB* network in emergency managing.

effectiveness of rescue and first aid activities, as well as for the security of operators themselves. A broadband TETRA network (Fig. 9-5) allows the OC to be in touch with agents not only with PMR voice services, but also with advanced services such as bidirectional video communication and remote data access. The system supports also DMO for peer-to-peer communications between terminals also in case they are not under network coverage. This feature assumes particular importance in case the infrastructure networks have been damaged during the disaster. Different city organizations and authorities can share the same network infrastructure for realizing virtual network to be used for communication between their own OCs and PPDR operators.

- *Restoring.* When data coming from sensor networks and from operators on the field confirm the end of the emergency phase, there is a *restoring* phase in which social media can be used to give useful information to the citizens together with the official channels (i.e. TV, Radio). PPDR operators are involved in restoring the normal life conditions.

9.4 Further developments

This section deals with some research open topics that require additional efforts in order to further improve the performance of the proposed Emergency Management platform.

9.4.1 Professional broadband communication system

One of the topics that will need of in-depth analysis is the modality of integration and migration of current PMR systems to the new one that is able to join the features of professional networks with high performance offered by new communication technologies like LTE. This theme is gaining interest in the international community, so that, triggered by operator interests and users request, ETSI and 3GPP are working to the standardization of the third generation of TETRA, i.e., TETRA Broadband [132]. This new system will integrate services and performance required from professional user with LTE broadband services. New specific features will be added in future Releases of the LTE-3GPP standard to meet critical communications requirements. However, some critical issues have to be investigated, indeed LTE has been conceived as a public network that in its traditional operation does not takes into account some specific features of mission critical communications. Hence, it becomes important to understand how these characteristics of professional network will be implemented. The two main open issues are represented by the Group call and the Direct Mode of operation. For what concerns Group calls, LTE supports a broadcast capability called evolved multimedia broadcast and multicast service (eMBMS) that could be used to provide group communications to large numbers of first responders in a geographic region. However, it requires investigation, in particular for what concerns the QoS control. The Direct Mode, that gives the ability of two or more public safety devices to communicate directly without the use of network infrastructure, requires the development of new solutions and to support device-to device (D2D) communications (also called as Proximity-based services).

9.4.2 Social networks

One of the strengths of the proposed framework consists in the use of social sensing to gather information. However at the moment there are no solutions to aggregate in an effective and dynamic way the heterogeneous information present on social networks. The lack of a coordinated approach often implies that the available content is fragmented and difficult to identify. In addition, the ability of Social Networks to gather a huge amount of information rises two main data security issues that need careful investigation: data reliability and users privacy.

Social networks due to their open nature can become a source of unconfirmed or misinformation that could cause panic situations. Therefore, it is necessary to validate the information flowing on social networks in order to verify its reliability and exclude both false and malicious alarms. A validation can be performed by filtering and correlating the data with those collected from official channels, making use of algorithms, processes and rules dynamically configurable. Another viable approach that receives great attention is *social validation*. It is possible to exploit the self-correcting ability of information in social media: users validate or deny the information diffused by another user. In addition by monitoring keywords specific for an emergency situation, it is possible to verify if the number of users that participate in a hot discussion under exam from the tool rapidly decreases that can suggest that the emergency has subsided or it never existed. Preservation of privacy is another important issue that must be carefully investigated. The right of citizens of privacy should be guaranteed in any case, otherwise citizens might refrain from using the platform. The proposed platform has to be able to monitor available open source data published on different social media. The analyzed information is only that make open by the user that has already accepted privacy rules of the social network. Data are collected to create statistics devoting particular attention to sensible information. Data non necessary for the analysis are not stored. However, problems related privacy can rise, due to the identity of the user, its localization, its queries, the security of stored data [133].

9.4.3 Wireless sensor networks

One of the key concepts associated with the computing of the next generation will be the idea that the physical world can be modeled, monitored and represented virtually through computer. In this direction, one the most interesting idea is the evolution of the IoT vision, towards the new paradigm of the Self-Aware IoT for which autonomy is an imperative property. Therefore, research efforts are needed on new techniques or methodologies on how to adapt and tailor existing research on autonomic communications and computing to the specific characteristics and requirements of smart disaster management systems. In particular, it will be necessary to define specific platforms compliant with the new Self-Aware IoT paradigm to accomplish data sensing, distributed data computing and information delivering, in a secure, intelligent, pervasive and ubiquitous manner. The huge number of objects in a smart environment and their heterogeneity require the introduction of techniques that can automatically select the relevance of the objects for a given application. Hence, a cognitive and autonomic management of the objects could be useful in certain environment [125]. IoT exploits the IPv6 protocol to obtain the interaction between heterogeneous devices with consolidated mechanism like http. A complete integration requires the definition of an adaptivity sublayer that enables efficient M2M protocols and the interoperability with the most common used communication standards in WSNs (like IEEE802.15.4).

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