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Ph.D Thesis

Reception Performance Studies for the Evaluation and Improvement of the New Generation Terrestrial Television Systems

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Specially dedicated to my parents, Begoña and Jose.
Thanks for every moment in my life.
Love you.

Resumen

La industria de la televisión (TV) ha experimentado grandes cambios durante las últimas dos décadas. El hecho de que las expectativas de los espectadores son cada vez mayores (mayor calidad de imagen y más entornos de recepción) y el espectro disponible para los servicios de TV se ha reducido drásticamente debido al dividendo digital, ha provocado la aparición de nuevos desafíos a los que deben responder los nuevos sistemas de Televisión Digital Terrestre (TDT), que deben ser cada vez más robustos.

El primer intento de dar respuesta a estos nuevos requisitos es el estándar europeo de nueva generación DVB-T2, publicado en 2009. La publicación de un nuevo estándar de TDT lleva consigo el inicio de un proceso completo de evaluación de su rendimiento antes de su comercialización. Al inicio de esta tesis, este proceso estaba prácticamente terminado para la recepción fija y móvil de DVB-T2; sin embargo, los escenarios de interiores aún no se habían estudiado en detalle. Por esta razón, esta tesis ha completado la evaluación de rendimiento del estándar DVB-T2 mediante el análisis de su recepción en interiores. Además, se ha definido una nueva metodología de evaluación de rendimiento optimizada para este tipo de escenarios.

A pesar de que las tecnologías utilizadas en DVB-T2 eran lo más avanzado en su momento, el sistema se definió hace casi diez años y durante el desarrollo de la tesis han aparecido nuevas técnicas avanzadas, como por ejemplo, nuevos códigos de corrección de errores o la nueva técnica de multiplexación por división en capas (LDM). Al igual que en el caso de DVB-T2, aún no se ha estudiado en detalle el rendimiento de estas nuevas técnicas en entornos de interior. Por ello, esta tesis incluye el análisis de rendimiento de estas nuevas técnicas para determinar su idoneidad para mejorar el sistema de DVB-T2 en escenarios de interior. Además, se ha probado que los algoritmos tradicionales implementados en los receptores de TDT no están optimizados para la nueva situación en la que se consideran señales multicapa y escenarios móviles. Por esta razón, se han propuesto modificaciones a los algoritmos existentes para mejorar el rendimiento de la recepción de TDT.

El último intento de hacer frente a los altos requerimientos para los nuevos sistemas de TDT, ha sido el estándar americano de nueva generación ATSC 3.0, finalmente aprobado en 2016. Igual que en el resto de sistemas de TDT, también es necesario un proceso completo de evaluación del rendimiento del sistema. Por ello se han realizado algunos de los primeros estudios del rendimiento de ATSC 3.0 en base a simulaciones por ordenador y pruebas de

laboratorio en diferentes escenarios.

Como resultado de todos estos estudios, se ha probado la viabilidad de ofrecer servicios de alta calidad en escenarios muy desafiantes para los sistemas DVB-T2 y ATSC 3.0, así como para algunas nuevas técnicas desarrolladas entre ambos procesos de estandarización. Para ello, se han obtenido los valores de de relación señal a ruido umbral (SNR) en diferentes escenarios de recepción. Estos resultados muestran el rendimiento de los sistemas en diferentes condiciones, lo cual es útil para la planificación de redes de TDT.

Todo este trabajo de investigación se ha difundido en una conferencia nacional y varias conferencias y revistas internacionales.

Laburpena

Telebista industriak aldaketa nabari jasan ditu azkenengo bi hamarkadetan. Ikuslegoaren itxaropenak gero eta zorrotzagoak dira (irudi kalitate handiagoa eta harrera ingurune gehiago) eta espektro erabilgarria Telebista zerbitzuetarako nabari murriztu egin da dibidendu digitala dela eta. Faktore hauek erronka berrien agerpena eragin dute eta Telebista Digitalaren Sistema berriek (DTT) erantzuna eman behar diete gero eta sendoagoak izanik.

Eskakizun hauei aurre egiteko lehenengo ekinaldia DVB-T2 belaunaldi berriko europar estandarra da, 2009-an argitaratuta hain zuzen. DTT estandar berriaren argitalpenak bere errendimenduaren ebaluaketa prozesu osoa eragin zuen estandarraren merkaturatzea hasi baino lehen. Tesi honen hasieran, aipaturiko DVB-T2-ren ebaluaketa, bai harrera finkorako bai harrera mugikorrerako, ia amaituta zegoen. Hala ere, eraikinen barne dauden agertokiak artean ez ziren zehazki aztertu. Guzti hau dela eta, tesi honek DVB-T2 estandarraren errendimendu ebaluaketa osotu du eraikinen barne dauden agertokiak aztertuz. Gainera, mota honetako agertokien errendimendu ebaluaketa hobetua egiteko metodologia berria definitu egin da.

Nahiz eta DVB-T2 estandarrak, bere garaian zeuden teknologiarik aurreratuenak erabili, sistema orain dela 10 urte definitu zen eta tesi honen garapenaren zehar teknologia aurreratu berriak agertu egin dira: erroreak zuzentzeko kode berriak eta geruza zatituen multiplexazioaren teknika berria (LDM) hain zuzen. DVB-T2-ren kasuaren antzera, teknika berri hauen errendimendua ez da zehazki aztertu eraikin barneko agertokietan. Hori dela eta, tesi honek teknika berri hauen errendimenduaren analisia barneratzen du, beraien egokitasuna determinatu ahal izateko DVB-T2 sistema eraikinen barneko agertokietan hobetzeko. Gainera, tesi honetan, telebista digitaleko hartzaileetan inplementaturiko ohiko algoritmoak, geruza anitzeko seinaletarako eta agertoki mugikorretarako hobetu daitezkeela frogatu egin da. Hau dela eta, ohiko algoritmoen zenbait aldaketa proposatu egin dira DTT-ren harreraren errendimendua hobetzeko.

Telebista Digitaleko sistema berrien eskakizun zorrotzei aurre egiteko azken ekinaldia, belaunaldi berriko ATSC 3.0 estandar amerikarra izan da, 2016. urtean aprobatuta. Telebista Digitaleko beste sistema guztien moduan, sistema honen errendimenduaren ebaluaketa prozesu osoa beharrezkoa da. Hori dela eta, ordenagailu simulazio eta agertoki desberdinetan egindako laborategi frogetan oinarritutako ATSC 3.0 errendimendua ebaluatzen duten lehenengo ikerketak egin dira.

Lehen aipaturiko ikerketa guztien ondorioz, agertoki erronkarietan kalitate handiko zerbitzuak eskaintzeko bideragarritasuna frogatu egin da, DVB-T2 eta ATSC 3.0 sistementzako eta garatu egin diren zenbait teknika berrientzako. Horretarako, seinale eta zarataren arteko erlazioak (SNR) lortu egin dira harrera agertoki desberdinetarako. Emaitza hauek aipaturiko sistemen errendimendua baldintza desberdinetan agerian utzi egiten dute eta guztiz beharrezkoak eta erabilgarriak dira Telebista Digitaleko sareen plangintzarako.

Tesiaren ikerketa lan osoa estatu mailako konferentzia batean eta zenbait nazioarteko konferentzia eta aldizkarietan argitaratu egin da.

Summary

The television (TV) broadcasting industry has experienced dramatic changes during the last two decades. The continuously increasing expectations of viewers and the reduced available spectrum for TV services due to the digital dividend issue have recently caused the emergence of new challenges. As a consequence, more robust Digital Terrestrial Television (DTT) systems are needed.

The first attempt to fulfill these high requirements was the European new generation DVB-T2 standard, published in 2009. The publication of a new DTT standard means the beginning of a whole performance evaluation process. At the beginning of this thesis, this process was almost finished for fixed and mobile reception. However, indoor scenarios had not been studied in detail. For this reason, this thesis has completed the performance evaluation of the DVB-T2 standard by means of analyzing the system indoor reception. In addition, a new performance evaluation methodology optimized for this kind of scenario has been defined.

In spite of the cutting-edge technologies used in DVB-T2, the system was defined almost ten years ago and new advanced techniques, such as new error correction codes or the novel Layered Division Multiplexing (LDM) technique have appeared during the thesis development. Similarly to DVB-T2, these new techniques performance has not been widely studied in indoor environments yet. Consequently, this thesis includes the performance analysis of these new techniques to determine their suitability to improve the DVB-T2 system performance in indoor scenarios. Moreover, it has been proved that traditional DTT receivers' algorithms are not optimized to the new situation where multilayer signals and mobile scenarios are considered. For this reason, some modifications to the receivers' algorithms have been proposed in order to improve the reception performance.

The last attempt to cope with the high desired requirements was the American new generation ATSC 3.0 standard, finally approved in 2016. As a complete performance evaluation process of the system is also necessary in this case, some computer simulations and laboratory measurements have been conducted so as to test the performance of this new DTT system in different scenarios.

As a result of all these studies, the feasibility to deliver high quality services in very challenging scenarios has been tested for DVB-T2 and ATSC 3.0

systems as well as some new techniques studied between both systems standardization processes. For this purpose, Signal to Noise Ratio (SNR) threshold values have been obtained under different reception scenarios features. These results show the systems performance under different conditions, which is helpful for broadcasters planning purposes.

All this research work has been disseminated in one national conference and several international conferences and journals.

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CHAPTER I: Introduction

In this chapter, a brief description of the evolution of the most important terrestrial broadcasting systems until the current situation is included. The reception scenarios and expected video quality have increased with the time resulting in high demanding requirements in terms of robustness and capacity for the next generation broadcasting systems. According to this context, the motivation and main objectives of this thesis are also defined. Finally, the thesis organization is also presented.

1. Research Context

It is widely agreed that television broadcasting is a fundamental element in the life of most of the people as it can be considered one of the most cost-effective technologies for informing, educating and offering entertainment all over the world, especially in the developed countries where 98% of the households had at least one TV set since 2010 according to the International Telecommunications Union (ITU). Besides, although new ways of delivering video and audio contents have appeared with the time, live TV viewing still suppose almost the 60% of the viewers according to Figure 1 [1]. Nowadays, it is hard to imagine our world without TV. In fact, young people find it difficult to understand a TV system with just one or two programs which ended their emissions at certain time every day and from that moment a test card was shown [2].

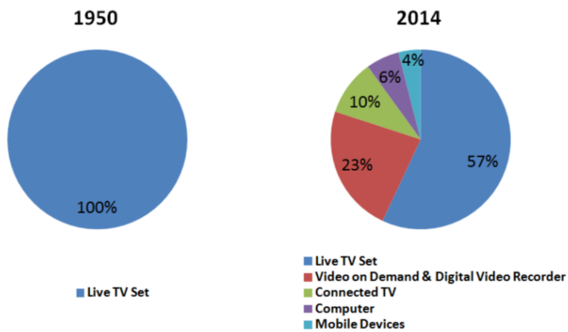


Figure 1. TV Contents Viewers

TV systems have highly evolved in their more than 100 years of history. In the beginning, the contents changed adapting to the viewers changing demands. However, the largest change was in the technical aspect. When the TV broadcasting started in the beginning of the last century, it was entirely based on analog terrestrial transmissions of sound and black and white images. The first regular television broadcasting services began in England by the British Broadcasting Corporation (BBC) in 1936. However, the color images were soon introduced in 1953 as well as new transmission methodologies such as satellite or cable and also digital systems. Indeed, prior to the development of the fully digital TV systems, there were several attempts to improve the existing analog systems, such as Eureka EU 95 Project [3] or PALplus [4] in Europe, Japan Broadcasting Corporation HDTV project in Japan [5], and Advanced Compatible Television in the United States [6]. However, they were still extremely inefficient as a complete radiofrequency (RF) channel was needed to transmit just one TV service. For this reason, complete digital systems were needed. In this context, the 21st century brought a substantial change, digitalizing both the contents and the transmission systems.

1.1 Birth of DTT Systems

As their analog predecessors, the first generation of digital television standards were also developed in parallel in different parts of the world due to a mixture of technical and geopolitical reasons. Although all the DTT systems had similar requirements, they were quite different in terms of technical features [7] [8] [9].

Nowadays, several regions of the world are in different stages of the television broadcasting systems digitalization process, implementing different broadcasting standards [10], as it is shown in Figure 2. The first generation DTT systems, whose main requirements are fixed reception and Standard Definition (SD) video quality, are described in Recommendation ITU-R BT.1306 [11].

Current DTT systems can be technologically classified in four different categories:

- Digital Video Broadcasting (DVB), which has been adopted in Europe, Australia and New Zealand.
- Advanced Television System Committee (ATSC), adopted by: The United States of America (USA), Canada, Mexico, South Korea, Dominican Republic and Honduras.
- Integrated Services Digital Broadcasting (ISDB), adopted in Japan and the Philippines. ISDB-T International is an adaptation of this standard using H.264/MPEG-4 AVC and has been adopted in most of South America and Portuguese-speaking African countries.
- Digital Terrestrial Multimedia Broadcasting (DTMB) adopted in the People's Republic of China, including Hong Kong and Macau.

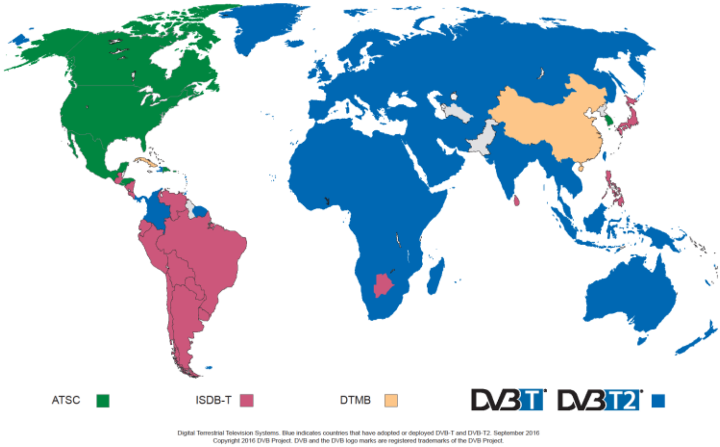


Figure 2. Worldwide DTT Systems [Source: dvb.org, 2016]

The first DTT standard was DVB-T which was standardized in March 1994 as the European Telecommunications Standards Institute (ETSI) EN 300 744 [12] [13]. It was firstly broadcasted in the United Kingdom (UK) in 1997 but it is nowadays the most popular DTT standard widely adopted by more than 60 countries in Europe, Asia and the Middle East. The adopted DVB-T configurations, such as for example in Spain, France and Germany, allow high bitrates up to 19.9, 24.8 and 13.3 Mbps (in 8 MHz bandwidth) with a required rooftop SNR of about 17.3, 18.5 and 15.3 dB, respectively.

Just one year later, in September 1995, the North American standard (document number A/53) [14] [15] was approved and it was launched in USA in November 1998. ATSC A/53 is still in use in America with a bitrate of about 19.9 Mbps (in 6 MHz bandwidth) and a required SNR of around 15 dB in rooftop reception.

Once in the 21st century, the specification for the Japanese DTT system, ISDB-T, was approved by Association of Radio Industries and Businesses (ARIB) in 2000 [11] and started to be commercially used in Japan in December 2003 [16]. Although ISDB-T can be considered an evolution of DVB-T because of their similar technical features and system parameters [17], it is also considered one of the first attempts for the simultaneous delivery of fixed and mobile services. For South America, a new version of the ISDB-T standard was developed, named ISDB-T International or ISDB-Tb (Brazilian version) [18]. In 2010, an interactive evolution named ISDB-Tmm [19] was also developed.

In China, the development of the digital television system started on 1996, but it was not until 2006 when the Government published the first DTT standard known as Digital Terrestrial Broadcast (DTMB) [20] [21] [22].

Summarizing, all the first generation DTT systems are flexible by choosing the appropriate system capacity (bitrate) and robustness (SNR) and determining the power of the transmitter for a given coverage area. Their maximum bitrate is about 30 Mbps, which was enough for HD services delivery at that moment. Besides, the most of them only target fixed reception; that is why, in general, the required SNR threshold is not low.

1.2 New Broadcasting Trends

At their time, the first generation broadcasting systems were the most innovative DTT standards with the main cutting-edge technologies included in their definitions. These systems have been widely adopted all over the world, providing high quality and cost effective services. However, new challenges have recently emerged, due to the continuously increasing expectations of viewers and the reduced available TV spectrum due to the digital dividend [23].

On the one hand, the importance of mobility in broadcasting systems and the desire for mobile and portable devices to be capable of correctly working are totally clear. This fact requires more robust DTT systems in order to be able to received TV services in very challenging scenarios [24]. Moreover, the deploying of higher resolution systems (from HD to UHD quality), which get a closer representation of reality, is highly desirable but means more necessary capacity. On the other hand, the efficiency of the existing standards was not enough for the growing spectrum scarcity. Finally, it must be taken into account that most of them were defined more than 10 years ago when processing capabilities were lower.

1.2.1 New Scenarios

By the early nineties, it was observed that the consumer habits were evolving very fast toward handheld and mobile devices due to the high popularity of mobile phones. The TV digitalization brought the very first steps for mobile TV. However, the first generation standards performance in mobility was not satisfactory because of the very challenging receiving conditions for handheld/mobile terminals on urban and indoor environments. Due to the limits of some of first generation broadcasting systems in mobility, new standards were defined providing broadcasting services for handheld receiving devices and using a dedicated transmission standard. In general, they take into account the special requirements of the handheld devices, such as the receiver small size and lightweight, and low battery [25]. These properties implicate severe restrictions, such as low power consumption and impossibility of pointing at the transmitter if the reception terminal is in motion.

One of the first attempts at mobile broadcasting was the European Handheld DVB-H. This is a specific standard for handheld terminals adopted as ETSI standard EN 302 304 in November 2004 [26] so as to solve the limits of DVB-T in mobility. One year later, in December 2005, T-DMB was launched in Korea as ETSI standard TS 102 427 and TS 102 428. By the end of 2009, T-DBM service had become the world's largest terrestrial broadcast mobile service with more than 20 million customers [27]. In North America, ATSC Mobile/Handheld, (document number A/153) [28] was launched in April 2009. It uses the physical layer of ATSC but with higher protection for being correctly received in mobile challenging scenarios. Regarding the Japanese standard, ISDB-T directly enables mobile reception.

The international market prospective of mobile broadcasting services is variable. They are successfully implemented in Korea and Japan using the T-DMB and ISDB-T respectively. However, in Europe and North America, their use was limited due to the limited market in that moment and the need to use a dedicated standard. However, the situation is currently totally different, and mobile reception is considered one of the most important reception scenarios. Besides, a practical solution could be to design more protected mobile services that can be understood as an additional capability of an already existing legacy system in a simultaneous delivery. This fact provides the broadcasters the advantage to maintain or even increase the profits without the need to change the network.

1.2.2 More Video Quality

The demand for high quality services is to a great extent driven by the growing number of households with flat panel displays, able to present not only SD (Recommendation BT.601 [29]), but also HD (Recommendation BT.709 [30]) and, more recently, UHD (Recommendation BT.2020 [31]). Consequently, broadcasters in many countries consider the distribution of HDTV or even UHDTV as essential for an enhanced quality for viewers. The tendency is to increase the video quality and limit the number of broadcasted SD services. However, some very robust SD programs are still desirable to be delivered in very challenging scenarios where the available bitrate is really limited.

Higher resolution and higher quality mean higher necessary bitrate. However, existing DTT systems cannot directly cope with the high bitrate increment needed for HD and especially UHD services. For this reason, video compression algorithms are needed to reduce the necessary bitrate while maintaining the same high video quality. Table 1 includes the different standards considered in DTT systems depending on the considered video quality.

Table 1. Main Video compression algorithms in DTT systems

Replacement Year	Quality	Standard	Improvement over the previous
--	SDTV	Moving Picture Experts Group (MPEG)-2 [32]	--
2003	HDTV	MPEG-4 [33]	50% bandwidth reduction [34]
2013	UHDTV	High Efficiency Video Coding (HEVC) [35]	50% (HD) or 60% (UHD) bandwidth reduction

In 2014, a scalable version is defined, named Scalable High Efficiency Video Coding (SHVC) and based on the division in two sub-bitstreams or scalable video coding layers [36]. The base layer (BL) must be HEVC compliant, while the enhancement layer (EL) can improve the video quality using one or a combination of six possible scalabilities: spatial, temporal, quality hybrid codec, bit depth and color gamut scalability. The main advantage of using SHVC is the bandwidth reduction (See Figure 3). By this way, more programs can be delivered or higher quality can be offered. Besides, less network congestion and reduced storage sizes are achieved. However, users that only need the EL increases the necessary bandwidth and the decoding complexity, as it can also be seen in Figure 3. In average, the bandwidth requirement of video streams reduces about an extra 30% for the EL if the BL quality is not changed [37].

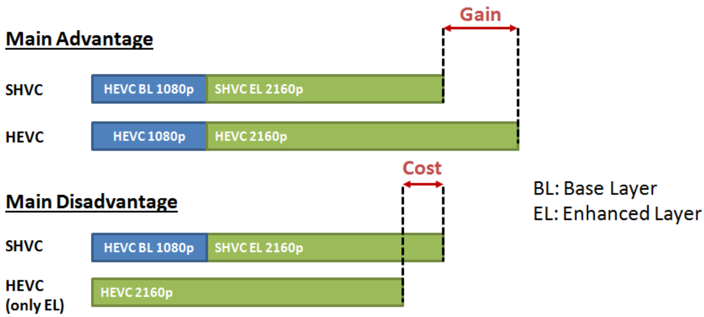


Figure 3. Main advantages and disadvantages of SHVC

Summarizing, video compression is necessary in order to be able to reduce the needed bitrate for video contents transmission. Raw video contents require very high bitrate, but with good compressing techniques, the required bitrate for the same video quality can be highly reduced. MPEG-4 standard was demonstrated to be efficient enough for HDTV transmissions but when UHDTV appeared, the HEVC next generation video coding standard was needed.

1.2.3 Spectrum Regulation

Another problem of the broadcast industry is the scarcity and high cost of the radiofrequency spectrum. The regulation establishes the allocation of some parts of the spectrum to specific services, such as mobile services, radars or broadcasting services. Due to the high problematic, several international bodies are in charge of the worldwide spectrum allocation, such as the ITU or the European Commission (EC).

The increment in about 60-75% in the transmission capacity after the digitalization process in comparison with previous analog TV systems meant that only about the 30% of the previous spectrum was needed to accommodate the existing services, and therefore, about the 70% of the previously occupied channels could be freed up to different services, especially mobile services [38], in a fact known as digital dividend [39]. This fact started in 2000 resulting in allocation of mobile services in the upper part of Ultra High Frequency (UHF) band (694-862 MHz). It seemed that this trend was going to be maintained during the following years, and consequently, the available spectrum for broadcasting services was going to decrease more and more. However, the situation has become stagnant in 2014 [40], when it was agreed that there would be no change to the allocation in the 470-694 MHz band. Moreover, the use of the entire UHF band is going to be reviewed in 2023. Despite this, the first generation DTT systems are not efficient enough to cope with the spectrum reduction and more efficient systems are needed.

1.3 New Generation DTT Systems

The new challenges that have recently emerged for the DTT systems cannot be correctly fulfilled with the first generation of DTT systems. For this reason, a new generation of broadcasting standards came to light with the aim of applying new processing techniques to achieve higher capacity, higher robustness and better power efficiency in comparison to the existing ones. For example, in England, capacity increased from 24 Mbps with DVB-T to 40 Mbps with DVB-T2, while transmitter power and coverage area remained the same. In addition, all of them include features to target any kind of reception scenario being able to offer simultaneously the highly protected mobile services and high capacity stationary programs over the same radiofrequency channel. By this way, the new desired mobile services can be transmitted over the same deployed network.

The Recommendation ITU-R BT.1877 [41] describes the new generation DTT systems. Chinese and Japanese standards have been set aside as they have not evolved as fast and clear as the American and European standards and no new generation systems have been developed. In fact, according to Figure 4 [42], they are the least used systems by far.

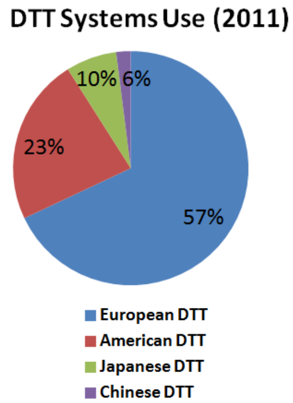


Figure 4. Worldwide DTT Systems Use

DVB-T2 was standardized as ETSI EN 302 755 in September 2009 [43] with the main objective of increase the maximum throughput up to around 50 Mbps (in an 8 MHz bandwidth) to be able to deliver higher quality services. Based on, but not compatible with its preceding standard DVB-T, DVB-T2 includes some of the most advanced techniques increasing the spectral efficiency by more than 30 percent. Additionally, a new profile named T2-Lite, which targets mobile reception, appeared in 2012. Basically, it is based on adapting some aspects of DVB-T2 for non static services with lower capacity resulting in cheaper, smaller and more efficient receivers. For this reason, T2-Lite can be considered as a patch in a system which primarily targets stationary reception.

During the thesis development, the document entitled "Call for Proposals for ATSC 3.0 Physical Layer. A Terrestrial Broadcast Standard" [44], established that a new next generation DTT standard must provide improvements in performance, functionality and efficiency. There were three main technical challenges to overcome: the provision of tools for a flexible and robust use of the spectrum, the use of the latest technology improvements to increase the spectrum efficiency and the maximum system capacity and the increment of the system robustness to improve the portable, mobile and indoor reception.

ATSC has just developed a new generation broadcasting standard, named ATSC 3.0 (document A/322) [45], with similar technical features to DVB-T2 but with some novelties. This new standard gives an answer to all the requirements for a next generation DTT system, presenting an important quality leap with a physical layer defined as a flexible, robust and efficient next generation tool for delivering high quality services (up to around 60 Mbps in a 6 MHz bandwidth) to fixed, portable, indoor and mobile receivers.

In addition, DVB has recently started to study the WideBand reuse-1 (WiB) concept [46]. This idea was described in the International Broadcasting Convention (IBC) in September 2016 in order to improve DVB-T2 system in terms of efficiency in the network planning.

1.4 Framework of the thesis research

Taking everything into account, Figure 5 shows the timeline related to the main worldwide DTT systems. First generation of DTT systems appeared in the middle of 90's widely improving the existing analog systems. Due to the differences in each market, four different systems were developed: DVB-T in Europe, ATSC A/53 in North America, ISDB-T in Japan and, more recently, DTMB in China.

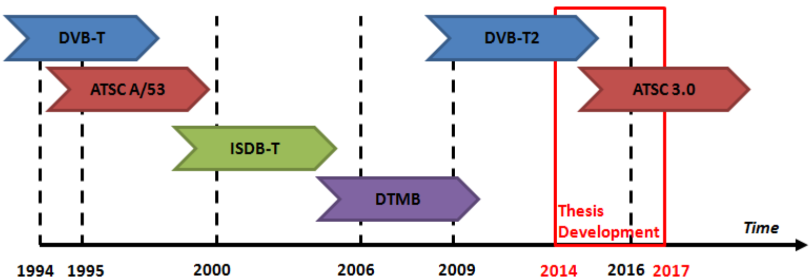


Figure 5. Thesis research Framework

However, although these systems were very efficient, they could not cope with the new requirements that have appeared with the time. At the beginning of the new century, viewers started to demand more video quality and more reception scenarios (mobile TV). In addition, the boom of the mobile communications caused the meant of more efficient DTT systems. Moreover, at the same time more efficient techniques appeared with the aim of improving the previous DTT systems. For this reason, DVB-T2 was developed in Europe and, more recently, ATSC 3.0 has been defined in USA.

In all the cases, some research time is needed before commercially launching the systems so as to determine the different configuration options performance. By this way, the most efficient configurations are considered for each reception scenario.

Figure 5 also includes the time period of the current research. The research context starts at the final steps of the DVB-T2 performance analysis. The research continues with the test and evaluation of new techniques that have appeared since the DVB-T2 standardization process up to the definition and first performance evaluation of the ATSC 3.0.

2. Motivation of this thesis

After explaining the current situation of the DTT systems, it is clear that the first generation cannot cope with the increasing need of higher quality services in more challenging scenarios demanded by current TV consumers. Additionally, the spectrum reduction imposed by the spectrum regulatory bodies is another disadvantage for the first generation DTT systems. However, new advanced techniques have appeared in the last decade to significantly improve the spectrum efficiency, system robustness and system capacity. As a result, a new generation of broadcasting systems that is supposed to deal with the current high viewers demands has started to appear. More specifically, the European DVB-T has evolved into the new generation DVB-T2 standard.

Any new system must be clearly evaluated before its commercial launching. For this purpose, a complete performance evaluation process to test its feasibility to fulfill its entire requirements is usually performed. This process includes computer simulations, laboratory measurements and field trials to check the system performance under different conditions.

The initial main motivation of this thesis is to complete the performance evaluation studies to check the feasibility of new generation DVB-T2 system to cope with the high quality contents delivery in challenging scenarios in a spectrum efficient way.

Furthermore, during the thesis development, a call for proposal for a new generation DTT in USA meant the appearance of new more efficient technologies, which were supposed to improve the DVB-T2 standard, motivating new evaluation research work to test their suitability for a new generation DTT system improving the existing ones. Besides, the uncompleted adaptation of traditional DTT receivers' to the current requirements has also motivated the study of new algorithms to improve the reception performance in the required challenging scenarios.

Finally, the thesis development also coincided with the definition of the new generation ATSC 3.0 standard in USA, which shares some technical features with DVB-T2. This fact motivated some research work in the first steps of the evaluation process testing some ATSC 3.0 technical features performance in order to test their suitability in terms of capacity, robustness and efficiency.

3. Objectives

The thesis aims to contribute with performance tests and research studies to evaluate and improve the most recent DTT systems. At the beginning of the thesis development, the most recent standard was the new generation DTT system DVB-T2. However, during the thesis development, new techniques to be applied in DTT systems appeared improving their performance and spectrum efficiency. Finally, making use of these new techniques, a new generation DTT standard, named ATSC 3.0, was standardized.

For this reason, the main objective can be further described by means of specific partial targets that can be grouped into three sequential different areas:

DVB-T2 Indoor Studies

The objectives about DVB-T2 indoor studies are focused on completing the necessary DVB-T2 performance evaluation:

- Study and performance evaluation of DVB-T2 fixed and portable indoor reception by means of laboratory measurements under different standardized channel models. The most appropriate channel model for indoor reception should be also studied.
- Study and evaluation of the DVB-T2 feasibility to offer indoor reception by means of coverage studies with field trials. The most appropriate DVB-T2 configuration features should be also determined.
- Study and performance evaluation of DVB-T2 fixed and portable indoor reception by means of field trials.
- Definition of a good methodology for performance analysis in indoor scenarios. The study of the time and location variability of real indoor scenarios is needed in order to obtain performance degradations that could be applied in any other indoor studies.

Studies of New techniques for the next generation DTT systems

Four objectives have been defined related to some of the new techniques to be possibly applied to a new next generation DTT system physical layer so as to improve its performance.

- Check the improvement in indoor scenarios in terms of robustness of the recently designed Quasi-Cyclic (QC) LDPC codes by means of field trials.
- Analyze the suitability of new Layered Division Multiplexing (LDM) technique to offer indoor services in an efficient way. The performance evaluation for the delivery of HD and UHD services should be also performed.
- Definition and evaluation of new decoding algorithms for improving multilayer signals reception performance.
- Definition and evaluation of ICI power estimators to implement in a DTT receiver improving portable, indoor and mobile reception.

ATSC 3.0 Studies

The objectives in the ATSC 3.0 research work are based on the completion of the first phase of the system performance evaluation:

- Development of an ATSC 3.0 emulation platform to create a basic tool to analyze the standard by means of computer simulations.
- Study and evaluation of the ATSC 3.0 BICM theoretical spectral efficiency and robustness.
- Study and performance evaluation of the ATSC 3.0 interleaving parameters by means of computer simulations.
- Study and evaluation of the ATSC 3.0 LDM parameters by means of computer simulations and laboratory measurements.
- Performance comparison of different ways to provide simultaneous services in ATSC 3.0 (TDM/FDM and LDM) with different decoding algorithms with computer simulations.

4. Thesis Organization

This thesis is organized in 5 different chapters. There is one introductory chapter to show the context of the present thesis. In addition, three main chapters dedicated to the three different DTT temporal contexts studied in this thesis are included. Each one of these chapters includes a state of the art as well as all the studies carried out organized according to the related dissemination. Finally, one conclusive chapter with the main contributions and some possible additional research related to the topic are included.

Chapter I. The context of the research is presented by means of a description of the evolution of the digital broadcasting systems until the most recent DTT systems (DVB-T2 and ATSC 3.0) appearance. In this context, the main motivations for this research work and the related objectives are presented.

Chapter II. In this part, the DVB-T2 performance studies are widely analyzed based on existing bibliography. In addition, new indoor studies have been carried out by means of simulations, laboratory measurements and field trials, as this target scenario has not been previously analyzed in detail.

Chapter III. This chapter introduces and studies some new techniques to improve the existing DTT systems performance and efficiency. More efficient code-rates and new multiplexing techniques have been analyzed. Moreover, new algorithms optimized for mobile and multilayer signals reception have been tested over a DVB-T2 system platform evaluating their performance gain.

Chapter IV. In this part, the ATSC 3.0 performance is widely analyzed, including fixed, indoor and mobile reception. Studies with simulations are performed related to the system interleavers and multiplexing techniques. Furthermore, some research related to LDM is also carried out by means of computer simulations and laboratory measurements.

Chapter V. In this final chapter, the main contributions of this thesis are summarized. The disseminating results of this thesis are also presented. Moreover, the identified future research topics as a follow up of this dissertation are also highlighted.

CHAPTER II: DVB-T2 Indoor Studies

This chapter surveys the previous studies for DVB-T2 standard available at the moment of the writing. As the main gap in the research work is about indoor reception performance, the three steps of a performance evaluation process over this scenario (computer simulations, laboratory measurements and field trials) are carried out in order to complete the existing DVB-T2 performance information.

1. Introduction

DVB-T2 was first standardized in September 2009 [47] with the aim of updating the way DTT services were broadcasted at that time. In other words, its main objective was to improve the existing DVB-T standard, which had appeared in 1993. In 16 years, new algorithms and signal processing techniques had been developed, while others already discovered a long time ago, became feasible with improvement of the related consumer electronics. As a result, the performance of this new generation broadcasting system highly improved the DVB-T system [23] [48] [49] and it was even very close to the Shannon limit [50] [51]. This gain is thanks to enhanced Orthogonal Frequency Division Multiplexing (OFDM) transmission, flexible frame structure, Low Density Parity Check (LDPC) [52] and Bose, Ray-Chaudhuri and Hocquenghem (BCH) codes, bit-interleaved coded modulation (BICM) with iterative decoding… What is more, a new standard version was published in April 2012 with a new profile, named T2-Lite, intended to broadcast mobile services.

The requirements for DVB-T2 standard were collected in DVB Document A114, Commercial Requirement for DVB-T2, released in April 2007 [53]. Twenty one requirements were defined grouped in several categories, such as transmission and receiving conditions, efficiency or robustness. All these requirements follow the current viewers' requirements asking for higher quality and target scenarios. The most prominent requirements for this thesis and their meaning are:

- DVB-T2 should increase the capacity of DVB-T in at least 30 % considering similar reception conditions. By this way, the service quality can be increased offering HD services instead of only SD ones.
- Each transmitted service should be configurable independently to provide varying degrees of protection and robustness. This fact refers to the possibility of simultaneously delivering different kind of services with different target scenarios and, consequently, different configuration parameters (specific capacity and robustness levels). For this purpose, the main novelty was the definition of Physical Layer Pipes (PLP) [54] [55] [56], allowing specific trade-off of capacity and robustness services with different coding and modulation schemes. Besides, the Future Extension Frames (FEFs) have been also included in order to transmit any other kind of

information inside the DVB-T2 data. These are the first European attempts to deliver simultaneously mobile and stationary services based on the Time Division Multiplexing (TDM)/ Frequency Division Multiplexing (FDM).

- DVB-T2 should be designed for stationary reception but it should be possible to design DVB-T2 networks for all kind of receiving conditions in fixed, mobile and indoor scenarios. On the one hand, fixed reception refers to the optimal situation in which the receiver is a roof-top antenna, so there is usually Line-of-Sight (LOS) with the transmitter. On the other hand, mobile reception refers to the situation in which the receiver is moving at any speed (from pedestrian to very high speed) in an outdoor environment. Finally, indoor reception refers to any situation that happens indoors both when the receiver is static or moving at a pedestrian speed.

Any new standard must be clearly tested before its commercial launching. In order to test the feasibility of this standard to fulfill all its requirements, a performance analysis is needed. The most common feature to evaluate a system performance is the Signal to Noise Ratio (SNR) threshold value under different reception scenarios conditions.

In the case of DVB-T2 standard, it was firstly tested with computer simulations using the Common Software Platform (CSP) [57] [58], which is a DVB-T2 emulation platform developed inside the DVB consortium but finally made public. The purpose of these first tests was to determine the system performance under ideal conditions by means of some common Validation & Verification (V&V) trials.

Next, and once some hardware prototypes were implemented, laboratory measurements are usually conducted so as to test the system performance under different reception conditions. These performance results are much closer to the real world as hardware (HW) equipment is involved in the performance evaluation. Finally, a series of field trials are usually needed. They are characterized by using HW equipment and antennas so as to analyze the feasibility and performance of a system under real conditions.

By the moment of the thesis writing, the DVB-T2 performance had been widely studied in fixed and even mobile scenarios, showing the feasibility of the system in this kind of scenarios. However, indoor reception, which was one of the target scenarios for DVB-T2, was still poorly studied, especially with hardware equipment. For this reason, in this thesis some laboratory measurements and field trials have been performed so as to test the DVB-T2 performance in indoor scenarios and check the feasibility of DVB-T2 to offer broadcasting services indoors.

Taking everything into consideration, this chapter includes a detailed state-of-the-art including the most prominent DVB-T2 performance studies. In addition, new research work related to the DVB-T2 performance analysis in indoor scenarios is also included by means of three different studies.

2. Previous Studies

As any new standard, DVB-T2 must be clearly tested before its commercial launching. For this purpose, computer simulations, laboratory measurements and, finally, field trials are needed.

2.1 Computer Simulations

The DVB-T2 theoretical system performance by means of computer simulations is provided along the DVB-T2 implementation guidelines document [59], which contains the first values of minimum required SNR for correct reception. The values appear in chapter 14 of the guidelines, entitled "Performance" and dedicated to the qualitative analysis of the DVB-T2 system performance based on simulations carried out with the publicly available CSP DVB-T2 software (SW) platform [57] [58]. These theoretical simulations were performed for fixed reception, which was the major aim of this standard. In this case, the receiver usually includes a roof-top antenna. This is the most common situation in many countries, such as in Spain, where the usually high gain directional receiving antenna is located on the building roof to deliver TV services to all the residents. In this case, there is usually line-of-sight with the transmitter and the receiver stays static. Although the Ricean channel model (F1) is the most similar to the real roof-top reception, there is another channel usually considered as reference (Additive White Gaussian Noise, AWGN). Similarly, more demanding scenarios are also analyzed (Rayleigh, P1 and 0 dB Echo). Moreover, the robustness of P1 synchronization symbol and the behavior of the P2 signaling symbols are also studied.

In these simulations, the minimum SNR threshold results gathered in [59] are given at a Bit Error Rate (BER) of 10^{-7} after LDPC decoding, which approximately corresponds with a BER of 10^{-11} after BCH decoding. To ensure reliable results, at least 1000 erroneous bits have to be detected. In other words, to reach a BER value of 10^{-7}, 10^{+10} bits have to be decoded, resulting in a considerably long simulation time. However, due to the sharp fall in the BER vs SNR curve for LDPC decoding, no so restrictive BER values can be considered with similar performance. Furthermore, the DVB-T2 OFDM parameters used for the simulations were chosen to be as similar as possible to those for DVB-T standard. These parameters are as follows: the Fast Fourier Transform (FFT) size is 8k with a guard interval of 1/32, and the bandwidth is 8 MHz with normal carrier mode. Rotated constellations were used and Peak-to-Average Power Ratio (PAPR) techniques were not applied [60]. All the combinations of constellation and code-rate have been considered for the two FEC block sizes. The simulations assumed ideal synchronization and ideal channel estimation and no phase noise. Besides, Genie-Aided Demapping is considered with a prior knowledge of the transmitted bits for the constellation demapping. In addition, no pilot carriers and neither special symbols (P1, P2 and frame closing symbols) are included.

Due to the lack of pilots in the simulations it is necessary to increase the obtained minimum SNR value. The reason is that the pilot carriers have slightly higher power than data carriers to increase their protection against AWGN. This implies that the total power of the OFDM symbol is higher than when only data carriers are considered in the simulations. The increment in SNR depends on the percentage of pilot carriers with respect to the data carriers in each symbol. Consequently, it depends on the FFT size, the use of normal or extended mode and the specific scattered pilot pattern.

As the conditions for the computer simulations are ideal, it is clear that the results are too optimistic and they are practically impossible to be achieved under normal reception conditions with clock mismatches, time synchronization errors and channel frequency shifts. Moreover, the channel estimation is performed by means of pilot carriers resulting in a not perfectly estimated frequency response as pilot carriers suffer from noise. Consequently, interpolation for the not so accurate pilot carriers must be applied for the data carriers. Moreover, although an iterative demapping process is used, the prior knowledge of the transmitted bits shows better performance. These degradations on the minimum SNR threshold, which are called implementation losses, depend on the specific receiver implementation.

Another information source that provide theoretical minimum SNR values for DVB-T2 fixed reception is the EBU Technical Report 3348 [61], entitled Frequency and Network Planning Aspects of DVB-T2. This document gathers the main features of the standard and some criteria for DVB-T2 networks planning. The considered minimum SNR thresholds are not innovative as it uses those from DVB-T2 implementation guidelines. Moreover, some DVB-T2 configurations are suggested for typical fixed reception scenarios. Additionally, there are some research works such as that in [23][62] [63] and [64] which includes additional simulation results for minimum SNR threshold in fixed reception. The presented results are comparable to those gathered in [59].

It must be noted that simulation results for T2-Lite are not provided in [59]. In this sense, on the one hand, code-rates that are also present in the DVB-T2 base profile keep similar SNR threshold values from the guidelines. On the other hand, the SNR threshold for fixed reception for the new code-rates introduced in T2-Lite (1/3 and 2/5), has been extrapolated from DVB-S2 simulations values [65].

DVB-T2 was also developed with mobile reception as one of the target user groups, including the time interleaving feature to benefit from time diversity. The term “mobile” has become a very broad meaning term, and thus, the National Association of Broadcasters (NAB) has recently presented a more focused definition of what is considered as a mobile service [66]. The NAB association considers mobile outdoor reception when pedestrian handheld receivers are used in outdoor environments. Furthermore, vehicular built-in receivers, handheld used in-vehicle devices and portable devices should also be considered in this scenario. Finally, the mobile reception should also be possible at ground speed of at least 150 kmph. Although there are several channel models to emulate portable or mobile outdoor reception [67], Typical Urban 6 paths (TU6) [68] is usually the considered channel model for the mobile reception despite being too pessimistic [69].

However, mobile reception was not one of the main focuses for DVB-T2. Consequently, DVB-T2 Implementation Guidelines [59] do not focus on portable and mobile outdoor performance. They only include some reference figures with minimum SNR thresholds. The defined figures shows the DVB-T2 performance at different Doppler frequencies (10 Hz and 80 Hz) and with different time interleaving depths. Besides, the use of subslicing and inter-frame time interleaving is also tested. The considered threshold criterion is based on 1% of BaseBand (BB) Frame Error Rate (FER) which takes into account the bit error correlation, unlike BER criteria which only indicate the percentage of erroneous bits. Considering that one single erroneous bit is enough to corrupt an entire BB frame, the final performance of the system depends highly on the bit error correlation.

Some additional minimum SNR values for mobile reception are provided in the EBU Technical Report 3348 [61] for different Doppler frequencies. The values for the new code-rates introduced in T2-Lite are obtained by extrapolation. Besides, some DVB-T2 configurations are suggested for several portable and mobile scenarios. However, it does not include any SNR threshold for mobile scenarios neither any methodology to obtain it as no receivers particularly optimized for portable and mobile reception were available at the moment. These results are not representative enough and more research was needed. For this reason, several studies were carried out in order to analyze the DVB-T2 theoretical performance in mobility. In [23] and [70] BER vs SNR curves are presented for several channel models intended for portable and mobile outdoor reception. Equally, in [64] some BER vs SNR curves for TU6 channel are presented. These curves are extrapolated in [71] using a sigmoidal shaped curve-fit so as to obtain the required SNR thresholds for portable reception.

Moreover, several research works focus on the flexible time interleaving parameter, whose length ranges from hundreds of milliseconds up to several seconds at the expense of increased latency and zapping time. In [72] and [73] several simulations are conducted so as to test the influence of the DVB-T2 time interleaving scheme that allows multiple tradeoffs in terms of time diversity for mobile scenarios, latency and power saving by means of inter-frame interleaving, frame hopping and subslicing. It concludes that the use of subslicing highly improves the system performance in fast fading scenarios whereas the use of intra-frame time interleaving improvement achieves very important gains in shadowing scenarios, especially at low speeds. In [48] the frequency, time and space diversity influence of DVB-T and DVB-T2 systems is studied in mobile reception showing similar results. In [74], a detailed study of the influence on the mobile performance of several waveform parameters, such as FFT size and pilot pattern, is presented.

In addition to the TU6 channel, [23] analyzes the DVB-T2 performance with some preliminary theoretical BER vs SNR curves with Vehicular Urban at 30 kmph (VU) and Motorway Rural at 100 kmph (MR) [75] [76] [77]. Channel models, which have been sometimes considered to emulate mobile outdoor performance.

Moreover, in case of portable outdoor reception, in addition to the TU6 at low speeds, the P1 channel model has been traditionally considered as well. However, its use is very optimistic as it is a stationary channel model and slow channel variations from portable reception cannot normally be avoided. Even if the receiver itself is stationary there are often other objects in the vicinity of the receiver which move, such as cars or people. For this reason, a new channel model has been recently defined, such as Pedestrian Outdoor at 3 kmph (PO) [75] [76] [77]. The DVB-T2 performance under this channel model is also included in [23].

Finally, indoor reception is also a target scenario for DVB-T2 system. It refers to the situation when any static or handheld device is used indoors [78]. This scenario is very common in some countries such as USA, where each resident has its own indoor TV antenna connected to its TV device. However, it is not so common in Europe and consequently, equally to mobile reception, it is not a primary target scenario. For this reason, no theoretical studies for specific indoor reception are included in the DVB-T2 Implementation Guidelines [59] or EBU Technical Report [61], where only DVB-T2 performance under P1 channel can be considered for indoor reception as this channel has been widely considered as reference of this kind of scenario. However, as it was demonstrated in WingTV project [75], P1 is too optimistic as no channel variations are taken into account: nor the receiver movement neither the surrounding elements movement. For this reason, other channel models have been recently defined to emulate indoor reception. The most common one is Pedestrian Indoor at 3 kmph (PI) [75] [76] [77]. Some preliminary theoretical BER vs SNR curves for PI channel model have been presented in [23].

2.1 Laboratory Measurements

In Europe, some laboratory measurements for testing DVB-T2 reception have been carried out within the development of Broadcast for 21st Century (B21C) [79] and Enabling Next GeneratIon NEtworks for broadcast Services (ENGINES) [80] projects. The latest one also analyzed the T2-Lite performance. Most particularly, in Spain Futura Red Integrada Audiovisual (FURIA) [81] and Nueva Generación de sistemas de Radiodifusión DigitAl Terrestre (NG-RADIATE) [82] deliverables also contain some performance results for DVB-T2 fixed reception.

The most of the laboratory measurements for fixed reception are gathered in [83], studying the DVB-T2 configuration parameters influence on the performance in detail. Additionally, [84] and [85] includes some results for

AWGN, F1, P1 and 0 dB Echo channel models. Besides, the influence of different DVB-T2 configuration parameters such as constellation order, code-rate, FFT size, the use of rotation constellations, guard interval lengths and pilot patterns is also tested. Finally, the influence of modifications on the second path delay, attenuation and frequency shift on the system performance is also studied. The results show that the implementation losses due to different receivers' implementation are about 2 dB in AWGN channels an up to 4 dB under F1 channel models. Besides, the EBU Technical Report [61] includes some laboratory measurement results for AWGN and F1 channel models carried out with four different receivers. The results are in line with those in [83] [84] and [85].

As DVB-T2 main target scenario is fixed reception, DVB-T2 receivers are not tested for mobile reception. Consequently it is important to carry out laboratory measurements for mobile and indoor reception to establish the losses due to the receiver implementation. For this reason, portable and mobile receptions have been widely tested in ENGINES [80], with special emphasis in T2-Lite performance as it is the profile intended for mobility.

On the one hand, in [86] the influence of some DVB-T2 configuration parameters (FFT size guard interval length and the use of rotated constellations) on the portable and mobile performance (PO and TU6 channel models) is tested in the laboratory. On the other hand, some preliminary laboratory measurements for indoor reception are also gathered in [86], showing the influence of some DVB-T2 configuration parameters (FFT size guard interval length and the use of rotated constellations) under PI channel model. The main conclusion is that indoor reception is more SNR demanding than fixed or mobile reception.

2.1 Field Trials

On the one hand, several field trials have been performed in order to test the DVB-T2 fixed reception in different countries with the objective of testing the feasibility of the DVB-T2 system in different type of cities. Various field strength measurements have been conducted in Thailand [87], in Oman [88], in Croatia [89] and in Mongolia [90] to test the DVB-T2 coverage in comparison with the previous television systems in each country. Furthermore, minimum SNR threshold were obtained by means of field trials carried out in Croatia [70] and in Spain [49] [83] [91] within the FURIA [81] and NG-RADIATE [82] projects. The results show that the SNR thresholds in fixed reception with roof-top antennas are between 1 to 2.5 dB higher than those based on computer simulations [59] [61].

On the other hand, some field trials have been conducted in order to test the DVB-T2 performance in mobile scenarios. Mobile urban coverage results and minimum SNR thresholds are presented in [92] and [93], respectively by means of field trials carried out in Spain. Moreover, in [94] and [95] the portable and mobile reception are analyzed in detail following several mobile routes in Germany for several DVB-T2 configurations and with special emphasis in the FFT size and time interleaving length influence on the system performance.

The performance in mobile scenarios is very dependent on the receiver implementation, especially in the channel estimation and equalization processes. Depending on the specific implementation, the degradation can be higher. This is the problem of commercial DVB-T2 receivers, which have not been created for mobile reception. For this reason, in [96] different channel frequency response implementations have been tested showing different performance in mobility.

3. Research Work

As it can be seen in the former section, indoor scenarios are still poorly studied. This is because although indoor reception is considered a booming scenario in some countries where each TV device has its own indoor receiver antenna, the most current broadcasting networks are still designed for rooftop reception. For this reason, there is still a lack of indoor DVB-T2 studies so as to complete the existing theoretical results and check the feasibility of the system for providing indoor services, as this kind of scenario is increasingly more and more common.

On the one hand, although some laboratory measurements have been performed under P1 and PI channel model conditions, there are other channel models that have been recently defined for portable indoor reception that can also be considered as reference. Some of them are Indoor Office A (IOA)/Indoor Office B (IOB) and Indoor Outdoor and Pedestrian A (IOPA)/ Indoor Outdoor and Pedestrian B (IOPB) [97], defined by the ITU. Finally, the TU6 [68] widely used for emulating mobile outdoor reception, is sometimes considered for portable indoor reception with a pedestrian speed of 3 kmph. On the other hand, some field trials in indoor scenarios are needed in order to test the DVB-T2 feasibility under real indoor conditions.

The technical information offered by most of the DVB-T2 commercial receivers was very limited from the researching point of view in the moment of the studies development. For this reason, a complete DVB-T2 and T2-Lite receiver framework, totally implemented in the University of the Basque Country, was considered for the analyses.

These results can improve the existing performance information and help the broadcasters to ideally design the broadcasting network including indoor coverage. Especial emphasis is needed in portable indoor reception because, although the indoor receiver can stay static, there are always several moving elements in an indoor scenario that can produce variations in the reception channel.

For this reason, in this thesis some laboratory measurements with different indoor channel models and some field trials have been carried out so as to evaluate the DVB-T2 performance in indoor scenarios by means of obtaining of SNR thresholds for a correct indoor reception.

More specifically, three studies have been carried out:

- Study A: The portable indoor reception has been tested in the laboratory.
- Study B: The portable indoor reception coverage has been tested in the field
- Study C: The fixed and portable indoor reception has been deeply analyzed by statistical studies with laboratory measurements and field trials.

3.1 DVB-T2 Receiver Framework

The TSR research group, within the UPV/EHU, has developed a professional Software Defined Radio (SDR) DVB-T2 Test Receiver Framework for demodulating DVB-T2, DVB-T2-Lite and combined signals [47]. This receiver has been widely used in this thesis for all the DVB-T2 indoor studies because, unlike existing commercial DVB-T2 receivers at the research time, it is totally compatible with all the DVB-T2 parameters. Figure 6 shows a diagram with the main blocks in which the SW receiver is organized. Green boxes describe the available measurements for each block.

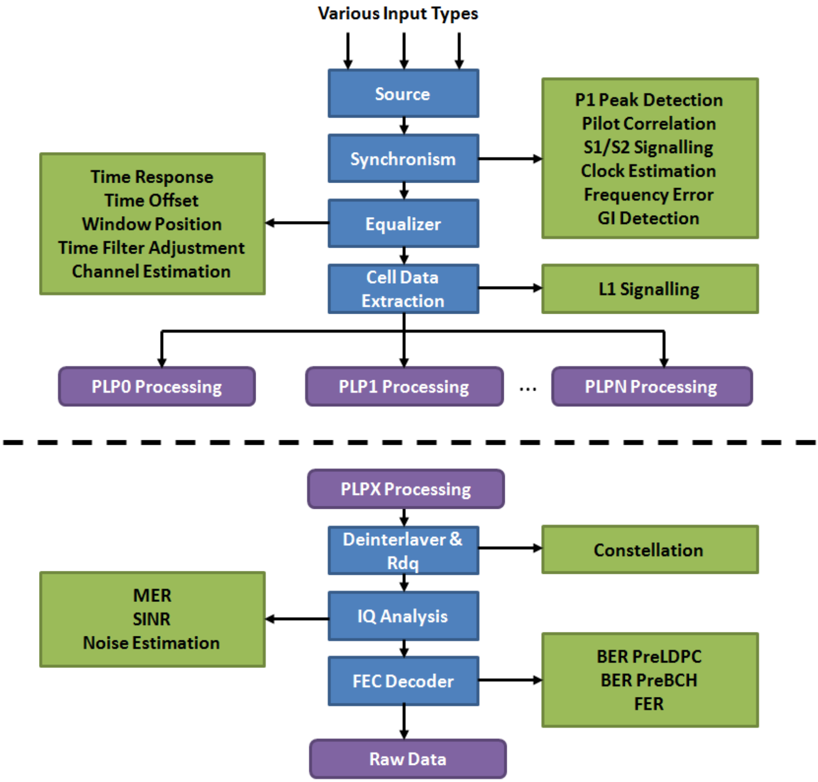


Figure 6. Main blocks of the UPV/EHU DVB-T2 Receiver

This receiver is a SW demodulator that can work in two different modes as it can be seen in Figure 7. The first one is based on an offline analysis, by the demodulation of In-phase and Quadrature (IQ) samples previously recorded into an IQ samples file. The second one consists on demodulating RF signals by using an additional RF module which receives the DVB-T2 or DVB-T2-Lite RF signal as baseband IQ samples. These samples can be saved in a file, which could be later demodulated by the SW demodulator, or can be sent directly to the SW demodulator by a TCP/IP socket, getting a pseudo-real time analysis of the received signal. This additional module can be a Universal Software Radio Peripherical (USRP) N-210 device from Ettus Research [98], which is connected to the computer using a Gigabit-Ethernet link.

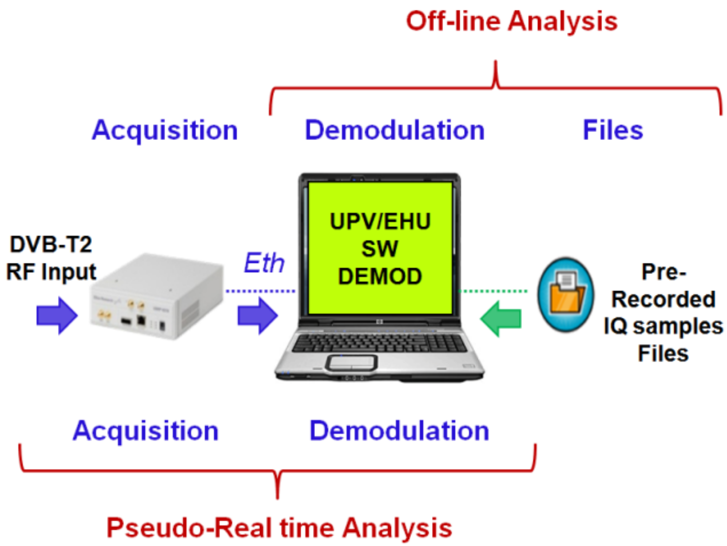


Figure 7. UPV/EHU DVB-T2 Receiver Operation Modes

When doing offline analysis, the input to the SW receiver is a file with the previously recorded IQ samples of the signal to be demodulated. Supported formats for the input files are:

- Binary files with double (IEEE) IQ samples.
- Text files with double IQ samples separated by spaces or newline.

- Binary file with IQ samples saved as signed Int16 little Endian.
- Binary file in the Tektronix IQT format.
- Binary file in the Tektronix TIQ format.
- Binary file in the HP VSA SDF format.
- Binary file in the HP VSA BIN format.
- Binary file in the ADIVIC TCX format after proprietary conversion.
- Binary file in the Anritsu DGZ format.

When doing pseudo-real time analysis, the inputs to the SW receiver are the IQ samples sent by a Transmission Control Protocol (TCP)/Internet Protocol (IP) socket. In this last case, the RF signal is received using the RF input interface the USRP has, whose supported format is:

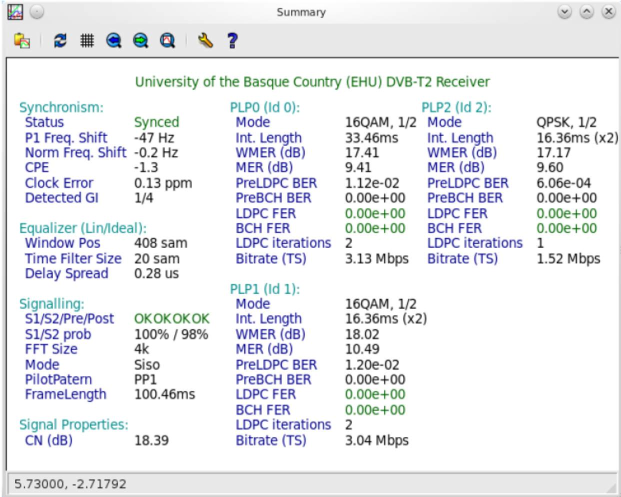
- Int16 IQ samples through TCP/IP socket.

As this is a SW demodulator, there are no physical input/output interfaces. The data input is by a binary file or a TCP/IP socket with the IQ samples of the signal to demodulate. In case of demodulating a RF signal using the USRP device, as done in this thesis, the input interface is:

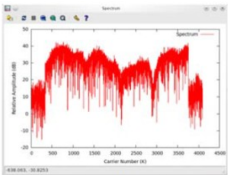
- **RF input (when using the USRP N-210 device)**

  General function: DVB-T2 or DVB-T2-Lite RF signal reception.
  Frequency range: From 50 MHz to 2.2 GHz.
  Level range: -90 dBm to -20 dBm
  Connector: SMA (SubMiniature version A) - 50 Ω (Female)
  Capture Bandwidth: 10 MHz

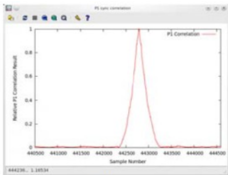
The demodulator gives some graphic information such as signal spectrum (Figure 8b), channel estimation (Figure 8c), P1 symbol detection (Figure 8d), impulsive response (Figure 8e), pilot carriers correlation (Figure 8f) and constellation (Figure 8g).



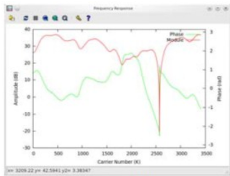
3a)



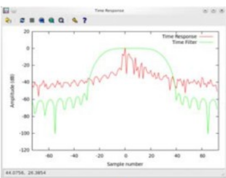
3b)



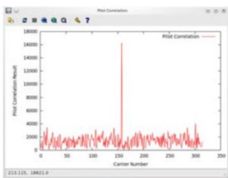
3c)



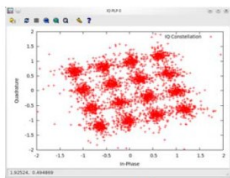
3d)



3e)



3f)



3g)

Figure 8. UPV/EHU DVB-T2 Receiver Provided Information.

Besides, Table 2 resumes the main text information monitored by the receiver and gathered in Figure 8a.

Table 2. DVB-T2 Receiver main provided text information

Synchronization	Status (synced or no)	**Signal Properties**	SNR (dB)
	P1 frequency shift		Modulation
	Norm. Frequency shift		Code-rate
	CPE		Time Interleaving Length
	Clock Error		WMER (dB)
	Detected GI		MER (dB)
Equalization	Window Position	**PLP Information**	Pre LDPC BER
	Time Filter Size		Pre BCH BER
	Delay Spread		LDPC FER
Signaling	S1/S2/L1 Pre/L1 Post OK		BCH FER
	S1/S2 Probability		LDPC Iterations
	FFT Size		Bit Rate (Mbps)
	Mode		
	Pilot Pattern		
	Frame Length		

The obtained information depends on the operation mode. When the offline analysis is carried out, all the information from Table 2 and Figure 8 is provided. However, there are two possible pseudo-real time analysis modes. In one of them, all the information from Table 2 and Figure 8 is provided but it takes between 1 and 3 s (depending on the DVB-T2 configuration parameters) to analyze the receiving signal before updating the whole information. However, it is possible to obtain more frequent updates of the information in Table 2 and Figure 8 by using the second pseudo-real time analysis mode. This could be named fast pseudo-real time analysis, as it only spends about 500 ms analyzing the received signal before updating the graphic and text information. Nevertheless, when using the fast pseudo-real time analysis mode it is not possible to obtain BER and FER measurements. Besides, if the rotated constellation feature is in used, it is not possible to obtain neither a constellation graphic nor Modulation Error Rate (MER) measurements. The other text and graphic results are always provided.

3.2 Indoor Performance Studies

Three different studies have been carried out in order to complete the DVB-T2 indoor performance information by means of laboratory measurements and field trials.

- **Study A: DVB-T2 portable reception in indoor environments**

This study analyzes and evaluates the DVB-T2 performance in indoor scenarios by means of laboratory measurements under different channel models intended for fixed and portable indoor reception.

- **Study B: DVB-T2 field trials for portable indoor reception**

This study includes a coverage analysis to test the feasibility of DVB-T2 system to offer indoor reception with a real broadcasting network. Besides, different DVB-T2 configuration parameters are tested determining which the most appropriate for indoor reception are.

- **Study C: Field Trials Based Planning Parameters for DVB-T2 Indoor Reception**

This study includes a detailed analysis and evaluation of the DVB-T2 performance in fixed and indoor scenarios by means of field trials. Furthermore, some performance results with laboratory measurements are also presented so as to determine the standardized channel model which better fits the real DVB-T2 indoor reception.

In addition, a complete performance evaluation methodology for indoor scenarios has been widely defined and tested, including the obtaining of time and location variability.

3.2.1 Study A: DVB-T2 portable reception in indoor environments

The main objective of this study is the obtaining of minimum SNR thresholds for correct DVB-T2 reception in portable indoor scenarios, testing the system performance in this kind of scenarios.

For this purpose, laboratory measurements with several channel models have been carried out including reference channels (AWGN, F1 and P1) for fixed reception and specific channel models, which includes some Doppler effect because of the movement, for portable indoor reception. These results improve the existing results as HW equipment is used and several indoor channel models are considered. Besides, three different DVB-T2 receivers have been used in order to see the difference in terms of performance.

3.2.1.1 Physical Layer Pipes (PLPs)

With the appearance of new target scenarios, one of the requirements for the new generation broadcasting systems was the possibility of offering several services for different scenarios at the same time. In DVB-T2, PLPs enable the definition of different service specific robustness levels, allowing different protection levels by means of different coding parameters, constellation orders and interleaving depths.

By this way, a single radiofrequency channel can therefore transmit one or more PLPs, configured for different bitrates and robustness. For example, one PLP can be configured for broadcasting high bitrate services to be received by roof-top antennas (low protection and high constellation order) whereas a second PLP can be configured for high robustness but low bitrate so as to be received by portable or mobile receivers. Nevertheless, it is important to be noted that waveform parameters (FFT size, Guard Interval (GI) length or pilot patterns) are common for all the PLPs.

This feature can be also used for researching purposes in order to test different DVB-T2 BICM parameters under the same channel conditions, as all the PLP configurations are transmitted in the same DVB-T2 frame.

3.2.1.2 DVB-T2 Configuration

This study takes advantage of the DVB-T2 capacity to simultaneously transmit different DVB-T2 configurations by means of multiple PLPs in order to test the performance under the same channel conditions.

Table 3 summarizes the main parameters of the three tested DVB-T2 configurations. These configurations are good examples of future possible configurations that could be used for the transmission of at least one or two HD services in complex indoor scenarios with H.264 video compressing standard [99] [100] [101] [102]. However, a bit-rate of 1.1 Mbps has been considered as a common reference for the tested configurations in the different PLPs. The considered modulation schemes are Quadrature Phase-Shift Keying (QPSK) and 16 Quadrature Amplitude Modulation (QAM) with low code-rates, so as to be robust enough for indoor reception.

Table 3. DVB-T2 tested Configuration modes for portable indoor reception

Parameters	Indoor Modes		
	#1	#2	#3
BW	8 MHz (Extended)		
FFT GI PP	16k 1/32 PP4		
Modulation	QPSK	QPSK	16QAM
Rotated	Yes		
Code-rate	1/2	2/3	1/2
Interleaving Time	247 ms		
Capacity (Mbps)	6.8	9.3	13.6
Considered Bitrate (Mbps)	1.1		

Most parameters are common in the three configurations. The combination of FFT size, guard interval and pilot pattern (FFT GI PP) takes an intermediate value between capacity and robustness, while modulation and code-rate values have been chosen to ensure the robustness needed for the high robustness needed in portable indoor scenarios. Besides, the time interleaving option is used with almost the highest possible interleaving length defined in DVB-T2 to improve the system performance in portable scenarios.

3.2.1.3 Laboratory Set-up

Figure 9 shows the set-up used for the portable indoor reception laboratory measurements. The transmitter includes a DVB-T2 signal modulating card generating the desired DVB-T2 profiles on 690 MHz frequency (channel 48). Although the modulating card includes the channel simulating option, an external Anite HW channel emulator (Radio Channel Emulator Propsim F8) with the capacity of emulating every channel model has been considered in this study.

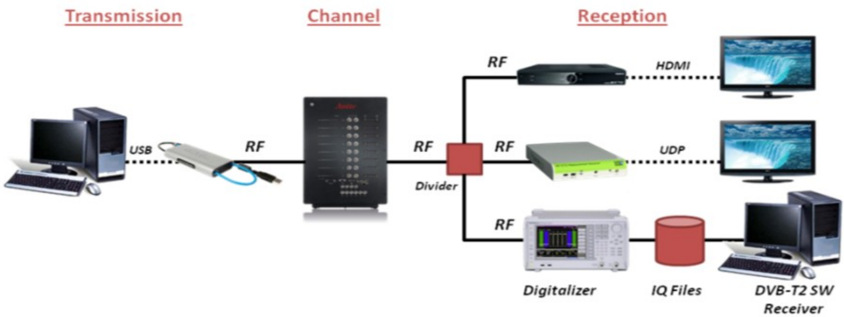


Figure 9. Laboratory Measurements Set-up

Table 4 shows the simulated channels models for portable indoor reception including the considered speed in each case. The output of the channel emulator is connected to a 1 to 3 splitter in order to simultaneously receive the DVB-T2 signal with 3 different receivers.

In this study, the traditional channel models related to fixed reception (AWGN, F1 and P1) [59] have been tested as reference. Additionally, some channel models specific to emulate portable indoor reception have also been considered. First, the PI at 3 kmph defined in DVB-H Implementation Guidelines [76] has been considered. Additionally, the IOA (with a delay spread low average), IOB, (with a delay higher average spread), IOPA (with a delay average spread bass), IOPB, (with a delay average spread high), defined by the ITU [97] have also been tested. Finally, the TU6 [68], widely used for emulating mobile reception, has also been tested with a pedestrian speed of 3 kmph.

Table 4. Simulated channel models for portable indoor reception

Name	Channel Model	Speed	Reference
AWGN	AWGN	--	DVB-T2 [59]
F1	Ricean	--	
P1	Rayleigh	--	
PI	Pedestrian Indoor	3 kmph	DVB-H [76]
IOA	Indoor Office A	3 kmph	ITU-R [97]
IOB	Indoor Office B	3 kmph	
IOPA	Indoor to Outdoor Pedestrian A	3 kmph	
IOPB	Indoor to Outdoor Pedestrian B	3 kmph	
TU6	Typical Urban 6 paths	3 kmph	GSM [103]

Three different DVB-T2 receivers were simultaneously used in the laboratory measurements so as to determine possible performance differences for the different channel models.

- The first receiver is a commercial Set-Top Box (STB) connected via High-Definition Multimedia Interface (HDMI) to a monitor allowing the visual observation of the decoded video contents.
- The second receiver is a professional HW receiver with information of signal quality parameters, such as SNR, BER or FER, among others. Furthermore, the decoded video contents can be sent to a video player by Real-time Transport Protocol (RTP) or User Datagram Protocol (UDP) and be able to visually observe the image on a screen.
- The third receiver is the professional SDR receiver [104] described in the former section in its offline mode. Similarly to the HW professional receiver, it includes information about the quality of the received signal. As it is a SW receiver, a previous digitalization process must be carried out in order to store the received signal IQ samples which are then analyzed by the SW receiver [105].

3.2.1.4 Laboratory measurements Description

The aim of these laboratory measurements is the obtaining of the minimum SNR threshold under different channel models considering a signal demodulation without error over a period of 30 s [106]. For this purpose, the DVB-T2 signal is generated based on the settings defined in Table 3. It is next introduced into the channel simulator, which emulates the channels defined in Table 4.

The first step consists on checking the correct signal reception with the first two receivers by a subjective analysis. In addition, a signal digitalization process is performed so that it can be further processed with the professional SDR receiver by an objective analysis. Consequently, two different processes are simultaneously performed.

On the one hand, the channel simulator allows the addition of AWGN in the signal band. For receivers 1 and 2, which monitor the demodulated received signal in real time, increasingly AWGN is added in steps of 0.2 dB until each receiver is unable to demodulate the signal without errors (the threshold situation has been reached). The threshold situation for both receivers is based on a direct visual observation of the video content in real time. Thus, the presence of errors in the received signal is considered when errors happen in the picture or the image is pixilated. The signal power and the added noise power are then measured with a power meter. Thus, the value of SNR threshold can be directly obtained.

On the other hand, the receiver 3 does not allow a direct real time demodulation and requires a previous digitalization process of the received signal. Consequently, the AWGN addition to the signal is carry out in the receiver (it also includes the capability of AWGN addition). Otherwise, the needed time for the laboratory measurements could be extremely long. The measuring process is similar to the one considered for the other two receivers, starting with very low AWGN power (high SNR) which is incremented in steps of 0.2 dB until the free-error threshold situation is achieved. In this case the error detection cannot be done by image observation. However, there is a direct relationship between errors in the visual image and the value of the FER after BCH as when FER is different from 0, errors appear in the visual image, reaching the threshold situation. In this case, the SNR threshold value is directly obtained in the receiver when the threshold condition is reached with a FER value different to 0.

Finally, these processes must be repeated for all channel models defined in Table 4 and for all the DVB-T2 configurations in Table 3.

3.2.1.5 Laboratory Performance Results

The results for minimum SNR threshold for the DVB-T2 configuration modes shown in Table 3 under channel models listed in Table 4 are presented in Table 5.

Table 5. SNR threshold for portable indoor reception on the laboratory

Receiver 1 (commercial STB)									
Modes\ Channels	**AWGN**	**F1**	**P1**	**PI**	**IOA**	**IOB**	**IOPA**	**IOPB**	**TU6**
#1	1.2	1.6	3.2	6.7	14.3	12.8	16.9	13.2	8.2
#2	3.8	4.0	6.6	10.2	17.9	17.0	21.6	15.8	12.0
#3	6.2	6.6	8.4	12.2	20.3	19.2	22.0	18.0	14.1
Receiver 2 (Professional HW receiver)									
Modes\ Channels	**AWGN**	**F1**	**P1**	**PI**	**IOA**	**IOB**	**IOPA**	**IOPB**	**TU6**
#1	1.2	1.6	3.2	6.7	14.3	12.8	16.9	13.2	8.4
#2	3.8	3.8	6.6	10.0	17.7	16.8	21.6	15.8	12.4
#3	6.2	6.6	8.4	12.0	20.3	19.2	22.0	18.0	13.9
Receiver 3 (Professional SW receiver)									
Modes\ Channels	**AWGN**	**F1**	**P1**	**PI**	**IOA**	**IOB**	**IOPA**	**IOPB**	**TU6**
#1	1.0	1.5	3.4	6.5	14.3	11.2	12.6	10.8	7.9
#2	3.2	3.3	6.8	9.1	17.9	14.7	15.4	13.6	10.8
#3	6.0	6.4	8.6	11.2	20.4	16.7	17.5	15.9	13.0

All in all, the performance of receivers 1 and 2 (commercial receivers) is very similar to each other, being also very similar to the receiver 3 (professional SDR receiver) performance for the reference stationary channel models (AWGN, F1 and P1). The existing small differences may be due to the 0.2 dB step in the noise level addition in the laboratory tests. However, these two receivers generally show lower performance than receiver 3 under portable indoor channel models conditions with differences of up to 4 dB in the SNR threshold. This fact may be because receivers 1 and 2, unlike receiver 3 [104], are not optimized for mobility, so its performance slightly decreases in portable scenarios. In addition, it can be observed that the P1 channel, which is sometimes considered as a suitable model for fixed indoor reception as non line-of-sight with the transmitter is considered, has a minimum SNR threshold at least 3 dB lower (and even up to 13 dB) than any portable indoor channel model.

On the one hand, if only portable reception is considered, the performance differences between different channel models can be up to 10 dB. PI channel model is the least restrictive with a performance degradation of about 3 dB with respect to model P1 channel. However, IOA and IOPA are the most restrictive with SNR thresholds of about 13 dB higher than P1 channel model. Moreover, if all the specific portable indoor channel models are compared with the mobile TU6, it can be stated that all of them, except for PI, have a SNR threshold at least 3 dB higher than TU6. PI, however, has a SNR threshold 1.5 dB lower than TU6.

On the other hand, the obtained SNR thresholds are always higher for 16QAM configurations, with values that can be excessively high. For example, in a country as Spain in which the existing DVB-T broadcasting network is designed for a SNR threshold of about 17 dB [49], mode #3 and sometimes even mode #2 SNR requirements are extremely high. Consequently, only mode #1, the most robust and with lower available bitrate, could be feasible for DVB-T2 portable indoor reception.

3.2.1.6 Conclusions of this study

This study includes the DVB-T2 performance characterization in indoor environments by means of laboratory measurements carried out with several channel models intended for this kind of scenario. Thus, the main performance differences between some of the channel models defined by various agencies (DVB, European COoperation in the field of Scientific and Technical Research –COST- 207, ITU) have been presented.

The results show that the TU6, usually used for mobile outdoor reception, and P1 channel model, typically used for fixed indoor reception, present very different performance to other specific portable indoor channel models. On the one hand, all of them are focused on portable reception, while the P1 channel model is focused on fixed reception, so its SNR is always lower (between 3 and 13 dB). Furthermore, the characteristics of an indoor scenario are very different from those in an outdoor environment for which the TU6 channel is oriented. Consequently, TU6 is more optimistic that the most channel models considered for portable indoor reception with up to 8 dB lower SNR thresholds. However, the PI channel model, proposed in DVB-H for portable indoor reception, is even more optimistic with SNR thresholds of about 2 dB lower than TU6. Thus, it can be concluded that there are a lot of standardized channel models intended for portable indoor reception. However, due to the high variability in the measured SNR thresholds, a single reference model for suitable portable indoor reception planning cannot be established.

Finally, the presented SNR threshold values can update and complete those in DVB-T2 implementation guidelines [59] and in the Frequency and Network Planning Aspects technical report of the EBU [61]. These results can be considered for future planning of digital broadcasting services in indoor environments.

3.2.2 Study B: DVB-T2 field trials for portable indoor reception

The main purpose of this study is the analysis of DVB-T2 coverage for portable indoor reception (with a moving receiver at pedestrian speed) by means of field trials conducted in Spain for different DVB-T2 configuration parameters.

These field trials were performed within the ENGINES project [80], being the first field trials including the analysis of T2-Lite. Besides, they are the first attempt to check the feasibility of the DVB-T2 system to offer portable indoor services in real scenarios.

3.2.2.1 T2-Lite & Future Frame Extension (FEF)

DVB-T2-Lite is the mobile profile included to the DVB-T2 specification in the release number 1.3.1 in 2012. It targets mobile and portable reception and consequently, it only contains the transmission modes suited for mobile reception while minimizing the amount of receiver complexity as it is usual in handheld receivers. For example, it establishes restrictions in terms of time interleaver memory, modulation orders, code-rate and bitrate. At the same time, the number of new elements has been restricted in order to maintain compatibility with DVB-T2.

DVB-T2 includes the possibility of use FEFs, which are part of the T2 frame that can be filled with any signal. Thanks to the FEFs feature, it is possible to introduce a T2-Lite signal optimized for mobile reception inside a T2 frame transmitting a mixed configuration. The combination of T2-Lite with DVB-T2 transmissions is one solution for the new requirements of broadcasting systems. For example, it would be possible to dedicate 75 % of the transmission time to fixed DVB-T2 and 25 % to mobile T2-Lite.

In this case, unlike the PLPs, each profile has total independency in the configuration parameters as they are individual signals transmitted in the same multiplex. By this way, a T2-Lite signal with 8k FFT size (with extended carrier mode), QPSK constellation 1/2 code-rate and PP1 pilot pattern, configuration parameters ideal for mobile scenarios, has a capacity of approximately 1.2 Mbps per channel (8 MHz bandwidth), which is enough for a HD service or 3 SD services at about 400 kbps. In general, the overall spectral efficiency gain depends on the percentage of time dedicated to the transmission of mobile services. Summarizing, the use of FEFs is a time multiplexing technique based on the transmission of two totally independent signals transmitted in the same T2 frame.

Equally to multiple PLPs, the FEFs can also be used for researching purposes in order to test different DVB-T2 waveform and BICM parameters under the same channel conditions.

3.2.2.2 DVB-T2 Configuration

Thanks to the FEFs, the influence of different DVB-T2 parameters on the coverage can be ideally tested, as all the configurations are transmitted at once. The DVB-T2 parameters that have been studied are:

- FFT size. DVB-T2 adds the 1k, 4k, 16k and 32k FFT sizes to those used in DVB-T (2k and 8k). Larger FFT sizes mean a greater delay tolerance for the same fraction of guard interval, allowing the planning of larger SFNs. However, smaller FFT sizes are preferable in mobile or portable reception with Doppler effects as larger FFT sizes have higher vulnerability to fast time-varying channels. For these reason 4k and 8k sizes have been studied.

- Pilot Pattern (PP).The performance in time varying channels can also be affected by the choice of the PP. In DVB-T2 eight PPs are available (named from PP1 to PP8) unlike DVB-T in which only one PP was available. The network planner has to choose a PP considering the expected channel and a tradeoff between capacity and performance.

  In the case of portable indoor reception with multipath effects, denser PPs are recommended so as to follow channel variations in both frequency and time. In this study PP1 and PP2 have been tested with the different FFT sizes mentioned above to obtain their influence on the performance.

- Code-rate and constellation. In DVB-T2 several code-rates (1/2, 3/5, 2/3, 3/4, 4/5 and 5/6) and constellation schemes (QPSK, 16QAM, 64QAM and 256QAM) are allowed. However, new high constellation schemes and code-rates are oriented to fixed reception because they require high SNR. In this case, 1/2 code-rate has been tested with QPSK and 16QAM so as to have a tradeoff between robustness and capacity.

Taking all above into consideration, two mixed modes were defined and tested. Both of them make use of the FEF and multiple PLP new features.

- Mode #1 (FFT Change). This is a mixed mode with two T2-Base components with two PLPs inserted in each one. Both components share the same configuration parameters with the exception of the FFT size. It allows testing in identical conditions of reception the effect of two FFT sizes (4k and 8k) and two modulation schemes (QPSK and 16QAM).

- Mode #2 (PP Change). This is also a mixed mode with two T2-Base components with two PLPs inserted in each one. Both components share the same configuration parameters with the exception of the pilot pattern. It allows testing in identical conditions of reception the effect of two different pilot patterns (PP1 and PP2) and two different modulation schemes (QPSK and 16QAM).

The main changing parameters of these modes are resumed in Table 6, while the other primary configuration parameters were the same in all cases: bandwidth of 7.61 MHz, guard interval fraction of 1/8, 1/2 code-rate, 16k for LDPC FEC length, the use of rotated constellations and around 32 ms of interleaving length. The capacity for QPSK configurations is about 6.0 Mbps whereas it grows up to 12 Mbps when 16QAM is considered. This capacity is enough for the transmission of several HD services with H.264 [99] [100] [101] [102]. However, as the time had to be divided into 4 different PLPs in each mode, the considered bitrate in all the configurations with QPSK constellation is 1.3 Mbps and exactly the double with 16QAM constellations constellation.

Table 6. DVB-T2 measured modes Main Changing Parameters

Mode	Comp	FFT	Pilot Pattern	Constellation	Considered Bitrate (Mbps)
#1	Base	8k	PP2	QPSK	1.3
				16QAM	2.6
	Lite	4k	PP2	QPSK	1.3
				16QAM	2.6
#2	Base	8k	PP1	QPSK	1.3
				16QAM	2.6
	Lite	8k	PP2	QPSK	1.3
				16QAM	2.6

3.2.2.3 Measurement Campaign Description

Experimental Network

Two sections shown in Figure 10 were involved in the experimental network. The first one was the T2 Modulator Interface (T2-MI) Generation section. For each mixed mode four services were combined using two T2-gateways, one for each component of the mixed signal. Each gateway combined two PLPs, so two Transport Streams (TS) players connected to each T2-gateway with Asynchronous Serial Interface (ASI) connections were needed.

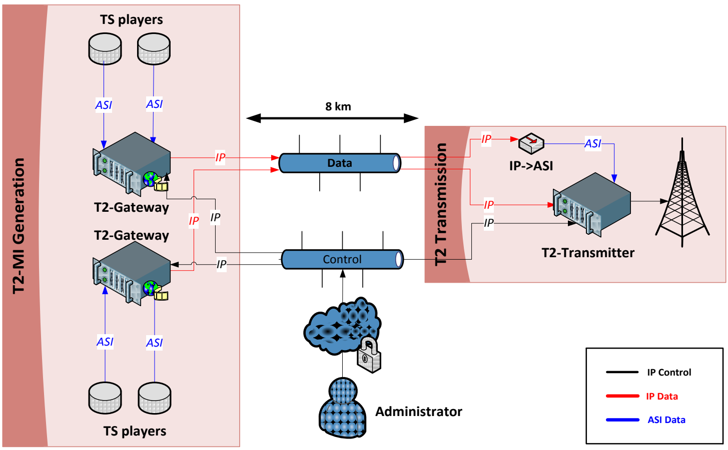


Figure 10. Network scheme for the mixed mode signals transmission

The two T2-MI signals generated were sent to the second section, the T2 Transmission section, via Ethernet links. The transmitter directly received one of the components to configure the mixed mode. However, the T2-MI signal of the second component was previously sent to an IP to ASI converter so as to convert the IP data signal into an ASI signal that was next sent to the ASI transmitter input. By this way, the transmitter was capable to combine both components and generate the mixed signal. As a result, four different T2 configuration modes were transmitted in each mixed mode. All the equipment was configured through web interfaces using an Ethernet link.

The transmitter centre was located on a mountain in Collserola

(Barcelona, Spain), 448 m higher and around 8 km away from the Barcelona Exhibition Centre reception building. Figure 11 shows an image of the transmitter and reception building in their locations.

The transmission system consisted on a 300 W transmitter with a radiating system based on a three dipole array panel with vertical polarization radiating in 482 MHz (UHF channel 22) with an antenna gain of 16 dBi.

Figure 11. Transmitter and reception building locations

Measurement System

The measurement system was based on an online IQ signal recording for posterior offline laboratory processing, which provided great flexibility [105] [107].

Figure 12 shows a scheme of the equipment used in the measurement system inside the mobile unit. The reception system had a vertical dipole tuned to the transmission frequency of 482 MHz. This antenna was on the roof of a self-powered mobile unit. The received signal was recorded in a high speed eSata hard disk as baseband IQ samples for a posterior offline processing stage using and IQ recorder equipment. Online measurements were also carried out at the same time with the professional SDR DVB-T2 demodulator developed by

UPV/EHU in its pseudo-real time mode, connected to an IQ digitalizer so as to verify the correct reception in real time. Besides, a signal analyzer was used to measure the received signal power. A custom SW system application, which remotely configures each measure, records the RF signal (IQ samples) and associates this information with the receiver speed and position information provided by a Global Positioning System (GPS) and a tachometer.

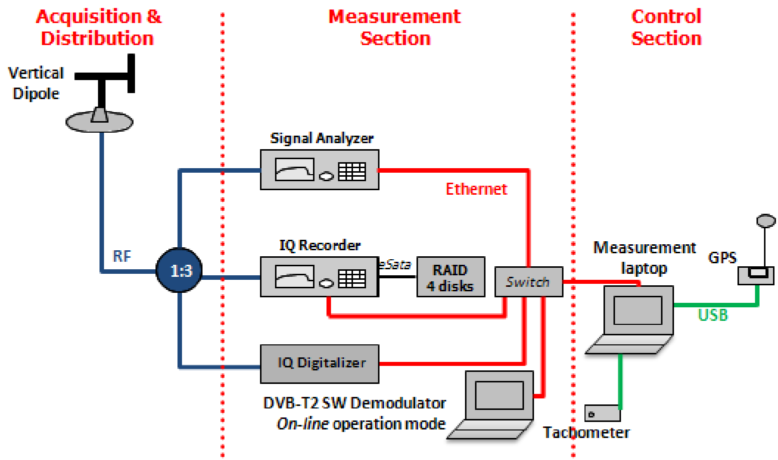


Figure 12. Measurement system scheme

Measurement environment

The measurement campaign was conducted inside the Barcelona Exhibition Centre. This is a group of one floor buildings in the outskirts of the city of Barcelona.

These buildings are obvious examples of typical industrial pavilions or public exhibitions buildings. As it is shown in Figure 13, the walls are of reinforced concrete with metallic beams and pillars on roof and walls. There are also some glass panels both on the roof and on the walls.

When the measurements were performed, the buildings were empty and it was possible to drive the mobile unit inside the pavilions at pedestrian speed (less than 5 kmph). Several routes were carried out along two of these buildings and recording the received signal on hard disks for a posterior offline analysis.

Figure 13. View inside the Barcelona Exhibition Centre building

Measurement Analysis Methodology

The measurements were divided into two phases. The first one consisted on online measurements carried out during the field trials. With these measurements some preliminary results of coverage were obtained and the signal for the posterior offline analysis was recorded on hard disks.

All the data stored on hard disks during the field trials were afterwards processed in the laboratory with the professional SDR receiver [104] defined in DVB-T2 Receiver Framework in its offline operation mode in order to obtain the performance results. It implements the whole reception chain, from synchronization to PLP extraction, providing detailed measurements of the main parameters of the recorded signal.

The offline processing generated multiple files with information about the quality of the received signal. The main objective of this offline analysis was to measure the amount of seconds with good reception and with errors, so as to obtain the percentage with no errors (correct reception).

Reception Threshold Criterion

The DVB-T2 standard sets the threshold criterion for DVB-T2 signal reception in getting to the Quasi Error Free (QEF) point, which is reached when the BER is lower or equal to 10^{-7} after LDPC [59].

However, due to the high measurement time required to obtain BER = 10^{-7} values and considering the receiver capabilities, the threshold criterion for the measurements was based on a FEC Blocks Error Rate (FBER) [108], which is defined as the ratio of erroneous FEC blocks received during an observation period established in a T2-frame time. In other words, the FBER value per frame is the number of FEC blocks with errors inside a T2-frame.

By this way, the number of T2-frames with FBER≠0 (any FEC Block in the T2-frame has errors) were measured for each configuration mode, obtaining the correct reception percentage.

Methodology

The measuring analysis methodology in outline is shown in Figure 14.

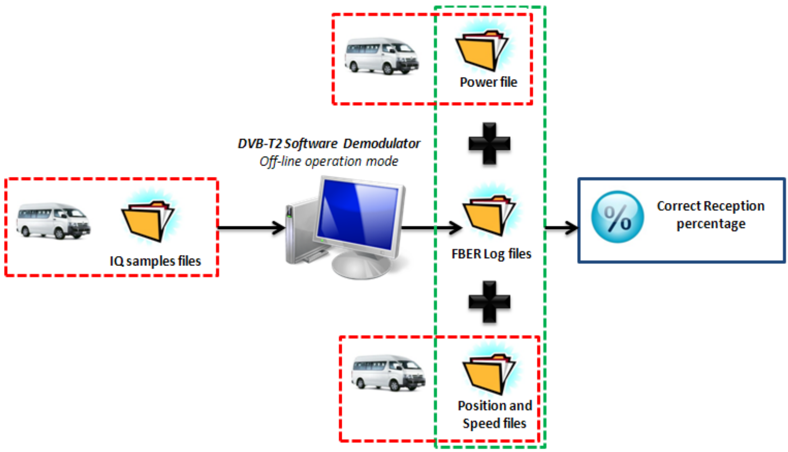


Figure 14. Measurement analysis methodology

The received signals recorded as IQ samples during the field trials were analyzed using a professional DVB-T2 demodulator. As result, many log files with the FBER information were obtained. This information was combined with the location data from the GPS (latitude and longitude coordinates) and the speed data from both the GPS and the tachometer. Besides, it was also analyzed in combination to the power measurements taken during the trials.

All in all, the number of correct frames (frames with FBER=0) and incorrect frames (frames with FBER≠0) were calculated, obtaining the correct reception percentage for each configuration mode in portable indoor reception.

The offline laboratory processing also enables the simulation of lower transmitted power levels by means of adding external white noise to the recorded signals. By this way, the SNR decreases as if the transmitted power during the trials was lower. 100 W, 10 W and 1 W simulated transmitted power levels were tested. The decrement in the SNR (*SNRdec*) needed to obtain the desired simulated transmitter power (*Ptx_des*) considering a real transmitter power (*Ptx_real*) of 300 W can be obtained obtained by (1).

$$SNRdec(dB) = 10 \times \log\left(\frac{Ptx\_real(W) - Ptx\_des(W)}{Ptx\_des(W)}\right) \quad (1)$$

This necessary decrement in the SNR cannot be done by decreasing the transmitted power level but it can be performed equally by adding an additional noise power level to the existing noise level. The existing noise power level (*Nex*) was measured in the received signal out of band samples, obtaining en empirical value of -97.8 dBm (measured in a bandwidth of 7.61 MHz). By this way, the real noise power level (*Nreal*) needed to obtain the SNR decrement necessary to simulate the decrement in the transmission power is obtained by (2).

$$Nreal(dBm) = 10 \times \log\left(10^{\frac{Nex(dBm)}{10}} + 10^{\frac{Nex(dBm) - SNRdec(dB)}{10}}\right) \quad (2)$$

3.2.2.4 Field Trials Coverage Results

The results presented are based on the correct received percentage, in other words, the percentage without errors of the measured signal, for each configuration mode under study along the tested routes inside the Barcelona Exhibition Centre buildings. Table 7 shows the performance results for the two measured modes (M1 and M2 respectively). The results are shown in terms of average percentage of frames without errors, considering erroneous frames those with FBER≠0.

Considering the results obtained in Table 7, which includes the general correct reception percentages (CRPs), the CRP is always higher than 95% (which is considered as a limit for "good" reception in portable reception [61]) with the entire tested configuration modes with a simulated transmission power of 300 W and 100 W. The CRP is also higher than the limit of 95% with 10 W and QPSK modulation scheme. However, it is decreased in at maximum 13% when the constellation order increases to 16QAM, but maintaining a CRP higher than 70% (which is a secondary limit for "acceptable" reception in portable reception [61]). The CRP decreases up to around 50% when a simulated transmission power of 1 W is considered. The situation worsens when the constellation order increases to 16QAM, reducing the CRP up to values of 7%.

Table 7. Correct Reception percentages (CRP) % for the measured DVB-T2 Modes for Different Transmission Powers

Mode	Comp.	Const.	Cap. (Mbps)	CRP (300W)	CRP (100W)	CRP (10W)	CRP (1W)
#1 (PP2)	Base: 8k	QPSK	1.3	100	100	99.9	43.7
		16QAM	2.6	100	99.9	87.5	7.3
	Lite: 4k	QPSK	1.3	100	100	100	67.5
		16QAM	2.6	100	100	95.6	9.5
#2 (8k)	Base: PP1	QPSK	1.1	100	100	100	54.7
		16QAM	2.2	100	100	87.7	12.1
	Lite: PP2	QPSK	1.3	100	100	100	53.8
		16QAM	2.6	100	100	87.8	11.3

Besides, analyzing the influence of the parameters under study, the following results have been obtained. The influence of the parameters is noticeable with lower simulated transmission power levels, especially with 10 W and 1 W:

- Increments in the CRP of up to 20% with QPSK and 8% with 16QAM are obtained with 4k over the CRP values obtained with 8k FFT size, with no decrement in the capacity.

- The Pilot Pattern has little influence on the results, obtaining increments of less than 1% using PP2 instead of PP1.
- The change in the constellation order from QPSK to 16QAM means a decrement in the CRP of between 5% and 13%.

3.2.2.5 Conclusions of this study

This study provides results about the feasibility of DVB-T2 system to offer HD portable indoor reception by means of field trials with a real broadcasting network. Field trials were conducted in real indoor scenarios at pedestrian speed so as to obtain performance measurements of DVB-T2 based on the correct reception percentage of the measured signal. These trials took advantage of some of the new features of the DVB-T2 standard, such as the multiple PLP and the FEF features, being able to measure different combinations of configuration parameters (such as the FFT size, the pilot pattern and the constellation order) at once, in other words, in exactly the same reception conditions.

Besides, it also provides performance results of the indoor reception depending on the transmitted power, obtaining reference results that could be helpful for future network planners. The obtained results demonstrate that a correct reception percentage of more than 95% is possible for SDTV with all the configurations tested both with a transmission power of 300 W and 100 W in portable indoor reception. However, a transmission power of 300 W is excessively high, as 100 W are enough to ensure a CRP higher than 95% for up to 2.6 Mbps of capacity. Indeed, if a capacity lower than 1.3 Mbps is desired for this reception scenario, it is possible to design the network with a transmission power of only 10 W.

On the other hand, considering lower simulated transmission power levels (10 W and 1 W), the influence of the FFT size, the pilot pattern and the modulation schemes in the indoor portable reception has been tested. The 4k FFT size is better than 8k because it allows a higher CRP without decreasing the capacity. The pilot pattern has an almost null influence on the CRP in portable indoor reception. Finally, the change in the modulation scheme from QPSK to 16QAM means an increment in the capacity at expense of a considerable decrement in the CRP.

3.2.3 Study C: Field Trials Based Planning Parameters for DVB-T2 Indoor Reception

The main objective of this study is to characterize in detail the performance of DVB-T2 signals in indoor environments by means of field trials so as to determine the required minimum SNR for a correct indoor portable reception and complete the DVB-T2 performance evaluation process in indoor scenarios. These results could be considered as reference information that could be taken into account by future network planners to establish the necessary transmitted power [109].

In addition, a detailed methodology to obtain the SNR threshold in indoor scenarios is defined. For this purpose, a correct reception criterion for both fixed and pedestrian indoor reception is defined. Moreover, the time variability of the received signal in each measured indoor scenario as well as the location variability of all the measurements is analyzed.

3.2.3.1 DVB-T2 Configuration

One mixed DVB-T2-Base and T2-Lite signal with multiple PLPs was defined and tested in order to test different signal parameters at once. This signal includes two profiles by means of the FEF feature. The main parameters of each tested configuration mode of the mixed signal are shown in Table 8.

The DVB-T2 Base profile has been selected to have high capacity (256QAM), because it is intended for HDTV fixed reception. The highest FFT size and the smallest guard interval have been selected because they provide higher capacity with no noticeable performance degradation in fixed reception. The pilot pattern is the least dense which means lower overhead and, consequently, higher capacity [85].

The DVB-T2 Lite profile has been selected to be more robust because it is intended for mobility reception in difficult scenarios. DVB-T2 Lite includes 3 PLPs with different modulation and code-rate combinations, including the new most robust code-rate of the DVB-T2 profile (1/3). In this case, the FFT size is the one which provides a trade-off between capacity and performance in mobility. The selected guard interval and pilot pattern are also a trade-off between overhead and performance in indoor scenarios [86].

Table 8. DVB-T2 and DVB-T2 Lite Parameters

Main Changing Parameters					
Profile	**Frame length (ms)**	**FFT / IG / PP**	**LDPC FEC length**	**MOD-COD**	**Considered Bitrate (Mbps)**
Base	209.7	32k /1/128 /PP7	64k	256QAM 2/3	23.0
Lite	249.6	8k /1/16 /PP4	16k	QPSK 1/3	1.1
				QPSK 1/2	1.1
				16QAM 1/2	1.1
Main Common Parameters					
Bandwidth (MHz)	Rotated Constelation.	L1 MOD	Interleaving length		
7.71	Yes	BPSK	T2-Frame length (subslicing ON)		

On the whole, the signal was defined so as to maintain a similar capacity (1.1 Mbps) in all the configuration modes dedicated to mobility services in the DVB-T2 Lite component. This bitrate is enough for one SD service when H.264 is used. Besides, if the whole T2-Lite slot is dedicated only to one PLP, the available bitrate will be higher and more SD or HD services could be allocated with H.264. The DVB-T2 Base component is dedicated to high capacity fixed services so its capacity increases up to 23 Mbps enabling the transmission of two or even three HD services with H.264 [99] [100] [101] [102].

This signal is a good example of possible future configurations that could be used to cover any services, from high capacity services in fixed locations, to low capacity services but more robust in challenging scenarios, such as indoor or mobile environments.

3.2.3.2 Experimental Network

Transmission System

Banderas transmission center, shown in Figure 15, was used to cover the urban core of the city of Bilbao, in the north of Spain. It is located on a hill of about 216 m high and about 3 km far from the city centre of Bilbao. Bilbao is a city with an urban centre with medium altitude of around 8 m over the sea level.

The most of the buildings are between 5 and 10 floors with very few skyscrapers.

The transmission equipment for the DVB-T2 mode consisted of:

- DVB-T2 and DVB-T2-Lite modulator that generated the signal at Intermediate Frequency (IF).
- Frequency converter and pre-amplifier that shifted the IF signal to the RF frequency channel 48 (690 MHz) and amplified it to achieve the required level power to input the power amplifiers.
- Power amplifier that provided a total power level of 300 W.
- Computer to configure the DVB-T2 configuration options (modulator) and the transmission power levels (up-converter and main amplifier).
- Radiating system in vertical polarization with a two panel array (16 dBi maximum gain), providing a 5kW Effective Radiated Power (ERP).

Figure 15. Banderas transmission center

The DVB-T2 modulator and the up-converter were connected to the computer, while the computer and the amplifier were directly connected to the transmission internal network. In that way, accessing the transmission internal network through a Virtual Private Network (VPN) by means of a Remote

Desktop application, the equipment was remotely controlled. Besides, a fast configuration of all the parameters associated with the transmitted signals was allowed. Figure 16 shows all the signal and control connections in the transmitter.

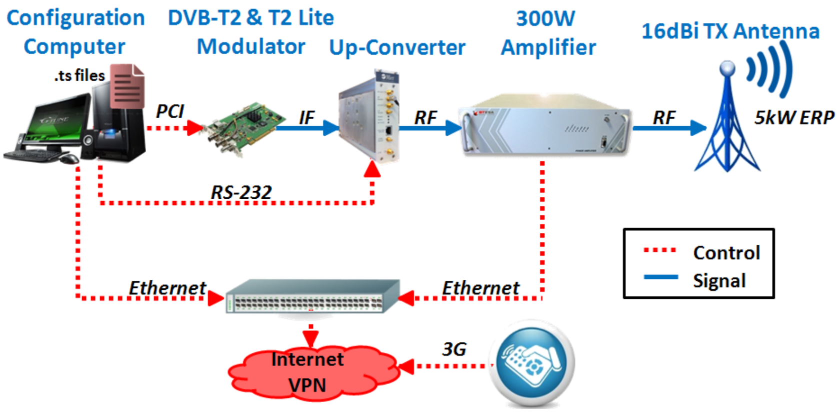


Figure 16. Transmitter scheme

Reception System

The reception system, shown in Figure 17, was based on an online IQ signal recording for posterior offline laboratory processing, as in the Study B [105].

The reception system consisted of a 2 dBi vertical monopole tuned to the transmission frequency of 690 MHz and connected to a channel filter, set to channel 48, to prevent other emissions interfere with desired signal measurement. The received signal was recorded using an IQ recorder equipment in a high speed eSata hard disk as baseband IQ samples for a posterior offline processing stage. Furthermore, during the measurements, information related to the environment was manually recorded (location, height, type of building, type of room, presence of furniture or people around, presence of windows in the room…) so as to obtain information about the reception environment.

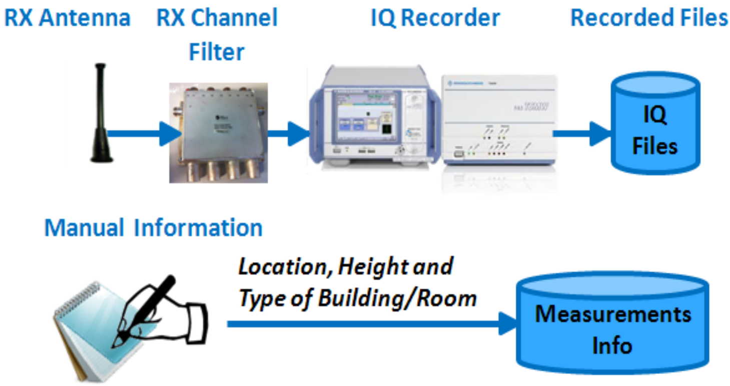


Figure 17. Reception system

Measurement locations

The measurement campaign was conducted in 10 buildings inside the city of Bilbao. Figure 18 shows a map of Bilbao with the location of the transmitter marked with a yellow triangle and the reception locations marked with purple circles.

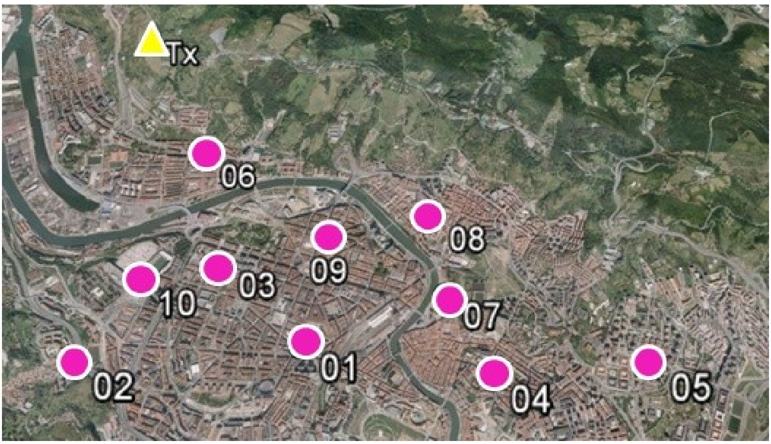


Figure 18. Measurement locations

Fixed and pedestrian indoor scenarios were measured. In a fixed indoor scenario the receiver is static although there could be people moving around it. Furthermore, in a pedestrian indoor scenario the receiver moves at near 3 kmph while there could also be people moving around it.

Eight of the locations consisted in housing buildings. Measurements were carried out in every room in the house, including rooms with and without windows, at different floors and with different wall materials. In addition, data was also recorded in common spaces of the building (landings, as well as vestibules). Figure 19 shows an example of this kind of scenarios. The other two locations were inside public buildings (university and primary school) with many people inside. All these locations have been considered as a representative sample of indoor environments.

Figure 19. View inside one of the measured locations

Table 9 summarizes all the main features of the measured locations. Altogether, 107 different fixed locations and 137 different pedestrian routes were measured in indoor environments.

The first column indicates each location number informing about the floor in Column 2. Column 3 shows the number of fixed measurements recorded in rooms inside its building. Column 4 indicates the number of pedestrian measurements carried out inside each building.

Table 9. Measurements Main Features

Point	Floor	Fixed locations	Pedestrian routes	Point	Floor	Fixed locations	Pedestrian routes
01	3	10	8	06	5	14	8
02	0	10	8	07	4	15	8
03	9	10	8	08	5	10	8
04	5	12	8	09	From 1 to 4	10	53
05	8	12	8	10	4	4	20
TOTAL		**107 fixed locations**				**137 pedestrian routes**	

Recording times

In order to define appropriate recording times in each type of measurements (fixed and pedestrian), different criteria were applied.

In the case of fixed reception in indoor scenarios, several measurements with different recording times were previously performed in the same locations with and without people moving in the surrounding area. Analyzing the signal power along the different tested recording times, it has been checked that the power time variation is almost the same in all the cases. For this reason, it is not necessary to record the received signal during long periods of time. Thus, 15 s recording time has been considered representative enough for the signal power time variability in indoor environments.

On the other hand, in the case of the pedestrian reception in indoor environments, the recording time has been chosen considering the typical size of a room in a housing building. Taking into account that routes of around 8 m length have been carried out inside each room and that 3 kmph is a typical speed of a person walking in an indoor environment, measurements during 10 s are considered to define a route inside a room.

3.2.3.3 Processing Methodology

The processing methodology is very similar to that in Study B with a first step of storing all the data on hard disks during the field trials and a posterior step of processing them in the laboratory with the UPV/EHU professional SDR DVB-T2 Receiver defined in DVB-T2 Receiver Framework in its offline operation mode. The receiver generates multiple quality files as shown in Figure 20. The most relevant were the error rate for each FEC block in a DVB-T2 frame and the SNR per symbol in a DVB-T2 frame.

Figure 20. Offline measurement system

Threshold Criterion

The first step to make a performance study of the measured DVB-T2 configuration mode in indoor environments is based on establishing the correct reception threshold criterion to be applied. As in the Study B: Study B it is based on the FBER [108], in other words the ratio of erroneous FEC blocks received during an observation period established in a T2-frame time.

Due to the establishment of the T2-frame time as the observation period for the defined threshold criterion, all the data obtained from the offline processing (FBER and SNR) has to be combined and reduced to an updating period corresponding to the T2-frame length time. Therefore, the FBER value per T2-frame has been obtained as the percentage of FEC blocks with errors in comparison with the total number of FEC blocks inside the corresponding T2-frame time. The representative value of SNR per T2-frame is the median value in a T2-frame.

As in several locations the reception was always correct and no erroneous FEC block was received, additional noise had to be added to the entire signal so as to simulate the effect of decreasing the SNR value for each T2-frame until several erroneous FEC Blocks appear. The process is based on adding increasing additional noise power to every recorded signal so as to locate the SNR threshold in each case. The limit in the noise power level is reached when erroneous FEC Blocks appear in every recorded signal for any of the configuration parameters sets.

Figure 21 depicts an example of the relation between SNR per T2-frame and the FBER percentage in a real indoor location for one PLP. FBER values per T2-frame (also considering different additional noise power levels in the specific location), are represented with crosses. As it can be seen, the relation is very sharp. This is because of the sharp drop in the relation between the error rate and the SNR of the LDPC codes, which means a change from correct reception to signal loss in just a few tenths of decibel in the SNR value [110].

The FBER percentage values can be easily approximated to a straight line. On that way, a linear approximation has been applied (shown in Figure 7 as a blue line) so as to obtain an instantaneous SNR threshold for each measured location based on the FBER vs SNR relation per T2-frame.

The selected FBER threshold criterion is based on the limit of the percentage of frame length time that can be erroneous. In that way and considering that having a coverage of 99% time is considered as good reception in mobility [61], a threshold criterion based on having 1% FBER has been established.

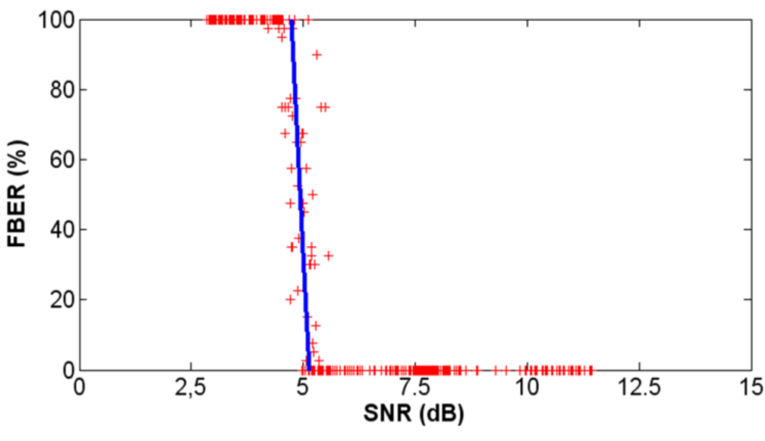


Figure 21. SNR vs FBER

Methodology for Obtaining the SNR threshold

The data obtained from the offline processing with the professional receiver (FBER and SNR per T2-frame) is the starting point for the performance analysis. The methodology to obtain the reference SNR threshold in fixed and pedestrian indoor scenarios is shown in Figure 22.

The starting point calculates the threshold criterion based on 1% FBER to every recorded signal (both in fixed and pedestrian measurements), obtaining an instantaneous SNR threshold per location/route. These instantaneous SNR thresholds should be referred to 50% of time as in planning tools. In this case, each location is mainly characterized with its median SNR or median power level in the time domain, which is the information usually used by broadcasters. Thus, the instantaneous thresholds referred to 50% time (Step 1 in Figure 22) are obtained.

The second step carries out a statistical study of the instantaneous SNR threshold from Step 1, obtaining the median value of the instantaneous SNR thresholds of all the locations/routes (*SNRinst*). By this way, the SNR threshold for the 50% of the time and 50% of the locations is obtained (Step 2 in Figure 22).

In digital services, it is typical to cover the 99% time [61], so it is important to determine the signal variation and obtain the SNR threshold value for this percentage of measured time. For this reason, the following step (Step 3 in Figure 22) consists in analyzing the time variability of the SNR in each location/route so as to obtain the necessary time correction factor between covering the 50% and 99% of the signal time in each location/route. Next, a statistical study of the time correction factors of each location/route is done in order to obtain a generic and common Time Correction Factor (TCF) to cover the 99% signal time in an indoor environment. This common time correction factor is applied to the SNR threshold for 50% time and 50% locations from Step 2 resulting in a SNR threshold value for the 99% time and 50% locations.

The final step (Step 4 in Figure 22) lies in making a statistical study of the instantaneous SNR threshold values for all the locations from Step 1 so as to obtain the Location Correction Factor (LCF) to cover the 95% of the locations. The LCF is applied to the SNR for 99% time and 50% locations from Step 3 resulting in a unique SNR threshold value that covers the 95% of the locations [61]. This threshold value is an overall SNR threshold value for the 99% of the time and the 95% of the locations.

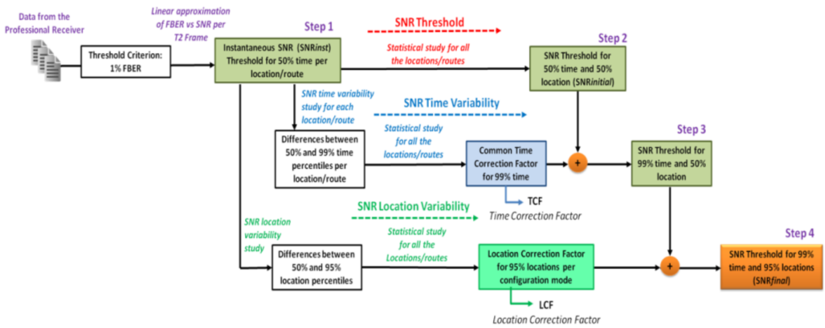


Figure 22. Methodology for obtaining the SNR Threshold

Taking all above into consideration, the SNR threshold in indoor reception is given by equation (3):

$$SNRfinal\,(dB) = SNRinitial\,(dB) + TCF(dB) + LCF(dB) \quad (3)$$

Equation (3) summarizes that the SNR threshold (*SNRfinal*) per each configuration parameters set is initially given by the SNR for 50% time and 50% locations (*SNRinitial*) obtained by means of 1% FBER criterion (Step 2). This value has to be updated with the *TCF* (Step 3) and the *LCF* (Step 4) so as to cover the 99% of the time and the 95% of the locations.

Laboratory Reference Values

Apart from the results obtained in the field trials, some laboratory measurements were also carried out so as to obtain the referenced values for indoor reception with emulated channel models. F1 and P1 theoretical channels [59] will be considered for fixed indoor reception. Additionally, it is shown in [75] that a specific channel model was needed in order to simulate the pedestrian indoor reception. This was the PI [76] channel model. Besides, the ITU also proposed two channel models intended for pedestrian indoor reception, IOA and IOB [97].

Figure 23 describes the laboratory measurements set-up using the same receiver as in the field trials but emulating the channels by means of the HW channel simulator used in Study A. The methodology applied is the one described in "Methodology for Obtaining the SNR threshold" section but considering that the time variation is null (TCF = 0 dB) for the channel models considered for fixed indoor reception, while the spatial variation is null (LCF = 0 dB) for every simulated channel model.

Figure 23. Laboratory measurements set-up

3.2.3.4 Studies and Results

SNR Threshold for 50% time and 50% locations

Applying the 1% FBER criterion for every measured location/route during the 15 s recording time in fixed reception and during 10 s in case of pedestrian reception, the instantaneous SNR threshold values per location/route (Step 1 in Figure 22) are shown in Figure 24 and Figure 25 for fixed and pedestrian reception, respectively. The straight lines show the median instantaneous SNR value for each DVB-T2 configuration mode considering all the measured locations.

Numbers in the abscissa axis of Figure 24 and Figure 25 indicate the amount of measurement locations on which the threshold situation has been reached for each DVB-T2 configuration parameters set. This number depends on the robustness of the configuration parameters, showing in general fewer locations for the least robust configuration mode (256QAM) on which it was not possible to reach the threshold situation because it was directly received with more errors than those of the 1% FBER criterion.

Furthermore, although the highest number of measurement locations should be for the most robust configuration mode (QPSK 1/3), the situation is different. This is because it is necessary to establish a tradeoff between the additional processing time required for reaching the threshold situation in new locations and the number of measurements that are statistically representative. Thus, the statistical analysis results will be the same with a higher number of measurements.

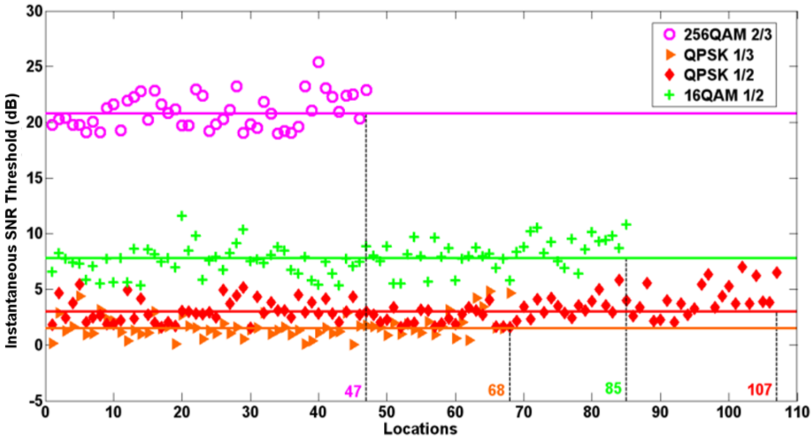


Figure 24. Instantaneous SNR thresholds for measured fixed locations

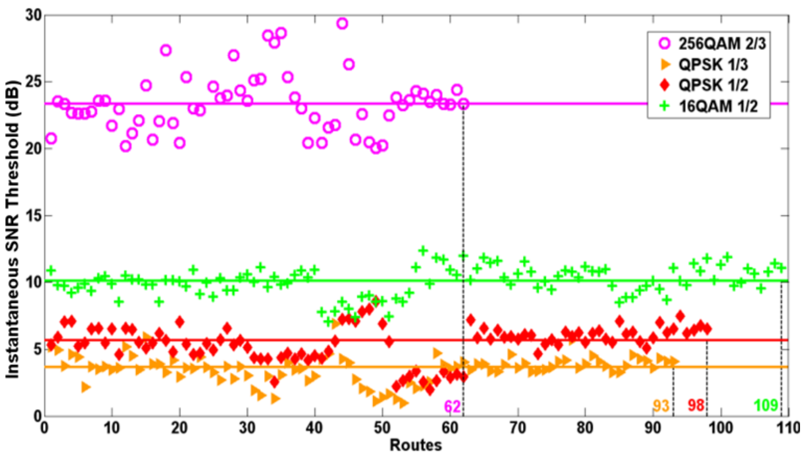


Figure 25. Instantaneous SNR thresholds for measured pedestrian routes

Table 10 shows the median values of the obtained instantaneous SNR values per location/route and for both fixed and pedestrian scenarios. These results are important due to the fact that they are the reference values broadcasters take into consideration and are also used in network planning tools. The results included in Table 10 are the SNR thresholds for 50% time and 50% locations (Step 2 in Figure 22) and its variance, obtained from the previous results in dB.

Table 10. Threshold for 50% time and 50% locations Main Statistical Features (dB)

Modes	Fixed Scenario		Pedestrian Scenario	
	Median	Variance	Median	Variance
Base: 256QAM 2/3	20.7	2.2	23.3	5.4
Lite: QPSK 1/3	1.5	1.0	3.7	1.0
Lite: QPSK 1/2	3.0	1.2	5.7	1.8
Lite: 16QAM 1/2	7.8	1.4	10.1	1.4

As it can be seen, the median value of the instantaneous SNR threshold in pedestrian scenarios increases between 2.2 and 2.7 dB depending on the configuration mode. However, the variance is mainly the same in pedestrian than in fixed scenarios. This is because the receiver movement has less influence on the SNR threshold than the changing surrounding environment. For this reason, the variance in static and pedestrian reception is similar because it is only affected by the surrounding scenario. The variance for the 256QAM parameters set is higher, especially in case of pedestrian reception. The reason for this high value may be the high SNR threshold of this configuration as it is not suitable for indoor reception.

SNR Threshold for 99% time and 50% locations

The instantaneous SNR per location for the 50% time obtained in "SNR Threshold for 50% time and 50% locations" section (Step 2) has to be updated to the 99% of the signal time [61]. For this purpose, the time variability of the SNR in each location has to be analyzed to obtain the necessary correction factor [111].

Figure 26 shows an example of the process of obtaining the time correction factor for the 99% time in one location/route. "D99" is the difference between the SNR for 99% and 50% time .In other words, the difference between the SNR value that is exceeded during the 99% of the time and during the 50% of the time. "D99" is the time correction factor that should be applied to the SNR threshold for the 50% time in each location/route (from Step 1) because of the time variability of the signal.

This correction factor has been obtained for all the measured locations/routes. "D99" is the same for every configuration parameters set in a DVB-T2 mode because it only depends on the signal power level received during the recording time (or SNR). Red crosses in Figure 27 and Figure 28 show the obtained "D99" values for each fixed and pedestrian measurements, respectively. As it can be seen, "D99" varies in each location as the SNR time variability also does.

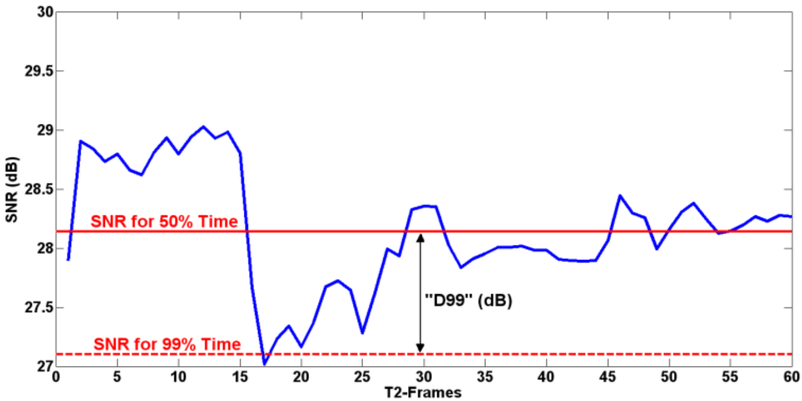


Figure 26. Time variability correction example

In order to have a unique and common correction value for all the locations that could be applied in any case, the variation of "D99" correction factor is analyzed for all the locations. For this reason, a statistical study of the "D99" variation has been done showing the main significant features (median, 75% and 95% percentiles) in Figure 27 for fixed reception and in Figure 28 in case of pedestrian reception.

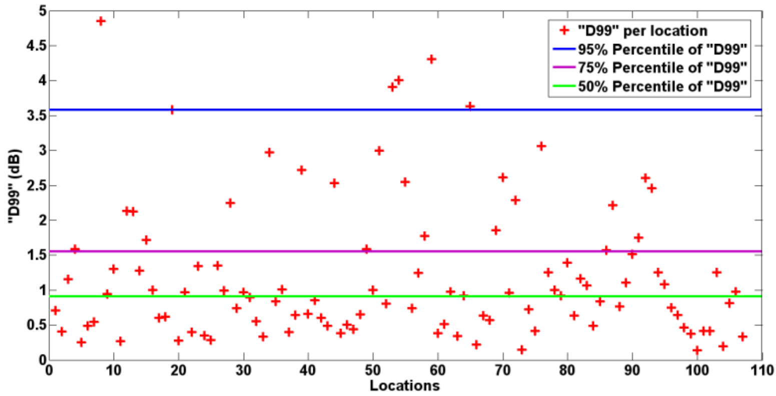


Figure 27. "D99" statistical analysis in fixed locations

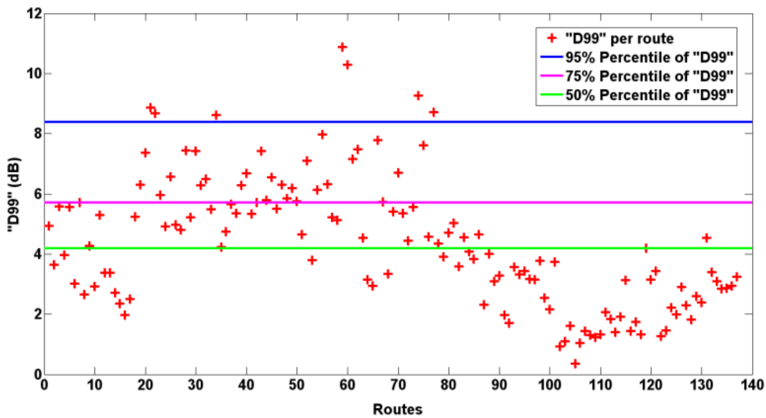


Figure 28. "D99" statistical analysis in pedestrian routes

In addition to obtaining the time variability correction factor for the 99% of the time ("D99") as it is typical in digital services, other time percentages can be also studied. The main percentages of time to consider besides 99% are 100% ("D100") and 95% ("D95"), because they are typical percentages to consider for the time variability [61]. The process to obtain them is the same as for "D99" but with different percentages of time.

Table 11 shows the main statistical features of the time correction factors (TCF) (“D100”, “D99” and “D95”) in fixed and pedestrian indoor locations.

Table 11. Time Variability Correction Factor (TCF) (dB): Statistical Analysis in Fixed and Pedestrian Scenarios

Statistical Features	Fixed Time Correction Factor (TCF)			Pedestrian Time Correction Factor (TCF)		
	“D100”	**“D99”**	**“D95”**	**“D100”**	**“D99”**	**“D95”**
100%	4.9	4.8	4.1	11.7	11.7	9.9
99%	4.6	4.6	4.0	10.6	10.6	9.2
95%	3.5	3.5	2.9	8.2	8.1	7.5
75%	1.7	1.6	1.4	5.7	5.6	4.9
50%	0.9	0.9	0.8	4.2	4.1	3.6

In fixed scenarios, as it can be seen in Table 11, the differences between “D100” and “D99” are always lower than 0.1 dB while the differences between “D99” and “D95” can be up to 0.7 dB. Equally, in pedestrian reception, the differences between “D100” and “D99” are always lower than 0.1 dB but this differences increase up to 1.8 dB when comparing “D99” with “D95”. This means that the time variability correction factor for the 99% time (“D99”), which is the typical one for digital services, is restrictive enough without involving huge degradation in the TCF in comparison with other time percentages (always lower than 0.1 dB).

The statistical features of “D99” in Table 11 show that 100% and 99% percentiles are too restrictive with increments of around 1.2 dB in fixed reception and between 2.5 and 3.6 dB in pedestrian reception in comparison with 95%. Considering that 50% and 75% percentiles are not restrictive enough, the 95% percentile statistical feature, which is considered as good reception [61], is the one to consider for the time variability correction factor in both fixed and pedestrian indoor scenarios.

Taking all into consideration, the time correction factor in a fixed indoor environment is established in 3.5 dB while it increases up to 8.1 dB in case of pedestrian indoor scenarios. For this reason, the time variability of the received signal in indoor environments has more influence on the pedestrian than on the fixed reception. The receiver movement in pedestrian indoor scenarios in comparison with fixed scenarios means an increment of 4.6 dB in the TCF for covering the 99% time.

The SNR thresholds values for 50% time and 50% locations from Step 2 are updated with this time correction factor to cover the 99% time. The results are summarized in Table 12, resulting in SNR thresholds for 99% time and 50% locations (Step 3 in Figure 22).

Table 12. SNR Threshold for 99% time and 50% locations (dB)

Modes	Fixed Scenario	Pedestrian Scenario
	Median	Median
Base: 256QAM 2/3	24.2	31.4
Lite: QPSK 1/3	5.0	11.8
Lite: QPSK 1/2	6.5	13.8
Lite: 16QAM 1/2	11.3	18.2

As it can be seen, the median value of the SNR threshold for 99% time in pedestrian scenarios increases between 6.7 and 7.3 dB in comparison with the fixed SNR thresholds, depending on the configuration mode. Besides, the variance remains as in "SNR Threshold for 50% time and 50% locations" section, as the time correction factor applied is common to all the DVB-T2 configurations.

SNR Threshold for 99% time and 95% locations

Once the TCF has been assessed, the location variability has also to be established [112]. The study of the location variability depends on the specific configuration parameters set because it is based on a study of the SNR thresholds in different locations.

The instantaneous SNR threshold from Step 1 is different in each measured location and a statistical study of the location variation has to be carried out. Therefore, the main statistical features of the location variability (75% and 95% percentile values) have been obtained for each configuration mode.

A good reception is supposed if the 95% of the locations are covered, while an acceptable reception is in case of 75% [61]. For this reason, "D95 and "D75" are obtained. "D95" is the difference between the SNR for 95% and

50% locations, while "D75" is the difference between the SNR for 75% and 50% locations. These are the location correction factors that should be applied to the SNR threshold for the 99% time and 50% locations (from Step 3) because of the location variability of the signal.

Figure 29 and Figure 30 show the Cumulative Distribution Function (CDF) for instantaneous SNR thresholds (Step 1) depending on the locations/routes for fixed and pedestrian reception, respectively.

The dotted black lines and their associated values are the SNR threshold for the 50% time and for typical percentiles of locations, such as 50%, 75% (acceptable reception) and 95% (good reception). For this reason, the LCF "D95" and "D75" are shown in Figure 29 and Figure 30 in blue for one exemplifying configuration parameters set. These LCF should be applied to the SNR threshold from Step 3 and are different for each configuration mode.

As it can be seen in Figure 29 and Figure 30, the slope of the CDF curve for each configuration parameters set indicates the influence of the location variability in each case. If the curve was perfectly vertical, all the measured locations/routes would have the same instantaneous SNR threshold for 50% time and the location correction factor in these cases would be null. However, different measured points have different instantaneous SNR threshold for 50% time, and for this reason, there is a slope in the CDF curve. The higher the slope is, there is more influence of the location variability on the SNR threshold.

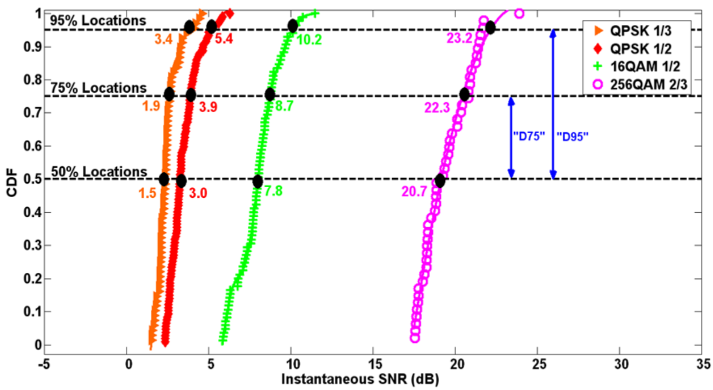


Figure 29. CDF of instantaneous SNR threshold for 50% time depending on the fixed locations

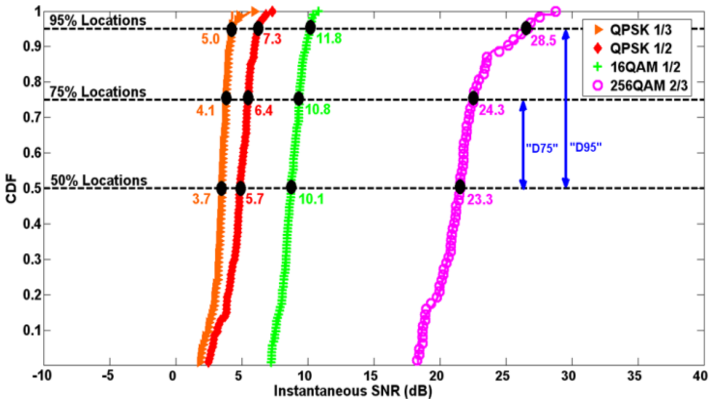


Figure 30. CDF of instantaneous SNR threshold for 50% time depending on the number of routes

Table 13 shows the LCF "D75" and "D95" for both fixed and pedestrian reception for each configuration mode.

Table 13. Location Variability Correction Factor (LCF) (dB)

Modes	Fixed Scenario		Pedestrian Scenario	
	"D95"	"D75"	"D95"	"D75"
Base: 256QAM 2/3	2.5	1.6	6.4	1.0
Lite: QPSK 1/3	1.9	0.4	1.3	0.4
Lite: QPSK 1/2	2.4	0.9	1.6	0.7
Lite: 16QAM 1/2	2.4	0.9	1.7	0.7

As it can be seen in Table 13, the more robust the DVB-T2 configuration is, the lower influence on the location variability is presented both fixed and pedestrian reception. As all the configuration modes were measured at once, the changes in the measuring location/route, and the consequent change on the propagation channel, has an increasing influence according to the decreasing configuration robustness. It has been previously demonstrated [83], that the higher the code-rate robustness is, the propagation channel has lower influence on the required SNR spatial variability. For this reason, modes with the same code-rate present similar variability results

All in all, it is not possible to establish a common and unique location variability correction factor to cover higher percentages of locations and thus, the location variability has to be considered individually for each configuration parameters set.

In fixed reception, considering an increment from 50% to 75% on the location percentage, increments on the SNR threshold between 0.4 and 1.6 dB should be considered. In case of covering 95% of the locations, the increment on the SNR threshold is between 1.9 and 2.5 dB. In case of pedestrian reception, the increment to cover 75% locations is between 0.4 and 1.0 dB, while the location correction factor to cover 95 % locations depending on the configuration parameters set varies from 1.3 and 6.4 dB. This last high LCF is for the 256QAM configuration mode, which presents a high variance on the SNR threshold for 99% time, due to the fact that its high instantaneous SNR threshold is not appropriate for pedestrian indoor reception. The results are similar to those proposed in [112].

If fixed and pedestrian LCF are compared, it can be seen that the location variability for pedestrian reception is mainly equal or even lower than in fixed scenarios. This might be because the receiver movement may counteract some of the changing environment effects on the reception, obtaining a similar or even lower LCF.

Anyway, from Table 13, the "D95" location correction factor, which is considered as a "good" reception [61], is the one that should be applied to the SNR threshold for 99% time from Step 3. Thus, the SNR threshold for 99% time and 95% locations obtained after applying all the processing methodology for both fixed and pedestrian indoor scenarios are obtained (Step 4 in Figure 22). These SNR thresholds can be a reference value to be considered for fixed and pedestrian indoor reception (with the exception, again, of the 256QAM configuration, whose high SNR threshold are not appropriate for pedestrian reception).

All in all, Table 14 and Table 15 summarize the main DVB-T2 planning parameters that can be considered as reference for broadcasters and can be taken into account in planning tools, for both fixed and pedestrian indoor reception. The first column is the median SNR threshold (SNR for 50% time and 50% locations) (Step 2 in Figure 22). The second and third columns are the TCF and LCF respectively. The forth column is the SNR threshold for 99% time and 95% locations (Step 4 in Figure 22).

Table 14. DVB-T2 Planning Parameters in Fixed indoor scenarios (dB)

Modes	Fixed Scenarios			
	50% / 50% SNR	TCF	LCF	99%/95% SNR
Base: 256QAM 2/3	20.7	3.5	2.5	26.7
Lite: QPSK 1/3	1.5		1.9	6.9
Lite: QPSK 1/2	3.0		2.4	8.9
Lite: 16QAM 1/2	7.8		2.4	13.7

Table 15. DVB-T2 Planning Parameters in Pedestrian indoor scenarios (dB)

Modes	Pedestrian Scenarios			
	50% / 50% SNR	TCF	LCF	99%/95% SNR
Base: 256QAM 2/3	23.3	8.1	6.4	36.6
Lite: QPSK 1/3	3.7		1.3	13.1
Lite: QPSK 1/2	5.7		1.6	15.4
Lite: 16QAM 1/2	10.1		1.7	19.9

As it is shown in Table 14 and Table 15, the location variability influence on the SNR threshold is lower than the time variability in both scenarios. This is because the indoor propagation channels (both fixed and pedestrian) are time variant but they are similar in every location.

Taking the 99% time and 95% locations requirements into consideration, the increment in SNR threshold between fixed and pedestrian indoor reception is between 6.2 and 6.5 dB with the exception of the 256QAM configuration mode (the increment increases up to 9.9 dB), which it is not ideal for indoors.

Comparison with reference values

The SNR threshold obtained for the fixed indoor reception (by means of field trials) will be compared with the F1 and P1 channels, considering results with laboratory measurements and the reference values included in the DVB-T2 implementation guidelines (IG) [59]. In pedestrian scenarios, the SNR threshold for the real pedestrian indoor channel (in field trials) will be compared with the PI channel [76] and IOA / IOB [97] (with laboratory measurements). In this case, there are no reference values in the DVB-T2 implementation guidelines.

The laboratory measurements were carried out following the same methodology as in the case of the field trials but without considering the Step 4 as the location variability is null (the emulated channel do not present this variability). In case of fixed reception (F1 and P1), the time variability is also null as the channel power is constant during all the measured time. For this reason, in these cases, the time correction factor is also null and Step 3 in Figure 22 does not have to be applied. However, in pedestrian reception (PI, IOA and IOB), the channel power varies with the time and the corresponding TCF has to be obtained and applied (Step 3 in Figure 22). Indeed, "D99" for the PI propagation channel model during 10 s is 1.9 dB while this value increases up to 4.8 dB in case of IOA and 4.2 dB for IOB channel models.

Table 16 summarizes the comparison of the SNR threshold results in fixed indoor scenarios between reference values with simulations (from the DVB-T2 Implementation Guidelines), laboratory measurements with typical fixed indoor propagation channel models and measured results by means of field trials. As the SNR thresholds for theoretical fixed channel models are for 50% time and 50% locations (TCF and LCF are null), the SNR thresholds for field trials considered to be compared are also for 50% time and 50% locations.

Table 16. SNR Thresholds in Fixed Indoor Scenarios (dB)

Modes	Reference value (IG)		Laboratory measurements		Field trials
	F1	P1	F1	P1	50% / 50%
Base: 256QAM 2/3	18.4	20.4	19.1	21.2	20.7
Lite: QPSK 1/3	-- (*)	-- (*)	-0.1	1.7	1.5
Lite: QPSK 1/2	1.4	2.5	2.0	3.5	3.0
Lite: 16QAM 1/2	6.2	7.4	6.6	8.3	7.8

(*)"--": SNR Threshold reference values in IG do not exist for T2 Lite new code-rates

As it can be seen in Table 16, the laboratory measurement results for F1 and P1 channel models compared with the reference values from IG show the implementation losses of the receiver and the FBER criterion applied. These losses are slightly different depending on the DVB-T2 configuration parameters although its value remains always lower than 1 dB with independence of the channel model [106][113].

Comparing the SNR threshold values obtained from laboratory measurements and those obtained from field trials, it can be stated that the actual fixed indoor channel is similar to the P1 theoretical channel model (with differences always lower than 0.5 dB). For this reason, it can be concluded that the P1 channel model, which is usually considered to model the fixed indoor reception, is appropriate for emulating an actual indoor reception, as it was similarly concluded in [114].

In case of pedestrian indoor reception, Table 17 summarizes the comparison of the SNR threshold results in fixed indoor scenarios between reference values by means of simulations (from IG), laboratory measurements with typical fixed indoor channel models and measured results based on field trials. As the SNR thresholds for theoretical fixed channel models are for 50% time and 50% locations (TCF and LCF are null), the SNR thresholds from field trials considered to be compared are also for 50% time and 50% locations.

Table 17. SNR Thresholds in Pedestrian Indoor Scenarios (dB)

<table>
<tr><th rowspan="3">Modes</th><th colspan="6">Laboratory measurements</th><th colspan="2" rowspan="2">Field trials</th></tr>
<tr><th colspan="2">PI</th><th colspan="2">IOA</th><th colspan="2">IOB</th></tr>
<tr><th>50% / 50%</th><th>TCF</th><th>50% / 50%</th><th>TCF</th><th>50% / 50%</th><th>TCF</th><th>50% / 50%</th><th>TCF</th></tr>
<tr><td>Base: 256Q 2/3</td><td>20.9</td><td rowspan="4">1.9</td><td>22.4</td><td rowspan="4">4.8</td><td>22.1</td><td rowspan="4">4.2</td><td>23.3</td><td rowspan="4">8.1</td></tr>
<tr><td>Lite: QPSK 1/3</td><td>1.2</td><td>3.4</td><td>2.5</td><td>3.7</td></tr>
<tr><td>Lite: QPSK 1/2</td><td>2.9</td><td>5.4</td><td>4.4</td><td>5.7</td></tr>
<tr><td>Lite: 16QAM 1/2</td><td>8.0</td><td>10.2</td><td>9.5</td><td>10.1</td></tr>
</table>

Comparing the SNR results for 50% time and 50% locations obtained from laboratory measurements and those obtained from field trials, it can be stated that the actual pedestrian indoor channel is similar to the IOA theoretical channel model, with differences lower than 0.3 dB (with the exception of the 256QAM configuration, where the high variability of the SNR threshold increases the differences up to 0.9 dB). However, the TCF factor in the actual pedestrian indoor scenario is much higher than that obtained in the laboratory for the three simulated channel models, with differences between 3.3 dB for IOA and 6.2 dB for PI. For this reason, the IOA channel model is the good estimation for obtaining the SNR threshold in pedestrian indoor scenarios but the time variability of the signal should be increased in 3.3 dB.

3.2.3.5 Conclusions of this study

This study completes the DVB-T2 performance evaluation and aims at improving the accuracy of the recommended SNR threshold values for the DVB-T2 system so as to be a good reference to be taken into account by broadcasters or in planning tools. For this purpose, a detailed methodology has been established and followed. This was based on three main phases: obtaining an instantaneous SNR threshold in each location/route; next, correcting the previous values considering the signal time variability and, finally, applying the location variability correction.

Table 14 and Table 15 summarize the main DVB-T2 planning parameters for fixed and pedestrian indoor reception, respectively. This information includes SNR thresholds for 50% time and 50% locations (which are necessary in planning tools), time and location variability correction factors, and SNR thresholds for 99% time and 95% locations (which are the actual thresholds to achieve the required coverage percentages in digital services.

The differences in the SNR thresholds for 99% time and 95% locations for fixed and pedestrian reception are mainly due to the different time variability of the received signal between fixed and pedestrian reception which supposes approximately an increase of 3.5 dB in fixed but 8.1 dB in pedestrian indoor scenarios. These time variability correction factors are valid not only for DVB-T2 but also for other digital 8 MHz OFDM technologies in UHF band.

Furthermore, the location variability influence is lower than the time variability. In fixed scenarios the location correction factor value reaches up to 2.4 dB, while in pedestrian the value is mainly up to 1.7 dB (with the exception of 256QAM configuration which, as mentioned before, is not a valid configuration for indoor environments).

Taking all into account, the increment in the SNR threshold so as to allow the pedestrian indoor reception in comparison to just fixed indoor reception is mainly between 6.2 and 6.5 dB (reaching 9.9 dB in case of 256QAM).

Apart from that, a comparison between the SNR threshold obtained in the field and those obtained in the laboratory for suggested theoretical channel models has been presented in Table 16 and Table 17. It is suggested that the P1 channel model perfectly fits the fixed indoor reception. In case of pedestrian indoor reception, the IOA channel model is the most appropriate. However, a time correction factor of extra 3.3 dB should be applied.

These results show the feasibility of DVB-T2 to offer HD quality services as configurations with a capacity of about 10 Mbps (QPSK 1/2) can be correctly received for SNR higher than about 8.9 dB or 15.4 dB in fixed and portable indoor scenarios, respectively. However, as rooftop services have often to be simultaneously transmitted, part of the frame time should be dedicated to the rooftop service, remaining at least about 3-4 Mbps, which is enough for at least one HD program with H.264.

4. Summary

In this chapter, the DVB-T2 system performance has been widely studied. First, a state of the art has been carried out concluding that fixed and mobile reception performance has been widely studied in the bibliography in the three phases of a performance evaluation process (computer simulations, laboratory measurements and field trials). However, DVB-T2 reception in indoor environments was an almost non-studied scenario that had to be analyzed more in detail, especially with hardware equipment (laboratory measurements and field trials) to check and evaluate the DVB-T2 indoor performance.

In order to complete the DVB-T2 indoor research work, three new studies have been carried out during the thesis development. These studies gather some performance reference results by means of laboratory measurements under several indoor channel models (Study A and Study C) as well as real environments by means of field trials (Study B and Study C). For this purpose, in addition to commercial receivers, a custom professional SDR receiver developed by the TSR research group and defined in DVB-T2 Receiver Framework has been considered.

First, several DVB-T2 configurations have been considered in the different studies concluding that a suitable configuration for indoor reception should have an appropriate combination of FFT size, guard interval and pilot pattern, taking into account that small FFT sizes and denser pilot patterns are more suitable for challenging portable scenarios indoors as concluded in Study B. Moreover, modulation and code-rate values should ensure the robustness needed. For this reason, low order constellations (up to 16QAM) and robust code-rates should be considered. Finally, due to the presence of slow fading in portable reception, a long time interleaving option should be considered so as to improve the system performance. These configuration features make the DVB-T2 indoor reception possible.

Furthermore, a new detailed methodology for performance study in indoor scenarios has been defined in Study C. This is based on three main phases: obtaining an instantaneous SNR threshold in each measured location; next, correcting the previous values considering the signal time variability and, finally, applying the location variability correction. The measured degradation due to the time variability is of 3.5 dB in fixed indoor reception and 8.1 dB in portable indoor reception. The degradation because of the location variability in indoor scenarios is of 2.4 and 1.7 dB for fixed and portable indoor reception, respectively. These degradations are valid not only for DVB-T2 but also for other digital 8 MHz OFDM technologies in UHF band.

Regarding to portable indoor reception, as there are so many different standardized channel models and with quite different required SNR thresholds (with differences up to 9.2 dB) as concluded in Study A, a reference model cannot be easily defined. The obtained results show that typically considered PI and TU6 channel models are very optimistic. However, it has been concluded in Study C that the IOA channel model can be considered appropriate to model the pedestrian indoor reception but with an extra degradation of 3.3 dB because of too low time variability present in this model. In general, the increment in the SNR threshold so as to allow the pedestrian indoor reception in comparison to just fixed indoor reception is about 6.5 dB.

Finally, Study C performance results conclude that DVB-T2 configurations with a capacity of about 10 Mbps can be correctly received in fixed indoor scenarios for SNR higher than about 8.9 dB. Similarly, these configurations require a SNR higher than about 15.4 dB to be correctly received in portable indoor scenarios based on Study A and Study C performance results. As rooftop services have often to be simultaneously transmitted, part of the frame time is dedicated to the rooftop service, remaining at least about 3-4 Mbps, which is enough for at least one HD program with H.264. Besides, preliminary coverage results from Study B show that the portable (and consequently also fixed) indoor reception is possible with currently considered transmission powers in existing broadcasting networks. In fact, the considered example shows that the power is oversized and DVB-T2 configurations with 10 Mbps capacity can show good coverage for much lower powers.

With this new research work, the DVB-T2 system performance, including some of its more advanced features, has been widely evaluated with laboratory measurements and field trials. Moreover, a new methodology for analyzing an OFDM system performance in indoor scenarios has been totally defined updating the existing methodologies.

CHAPTER III: Studies of New Techniques for Next Generation DTT Systems

In this section, some modifications to the DVB-T2 standard are suggested and tested in order to check its influence on the reception performance, especially for the new target scenarios, including mobile and indoor reception.

First, the most recently developed FEC codes and modulation schemes will be applied to the current DVB-T2 system. Next, the new Layered Division Multiplexing (LDM) technique will be tested so as to offer mobile and UHD services simultaneously. Finally, some optimizations to the current DTT receivers' algorithms will be presented in order to improve the performance with multilayer signals and portable and mobile new target scenarios.

1. Introduction

The DVB-T2 standard was first published in September 2009 with the cutting-edge technologies of the moment, what supposed high improvements in comparison to the previous standards. However, since then, new processing capabilities and new technologies have been developed enabling a new broadcasting system with more available capacity and robustness. By this way, the system performance can be improved over the DVB-T2 performance.

For this reason, the thesis includes some research work about some of the new technologies or features that have been developed since the DVB-T2 standardization process, with a detailed state of the art of their performance. Besides, some new performance analyses have been also included in the thesis in order to test the performance gain over the original DVB-T2 specification checking their suitability for a new DTT system.

On the one hand, the performance of new LDPC codes has been presented by means of field trials. These new LDPC codes improve the performance with a BER vs SNR relation nearer Shannon while highly reduce their complexity at the receiver. On the other hand, the LDM and SHVC techniques are presented and tested in the laboratory so as to provide different services at the same time with a better capacity and robustness trade-off than other multiplexing techniques. In both cases, the considered scenario is indoors as it was not as widely analyzed as fixed and mobile scenarios, whose performance is included in the state of the art analysis.

Moreover, some improvements to the typical DTT receivers' algorithms implementation have been presented and tested. First, a new LLR reliability formula optimized for multilayer signals such as those using LDM has been presented. Next, the inter-carrier interference (ICI) effect on mobile reception has been considered as an additional interference source. Several low complexity ICI noise estimators have been evaluated to establish the one with the best performance. Besides, a new method has also been proposed considering an experimental analysis.

Taking everything into consideration, this chapter includes a detailed state-of-the-art including the performance studies related to the considered new techniques for the next generation DTT systems. Additionally, new research work related to the performance of new LDPC codes and LDM technique is also included by means of two different studies. Finally, two new algorithms have been defined and tested in other two different studies in order to improve the reception performance.

2. Previous Studies

Since the DVB-T2 standard definition, several new technologies have been developed for the next generation DTT systems so as to improve the capacity, robustness, efficiency or performance of the systems. However, not all of them have shown real improvement over the existing ones. In this section, the state of the art of the most promising innovations has been included.

The major improvements have been performed in the BICM module. On the one hand, the traditional uniform constellations (UCs), characterized with uniform spacing between constellations points and square shape, and considered in several systems such as DVB-T2 because of their simplicity for encoding and decoding operations, have been improved with the development of non-uniform constellation (NUCs). NUCs have been proposed to reduce the gap between UCs performance and the Shannon limit, and provide a better performance, reducing the required SNR [115] [116] [117]. It has been demonstrated that the capacity of a communication system can be increased if all the constellation points are adapted to the SNR range in which the system is expected to work. This means that for each SNR value, there is one location for each constellation symbol which maximizes the system capacity. Consequently, different NUC can be defined for each specific coding value, since the optimum shape of the constellation depends on the operating SNR.

Indeed, NUCs can be optimized with respect to one of the axis (dividing the QAM in two pulse amplitude modulations and optimizing one of them) so, resulting in one dimension non uniform constellations (1D-NUC), with still rectangular but not uniform shape as it can be seen on the left side of Figure 31. Afterwards, an additional degree of freedom has been considered in the design of NUC resulting in two dimensional non uniform constellations (2D-NUC), with circular shape as it can be seen on the right side of Figure 31. For most cases, the capacity gain of 2D-NUC is higher than 1D-NUC. However, the calculation and processing complexity is higher, especially for high order constellations, due to the high number of variables involved as each symbol is optimized with respect to both I and Q axes [118].

Additionally, one of the special features of DVB-T2 relies on the possible use of rotated constellations [119] [120], also known as Signal Space Diversity (SSD). This technique usually leads to an additional diversity improving the receiver performance when severely faded channels are encountered at the receiver side, especially with low order constellations and high code-rates. It has been demonstrated that its combined use with NUC degrades the system performance [121]. More studies related to the NUCs and modifications over them are being conducted at the moment in another thesis research work.

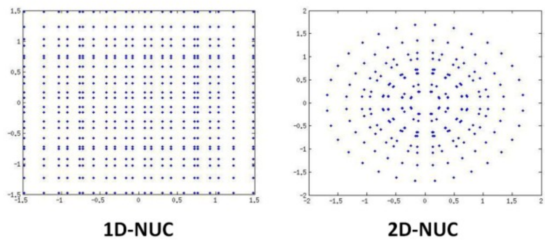


Figure 31. Examples of 1D-NUC (on the left) and 2D-NUC (on the right)

On the other hand, many efforts have been dedicated to the improvement of FEC codes in order to withstand with very noisy environments, such as complex mobile and indoor scenarios, where the total interference power (noise, multipath and co-channel interference) may be even larger than the desired signal level. In this context, newly designed QC LDPC codes [122], which can be easily shortened from their mother code to higher code-rates with a relatively good performance, have been developed. For example, when the SNR is high, the 1/4 code-rate can be truncated in a 50% of its length to form a 1/2 code-rate. By this way, the latency and computation complexity are reduced by 50%. In addition, when the SNR is 3 dB above the threshold value of the specific LDPC code, the required number of iterations is reduced by 90%.

Taking into account that the most of the broadcasting coverage area has SNR values about 3 dB higher than the required, full error correction capabilities that are designed for the lowest SNR are not needed in those cases. In this area, receivers can take advantage of the shortening capability of the LDPC code to achieve better power efficiency, in other words, longer battery life. Furthermore, the recently developed codes, specially designed to work under very low SNR scenarios, can significantly outperform those included in the DVB T2 standard.

Taking advantage of the puncturing capability of the rate compatible QC LDPC codes, they can be used to perform product code FEC blocks with lower complexity than traditional concatenated codes [123], such as a two-dimensional LDPC-RS error correction code, where the LDPC encoding is performed vertically and the Reed Solomon (RS) code implemented horizontally [124]. The advantage of a two dimensional error correction structure is that, even if its error correction capability is equivalent to a concatenated error correction code, a less complex decoding process is allowed thanks to the puncturing capability. At the receiver, the bidimensional product code works as follows: the received data are written in row by row, top to bottom; afterwards, the LDPC code is operated column by column, followed by RS (or BCH) code operated row-by-row [123].

The most of the studies have been performed to compare the new QC LDPC performance with other traditional LDPC codes performance. On the one hand, in [122], [125] and [126] a detailed theoretical performance study for these new codes is performed, showing the improvement over the DVB-T2 LDPC codes. In [127] some QC LDPC codes performance is tested in AWGN and 0 dB Echo channels whereas the mobile performance is tested in [124] under TU6 channel model. On the other hand, the performance of these LDPC has not been highly tested with HW equipment. Only some laboratory measurement results are resumed in [128] as well as some field trials in mobile environments [129]. However, indoor reception, which is a high SNR demanding scenario that can take real advantage of the new QC codes, had not been tested in the moment of the thesis writing.

Another new very efficient multiplexing technique, based on the high robustness of QC LDPC codes, is LDM [130], [131]. It is based on the simultaneous transmission of two (or even more) synchronized signals on the same RF television channel. LDM suitability to simultaneous delivery of mobile HD and rooftop UHD services has been widely analyzed theoretically and in the laboratory [132] [133] [134] [135] [137] showing much better performance than TDM and FDM multiplexing techniques present in current DTT systems with small implementation losses [136]. However, the possibility to offer indoor services by means of LDM had not been tested by the time of the thesis writing in spite of being a worldwide booming scenario. For this reason, LDM needs to be analyzed in more detail to check its feasibility to offer indoor services by means of this promising technique.

Finally, it is important to adapt the traditional DTT receivers' algorithm implementation to the new generation broadcasting systems requirements. On the one hand, the traditional DTT receivers have been always designed for single layer systems [138] and, consequently, the LDM receiver implementation needs to be improved in order to optimize the multilayer signals reception performance. On the other hand, DTT receivers have been always designed only for static reception, where only AWGN is considered as noise source. However, it has been demonstrated in [139] that new target scenarios such as indoor and especially mobile reception, include an additional noise source due to the presence of ICI.

3. Research Work

As it can be seen in the former section, the techniques to improve the next generation DTT physical layer have been poorly tested in indoor scenarios. For this reason, a detailed indoor analysis is needed related to the new more robust LDPC codes to allow higher capacity in indoor reception. Furthermore, LDM technique needs to be tested in indoor scenarios in order to check the feasibility of this technique to deliver services indoors.

Moreover, some modified algorithms need to be applied in the next generation DTT receivers so as to adapt to the new requirements improving the reception performance. On the one hand, the new systems target mobile and portable indoor scenarios, where an ICI noise is presented that should be taken into account in the receiver. There are several ways to measure the ICI noise, such as [140] [141] [142], so it is important to determine which one shows the best performance once implemented in a DTT receiver. On the other hand, the use of the traditional decoding algorithms designed for single layer signals with new multilayer signals means performance degradation. Besides, as LDM seems to be more efficient than single layer signals for the simultaneous delivery of several services, new decoding algorithms can be defined to optimize the multilayer performance.

For this reason, on the one hand, two studies have been carried out during the thesis development in order to test the performance improvement of new technologies over the traditional DVB-T2 standard in indoor scenarios.

- Study D: The portable indoor reception of new more robust FEC codes has been tested in the field.

- Study E: The portable indoor reception of LDM signals has been tested by means of laboratory measurements.

On the other hand, two modifications related to the reception algorithms have been defined and theoretically tested.

- Study F: Definition and computer simulations of a new LDM optimized LLR reliability formula.

- Study G: Definition and analysis with computer simulations of considering the ICI as an additional noise in mobile reception.

3.1 Performance Studies

Two studies have been carried out so as to check the suitability of recent techniques to improve the next generation DTT system performance.

- **Study D: Cloud Transmission System Performance in Portable Indoor Environments**

This study evaluates the new FEC codes performance in indoor environments by means of field measurements in order to check the performance gain in comparison to existing FEC codes used in DVB-T2.

- **Study E: SHVC and LDM Techniques for HD/UHD TV Indoor Reception**

This study verifies the transmission of HD and UHD services in indoor scenarios by means of LDM technique and SHVC video coding standard. In addition, a performance evaluation is performed in the laboratory by means of the emulation of different channel models.

3.1.1 Study D: Cloud Transmission System Performance in Portable Indoor Environments

The main aim of this study is to evaluate the new FEC codes performance in indoor environments by means of field trials to check the performance gain in comparison to current FEC codes used in DVB-T2. These results complete the existing ones with simulations and laboratory measurements and could be considered as reference that could be taken into account by future network planners.

To this purpose, coverage studies, as well as preliminary SNR threshold values for correct indoor reception, are obtained with the new FEC codes.

3.1.1.1 Configuration Mode

A DVB-T2 based configuration with modified FEC modes has been tested. The main parameters of this mode are shown in Table 18. On the whole, the mode was defined so as to support robust portable indoor reception. For this reason, the bits are mapped to a QPSK constellation and a low code-rate (CR=1/4) is used. Besides, a low FFT size (8k) is used so as to be robust enough in mobility. Considering a 6 MHz bandwidth channel, the data bit rate is finally set to 2.4 Mbps, which should be enough for two SD program when H.264 is considered or even one HD program if HEVC is considered [143] [100] [144] [145].

Table 18. Main configuration parameters

BW	FFT Size	Modulation	CR	LDPC Block	Capacity
6 MHz	8k	QPSK	1/4	64800	2.4 Mbps

3.1.1.2 Experimental Network

Banderas transmission center was used to cover the urban core of the city of Bilbao, in the north of Spain. It is located on a hill of about 216 m high and about 3 km far from the city centre of Bilbao. Bilbao is a city with an urban centre medium altitude of around 8m over the sea level. The most of the buildings have between 5 and 10 floors with very few skyscrapers.

As the new FEC codes were under study when the field trials were carried out, there was no HW transmitter yet. For this reason, a SW tool was used for the offline generation of the IQ samples file to be played afterwards. The transmission equipment, which is shown in Figure 32, consisted of:

- IQ samples file player at IF.
- Frequency up-converter and amplifier that shifted the IF signal to the RF frequency channel 48 (690 MHz) and amplified it to achieve the required level power to input the power amplifiers.
- Power amplifier that provided a total power level of 300 W.
- Radiating system with a two panel array providing 16 dBi maximum gain using vertical polarization.

Considering the features of all the transmission equipment, the ERP is 5kW.

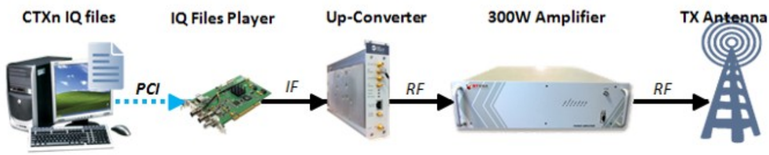


Figure 32. Transmitter main blocks

3.1.1.3 Reception System

The reception system, shown in Figure 33, was based on an online IQ signal recording for posterior offline laboratory processing, which provided great flexibility [146].

The reception system consisted of a 2 dBi vertical monopole tuned to the transmission frequency of 690 MHz connected to a channel filter, set to channel 48, to prevent other emissions interfere with desired signal measurement. The received signal was recorded in a high speed eSata hard disk as baseband IQ samples for a posterior offline processing stage using and IQ recorder equipment.

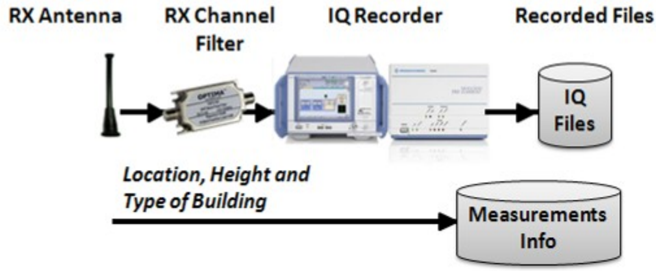


Figure 33. Reception system

Besides, during the measurements, information related to the environment was manually recorded (location, height, type of building, presence of windows, presence of furniture and presence of people moving around).

3.1.1.4 Measurement environment

The measurement campaign was conducted inside 5 buildings in the city of Bilbao. Figure 34 shows a map of Bilbao with the location of the transmitter marked with a green triangle and the reception locations marked with purple circles. Four of the locations were housing buildings at different levels with no more than 3 or 4 people inside, while the other one was a public building (school) with much more people inside. All these locations have been considered as representative samples of indoor environments.

Figure 34. Transmitter and locations for indoor reception

In all the locations, between 10 and 15 fixed measurements were carried out as well as one pedestrian route covering the whole building floor. In the public building, 5 different pedestrian routes were conducted.

Fixed measurements were taken during 10 s, which is considered enough time as the received power level variation in a static location is almost constant along this recording time. In total, more than 12 minutes of fixed indoor reception has been analyzed. The pedestrian measurements were performed during 2 minutes, which was the necessary time to cover each whole building floor at a pedestrian speed. In total, approximately 20 minutes were analyzed in pedestrian indoor reception.

3.1.1.5 Analysis Methodology

The main objective of the processing analysis was to analyze the coverage of the DVB-T2 system with new FEC codes by measuring the correct reception time percentage. Besides, preliminary SNR thresholds for correct reception in indoor and mobile environments are obtained. The correct reception threshold criterion for obtaining the coverage area and the SNR threshold is based on obtaining a FBER = 0 in a T2 frame time as described in Study C.

The used methodology is the same as considered in Study B and Study C, based on the data storing during the field trials for a posterior analysis in the laboratory with a modified version with new FEC codes capabilities of the professional SDR DVB-T2 receiver framework developed in the University of the Basque Country and described in detail in section 3.1 in Chapter II.

As the receiver provided data every OFDM symbol, all the measurements were gathered to obtain data every DVB-T2 frame. This is because it is necessary to analyze uncorrelated data and due to the signal interleaving value, this condition is fulfilled if the data analysis is carried out every DVB-T2 frame. The field power and SNR median values along a frame have been taken as representative.

In order to calculate the optimum transmitted power to obtain a good coverage over the target area, the offline laboratory processing also enables the simulation of lower transmitted power levels by means of adding external white noise to the recorded signals when 300 W were transmitted (*Ptx_real*). By this way, the SNR was decreased as if the transmitted power during the trials was lower. Desired simulated transmitted power levels (*Ptx_des*) of 200 W, 100 W, 10 W and 1 W were tested. The noise power level increment (*Nincr*) to simulate the decrement in the transmission power level is obtained by (4).

$$N_{incr}(dB) = 10\log\left(P_{tx\_real}(W) - P_{des}(W)\right) \quad (4)$$

The existing noise power level (*Nex*) was measured in the received signal out of band samples, obtaining an empirical value of -98.7 dBm (for 5.71 MHz noise bandwidth). By this way, the noise power to be added to the recorded signals (*Nadd*) is given by (5).

$$N_{add}(dBm) = N_{ex}(dBm) + N_{incr}(dB) \quad (5)$$

3.1.1.6 Results

The presented results show the main coverage features of the DVB-T2 system with new FEC codes. They are based on the correct received percentage, in other words, the percentage of the measured signal without errors, considering all the measured indoor locations. In [61] the limit for good portable reception is defined for coverage values higher than 95% while the limit for acceptable portable reception is achieved with coverage values higher than 70%.

Figure 35 shows a histogram of the received power level in the measured indoor scenarios for fixed reception, considering a transmission power level of 300 W (5 kW ERP). Each sample is the median received power level value each about 200 ms. The received power level in a particular fixed location does not vary significantly. However, as the measurements were conducted in different indoor locations (with different type of materials in walls, windows, furniture and presence of people moving around), the received power level varies significantly from one point to another. For this reason, there are huge differences between different measured locations.

The majority of the total received power level samples are between -90 and -50 dBm. It is important to outline that the existing noise power level is around -98.7 dBm. Consequently, the received SNR are mainly between 8.7 and 48.7 dB.

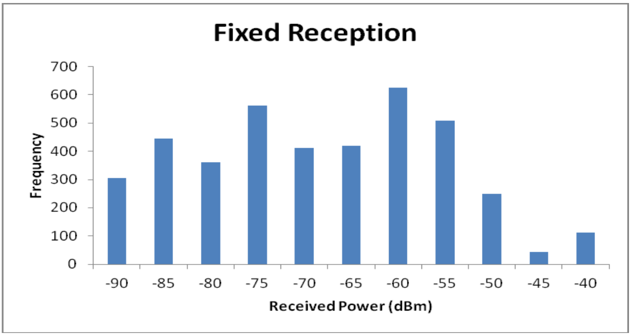


Figure 35. Received power histogram for fixed indoor reception

Figure 36 shows the histogram of the power level (per about 200 ms) received in the measured pedestrian indoor scenarios, considering a transmission power level of 300 W.

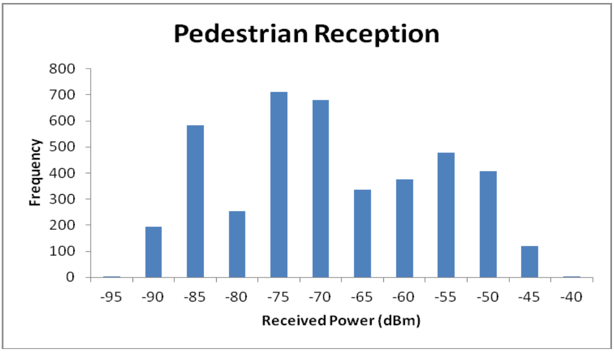


Figure 36. Received power histogram for pedestrian indoor reception

In this case, due to the receiver movement inside each housing building, the received power level in a particular pedestrian route varies significantly. However, as the pedestrian routes were measured along the same rooms than those were the fixed recordings were carried out, the range of received power level samples is similar to the fixed reception case.

Table 19 shows the main coverage features results in fixed and portable indoor reception in terms of average percentage of frames without errors, considering erroneous frames those with FBER ≠ 0, for different simulated transmission power levels (300 W, 200 W, 100 W, 10 W and 1 W).

Table 19. Correct Reception percentages (CRP) for different transmission powers

Scenario	CRP (300W)	CRP (200W)	CRP (100W)	CRP (10W)	CRP (1W)
Fixed	100.0	98.6	95.7	52.1	35.1
Portable	100.0	93.1	88.7	49.6	28.6

The CRP obtained in Table II show that in fixed reception, the CRP is always higher than 95% for a simulated transmission power level of 100 W or higher. To ensure an acceptable reception (CRP higher than 70%) a power level between 100 and 10 W should be transmitted. For lower transmission power levels, the CRP is always lower than the required to achieve an acceptable or good quality reception.

In portable reception, the good reception requirement [61] is achieved with a transmission power of 300 W (200 W power level is in the limit of good quality coverage features) while an acceptable coverage value is obtained with 100 W or higher. When the transmission power level is lower, the coverage area in pedestrian indoor scenarios reduces considerably, obtaining unacceptable reception coverage.

Figure 37 and Figure 38 show a preliminary study of the SNR values received (in blue) vs. the erroneous (“1”) or correct (“0”) demodulated frames (in red) considering one of the measured fixed locations and pedestrian indoor routes, respectively. The represented SNR value in is the median SNR value in a frame, so the instantaneous SNR value may differ. In fixed reception, this is because, although the receiver is static, the presence of people moving around may mean changes on the received power level that might sometimes be very sharp. On the contrary, in pedestrian routes, the receiver movements makes the received power vary.

This preliminary analysis of the SNR threshold shows that the modified DVB-T2 with new FEC codes fixed indoor reception is feasible with a median SNR per frame higher than 2.3 dB. Theoretically it requires a SNR threshold of around -3.4 dB for AWGN channels [147]. In this case, the propagation channel is fixed indoor and it is expected to show multipath and even Doppler effects due to the presence of people moving around, so the threshold SNR value is higher than the theoretical one. The portable reception is also feasible with a median SNR per frame higher than 3 dB. The increment in the SNR threshold because of the receiver movement is about 0.7 dB.

These measured SNR values are for the 50% time and 50% locations, and should be updated with the TCF and LCF determined in Study C for a city as Bilbao. However, the QC LDPC codes performance by means of field trials have been deeply studied in [128].

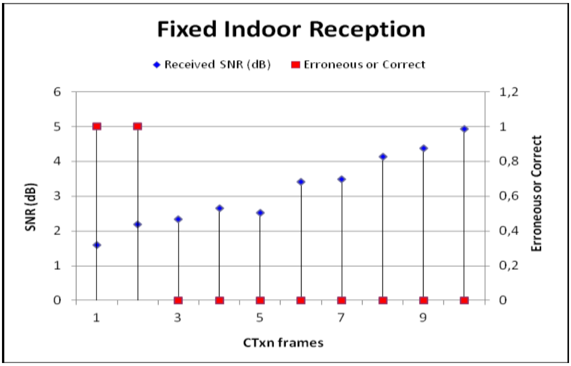


Figure 37. SNR vs. Erroneous Cloud-Txn frames in fixed indoor environments

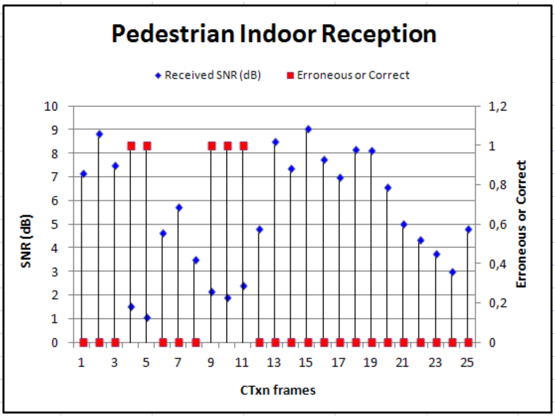


Figure 38. SNR vs Erroneous Cloud-Txn frames in pedestrian indoor environments

3.1.1.7 Conclusions of this study

This study provides the coverage results of the indoor (fixed and pedestrian) reception for a HD program (or several SD programs) by means of a DVB-T2 signal modified with new FEC codes, by means of field trials. Field trials were conducted in real scenarios in the city of Bilbao, so as to obtain performance values based on the CRP of the measured signal.

On the one hand, this study has provided the main coverage features depending on the transmitted power level, obtaining reference results that could be helpful for future network planners. The obtained results demonstrate that with a transmission power level of 300 W (5 kW ERP) a CRP higher than 95% is always possible in both fixed and pedestrian indoor reception. However, the use of lower transmission powers highly affects the CRP in indoor reception. An emulated transmission power of 100 W (1.67 kW ERP) is enough to ensure a good reception (CRP > 95%) in fixed indoor environments. However, in pedestrian indoor scenarios, the good quality requirement is nearly reached with 200 W (3.33 kW ERP). The indoor reception is much more demanding than mobile reception in terms of transmission power according to results in [129].

On the other hand, some preliminary results of the SNR thresholds for fixed and pedestrian indoor reception have also been presented. The results show that the new FEC codes enable a correct reception for SNR thresholds higher than 2.3 dB for fixed indoor scenarios and higher than 3 dB for pedestrian indoor scenarios. Comparing these results with those from DVB-T2 and also T2-Lite [148] it can be seen that the use of these new FEC codes improves the indoor performance in more than 0.7 dB.

3.1.2 Study E: SHVC and LDM Techniques for HD/UHD TV Indoor Reception

The main objective of this study is to verify whether the transmission of HD and UHD services in indoor scenarios is possible considering the bitrate and SNR threshold requirements. For this purpose, the new LDM technology and the recently developed SHVC standard are considered as they are considered optimal technologies at the moment.

Besides, the optimal configuration parameters of the LDM signal will be theoretically defined to fulfill the bitrate and SNR requirements. Afterwards, these configurations will be tested in the laboratory by means of the emulation of different channel models. By this way, the real performance of the selected configurations will be tested in practice for indoor scenarios.

3.1.2.1 Layered Division Multiplexing (LDM)

The current radio spectrum scarcity has led to the development of new technologies that maximize the spectrum efficiency. In this scenario, the reutilization of the limited existing spectrum to transmit superimposed services stacked in the same RF channel, through power allocation multiplexing, is presented as an interesting solution [149]. Moreover, mandatory requirement for new generation broadcasting system is the capability of simultaneously delivering services with different capacities [150], such as mobile and HDTV, or even UHDTV. DVB-T2 standard uses TDM by means of PLPs or the insertion of the mobile service within the FEFs of the fixed services [59] to fulfill this requirement. In ISDB-T [151], on the contrary, FDM is used for delivering different contents within the same frame. Recently, a new multiplexing technique has been developed from the concept behind Cloud-Txn [130], [131]. It is called Layered Division Multiplexing (LDM).

LDM is a new resource sharing technique based on constellation superposition that combines multiple PLPs at different power levels, often with different modulation and channel coding schemes before transmission in one RF channel at it can be seen in Figure 39. The most typical and realistic case is a two layer LDM system with two different synchronized signals (in time and frequency), which are broadcasted on the same channel and at the same time reusing the existing spectrum in a flexible way. The second layer is injected certain dBs below (Injection Level, IL). These are named Core Layer (CL) and Enhanced Layer (EL) [152].

Each Layer is a different PLP with its specific modulation and code-rate for obtaining different capacities and robustness depending on the target service of each signal. However, TI and waveform parameters are common for all of them in order to simplify the receiver implementation. The CL is the layer with more power (UL) intended, in general, for mobile services with high robustness and low capacity, whereas, the EL is usually the layer with lower power (LL) intended to very high bitrate services for UHD fixed services. In general, it has a better capacity and robustness trade-off for the delivery of fixed UHD and mobile HD services than other multiplexing techniques such as TDM or FDM [132] [153] [154].

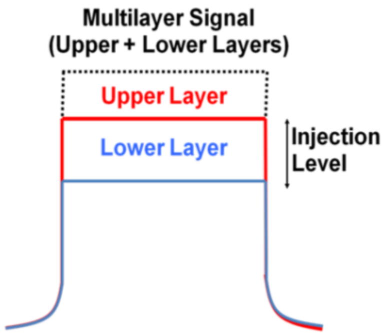


Figure 39. LDM Description

In general, LDM is feasible due to the high robustness provided by the current FEC techniques, such as those studied in Study D, which allows a correct reception even with negative SNR, and an optimal cancellation process for data demodulation [155].

The LDM signal, $x_{LDM}[k]$, in the frequency domain can be represented as (6), where $x_{UL}[k]$ is the UL modulated data stream, $x_{LL}[k]$ is LL one, and k is the sub-channel index. *IL* defines the injection level, between both layers, in other words, the relative power of the signal between the different transmission layers. For example, an IL of -5 dB sets the LL 5 dB lower the UL signal. In other words, the 80% of the LDM signal power is allocated to the mobile service.

Consequently, IL must get values between -∞ and 0 as the UL requires higher power because it targets mobile and indoor services.

$$x_{LDM}[k] = x_{UL}[k] + x_{LL}[k] \times 10^{\frac{IL}{20}} \quad (6)$$

LDM Transmitter

In an LDM transmitter, the major parts of the transmission modules are shared by both layers, and therefore, there is no significant complexity increase. A detailed block diagram of the transmitter is shown in Figure 40.

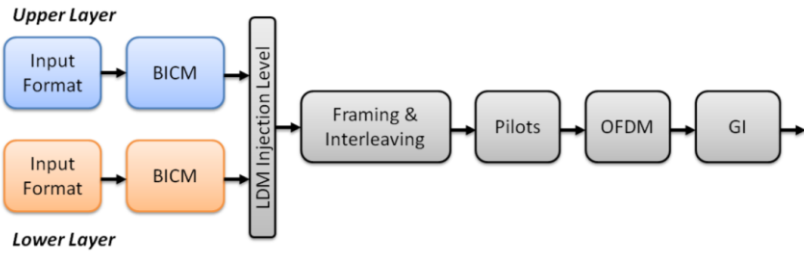


Figure 40. LDM transmitter system main bloks

The first important outcome is that each stream has its own Input Format and BICM modules, and consequently, data streams can be separately configured taking into account the different services that they target. Once signals are modulated, the UL is considered as the primary signal at the mapper output, and thus, it is superimposed to the LL signal staying frequency locked and clock synchronized with it.

In this architecture, the Injection Level (IL) is the key parameter indicating how deep the lower layer signal is embedded. The OFDM-based physical-layer is the same for both layers so the waveform parameters, such as the FFT size, guard interval length and pilot pattern are shared. This concept can be easily extended to more than two layers by means of more layers addition, or to a single layer by means of not adding (IL is -∞) the LL signal.

LDM Receiver

In the receiver, consecutive signal cancellation processes must be applied to correctly decode the overlaid signals [155]. The LDM receiver main blocks diagram is shown in Figure 41, which can be easily adapt to single layer signals if the UL regeneration and cancellation process is not performed.

The initial blocks after the antenna are the same as standard OFDM receivers. The synchronization, Waveform Detection (GI removal and OFDM) and equalization blocks are common for the two layers while an independent decoding block (DeBICM) is needed for each layer as different modulation schemes can be applied on different layers or even on different data carriers in the same layer. The PLP concept can also be applied on each layer. Actually, the LDM approach is equivalent to a layered PLP.

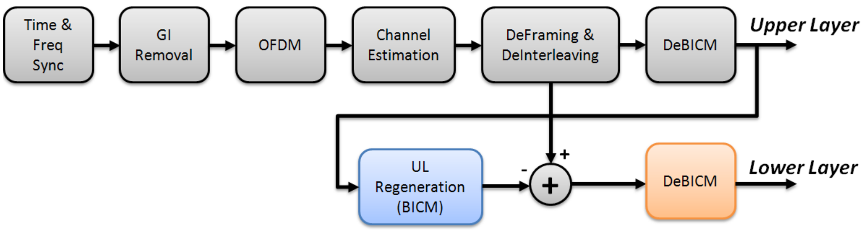


Figure 41. LDM receiver system main bloks

The decoding process of the UL is the same as a single layer signal but considering the LL signal as an additional noise source being the impact of this interference controllable by assigning different injection levels to the LL signal. However, for the LL decoding process, the receiver first correctly decode the UL, then, re-modulate the decoded data, and last, cancel UL from the received signal [155]. Once the UL has been removed, the decoding of the LL signal can proceed as a single layer signal. What is more the receiver first performs channel estimation and signal detection of the UL signal. It is important to note that, considering that the required SNR for LL decoding is much higher than what is required for error-free decoding of the UL, it is reasonable to assume that the UL signal can be perfectly reconstructed.

3.1.2.2 Theoretical Study

Bitrates requirements

In this study the considered coding algorithm is SHVC with HEVC compatibility for the two video coding layers both for HD and UHD services [36]. Based on the literature, the coding output bitrate is still a matter of discussion depending mainly on the perceptual quality [156] [100] [101] [102]. The required bitrates are presented in Table 20 for both HD and UHD services [144] [145]. Furthermore, as there is still not much information about SHVC requirements, the required bitrates for the BL and EL of the SHVC system have been proposed. Finally, the considered bitrates in this study for each service have been suggested.

On the one hand, in the case of offering HD content in the BL of SHVC scheme, a bitrate of around 3.5 Mbps has been considered, as it is an intermediate value between the suggested. On the other hand, two possible bitrates for the delivery of UHD services in the EL of a SHVC scheme have been considered: 10 Mbps and 20 Mbps, depending mainly on the desired frame rate value, 30 and 60 Hz respectively [144].

Table 20. HD and UHD Coding Bitrates with HEVC & SHVC (Mbps)

Format	Suggested Bitrates with HEVC	Suggested Bitrates with SHVC	Considered Bitrates in this study
HD	2.5-4.5	2.5-4.5 (BL)	~3.5 (BL)
UHD	15-32	10-23 (EL)	10 and 20 (EL)

Reception Scenarios

In a near future, it is expected that broadcasters will have to provide services in different scenarios [157]. First, the capacity of offering static and mobile reception (both in portable and high speed scenarios) will be essential. Second, the receivers will not only be located outdoors, but also inside the buildings. All in all, and considering all the possible combinations, 4 different scenarios will have to be considered [158]: portable and indoor (PI), mobile and outdoor (MO), static and indoor (SI), and static and outdoor (SO). The name of each scenario clearly defines its reception environment. The main difference between all of them is that the SNR requirements are completely different in each scenario, which means a consequently different set of configuration parameters.

Table 21 resumes the four potential scenarios and their requirements in terms of SNR threshold (for AWGN channels) to be correctly received [158]. Besides, the maximum SNR threshold value considered in this study is also stated. It is, always an intermediate value in the suggested range in [158] but closer to the upper extreme, to take advantage of all the SNR suggested range.

The main target of this study is to offer HD or also UHD services in indoor scenarios. This refers not only to static indoor scenarios (SI), but also to portable indoor environments (PI). Besides, to maintain a realistic broadcasting services delivery, it should be also necessary to also offer UHD services in SO and HD services in MO scenarios [159].

Table 21. SNR Requirements for Future Broadcasting Scenarios (dB)

Reception Type	Static /Mobile	Indoor/ Outdoor	Required SNR	Considered max. SNR
PI	Portable	Indoor	-6 – 1.5	1
MO	Portable/Mobile	Outdoor	1-5 - 8	6
SI	Static	Indoor	8 - 14	13
SO	Static	Outdoor	14 - 24	22

LDM Configuration parameters

Taking the bitrate and SNR threshold requirements included in Table 20 and Table 21 respectively, the appropriate LDM configuration parameters will be defined.

As the SHVC scheme is based in the transmission of two video coding layers (HD services in BL and scalability information for UHD services in EL), LDM technology completely fits. The video coding BL will be transmitted in the physical UL and the video coding EL will do in the physical LL. By this way, when only the HD services are required, the decoding of the video coding BL is necessary and, consequently, only the demodulation of the physical UL is needed. On the contrary, when both HD and UHD services are desired, the video coding EL for the UHD service requires first access to the HD service and, equally, the physical LL first needs the correct demodulation of the physical UL. So, taking all into consideration, the physical UL of the LDM system will deliver HD services (BL in SHVC) of about 4 Mbps. The physical LL will include UHD services (by means of EL in SHVC) with 10 or 20 Mbps depending on the desired frame rate. By this way, the feasibility of LDM to deliver HD and UHD services in indoor scenarios is demonstrated.

Taking all above into consideration, four potential LDM configurations have been proposed for four different scenarios that include the delivering of HD or UHD services in indoor scenarios. These four scenarios are summarized in Table 22.

Table 22. HD/UHD TV Services Delivering Cases Using LDM and SHVC

Scenario	UL minimum bitrate (Mbps)	UL max. SNR threshold (dB)	LL minimum bitrate (Mbps)	LL max. SNR threshold (dB)
1	3.5 (HD)	1 (PI, MO, SI, SO)	10 (UHD)	22 (SO)
2	3.5 (HD)	1 (PI, MO, SI, SO)	20 (UHD)	13 (SI, SO)
3	3.5 (HD)	6 (MO, SI, SO)	10 (UHD)	22 (SO)
4	3.5 (HD)	6 (MO, SI, SO)	20 (UHD)	13 (SI, SO)

Scenario 1 includes the transmission of HD services in mobility both in indoor and outdoor scenarios while UHD services will be only offered to static outdoor receivers. Scenario 2 includes the same HD services transmission, but in this case, UHD services will be also offered in static indoor environments. Scenario 3 and 4 includes the transmission of HD in mobility only for outdoor receivers, but also for static indoor and outdoor receivers; the UHD services will be offered in outdoor and in indoor and outdoor scenarios, respectively.

Between all the configuration parameters in a LDM system, those which really mean a change in maximum offered bitrate or/and SNR threshold will be studied (different types of modulation and code-rates for both physical layers). Besides, the possible injection levels between UL and LL will be analyzed. On the contrary, the remaining parameters of the LDM signal stay unchanged. Some of them are 5.7 MHz of *BW*, *PP* of $PP_{12,2}$, *GI* of 1/16 and FFT size of 16k, which is a tradeoff between capacity and robustness when in mobility.

The considered modulation schemes (*M*) are the new non-uniform constellations ranging from QPSK to 256QAM. The analyzed code-rates (*CR*) are QC LDPC codes, similar to those in Study D, ranging from 2 to 13 fractions of 15 and the recommended IL in the LDM system ranges from -3 to -6 dB. In

LDM, the *%Time* for each service in 100%.

The capacity calculation for a specific configuration based on the main configuration parameters of an OFDM signal is defined as defined in equation (7) [132] [136].

$$C(Mbps) = \log_2 M \times CR \times \frac{1}{1+GI} \times (1 - PPov.) \times BW \times \%Time \quad (7)$$

The SNR threshold of each possible combination of modulation and coding schemes, defined as SNR for the single-layer system (*SNR-SL*), has been obtained by ideal computer simulations. These values must be corrected depending on the injection level value as defined in equation (8) and (9) for the UL and LL respectively [153].

$$SNR - UL(dB) = SNR - SL + 10 \times \log 10(1 + 10^{\frac{IL}{10}}) - 10x \log 10(1 + 10^{\frac{IL+SNR-SL}{10}}) \quad (8)$$

$$SNR - LL(dB) = SNR - SL - IL + 10x \log 10(1 + 10^{\frac{IL}{10}}) \quad (9)$$

After a deep analysis of all the possible configuration of LDM to provide the services resumed in Table 22, it can be assured that there are many configuration sets that fit all the requirements in each case. However, those providing the maximum bitrate in the LL have been considered as better because they enable the improvement of the quality of the UHD signal. The optimal configuration set in each case is resumed in Table IV, including its maximum transmission bitrate and the *SNR-UL* and *SNR-LL* thresholds under AWGN channel conditions. All the configurations show in Table 23 fulfill the requirements of the four scenarios defined in Table 22 for the delivery of only HD or also UHD services in indoor environments.

The capacity of the physical UL configuration of modes 3 and 4 fits perfectly with the HD services bitrate requirements, while the capacity of the UL of modes 1 and 2 is much more limited. The physical LL of mode 2 is also very limited to offer UHD services, while the LL of the other modes fits perfectly well. For this reason, mode 2 is the more limited for both services while mode 1 is only limited for HD services and modes 3 and 4 completely enables the transmission of HD and UHD services in indoor scenarios.

Table 23. Optimal LDM Configuration To Provide HD/UHD Indoor Services

Scenario	UL			IL (dB)	LL		
	Conf.	Bitrate (Mbps)	SNR (dB)		Conf.	Bitrate (Mbps)	SNR (dB)
1	QPSK 5/15	3.4	0.6	-6	64QAM 11/15	22.6	21.6
2	QPSK 5/15	3.4	0.6	-6	64QAM 5/15	10.2	13
3	QPSK 6/15	4.1	4.2	-3	256QAM 10/15	27.3	22
4	QPSK 6/15	4.1	4.2	-3	64QAM 6/15	12.3	12.5

3.1.2.3 Laboratory Measurements Description

Once that the optimal configuration modes, resumed in Table 23, have been theoretical determined, they must also be tested in laboratory measurements so as to test their real performance in indoor scenarios.

Channel models

Some of the channel models that represent the indoor reception which is target in this study are presented. First of all, the stationary channels F1 and P1, defined in the DVB-T2 Implementation Guidelines [59], will be considered because they are the classically reference of stationary reception in indoor and outdoor scenarios, respectively. Besides, the mobile indoor reception will be tested by the following channel models: PI, defined in the DVB-H Implementation Guidelines [76] and IOA (with low delay spread) and IOB (with medium delay spread), defined by the ITU [160]. Moreover, the TU6 [161] highly recommended for mobile outdoor scenarios but also used for indoor reception, will be also tested at pedestrian speed. Finally, the AWGN channel model will be also tested as reference.

Set-up

The set-up used in the laboratory measurements is the one based on recording for posterior analysis processing [146], as shown in Figure 42. For the first phase, it is necessary a LDM transmitter, a channel emulator and a recording device. The LDM signal is generated by means of a modified DVB-T2

SW transmitter including the novelties that have been considered in this study (QC LDPC codes, NUC and LDM). The transmitter consists in playing the desired LDM signal (previously generated for each configuration in Table IV) in a radiofrequency channel by means of a generic IQ samples file player. The transmitter output is connected to the channel emulator input in which the channel models defined in "Channel Models" section will be implemented by HW. The length of the measurements is about 5 s for the stationary channels (AWGN, F1 and P1) and 10 s for the pedestrian scenarios (TU6, PI, IOA and IOB) [148]. Finally, the channel simulator output is directly connected to an IQ digitalizer in which the LDM signal affected by a specific channel model is recorded as an IQ samples file.

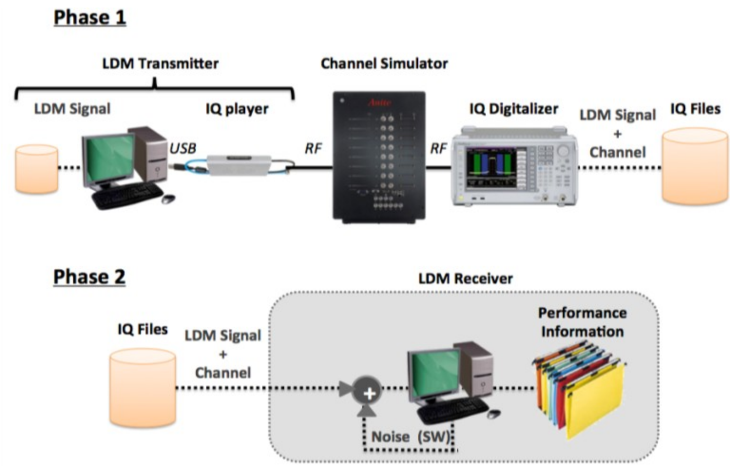


Figure 42. Laboratory measurements set-up

The phase 2 consists on a performance analysis of the previously recorded IQ samples files by means of an adapted LDM professional SDR receiver completely developed in the University of the Basque Country described in detail in section 3.1 in Chapter II. This receiver provides performance information, such as the received power level, BER, MER or FBER.

The reception threshold criterion to establish if the reception is correct or not, is based on FBER [148] [162] as described in Study C. In order to obtain the threshold criterion, increasing AWGN with steps of 0.25 dB, has been added by SW until the first errors appear in the receiver (FBER not equal to 0). At this time, the SNR threshold can be obtained as the relation between the mean signal power level per frame and the mean external AWGN power level per frame.

3.1.2.4 Results

The SNR thresholds obtained in the laboratory measurements are presented in Table 24 and Table 25, for UL and LL respectively.

Table 24. SNR Threshold based on Laboratory Measurements for UL (dB)

Scenario	UL						
	Stationay channels			Portable Indoor channels			
	AWGN	F1	P1	PI	IOA	IOB	TU6
1	0.9	1.3	2.4	2.9	1.5	2.3	4.7
2	1.0	1.3	2.6	3.1	1.8	2.6	4.5
3	4.7	4.8	7.6	8.2	5.3	8.6	8.7
4	4.6	4.9	7.6	8.1	6.1	9.8	8.1

Table 25. SNR Threshold based on Laboratory Measurements for LL (dB)

Scenario	LL						
	Stationay channels			Portable Indoor channels			
	AWGN	F1	P1	PI	IOA	IOB	TU6
1	22.0	22.2	24.9	26.7	22.8	24.7	>35
2	13.4	13.8	16.2	18.3	16.4	19.2	19.5
3	22.6	23.0	25.4	29.5	25.2	31.3	>35
4	12.9	13.2	16.0	18.4	17.0	21.7	20.3

On the one hand, the performance of the F1 channel model, which can be considered as a reference channel model for the static outdoor reception, because it includes the line-of-sight, suffers a degradation always lower than 0.4 dB in comparison to the AWGN channel model. For this reason, in the static outdoor reception with line-of-sight, the degradation is very small in practice.

On the other hand, the performance of the P1 means a degradation of between 1.5 and 3.1 dB with reference to AWGN. This channel model is usually considered the reference channel model for static indoor reception, as there is no line-of-sight. Besides, it can be also a reference for the static outdoor in case the reception antenna has no direct sight with the transmission antenna because of any intermediate obstacle. This fact especially affects the considered LL with highest capacity (modes 1 and 3), as the degradation means SNR threshold of near 25 dB, which starts to be quite high.

The four considered channel models defined as reference for portable indoor reception (at 3 kmph) have completely different performance. This means that the real performance of the specific LDM configuration depends on the type of mobile indoor scenario and no exact SNR threshold for PI scenarios can be established. However, these results are very helpful for planning purposes. The TU6 at 3 kmph and the IOB channel models are in general the most demanding in terms of SNR. However, the IOA has a similar performance to that with the static F1 channel model. The performance for the PI channel model takes intermediate values.

The UL, especially for modes 1 and 2, still maintains low SNR thresholds in MI scenarios for all the considered channel models. However, in case of modes 3 and 4 UL, the SNR takes higher values, up to around 8 and 9 dB, with the exception of the IOA channel, which is less SNR demanding.

Although, the LL target scenario is not the PI, the LL could be correctly received in this kind of scenario for SNR higher than about 20 dB. However, the SNR demanded for the LL of modes 1 and 3 is in general higher than 25 dB, not being ever possible to be correctly received.

Taking all above into consideration, mode 1 is the best option as the bitrate for the UHD services is higher, so does the quality, and the SNR threshold for the physical LL in static reception and for the physical UL in mobile reception are acceptable.

3.1.2.5 Conclusions of this study

In this research, the most efficient way to deliver HD and UHD services in indoor environments, both in fixed and portable scenarios, has been presented. For this purpose, two new technologies in combination have been considered: LDM and SHVC. By this way, the transmission of HD services is made in the physical UL of the LDM system as the video coding BL of the SHVC scheme while the delivering of UHD services is made in the physical LL as the video coding EL of the SHVC scheme. For this purpose, the bitrate and

SNR necessities have been theoretically studied to define the best configuration parameters that fulfill these requirements.

In addition, some laboratory measurement results have been presented in order to show the performance of the selected LDM configurations for the delivery of HD and UHD services in indoor scenarios. The new very robust LDPC codes used with LDM enables always the indoor HD and even UHD, reception. These results, shown in Table V and Table VI, can be helpful for broadcasters for network planning purposes.

3.2 Improvement Studies

Two research works have been carried out in order to adapt the traditional DTT receiver implementation to the current scenarios, improving the DTT systems performance.

- **Study F: LLR Reliability Improvement for Multilayer Signals**

This study defines a new LLR reliability formula optimized for multilayer signals such as LDM signals. Additionally, computer simulations are performed to determine the performance improvement over the traditional approach.

- **Study G: Performance Evaluation of Different Doppler Noise Estimation Methods**

This study defines a new ICI power noise estimator so as to determine the ICI noise to be considered in DTT receivers in order to improve their performance in mobility. In addition, other low complexity ICI power noise estimators are also compared in terms of receiver performance.

3.2.1 Study F: LLR Reliability Improvement for Multilayer Signals

The aim of this study is to adapt the current DTT receivers, optimized to single layers signals, to multilayer signals such as LDM signals. For this purpose, the design of a new LLR reliability formula is proposed.

Furthermore, several computer simulations are conducted in order to establish the performance improvement over the traditional approach optimized for single layer systems. These studies have been carried out depending on several signal parameters (LL and UL constellations and injection level). In fact, the new LLR reliability formula has been also proved in multilayer systems with more layers than usual (three layers).

3.2.1.1 Traditional LLR reliability formula

Considering, for simplicity, a scenario of two simultaneous signals, the reception of the most powerful signal, also called upper layer (UL), is corrupted by the superimposed signal with lower power, also called lower layer (LL). The reception of the LL signal, in contrast, is not degraded as the UL cancellation is first performed and no other signals are present in the decoding process.

It is important to note that it is well known that soft decision decoding algorithms outperform hard decision decoding algorithms since they make use of reliability measures to gain knowledge of the transmitted codewords [163]. LLR has been shown to be a very efficient measure.

In particular, since LLR is the input to the soft input decoder, knowledge of its probability density function (*PDF*) is required. Gaussian modeling of the PDF has traditionally been used. *It* depends on the channel estimation, ρr , and the existing noise power, N_0, as shown in (10) [138], where (It,Qt) and (Ir,Qr) represent the transmitted and received in-phase (I) and quadrature (Q) codewords, respectively.

$$PDF_i\,(I_r,Q_r\,/\,I_t,Q_t) = \frac{1}{2\pi \times N_0} e^{-\left(\frac{(I_r-\rho_r I_t)^2-(Q_r-\rho_r Q_t)^2}{2\times N_0}\right)} \tag{10}$$

In the absence of any other noise source, only AWGN power is considered ($N_0 = N_{AWGN}$). This equation fits well for single layer systems but it does not take into account the LL influence on the UL decoding process in multilayer systems.

3.2.1.2 Theoretical Study

The theoretical study presented in this work applies for a two layers system; nevertheless it can be directly extended to more layers. The first approach presented in the literature considers the LL as a white noise interference added to the external AWGN power (N_{AWGN}) [164] [154].

Considering that the multilayer (ML) signal power (P_{ML}) is the sum of the UL signal power (P_{UL}) and LL signal power (P_{LL}), which is lower in an Injection level (IL) factor, the overall noise power to be considered is defined in (11):

$$N_0 = N_{AWGN} + P_{LL} = N_{AWGN} + 10^{-IL/10} \times \frac{P_{ML}}{(1+10^{-IL/10})}. \quad (11)$$

Thus, the LLR PDF calculation in the decoder was done based on (10) taking N_0 as indicated in (11). However, the LL is an OFDM signal and its influence on the UL is different from that caused by the AWGN. For this reason, this coarse approximation of the LL signal as AWGN leads to degradation in the UL decoding process and, consequently, a new approach should be considered.

The new proposal for the LLR PDF in the UL decoding process is based on the study of the PDF of the IQ components of the received multilayer signal for different LL powers (depending on the specific IL) and different AWGN power levels. By means of curve-fitting, the PDF analytical solution in each case is calculated. This solution is based on the sum of as many Gaussian PDFs as LL constellation I/Q samples are. These Gaussian distributions consider only the external noise power level for its variance ($N_0 = N_{AWGN}$), and their average values depend on the LL constellation and the IL between the UL and LL of the multilayer signal.

PDF for the LLR reliability in the decoder based on the new approach should be modified according to the following more accurate formula shown in (12). In this case, ($I_{ML,t}$,$Q_{ML,t}$) and ($I_{ML,r}$,$Q_{ML,r}$) represent the ML transmitted and received IQ codewords, respectively, as defined in (13), where k spans each of the possible I_{LL}, Q_{LL} samples depending on the LL constellation.

$$PDF_i\,(I_{MLr}, Q_{MLr} \,/\, I_{MLt}, Q_{MLt}) = \frac{1}{2\pi \times N_{AWGN}} e^{-\left(\frac{(I_{MLr}-\rho_r I_{MLt})^2-(Q_{MLr}-\rho_r Q_{MLt})^2}{2\times N_{AWGN}}\right)} \quad (12)$$

$$\begin{aligned}(I_{MLr}, Q_{MLr}) &= (I_{ULr}, Q_{ULr}) + 10^{-IL/20} \times (I_{LLr}, Q_{LLr}) \\ (I_{MLt}, Q_{MLt}) &= (I_{ULt}, Q_{ULt}) + \sum_k 10^{-IL/20} \times (I_{LL_k}, Q_{LLk})\end{aligned} \quad .(13)$$

Figure 43 and Figure 44 show, in black dots, some example of the PDF of the in-phase component of the received multilayer signal for a "noise-free" (SNR = 30 dB) and a "noisy" (SNR = 0 dB) environment, respectively. In blue, the resulting LLR PDF of a Gaussian distribution based in (10) considering that the LL is an additional Gaussian noise added to the existing AWGN as shown in (11). In red, the improved LLR PDF of the received signal following (12) and (13) is shown. In this case, the IL between the UL and LL is 3 dB and the LL uses a 16QAM modulation (four possible in-phase samples).

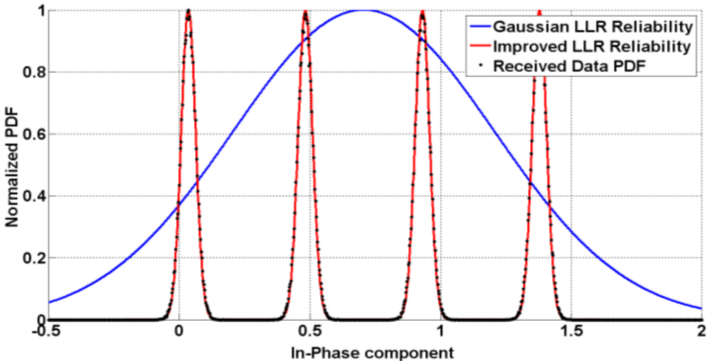


Figure 43. Example of the PDF of the received multilayer signal (black dots), the LLR PDF considering the LL as AWGN (in blue) and with the new approach (in red) in a "noise-free" environment (SNR = 30 dB).

As it can be seen, the new approach (red) based in equation (12) and (13) is much more accurate to the received signal PDF (black) than the original approach (blue) based in (10) and (11), especially for high SNR. In low SNR environments, the existing AWGN power is higher than the LL power and, consequently, the major influence is due to the AWGN. In this situation, the proposed distribution from equation (12) and (13) is similar to the Gaussian one, as it can be seen in Figure 44.

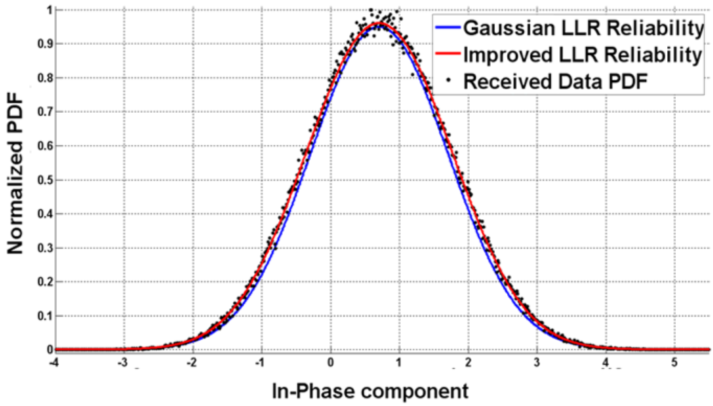


Figure 44. Example of the PDF of the received multilayer signal (black dots), the LLR PDF considering the LL as AWGN (in blue) and with the new approach (in red) in a noisy environment (SNR = 0 dB).

3.2.1.3 Simulation results

In this section, several simulations have been carried out to validate the new approach for the LLR PDF defined in (12) and (13), measuring the gain over the use of (10) and (11). For this purpose, the value of the improvement gain, in terms of multilayer SNR thresholds for correct UL decoding, is calculated with the two options. Four different studies have been carried out. First, the influence on the gain of the specific IL between UL and LL is measured. Next, the influence of the LL constellation and the specific UL constellation influence on the performance gain are also analyzed for a two layers system. Finally, the study is extended to a three layers system.

The SW implementation of a multilayer system (LDM profile) has been used, superimposing each layer with specific injection levels, after being separately formatted, encoded and modulated. Perfect time and frequency synchronizations are assumed, and ideal channel estimation is considered. In addition, the multilayer SNR is perfectly estimated as the average value per FEC block.

Thus, the minimum possible SNR thresholds are defined as the SNR points satisfying the condition that the FBER at the output of the outer coder is null. They are numerically obtained by a brute-force search with a simulation step of 0.1 dB [165]. The considered simulation step is 0.1 dB. The evaluation has been conducted for a P1 channel model [68], which is widely used as reference for portable indoor reception in wireless communications, which is one of the potential target of the UL service [166] [167].

Injection Level influence

Table 26 shows the power distribution of the UL (PUL) and LL (PLL) signals in a normalized multilayer signal for different IL values.

Table 26. Power Distribution on a Two Layers Signal depending on the IL

IL (dB)	P_{UL} (dBm)	P_{LL} (dBm)
1	0.56	0.44
2	0.61	0.39
3	0.66	0.34
4	0.72	0.28
5	0.76	0.24
6	0.80	0.20
7	0.83	0.17
8	0.86	0.14
9	0.89	0.11
10	0.90	0.10

For testing purposes, five different UL configurations have been considered: QPSK modulation with 3/15, 4/15, 5/15, 6/15 and 7/15 code-rate. The LL specific configurations are detailed in each section. Table 27 shows the theoretical SNR threshold for a P1 channel and the equivalent acceptable N_{AWGN} (dBm) of the normalized UL configurations when no LL is considered (Single Layer (SL) profile with P_{UL} = 0 dBm).

In a multilayer system, the UL suffers from two different effects. On the one hand, the P_{UL} decreases with lower values of IL between UL and LL signals. Consequently, the acceptable N_{AWGN} for UL correct decoding at the threshold situation is also decreased. On the other hand, as it has been stated before, the LL acts as an additional interference power that cannot be eliminated.

Considering in a first approach the LL as white noise, it also reduces the acceptable N_{AWGN} at the threshold situation in order to maintain the same SNR threshold for UL. In consequence the theoretical acceptable N_{AWGN} at the threshold situation for UL correct decoding depends on the UL threshold SNR in SL profile and on the IL between both layers. Figure 45 shows the theoretical N_{AWGN} depending on the IL between UL and LL for the five considered UL configurations when the LL is considered as an additional white noise. The P_{LL} for its IL value is also shown.

Table 27. UL Configuration SNR Threshold (dB) in Single Layer Profile

UL	P_{UL} (dBm)	SNR (dB)	N_{AWGN} (dBm)
QPSK 3/15	(Normalized SL profile)	-3.6	2.29
QPSK 4/15		-2.0	1.58
QPSK 5/15		-0.6	1.15
QPSK 6/15		0.8	0.83
QPSK 7/15		1.9	0.65

Figure 45 shows that the N_{AWGN} needed for reaching the correct reception threshold situation is almost constant after a certain value of IL. From this IL value, the LL power influence on the SNR threshold is almost negligible and, consequently, the new proposal based on using (12) and (13) does not provide any gain. This limit IL value depends on the specific robustness of the UL configuration, as the less robust the configuration is, the influence of the LL on the UL becomes negligible for higher IL values.

From Figure 45, it can be stated that the major LL influence for the considered UL configurations is with IL of up to 5 dB. From this reason, so as to the gain of using the new approach based on (12) and (13) depending on the IL, some computer simulations have been performed for IL ranging from 1 dB to 5 dB.

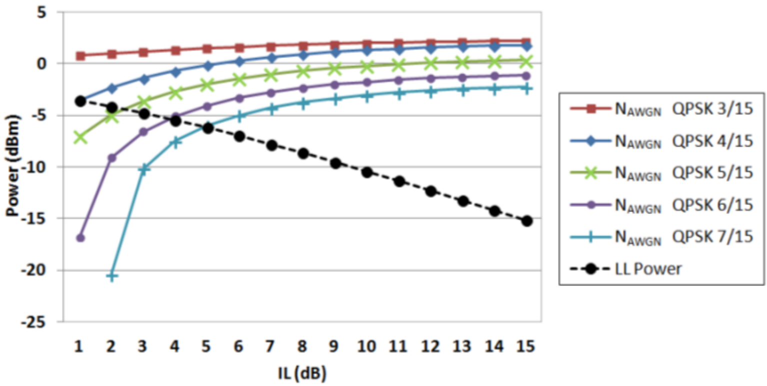


Figure 45. Acceptable N_{AWGN} (dB) for UL correct decoding for different ILs.

Table 28 shows the SNR threshold for correct UL decoding when using the original approach (equations (10) and (11)) and the new approach (equations (12) and (13)) for the considered UL configurations.

By this way, the real improvement gain of using the new suggested approach can be measured. Besides, the theoretical SNR threshold considering the acceptable N_{AWGN} from Figure 45 is also shown in order to make easier the understanding of the gain value. In this case, the LL signal is a 16 NU-QAM 11/15. The (*) means that the decoding process is incorrect for a multilayer SNR of up to 20 dB, so no improvement gain can be assessed, showing an "Undefined" value.

As it can be stated from Table 28, the UL decoding using the new approach in (12) and (13) does not mean degradation in any of the analyzed cases. In fact, its use almost always means gain for the considered IL values. The gain value increases with lower IL values because, as expected, the effect of the LL on the UL is higher for low IL (1 dB) and the influence of the new LLR PDF formula is more significant.

Besides, the gain improvement for the same IL is higher for less robust configurations, as the acceptable N_{AWGN} is lower, (see Figure 45) and consequently, the existing P_{LL} is comparatively higher than the N_{AWGN}. Under that circumstance, the use of the new approach based on (12) and (13) is more noticeable with less robust configurations and the gain significantly increases.

Table 28. SNR Threshold and Improvement Gain (dB) for different IL

UL	IL	Theory	(10)+(11)	(12)+(13)	Gain (dB)
QPSK 3/15	1	0.8	1.8	1.2	0.6
	2	0.0	0.6	0.2	0.4
	3	-0.8	-0.4	-0.6	0.2
	4	-1.3	-1.0	-1.2	0.2
	5	-1.8	-1.6	-1.6	0.0
QPSK 4/15	1	3.6	5.5	4.0	1.5
	2	2.3	3.3	2.6	0.7
	3	1.4	2.0	1.7	0.3
	4	0.7	1.1	1.0	0.1
	5	0.2	0.4	0.3	0.1
QPSK 5/15	1	7.1	11.4	6.9	4.5
	2	5.0	6.5	5.0	1.5
	3	3.7	4.4	3.9	0.5
	4	2.7	3.2	2.8	0.4
	5	2.0	2.2	2.0	0.2
QPSK 6/15	1	16.8	(*)	11.6	Undefined
	2	9.1	12.6	8.2	4.4
	3	6.6	7.7	6.3	1.4
	4	5.1	5.7	5.0	0.7
	5	4.1	4.4	4.1	0.3
QPSK 7/15	1	(*)	(*)	17.8	Undefined
	2	(*)	(*)	10.8	Undefined
	3	10.2	10.8	8.2	2.6
	4	7.5	7.6	6.6	1.0
	5	6.0	6.0	5.5	0.5

Moreover, if the new expression to calculate LLRs (equations (12) and (13)) is considered, the SNR threshold for correct UL detection is closer to the theoretical one. In fact, the performance of the new approach sometimes overtakes the theoretical performance based on considering the LL as AWGN. This fact happens when the P_{LL} is higher than the theoretical acceptable N_{AWGN}, as it can be seen in Figure 45, so the new LLR PDF formula optimizes its performance.

Figure 46 shows the theoretical acceptable N_{AWGN}, the value obtained with (10) and (11) and the one obtained using (12) and (13), for all the UL tested configurations and IL values.

As it can be seen in Figure 46, the acceptable N_{AWGN} obtained with the original LLR PDF formula based on (10) and (11) differs from the theoretical value. The difference increases for low IL and less robust configurations, when the LL has a major impact. In this situation, the new LLR PDF formula based on (12) and (13) especially increases its performance, getting more accurate N_{AWGN} values.

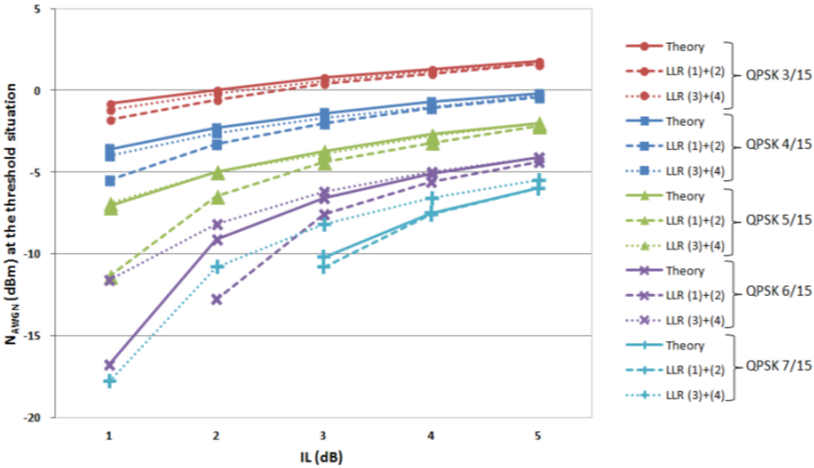


Figure 46. SNR threshold in theory, with (10) + (11) and with (12) + (13)

LL Constellation influence

As the new proposal based on (12) and (13) depends on the number of I/Q samples of the LL constellation, the improvement gain over using the original approach based on (10) and (11) depending on the specific LL constellation size has also been tested. This analysis has been performed for 16 NU-QAM 11/15 (4 bits), 64 NU-QAM 11/15 (6 bits), 256 NU-QAM 11/15 (8 bits) and 1024 NU-QAM 11/15 (10 bits) [168]. The considered IL value ranges from 1 to 5 dB.

Table 29 shows the improvement gain of using the new LLR PDF formula (equations (12) and (13)) for QPSK 3/15 UL configuration. Similar results have been also obtained for the rest considered UL configurations, so they are not included in Table 29.

As it can be stated from Table 29, although the definition of the new approach in (12) and (13) depends on the specific LL constellation, there are no high differences (always lower than 0.1 dB) on the gain values for different LL constellations. This is because the specific LL constellation is considered in the shape of the LLR PDF formula in each case.

Table 29. SNR Threshold and Improvement Gain (dB) for different IL and LL constellations (UL: QPSK 3/15)

LL	IL	(10)+(11)	(12)+(13)	Gain (dB)
16 NU-QAM	1	1.8	1.2	0.6
	2	0.6	0.2	0.4
	3	-0.3	-0.6	0.3
	4	-1.0	-1.2	0.2
	5	-1.6	-1.7	0.1
64 NU-QAM	1	1.8	1.2	0.6
	2	0.6	0.2	0.4
	3	-0.4	-0.6	0.2
	4	-1.0	-1.2	0.2
	5	-1.6	-1.7	0.1
256 NU-QAM	1	1.8	1.3	0.5
	2	0.6	0.3	0.3
	3	-0.4	-0.6	0.2
	4	-1.0	-1.2	0.2
	5	-1.6	-1.6	0.0
1024 NU-QAM	1	1.8	1.2	0.6
	2	0.7	0.3	0.4
	3	-0.4	-0.6	0.2
	4	-1.1	-1.2	0.1
	5	-1.6	-1.7	0.1

UL constellation influence

The UL constellation influence on the improvement gain of using (12) and (13) instead of (10) and (11) has also been tested. Different UL constellations sizes have been considered: QPSK (2 bits), 16 NU-QAM (4 bits) and 64 NU-QAM (6 bits). The specific code-rate in each case is the one that means a similar SNR threshold for correct decoding in a P1 channel model when no LL is present (SL profile). By this way, the influence of the UL specific constellation is studied with no influence of the different robustness level. Moreover, as it has been demonstrated in "LL Constellation influence" section, the specific LL constellation has very low influence on the improvement gain value, so the last LL constellation from the previous study (1024 NU-QAM) has been considered for the simulations. In this case, the IL also ranges from 1 to 5 dB.

Table 30. SNR Threshold and Improvement Gain (dB) for different IL and UL constellations

UL	IL	Theory	(10)+(11)	(12)+(13)	Gain (dB)
QPSK 6/15	1	16.8	(*)	11.6	Undefined
	2	9.1	12.6	8.2	4.4
	3	6.6	7.7	6.3	1.4
	4	5.1	5.7	5.0	0.7
	5	4.1	4.4	4.1	0.3
16 NU-QAM 3/15	1	15.0	(*)	14.8	Undefined
	2	8.7	(*)	8.4	Undefined
	3	6.3	8.6	6.5	2.1
	4	4.9	6.2	5.3	0.9
	5	3.9	4.8	4.3	0.5
64 NU-QAM 2/15	1	16.8	(*)	13.0	Undefined
	2	9.1	(*)	8.5	Undefined
	3	6.6	9.2	6.8	2.4
	4	5.1	6.4	5.3	1.1
	5	4.1	4.8	4.2	0.6

Table 30 includes the SNR threshold when using the original approach ((10) and (11)) and the new LLR PDF formula ((12) and (13)). By this way, the gain improvement is assessed for different UL constellations. Besides, the theoretical SNR threshold has also been included. The (*) means that the decoding process is incorrect for a multilayer SNR of up to 20 dB, so no gain can be assessed, showing an "Undefined" value.

As it can be seen, for similar robust UL configurations the gain is higher for higher order modulations. This is because the distance between consecutive IQ points is smaller in higher constellations. For this reason, the LL increases its influence on the UL for the same IL. Consequently, the improvement because of the use of the new LLR PDF approach ((12) and (13)) instead of the original one ((10) and (11)) increases with the UL constellation order.

However, the SNR threshold using (12) and (13) is similar for the three UL constellations orders, as the tested configurations have also similar performance when no LL is present. The differences in the gain value are due to the different SNR threshold when (10) and (11) are considered. This value increases with the constellation order resulting in higher degradation for UL higher constellations.

Therefore, the UL higher order constellations suffer more degradation because of the presence of a LL when (10) and (11) are considered. However, the improvement of using (12) and (13) is also higher for high order UL constellations so as to correct the specific degradation in each case.

Three Layers performance

If more than two layers are transmitted in the same radiofrequency channel at once, the influence of all of them should be taken into account [169]. If three layers are considered, the UL is influenced by LL1 with an *IL1* and LL1 constellation; and LL2 with an *IL2* and LL2 constellation.

In this case, the LLR formula defined in (12) is completely valid but with the new definition in (14) of transmitted and received IQ codewords of the multilayer signal, (I_{MLt}, Q_{MLt}) and (I_{MLr}, Q_{MLr}), respectively, where k_1 spans each of the possible I_{LL1}, Q_{LL1} samples depending on the LL1 constellation and k_2 spans each of the possible I_{LL2}, Q_{LL2} samples depending on the LL2 constellation

$$(I_{MLr}, Q_{MLr}) = (I_{ULr}, Q_{ULr}) + 10^{-IL_1/20} \times (I_{LL_1r}, Q_{LL_1r}) + 10^{-IL_2/20} \times (I_{LL_2r}, Q_{LL_2r})$$
$$(I_{MLt}, Q_{MLt}) = (I_{ULt}, Q_{ULt}) + \sum_{k_1} 10^{-IL_1/20} \times (I_{LL_{1k1}}, Q_{LL_{1k1}}) + \sum_{k_2} 10^{-IL_2/20} \times (I_{LL_{2k2}}, Q_{LL_{2k2}}) \quad (14)$$

In order to measure the improvement gain of using the new LLR PDF approach ((12) and (14)) instead of the original one ((10) and (11)), the used ATSC 3.0 multilayer system has been adapted to the simultaneous transmission of three layers instead of only two as defined in the standard.

Table 31 shows the power distribution of the UL (PUL) and LLs (PLL1 and PLL2) signals in a normalized multilayer signal for different IL1 and IL2 values ranging between 1 and 5 dB. Equally to the two layers system, the acceptable N_{AWGN} at the threshold situation depends on the IL1 and IL2 and the UL configuration robustness as it can be seen in Figure 47. Consequently, the theoretical SNR also depends on IL1 and IL2.

Table 31. Power Distribution on a Three Layers Signal depending on the IL

IL1 (dB)	IL2 (dB)	P_{UL} (dBm)	P_{LL1} (dBm)	P_{LL2} (dBm)
-1	1	0.39	0.30	0.30
	2	0.41	0.33	0.26
	3	0.43	0.35	0.22
	4	0.46	0.36	0.18
	5	0.47	0.38	0.15
-2	2	0.44	0.28	0.28
	3	0.47	0.30	0.23
	4	0.49	0.31	0.20
	5	0.51	0.32	0.16
-3	3	0.50	0.25	0.25
	4	0.53	0.26	0.21
	5	0.55	0.28	0.17
-4	4	0.56	0.22	0.22
	5	0.58	0.23	0.18
-5	5	0.61	0.19	0.19

As it can be seen in Figure 47, the influence of the IL1 and IL2 on the most robust configurations is low, especially for high IL values. However, less robust UL configurations (QPSK 6/15 or 7/15) are more influenced by the specific IL and, in general, cannot be correctly decoded with two layers, especially if both LL1 and LL2 are very low.

Taking everything into consideration, in this case, a QPSK 4/15 UL has been analyzed as it is robust enough to be correctly decoded with LLs inserted with low IL and it is no so robust to be able to detect the influence of the LLs over the existing AWGN in the threshold situation. As it has been demonstrated in the "Injection Level Influence" section, the specific LL constellation has very low influence on the improvement gain, so only two LL constellations have been considered (16 NU-QAM for LL1 and 64 NU-QAM for LL2).

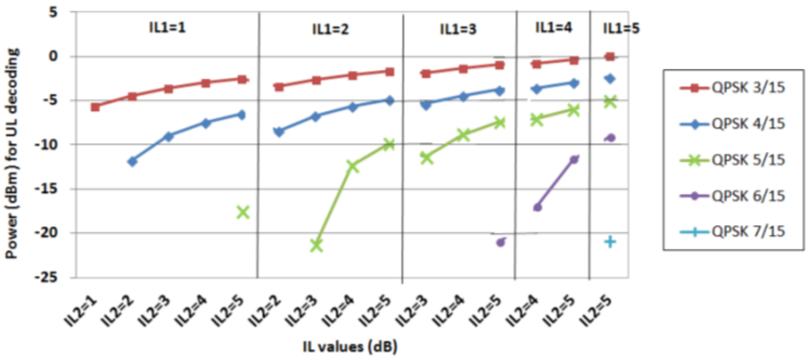


Figure 47. Acceptable N_{AWGN} (dB) for UL correct decoding for different ILs in a three layers system

Table 32 shows the improvement gain of using (12) and (14) instead of (10) and (11) with different IL1 and IL2 values. The (*) means that the decoding process is incorrect for SNR lower than 20 dB, and consequently the gain stays "Undefined".

The results follow the same tendency than with only one LL. On the one hand, the higher the IL1 and IL2 are, the lower the improvement is as the LLs have lower influence on the UL. In fact, when IL1 is very low, the use of (12) and (14) enables an UL correct decoding at the receiver whereas it is not possible with the original approach based on (10) and (11), obtaining a very high improvement gain.

Moreover, although the gain cannot be exactly estimated for very low IL1 (such as 1 or 2 dB), a minimum gain can be deduced with value from 3.7 dB up to 12.4 dB depending on the specific IL2 value.

The LL1 decoding improvement is not analyzed as once the UL has been cancelled, a two layers system is present and the results will follow the same tendency than in "Injection Level Influence", "LL Constellation Influence" and "UL Constellation Influence" sections.

Table 32. SNR Threshold and Improvement Gain (dB) for different IL1 and IL2 in a Three Layers System (UL: QPSK 4/15)

IL1 (dB)	IL2 (dB)	Theory	(10)+(11)	(12)+(14)	Gain (dB)
1	1	(*)	(*)	(*)	"Undefined"
	2	11.8	(*)	16.3	"Undefined"
	3	9.0	(*)	10.3	"Undefined"
	4	7.5	(*)	8.6	"Undefined"
	5	6.5	(*)	7.6	"Undefined"
2	2	8.4	(*)	9.7	"Undefined"
	3	6.7	(*)	7.6	"Undefined"
	4	5.6	11.6	6.4	5.2
	5	4.8	8.2	5.3	2.9
3	3	5.4	9.9	6.0	3.9
	4	4.4	7.2	5.0	2.2
	5	3.7	5.8	4.1	1.7
4	4	3.6	5.4	4.0	1.4
	5	2.9	4.2	3.1	1.1
5	5	2.3	3.4	2.7	0.7

3.2.1.4 Conclusions of this study

This study has proposed a new LLR reliability algorithm to be used in the reception of signals using multilayer multiplexing techniques to improve the receiver performance with this kind of signals. This new formula takes into account the degradation of a superimposed layer on the desired signal through a change in the expression of the LLR PDF formula. The new expression, defined in (12) and (13), depends on the injection level between layers and the specific lower layer constellation.

Several computer simulations have been conducted with an ATSC 3.0 transmission/reception platform, using the LDM profile. As a result, the multilayer SNR threshold for the correct upper layer decoding performance is never degraded and can be improved in up to 4.5 dB, depending mainly on the signal robustness, the specific signal constellation and the injection level. Besides, it has also been confirmed that the higher SNR requirement the signal has, the higher the new proposal gain, as the AWGN present in the threshold situation is lower and the lower layer has more impact on the performance. For this reason, these results can be extended for more challenging scenarios resulting in higher gain values. Furthermore, it has been also proved that the new LLR reliability expression can be extended to more than two layers with even higher performance gain.

Thus, the implementation of this new algorithm means that the receivers could correctly decode the upper layer of a multilayer signal with lower SNR thresholds, improving the receiver performance and increasing consequently the coverage area.

3.2.2 Study G: Performance Evaluation of Different Doppler Noise Estimation Methods

The main objective of this study is to improve the DTT receivers' performance under portable or mobile conditions, the new target scenarios defined for the new generation DTT systems. For this purpose, the effect of the ICI power resulting from the movement has to be considered the receiver as additional noise source. In this context, a new ICI power estimator has been proposed. Additionally, several low complexity ICI power noise estimators have been comparatively evaluated in a DVB-T2 receiver by means of the obtaining of theoretical minimum SNR thresholds for correct reception.

3.2.2.1 ICI Interference

In OFDM systems [170], such as DVB-T2, the ICI caused by Doppler frequency in mobile wireless channels degrades the reception performance. Due to the current importance of mobile reception for future broadcasting systems, the effect of the ICI must be studied in order to improve the system performance in mobile scenarios.

The ICI can be mainly reduced by increasing the space between carriers, which is inversely proportional to the FFT size. In addition, the space between carriers is also inversely proportional to the OFDM symbol length; in other words, the space between carriers increases with shorter symbols [139]. However, the symbol length cannot be highly reduced because the overhead due to the need of long cyclic prefix to avoid intersymbol interference (ISI) would become too large with short symbols.

The ICI has two main consequences on the received OFDM signal: on the one hand, it reduces the signal power converting it into ICI power, and on the other hand, it also introduces an additional noise that should be taken into account [171]. For this reason, the influence of the considering the ICI effect on the mobile performance should be tested.

3.2.2.2 ICI Power as Noise Source

The Doppler frequency shift (f_D) present in mobile channels leads to the ICI. Figure 48 shows an example of IQ components of a received OFDM signal

(7.71 MHz bandwidth, QPSK constellation and 8k FFT size) for different f_D and null AWGN. For this example the TU6 [161] channel model is used to simulate mobile reception at different receiving speeds.

As it can be seen, the higher f_D is, the more the constellation points spread. The effect of this degradation can be considered similar to the effect of the AWGN degradation.

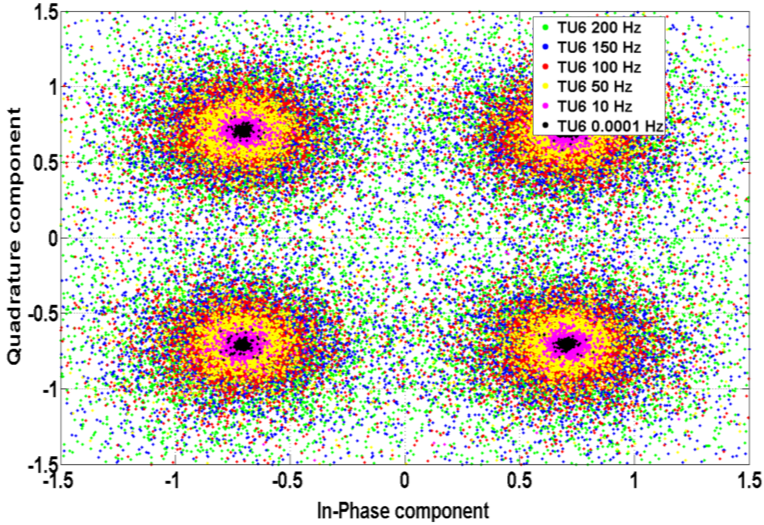


Figure 48. Example of the IQ components of a received signal with 8k FFT and QPSK constellation for different Doppler values TU6 channel model.

A DVB-T2 receiver architecture includes a LDPC channel coding block which relies heavily on the soft decision algorithms for the decoding stage. In particular, it makes use of reliability measurements, such as LLR [172], to gain knowledge of the transmitted signal. The reliability of the LLR depends on the PDF of the channel model, which depends on the channel estimation (ρr) and the noise power (N_0), as shown in (10), where (It,Qt) and (Ir,Qr) representing the transmitted and received in-phase and quadrature symbols, respectively.

In static channels, the unique present source of noise to be considered in (10) is the N_{AWGN}. However, in the case of mobile reception, an additional source of distortion has to be considered so as to improve the reliability of the transmitted signals. For this reason, the overall noise power to be considered in Eq. (15) is:

$$N_0 = N_{AWGN} + N_{ICI} \quad (15)$$

The AWGN noise power (N_{AWGN}) estimation has been comprehensively studied in the literature [173]. However, the ICI noise power (N_{ICI}) estimation has not been so deeply analyzed. The independently calculations of N_{AWGN} and N_{ICI} allows the extension of theoretical SNR thresholds in fixed scenarios (with only N_{AWGN}) to mobile scenarios, by the consideration of the additional N_{ICI}.

3.2.2.3 Experimental ICI Noise Estimator

An experimental method for estimating the Doppler noise power is proposed in this section. For this purpose, the Probability Density Function (PDF) of the IQ components of a DVB-T2 signal after being affected by a TU6 channel are obtained. Different f_D values ranging from very low speeds (f_D of 2 Hz, or ~1 kmph at 600 MHz) to very high speeds (f_D of 200 Hz, or ~360 kmph at 600 MHz) have been considered.

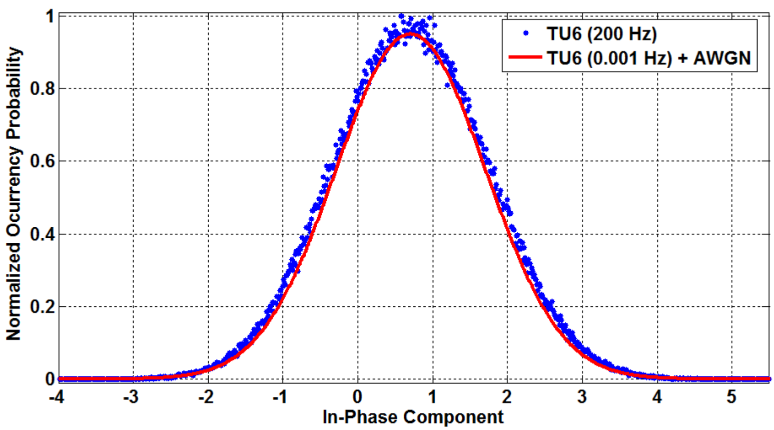


Figure 49. Example of the PDF in case of 200 Hz maximum Doppler frequency shift (in blue) and in case of no Doppler frequency shift with AWGN (in red).

It is important to outline that the PDF width increases with the increment of the f_D value. Equally, providing AWGN is increasingly added to the signal in a stationary channel model, the PDF also increases in width. Consequently, there is one value of AWGN power (N_{AWGN}) for which its PDF fits in shape to that for each f_D value. This value of AWGN can be considered as the equivalent ICI noise power (N_{ICI}) caused by mobile reception. In that point, as it can be seen in Figure 49, the pure N_{AWGN} case (in red) is similar to the estimated N_{ICI} from the Doppler frequency shift of the TU6 channel (in blue).

Several measurements for different f_D values, for different FFT sizes (8k, 16k and 32k), and consequently, for different space between carriers, have been carried out. As a result, the N_{ICI} for each case has been estimated as the equivalent N_{AWGN} for obtaining similar PDF shapes. By means of curve-fitting, a new analytical solution for the N_{ICI} depending on the maximum Doppler frequency shift (f_D), signal power (Ps) and the space between carriers (Δf) has been obtained as shown in equation (16):

$$N_{ICI\_new} = P_s \frac{74\pi}{5}\left(\frac{f_D}{\Delta f}\right)^{2.88} \tag{16}$$

This proposal is proportional to f_D and inversely proportional to Δf in a power factor of 2.88. Besides, it includes an amplitude factor of value $74\pi Ps/5$. The Ps is usually normalized to 1mW and the Δf is dependent on the FFT size and the signal bandwidth of the OFDM signal. A decrement in the space between carriers can be caused by an increment in the FFT size or a decrement on the signal bandwidth, causing an increment in the N_{ICI_new}.

3.2.2.4 Comparison to Existing ICI Estimators

In the literature there are several proposals to estimate the N_{ICI} power that affects OFDM signals [140]-[177]. On the one hand, several of the studied proposals require hard calculations in exchange of good precision and, on the other hand, there are also some fast estimators that can be implemented easily in a receiver. It is important to note that some of the most remarkable solutions are based on different formulas that mainly depend on the signal power (Ps), maximum Doppler frequency shift (f_D) and the space between carriers (Δf) in an OFDM system [140]-[142]. In particular, some of these studies establish that the N_{ICI} can be measured with the formulas (17), (18) and (19) respectively.

$$N_{ICI[140]} = P_s \frac{\pi^2}{6} \left(\frac{f_D}{\Delta f} \right)^2 \qquad (17)$$

$$N_{ICI[141]} = P_s \frac{\pi^2}{3} \left(\frac{f_D}{\Delta f} \right)^2 \qquad (18)$$

$$N_{ICI[142]} = P_s \left(\frac{f_D}{\Delta f} \right)^2 \qquad (19)$$

The three methods are very similar sharing the same square proportional dependency with f_D and inversely proportional with Δf, being the main difference between them the amplitude factor.

These three methods are different to the proposal described in section 3.2.2.3 (Experimental ICI Noise Estimator). Figure 50 shows the N_{ICI} (dBm) obtained for the three methods and for the proposal presented in this study for Doppler frequencies ranging from 0 to 250 Hz and considering Δf = 558 Hz and Ps = 1mW.

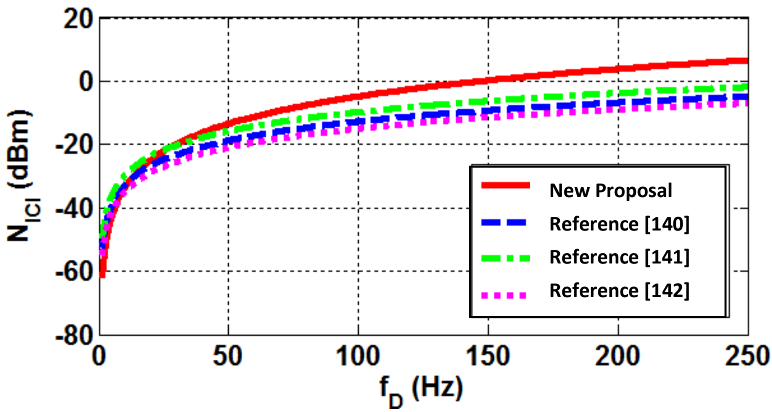


Figure 50. N_{ICI} (dBm) for different f_D with different methods (Δf of 558 Hz).

As it can be seen in Figure 50, for the same signal configuration, the three methods from the bibliography show the same dependency with f_D but with different amplitude, as this is the unique difference between them. The new proposal shows a different tendency as the power factor is different. This method estimates higher N_{ICI} values than other methods, especially for high f_D. For low f_D (lower than 25 Hz), it estimates lower N_{ICI} than the other considered methods.

3.2.2.5 Influence of N_{ICI} on the SNR Threshold

In this section, the mobile reception performance considering the N_{ICI} as an additional noise to the N_{AWGN} has been measured with the four considered estimators. Furthermore, the reception performance for stationary reception conditions is also measured. This refers to the typical receiver implementation without considering ICI noise. For this purpose, the SNR threshold, defined as the relation between signal power level (Ps) and the noise power (N_{AWGN}) is obtained when only AWGN is considered ($N_0 = N_{AWGN}$) and when ICI noise is also considered ($N_0 = N_{AWGN} + N_{ICI}$) for the four estimators.

A DVB-T2 transmitter and receiver have been fully implemented in a simulation platform. It includes a LDPC coding/decoding block that makes use of the LLR PDF formula defined in (10). Perfect time and frequency synchronizations are assumed, and ideal channel estimation is considered. In addition, at the receiver the maximum Doppler shift is known and the SNR is perfectly estimated. The signal reception will be considered error free when the BER value at the output of the outer coder is lower than 10^{-6} [59]. The computer simulations have been carried out for an 8 MHz bandwidth signal with 16k FFT size (Δf of 558 Hz). The signal power level is normalized (Ps = 1 mW) and different f_D values have been measured (50, 100, 150, 200 and 250 Hz). The tested configurations are a QPSK constellation with 1/4 and 2/3 code-rate, testing different robustness levels.

Table 33 shows the SNR threshold for f_D values from 50 to 250 Hz for the two considered configurations. The (*) indicates that the considered DVB-T2 configurations were not correctly decoded for SNR lower than 30 dB.

As it can be seen, the higher the f_D, the more critical the Doppler influence is in the receiver performance. What is more, there might be the case where the Doppler noise power is similar or even higher than the required AWGN for reaching the correct threshold situation, and its influence is, consequently, higher.

Table 33. SNR Threshold for different f_D

QPSK 1/4					
f_D (Hz)	$N_{ICI} = 0$	N_{ICI_new}	N_{ICI} [140]	N_{ICI} [141]	N_{ICI} [142]
50	-1.2	-1.2	-1.2	-1.2	-1.2
100	-0.8	-0.8	-0.8	-0.8	-0.8
150	0.3	0.3	0.3	0.3	0.3
200	1.9	1.6	1.7	1.7	1.7
250	3.9	3.6	3.8	3.7	3.9
QPSK 2/3					
f_D (Hz)	$N_{ICI} = 0$	N_{ICI_new}	N_{ICI} [140]	N_{ICI} [141]	N_{ICI} [142]
50	6.6	6.5	6.5	6.5	6.5
100	8.7	8.5	8.5	8.5	8.5
150	(*)	19.5	19.8	19.5	(*)
200	(*)	(*)	(*)	(*)	(*)
250	(*)	(*)	(*)	(*)	(*)

Table 34 gathers the N_{AWGN} powers for each SNR receiving threshold and, additionally, the N_{ICI} power for the different considered f_D. At this point, only QPSK 1/4 is considered as it is the only configuration robust enough to be correctly received at high Doppler frequency shift levels.

Table 34. AWGN and ICI Noise Power Levels (dBm)

QPSK 1/4					
f_D (Hz)	$N_{ICI} = 0$	N_{ICI_new}	N_{ICI} [140]	N_{ICI} [141]	N_{ICI} [142]
50	1.2	-13.5	-18.8	-15.8	-20.1
100	0.8	-4.8	-12.8	-9.8	-14.9
150	-0.3	0.2	-9.2	-6.2	-11.4
200	-1.9	3.8	-6.7	-3.7	-8.9
250	-3.9	6.6	-4.8	-1.8	-7.0

As it can be seen, for low Doppler noise (f_D < 50 Hz) the Doppler frequency is much lower than N_{AWGN}, and consequently, it can be considered negligible. Nevertheless, when high Doppler noise is presented, (f_D =250 Hz), the N_{ICI} has more influence than the N_{AWGN}, and N_{ICI} estimation method has strong influence on the performance results.

In general, the method in [141] and the new proposal have a similar performance in terms of SNR threshold when the N_{ICI} is considered in the LLR reliability formula. They match with the highest N_{ICI} power levels. The gain increases with the f_D value as the influence of the specific N_{ICI} is higher.

On the one hand, for considerable N_{ICI} power levels, they present a gain of up to 0.2 dB in comparison to method in [140] and of more than 1 dB in comparison to the method in [142]. On the other hand, the consideration of the N_{ICI} in the LLR reliability can improve the SNR threshold in about 0.5 dB, for a medium f_D (about 150 Hz). However, it can exceed 10 dB gain for high f_D (about 200 Hz), being able to correctly decode some configurations that could not otherwise be received (QPSK 2/3 at f_D 150 Hz).

3.2.2.6 Tolerance to a Doppler Frequency Shift miscalculation

As it has been stated before, the three considered methods for estimating the noise power level and the new proposal suggested in this study depend on the f_D. This value must be usually estimated and can be miscalculated with an ε error value. This error can be valued up to between 35 and 65 % of the real f_D value [178]. The error in the f_D estimation, leads to an error in the N_{ICI} of value ΔN_{ICI}.

This section studies the influence of an ε error in the estimation of the f_D value for the four considered methods, both theoretically and practically. In order to do that, SNR threshold value is obtained when the resulting erroneous N_{ICI} is considered in the LLR reliability formula.

If the f_D is measured with an ε error, the N_{ICI} suffers also an ΔN_{ICI} error which is different for each N_{ICI} estimation method. The theoretical ΔN_{ICI} for a f_D of 150 Hz considering the four studied methods is graphically represented in Figure 51. The considered ε error in the f_D is between -65 and +65% of the real f_D value. The ΔN_{ICI} for lower and higher f_D follow the same tendency but with lower and higher values, respectively.

As it can be seen, the theoretical ΔN_{ICI} is much higher for the new proposal while the lowest ΔN_{ICI} is obtained for the method in [142] but with a similar tolerance to method in [140]. The difference in ΔN_{ICI} between the considered methods increases with the f_D as the error tendency of the each method remains unchanging while the absolute value increases.

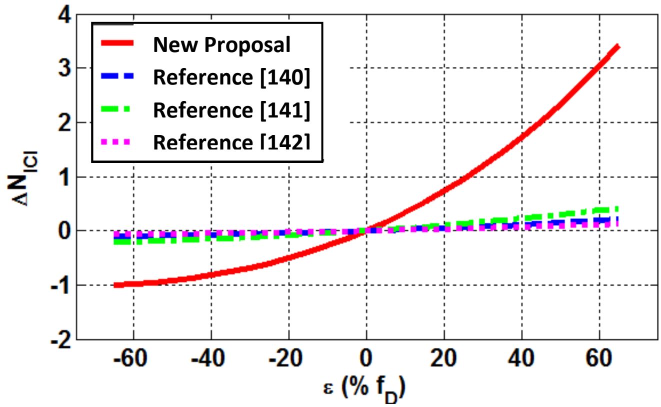


Figure 51. ΔN_{ICI} with up to 65% ε error for f_D = 150 Hz

The changes in the considered N_{ICI} in the LLR reliability formula may affect the SNR threshold for correct decoding. For low SNR thresholds (QPSK 1/4), an f_D error estimation leads to a degradation in the SNR threshold of up to 0.7 dB only for highest percentage of error (65%). In this case, the worst tolerance is for the new proposal and for the method described in [141]. For the remaining percentage errors, no degradation is present.

Nevertheless, when high SNR thresholds are considered, the ICI noise power is remarkable. Table 35 resumes the SNR threshold for different percentage errors in the estimation of 150 Hz f_D for QPSK 2/3 configuration. The (*) indicates that the received signal was not correctly decoded for SNR lower than 30 dB.

As it can be seen, the best tolerance in terms of SNR threshold is for method described in [140]. Besides, the degradation in the SNR threshold is, in general, higher for negative errors. A consideration of N_{ICI} higher than the real one degrades less the SNR threshold than an underestimation.

All in all, for very negative errors (-65%), the four analyzed methods have a similar performance. For low negative percentage errors (-15%) the method in [141] has the best performance while for low positive percentage errors (15%) the best performance is with the new proposal. Finally, for high positive errors (65%), the method in [142] presents the lowest SNR threshold.

Table 35. SNR (dB) Threshold for Different f_D Estimation Errors (dBm) considering f_D 150 Hz

QPSK 2/3					
ε (%f_D)	N_{ICI} = 0	N_{ICI_new}	N_{ICI} [140]	N_{ICI} [141]	N_{ICI} [142]
-65	(*)	(*)	(*)	(*)	(*)
-30	(*)	(*)	(*)	(*)	(*)
-15	(*)	(*)	(*)	18.1	(*)
0	(*)	19.5	19.8	19.5	(*)
15	(*)	16.5	16.7	(*)	(*)
30	(*)	(*)	17.9	(*)	18.1
65	(*)	(*)	(*)	(*)	19.0

3.2.2.7 Conclusions of this study

This study presents a comprehensive comparison framework for various low complexity theoretical methods to estimate the ICI power level so as to improve reception performance under mobile conditions. This is an important contribution for the soft decoding algorithms which rely on the LLR formula. Three methods have been analyzed from the literature, whereas a new one has been proposed based on an experimental study of the received signal PDF shape. All of them mainly depend on the maximum Doppler frequency shift and the space between carriers.

In order to compare the four methods, some computer simulations have been conducted. The method in [141] and the presented proposal show the best performance. With both ICI power estimators, performance improvements of about 0.5 dB (for medium receiver speeds) or up to more than 10 dB (for high speeds) have been measured.

Finally, the tolerance error of misestimating the Doppler frequency shift for the four methods has been also tested. Although, theoretically, the method in [142] shows the best tolerance, when practically analyzed it only offers gain for high positive f_D estimation errors. For low f_D estimation errors, the new proposal and method in [140] present the best performances.

Although this study has been carried out for the DVB-T2 standard, it is also applicable to other DTT standards based on OFDM and LLR reliability formulas for the decoding process.

4. Summary

In this chapter, the state-of-the-art of some new technologies, which have appeared since the DVB-T2 standard definition, has been analyzed. Besides, some additional research studies have been carried out to complete the information available in the bibliography. These new technologies have been applied to the definition of the DVB-T2 standard to check the improvement in terms of robustness and efficiency over the new generation existing techniques.

On the one hand, QC LDPC codes had been proved to show high robustness by means of simulations, laboratory measurements and field trials for fixed and mobile reception. However, they had not been tested in indoor environments. For this reason, some field trials, included in Study D, have been conducted in order to test the suitability of these new codes for indoor reception. For this purpose, these new codes have been applied as the LDPC codes in the DVB-T2 physical layer. The results show good coverage in indoor scenarios for low transmission powers due to their high robustness. Furthermore, instantaneous SNR thresholds for fixed and portable indoor reception have been also presented showing that correct reception is feasible for SNR thresholds higher than 2.3 dB in fixed and 3 dB in pedestrian indoor scenarios, 0.7 dB lower than the most robust DVB-T2 FEC code. By this way, the new QC LDPC codes have been proved to improve the system performance in indoor scenarios in comparison to existing DVB-T2 FEC codes.

In addition, LDM technology has been widely studied theoretically and in the laboratory in Study E, adapting the DVB-T2 physical layer with the new QC LDPC codes to LDM technique by means of layers addition in the transmission and layers cancellation in the receiver. This study shows the high suitability of LDM in combination with the new SHVC coding technique for providing HD indoor services for SNR values from 4.7 dB or even UHD indoor services with SNR values from 19.5 dB.

On the other hand, existing DTT receivers are not optimized for these new multilayer signals, such as when LDM technique is used. For this reason, a new LLR reliability algorithm optimized for multilayer signals has been proposed in Study F, taking into account the injection level and the LL specific constellation. Some simulations have demonstrated that with the new suggested LLR formula an improvement of up to 4.5 dB can be obtained in comparison to the traditional approach, depending mainly on the signal robustness, the specific signal constellation and the injection level. It has been also proved that the new LLR reliability formula can be easily extended to more layers systems.

Moreover, neither are current DTT receivers adapted to new target mobile scenarios as only AWGN is considered as noise in the receiver. However, the ICI is an additional noise source due to the Doppler effect present in mobility that should be considered so as to improve the performance. A wide bibliography study about low complexity theoretical methods to estimate the ICI noise power level present in mobile reception has been carried out in Study G. Besides, a new method has also been proposed based on the signal PDF shape. In order to compare the different estimators, some computer simulations have been carried out considering the ICI power as an additional interference in the receiver decoding process. The studied methods show, in general, performance gains of about 0.5 dB (for medium receiver speeds) or up to more than 10 dB (for high speeds) in comparison to the traditional receivers' implementation with no consideration of the ICI noise power, especially for less robust DVB-T2 configurations and high Doppler situations. The method in [141] and the new proposal show the highest gains in terms of SNR threshold.

CHAPTER IV: ATSC 3.0 Studies

This part of the thesis analyzes the new generation DTT system ATSC 3.0. Some performance evaluation computer simulations are included so as to theoretically analyze the influence of different ATSC 3.0 signal parameters reception under different scenarios conditions.

Moreover, different ways to deliver simultaneous fixed UHD and mobile or indoor HD services are also studied.

1. Introduction

From Chapter III, it is clear that the application of new techniques and receivers' modifications highly improves the robustness and efficiency of the DVB-T2 new generation DTT system. Consequently, another new generation DTT system can be defined making use of these new technologies.

Taking into consideration some of the physical layer features from the current European new generation DVB-T2 system and some of the new techniques that have been proved to improve current DTT standards, ATSC has just developed a DTT candidate standard, named ATSC 3.0 (document A/322) [45], which gives an answer to the current requirements. As any other DTT standard, it is necessary to test the system feasibility to fulfill all the requirements by means of a performance analysis before launching the system commercially.

As the thesis was developed during the ATSC 3.0 standard definition, research work on the first phase of the evaluation process (computer simulations) is needed. In addition, some of the first ATSC 3.0 laboratory measurements have been also conducted during the thesis writing.

More specifically, as there is no common simulation platform, unlike the case of DVB-T2 standardization process, an ATSC 3.0 emulation platform has been totally implemented to perform ATSC 3.0 computer simulations. Besides, a robustness study of the ATSC 3.0 BICM has been performed in order to determine the most important robustness reference values of the system.

Taking everything into consideration, this chapter includes a state-of-the-art including the existing ATSC 3.0 performance studies. Furthermore, new research work related to the ATSC 3.0 performance analysis is also included by means of an emulation platform implementation and four different studies.

2. Previous Studies

During the thesis development, the definition of the ATSC 3.0 system [45] was carried out, being finally approved as a standard in September 2016. As in the case of the previous DTT standard, a complete performance evaluation process is needed before launching a system in a real network.

As in the case of DVB-T2, the first step in a system performance analysis is by means of computer simulations with a SW emulation platform so as to obtain the theoretical minimum SNR thresholds under ideal conditions. Unlike the case of DVB-T2 standardization process, in which the CSP SW platform [59] [57] was implemented and worldwide available for carrying out computer simulations, in the ATSC 3.0 standard definition process no common simulation platform was neither implemented nor defined. For this reason, every company and research group involved in the standard definition, performed research works related to their own implementation of some of the ATSC 3.0 physical layer modules, widely defined in [179]. However, there have been several verification phases to check that, with independence of the specific implementation, each ATSC 3.0 module generates the same correct output signal.

Once the correct implementation of the standard is checked, the system performance of the different ATSC 3.0 modules has to be checked under ideal conditions. It is important to address that due to the high importance of this standard, several companies took part in the performance evaluation process.

The first module in an ATSC 3.0 is the input formatting block in order to generate ATSC Link layer Protocol (ALP) packets according to [180] and [181], so no performance analysis is needed.

The next block is the BICM, widely defined in [182], where the specific modulation and coding is applied and, consequently, also the bit interleaving. More specifically, LDPC codes based on the recent LDPC codes studied in Study D, are included with twelve code-rates ranging from 2/15 to 13/15. Equally, NUCs are used, using a uniform QPSK modulation and five NUCs: 16 NU-QAM, 64 NU-QAM, 256 NU-QAM, 1024 NU-QAM and 4096 NU-QAM. Some comparative studies have been presented in [183] and [118], measuring the new LDPC codes and new NUCs performance, respectively in comparison to previous DTT LDPC codes and UCs. What is more, [184] and [121] study some possible modifications to the defined NUCs based on the application of SSD or constellation condensation techniques. However, the complete ATSC 3.0 BICM module performance had not been analyzed yet in spite of being the module with the most influence on the system performance.

The following module is formed by the framing and interleaving processes as well as the signaling module, whose performance under AWGN and P1 channel conditions has been widely studied in [185] in comparison to the DVB-T2 signaling. Some preliminary computer simulations have been presented in [186] about the time interleaving influence on the performance and latency of the system under TU6 channel. More recently, the frequency interleaver performance has also been analyzed in [187]. However, more research work is needed so as to determine the optimal use of all the interleavers included in the ATSC 3.0 standard definition depending on the specific target scenario.

Another module is the waveform generation process, whose performance is expected to be very similar to other OFDM based DTT standards as the configuration parameters are quite similar (FFT, guard interval, PAPR...). However, some research studies have been conducted in [188] about the different pilot pattern and boosting performance in fixed and mobile scenarios in order to determine the optimal election depending on the specific scenario.

In addition, the newly designed bootstrap symbol for the system discovery, synchronization and transmitted signals identification has been totally defined in [189]. Besides, its high robustness under different kind of scenarios has been tested in [190].

Finally, there is an additional module related to the new LDM technique, introduced in Study E, useful for the delivery of simultaneous UHDTV to rooftop antennas and HD services to mobile or indoor receivers. However, its suitability within ATSC 3.0 has not been tested by the moment of the thesis writing. Additionally, the modification suggested in Study F to optimize the multilayer systems performance presents a practical implementation problem. The suggested new LLR reliability depends on the specific LL constellation points, so the mobile receiver should keep the whole constellations points although they are not needed with the classical LLR approach. Consequently, the memory requirements and system complexity increase.

Once the theoretical SNR thresholds are obtained with ideal computer simulations, the performance analysis must be brought closer to the real world by means of laboratory measurements and, next, field trials. However, by the moment of the thesis writing, the efforts were dedicated to the first phase of ATSC 3.0 (computer simulations) as no HW equipment had been still developed.

3. Research Work

As it can be seen in the former section, as ATSC 3.0 is a standard defined during the thesis development, there is much research work to do. First of all, an ATSC 3.0 emulation platform was designed and implemented in the thesis in order to have a tool to carry out the performance evaluation computer simulations.

With this emulation platform, research work about different system parameters need to be done. It is important to study the influence of the different configuration parameters in the theoretical threshold to determine if its use is recommended.

On the one hand, current DTT systems are very close to the Shannon theoretical limit by means of LDPC codes and uniform QAM constellations up to 256QAM. However, ATSC 3.0 candidate standard has included new elements in the BICM module in order to increase its closeness to the Shannon limit, being highly efficient without additional transmission power or bandwidth. What is more, the BICM module performance evaluation is especially significant as its influence is the highest. For this reason, all the BICM possible combinations' spectral efficiency and robustness have been obtained in this thesis so as to establish the base performance information about the system.

On the other hand, although some interleavers influence on the performance had been previously analyzed, more research was needed in order to test all the existing interleavers (time, frequency, cell and subslicing) under the same channel conditions, with special emphasis in mobile scenarios where their influence is much higher. What is more, as the implementation of each interleaver increases the system complexity, latency, necessary memory and, consequently, the power consumption in the receiver, the real performance improvement of using each interleaver has to be measured. Thus, the trade-off between complexity and performance gain can be maximized.

The remaining configuration parameters' performance, such as those from waveform OFDM module (guard interval length, pilot pattern, bandwidth…) had been previously studied in detail or can be extrapolated from other DTT standards due to their high similarity.

In addition, as the optimization of simultaneous fixed and mobile or indoor services has been considered equally important in the requirements for the ATSC 3.0 standard, LDM technique, which has been previously shown to be the most efficient multiplexing technique [132] [191], is also analyzed for this purpose. Furthermore, some of the first ATSC 3.0 performance results by means of laboratory measurements are also presented for LDM signals. What is more, a comparative performance study between TDM and LDM has been also carried out to test the advantage in ATSC 3.0 of LDM over TDM for the simultaneous UHD fixed and HD mobile or indoor services delivery. In fact, the optimized LDM receiver implementation suggested and tested in Study F (Chapter II) has been also applied so as to increase the advantage of LDM over TDM. Moreover, some pseudo-optimal approaches have also been suggested in order to reduce the high complexity and memory requirements.

All in all, three studies have been carried out in this thesis related to the performance analysis of ATSC 3.0 configuration parameters.

- Study H: The spectral efficiency and robustness of all the ATSC 3.0 BICM combinations have been obtained with computer simulations.
- Study I: The influence on the performance of all the ATSC 3.0 interleaving options has been tested in fixed and mobile scenarios with computer simulations.
- Study J: The fixed and mobile ATSC 3.0 LDM performance has been tested by means of computer simulations and laboratory measurements.

In addition, one study has been carried out in order to improve the LDM operation in ATSC 3.0.

- Study K: The gain of LDM over TDM has been theoretically tested considering different decoding algorithms.

3.1 ATSC 3.0 Emulation Platform

A complete ATSC 3.0 emulation platform, including a complete transmission and reception chain as well as configuration functions with all the features described in [45], has been developed in this thesis with the objective of creating a tool similar to the DVB-T2 CSP [57] [58]. This tool is needed for the first phase of the performance evaluation process of any DTT standard, which is based on computer simulations. It also enables the simulation of different channel models.

3.1.1.1 Platform Structure

The ATSC 3.0 emulation platform structure can be seen in Figure 52.

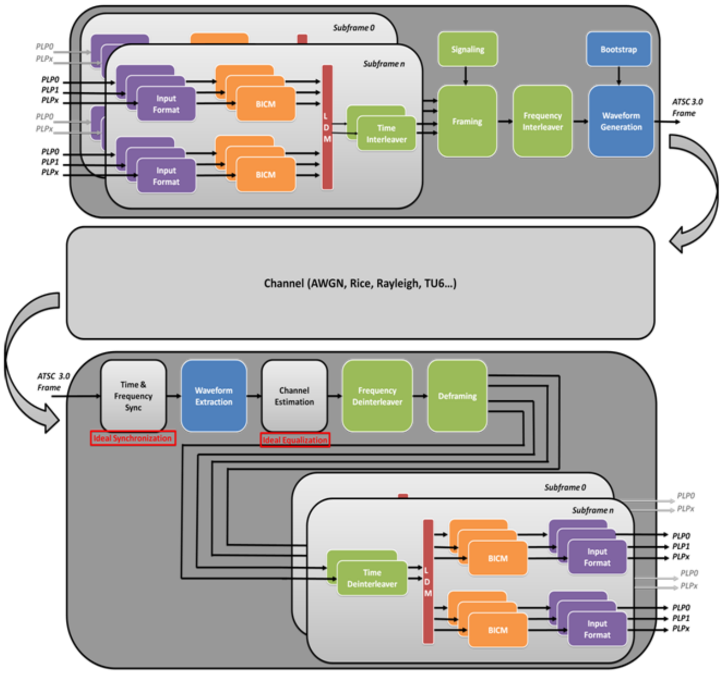


Figure 52. ATSC 3.0 Simulation Platform

All the baseline configuration parameters defined in the ATSC 3.0 A/322 standard [45] have been included in the platform. The design is so modular that it is a very flexible tool for research purposes and performance tests. In the platform, any module can be passed or even exchanged with a different module. By this way, the platform design allows any modification to the standard, as well as any block activation. Each module has some inputs (input data and ATSC 3.0 configuration) and some outputs (output data).

Furthermore, some V&V test points have been also included at the output of each module in the transmitter chain in the form of plain text files with data sample values. These V&V text files fulfill the required format defined for ATSC V&V [192] group in each point and have been correctly checked with V&V Plug Fest #2 testbenches (March 2016) [193].

3.1.1.2 Transmission Chain

The ATSC 3.0 implemented transmission chain includes the main configuration options defined in the standard and summarized in Table 36.

Table 36. Main ATSC 3.0 parameters supported in the SW platform

Waveform Generation Parameters		**Interleaving Parameters**	
Bandwidth	All	Time Interleaver	CTI & HTI
Pilot Patterns	All	Frequency Interleaver	Yes
MISO / MIMO	No	Extended Interleaver	Yes
PAPR	Tone Reservation	**LDM Parameters**	
Guard Interval	All	LDM	Yes
Time Aligned Mode	Yes	Injection Levels	All
Bootstrap	Yes	**BICM Parameters**	
TxID	No	Inner Code (LDPC) Length	All
Framing Parameters		Inner Code (LDPC) code-rate	All
Multiple Frames	Yes	Outer Code	All
Multiple Subframes	Yes	Constellations	All
Symbol Types	All	Bit Interleaver	Yes
FFT Size	All	**Input Parameters**	
Reduced Carriers	All	Multiple PLP	Yes
Subslicing	Yes	Channel Bonding	No
Signaling	All		

The input to the Input Format block in the transmission chain is the data to be transmitted using the ATSC 3.0 physical layer. As shown in Figure 53, this process starts with a PseudoRandom Binary Sequence (PRBS) generated using an external Linear Feedback Shift Register (LFSR) with the polynomial generator x23+x18+1, as specified by ITU-T recommendation O-151 [194]. The generator initialization value depends on the specific PLP identification [192]. The payload results from dividing the PRBS output into fixed length portions of 1200 bytes with UDP and IP headers so as to generate the IP Packet for each PLP that should be considered as the input to the Input Format block.

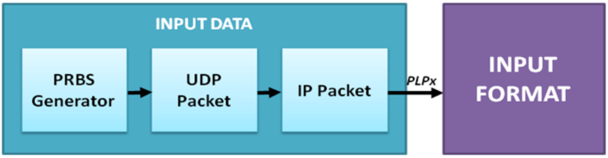


Figure 53. PRBS Input Data Generator

The output of the Waveform Generation block in the transmission chain is a file with the IQ samples of the generated ATSC 3.0 signal. This can be:

- Binary file with I/Q samples saved as signed Int16 Little Endian.
- Binary files with double (IEEE) IQ samples.

The sampling rate of the IQ samples depends on the bandwidth of the signal as shown in Table 37.

Table 37. Output IQ file sampling-rate

Bandwidth (MHz)	Sampling-rate (Mbps)
4.5 (Bootstrap)	6.144
6	6.912
7	8.064
8	9.216

This transmission chain can also be used in a SDR system in combination with a general purpose HW device with the capability of transmitting radiofrequency signals. One example of general purpose device is USRP.

3.1.1.3 Channel

The standardized channel models that have been included in the SW platform are mostly resumed in [195]:

- AWGN [59]. This channel model simulates the ideal reception, with nor multipath neither Doppler and only AWGN is added.
- Rice (F1) [59]. This 21 path channel model refers to a fixed scenario with LOS.
- Rayleigh (P1) [59]. This 20-path channel model refers to a fixed scenario with no LOS.
- 0 dB Echo [59]. This channel model refers to a Single Frequency Network (SFN).
- Pedestrian Indoor (PI) [76], Indoor Office A/B (IOA, IOB) [97] and Pedestrian A/B (PA, PB) [195]. All these channel models refer to a portable indoor reception at 3 kmph.
- Pedestrian Outdoor (PO) [76]. This channel model refers to a portable outdoor reception at 3 kmph.
- Vehicular A/B (VA, VB) [97]. This channel model refers to a mobile reception at 120 kmph.
- Typical Urban 6 paths (TU6) [196]. This is the most used channel model intended to simulate mobile reception. The Doppler value is a configuration parameter.
- Other. In addition to, all the channel models included in [195] have been implemented (Brazil, Communications Research Centre Canada and ATSC ensembles).

3.1.1.4 Reception Chain

The receiver implements the inverse ATSC 3.0 block chain than the transmitter. Besides, in this SW platform, ideal time and frequency synchronization is considered and perfect channel estimation is assumed.

In the receiver, several quality measurements can be obtained at several points of the receiver: BER, FER and SNR can be calculated, comparing the obtained data samples with those generated in the transmitter at the same point of the chain.

3.2 Performance Studies

Three different studies have been carried out during the ATSC 3.0 definition process in order to test the system performance for different parameters in all kind of scenarios.

- **Study H: ATSC 3.0 BICM Analysis**

This study determines the reference values for spectral efficiency and robustness of ATSC 3.0 BICM by means of computer simulations under ideal conditions. These results show the basic relation between capacity and robustness.

- **Study I: ATSC 3.0 Interleavers Influence in Reception Performance.**

This study evaluates the influence of each ATSC 3.0 interleaver in the system performance for different scenarios.

- **Study J: LDM Core Services Performance in ATSC 3.0**

This study includes a theoretical study of ATSC 3.0 LDM different parameters as well as some performance evaluation results based on laboratory measurements.

3.2.1 Study H: ATSC 3.0 BICM Analysis

The main objective of this research work is to analyze the reference values for spectral efficiency and robustness of ATSC 3.0 by means of computer simulations under ideal conditions. For this purpose, the BICM component of ATSC 3.0 is studied in detail, analyzing its spectral efficiency and performance as its influence in the system is the highest.

3.2.1.1 Spectral Efficiency

The spectral efficiency (*Eff*) can be obtained with equation (20), where M_{bits} refers to the bits of the modulation and *CR* refers to the considered code-rate.

$$Eff\ (bps/Hz) = M_{bits} \times CR \quad (20)$$

Table 38 shows the spectral efficiency of all the possible modulation and code-rate combinations in ATSC 3.0. The results are round to the first decimal value.

Table 38. Spectral Efficiency (bps/Hz) of ATSC 3.0 modulations and code-rates

	2/15	**3/15**	**4/15**	**5/15**	**6/15**	**7/15**
QPSK	0.3	0.4	0.5	0.7	0.8	0.9
16 NU-QAM	0.5	0.8	1.1	1.3	1.6	1.9
64 NU-QAM	0.8	1.2	1.6	2.0	2.4	2.8
256 NU-QAM	1.1	1.6	2.1	2.7	3.2	3.7
1024 NU-QAM	1.3	2.0	2.7	3.3	4.0	4.7
4096 NU-QAM	1.6	2.4	3.2	4.0	4.8	5.6
	8/15	**9/15**	**10/15**	**11/15**	**12/15**	**13/15**
QPSK	1.1	1.2	1.3	1.5	1.6	1.7
16 NU-QAM	2.1	2.4	2.7	2.9	3.2	3.5
64 NU-QAM	3.2	3.6	4.0	4.4	4.8	5.2
256 NU-QAM	4.3	4.8	5.3	5.9	6.4	6.9
1024 NU-QAM	5.3	6.0	6.7	7.3	8.0	8.7
4096 NU-QAM	6.4	7.2	8.0	8.8	9.6	10.4

As it can be seen in Table 38, some of the modulation and code-rate combinations have the same spectral efficiency, and they are consequently redundant. For this reason, the Eb/No of each modulation and code-rate combination is obtained in order to know which the most robust configurations for the same spectral efficiency are.

3.2.1.2 Robustness

The Eb/N_0 is defined as the ratio between the signal energy per bit and the noise power spectral density. It can be directly related to the SNR with equation (21).

$$SNR\ (dB) = \frac{Eb}{N0}(dB) + 10 \times log_{10}(Mbits) + 10 \times log_{10} CR \quad (21)$$

Table 39 and Table 40 resume the Eb/N_0 thresholds of the BICM block for all the possible modulation and code-rate combinations with long length LDPC codes for AWGN and P1 channel models respectively. As it can be seen, there are in total 72 modulation and code-rate combinations for long length LDPC codes (64800 bits). However, short length LDPC codes, which are usually used in mobile scenarios due to their less latency, have no sense with high modulations (1024 and 4096 NU-QAM). For this reason, short length LDPC codes (16200 bits) analysis is limited up to 256 NU-QAM.

Table 39. Eb/N_0 (dB) of ATSC 3.0 modulation and code-rate combinations for long LDPC codes under AWGN channel

	2/15	**3/15**	**4/15**	**5/15**	**6/15**	**7/15**
QPSK	-0.6	-0.4	-0.2	0.0	0.4	0.6
16 NU-QAM	-0.1	0.7	1.1	1.5	2.1	2.5
64 NU-QAM	0.6	1.4	2.1	2.9	3.8	4.4
256 NU-QAM	1.2	2.1	3.2	4.2	5.4	6.3
1024 NU-QAM	1.8	3.1	4.4	5.8	7.3	8.5
4096 NU-QAM	2.5	4.0	5.5	7.3	9.1	10.6
	8/15	**9/15**	**10/15**	**11/15**	**12/15**	**13/15**
QPSK	0.8	1.1	1.5	1.9	2.5	3.1
16 NU-QAM	3.0	3.5	4.0	4.8	5.5	6.4
64 NU-QAM	5.2	5.9	6.8	7.8	8.7	9.8
256 NU-QAM	7.6	8.6	9.8	11.0	12.3	13.8
1024 NU-QAM	10.1	11.6	13.1	14.7	16.4	18.3
4096 NU-QAM	12.5	14.4	16.4	18.3	20.4	22.6

Table 40. E_b/N_0 (dB) of ATSC 3.0 modulation and code-rate combinations for long LDPC codes under P1 channel

	2/15	3/15	4/15	5/15	6/15	7/15
QPSK	-0.1	0.3	0.7	1.2	1.8	2.2
16 NU-QAM	0.8	1.7	2.4	3.0	3.9	4.4
64 NU-QAM	1.8	2.8	3.8	4.7	5.8	6.5
256 NU-QAM	2.5	3.9	5.1	6.2	7.7	8.8
1024 NU-QAM	3.3	4.9	6.5	7.9	9.7	11.1
4096 NU-QAM	4.0	5.9	7.7	9.6	11.9	13.4
	8/15	**9/15**	**10/15**	**11/15**	**12/15**	**13/15**
QPSK	2.9	3.5	4.3	5.3	6.7	8.5
16 NU-QAM	5.3	6.1	7.0	8.0	9.4	11.3
64 NU-QAM	7.6	8.6	9.7	10.9	12.5	14.5
256 NU-QAM	10.2	11.3	12.6	14.1	15.8	18.1
1024 NU-QAM	12.7	14.4	16.2	17.8	19.7	22.2
4096 NU-QAM	15.4	17.6	19.5	21.6	23.9	26.2

Table 41 and Table 42 resumes the E_b/N_0 thresholds of the BICM block for all the possible modulation and code-rate combinations with short length LDPC codes for AWGN and P1 channel models respectively. As it can be seen, ATSC 3.0 varies from the most robust mode (QPSK 2/15) operating below -0.6 dB E_b/N_0 with a spectral efficiency of 0.3 bps/Hz, up to the highest capacity mode (4096 NU-QAM 13/15) with 10.4 bps/Hz but E_b/N_0 of 26.2 dB.

Table 41. E_b/N_0 (dB) of ATSC 3.0 modulation and code-rate combinations for short LDPC codes under AWGN channel

	2/15	3/15	4/15	5/15	6/15	7/15
QPSK	0.0	0.1	0.4	0.5	0.7	0.7
16 NU-QAM	0.4	1.1	1.6	1.8	2.3	2.7
64 NU-QAM	1.0	1.8	2.5	3.1	3.9	4.7
256 NU-QAM	1.8	2.6	3.7	4.5	5.6	6.7
	8/15	**9/15**	**10/15**	**11/15**	**12/15**	**13/15**
QPSK	1.1	1.4	1.7	2.2	2.7	3.4
16 NU-QAM	3.1	3.7	4.2	5.0	5.6	6.5
64 NU-QAM	5.6	6.1	7.0	8.2	8.9	10.0
256 NU-QAM	8.0	8.8	10.0	11.3	12.7	14.0

Table 42. E_b/N_0 (dB) of ATSC 3.0 modulation and code-rate combinations for short LDPC codes under P1 channel

	2/15	3/15	4/15	5/15	6/15	7/15
QPSK	0.5	0.8	1.2	1.7	2.1	2.5
16 NU-QAM	1.4	2.2	2.9	3.3	4.1	4.7
64 NU-QAM	2.4	3.2	4.3	5.0	6.0	6.9
256 NU-QAM	3.1	4.5	5.8	6.7	7.9	9.2
	8/15	**9/15**	**10/15**	**11/15**	**12/15**	**13/15**
QPSK	3.0	3.8	4.7	5.6	6.9	9.2
16 NU-QAM	5.5	6.3	7.2	8.4	9.7	11.7
64 NU-QAM	7.8	8.8	10.0	11.2	12.8	15.0
256 NU-QAM	10.6	11.5	13.0	14.5	16.5	18.5

Compared to ATSC A/53 [14], ATSC 3.0 is almost 4 dB and 7 Mbps closer to the Shannon limit in a 6 MHz RF channel [50] [51]. Compared to the current DVB-T2 standard the gain reaches up to 1 dB in some cases.

Change of constellation order

The increment in the constellation order means degradation in terms of E_b/N_0, because the highest the constellation order is, the nearest the constellation points are and the robustness decreases. Figure 54 shows the degradation in E_b/N_0 because of the change between a low order constellations to the consecutive one in size for all the defined code-rates.

As it can be seen in Figure 54, the degradation increases with the code-rate, from 0.5 dB for the lowest code-rate (2/15) up to 4.5 dB for the highest code-rate (13/15). Besides, for code-rates lower than 8/15, the degradation is similar (differences always lower than 0.5 dB) independently of the constellation orders. However, for code-rates higher than 8/15, the degradation of increasing the constellation in one order slightly increases with the order value. For example, for 12/15 code-rate, the degradation in E_b/N_0 of changing from QPSK to 16 NU-QAM is 3 dB while the increment between 1k to 4k constellations is 4 dB.

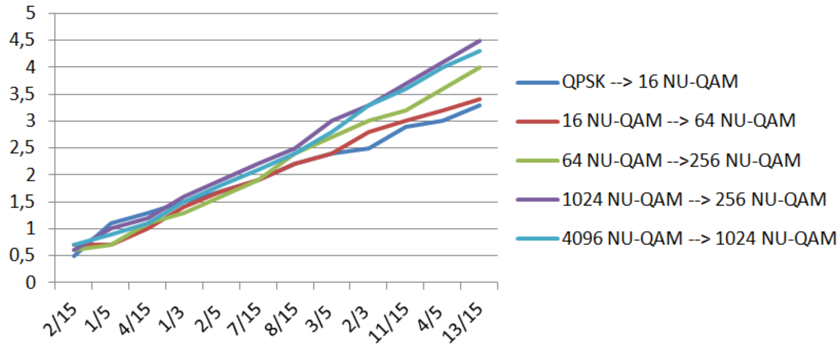


Figure 54. Eb/N_0 increment for increments on consecutive constellation orders

Change of code-rate

The degradation in the Eb/N_0 because of the change between a low code-rate to the consecutive one is shown in Figure 55 for all the ATSC 3.0 defined constellations.

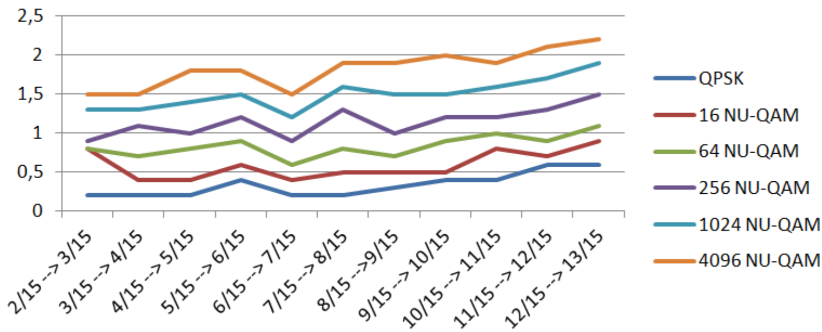


Figure 55. Eb/N_0 increment for increments on consecutive code-rates

As it can be seen in Figure 55, the degradation because of the change in code-rate is higher for higher order constellations. Besides, Eb/N_0 increment because of the change between consecutive code-rates from 10/15 is slightly higher than for lower code-rates. The change between 6/15 and 7/15 is an special case that decreases the tendency of the degradation in Eb/N_0 for all the modulation orders, This is because in addition to the change in code-rate, there is also a change in the LDPC codes internal type [183].

Change of LDPC length

The degradation in the Eb/N_0 because of the change in the LDPC size between long length (64800 bits) to short length (16200) is shown in Figure 56 for all the ATSC 3.0 defined modulation and code-rate combinations. This degradation is only shown from QPSK to 256 NU-QAM as higher constellation orders are not defined for short length LDPC codes.

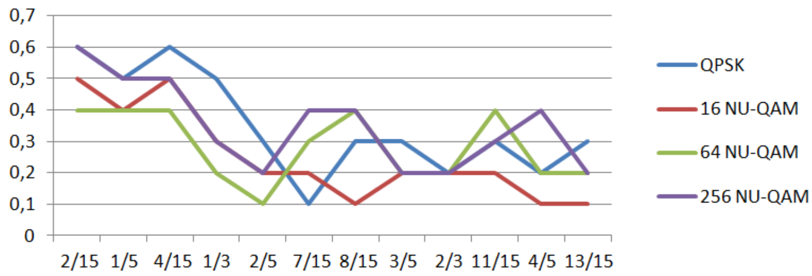


Figure 56. Eb/N_0 increment for decrement on the LDPC length

As it can be seen, using 16200 bits means always degradation for all the modulation and code-rate combinations defined in ATSC 3.0. This degradation is higher for low code-rates up to 5/15 (0.5-0.6 dB) while it remains lower (around 0.3 dB) for higher code-rates. The constellation order has no real influence in the Eb/N_0 degradation when the LDPC length is changed.

Change in the target scenario

The degradation in the Eb/N_0 because of the change in the target scenario from an ideal AWGN to a more challenging P1 channel model is shown in for all the modulation and code-rate combinations defined in ATSC 3.0.

As it can be seen in Figure 57, the degradation because of the more challenging channel model has the same tendency for all the constellation orders, increasing its value with the increment in the considered code-rate. Up to 9/15 code-rate, the degradation is lower for lower constellation orders. For 10/15 code-rate, the degradation is similar for all the modulation orders. However, for code-rates higher than 10/15, the degradation increases for lower constellations.

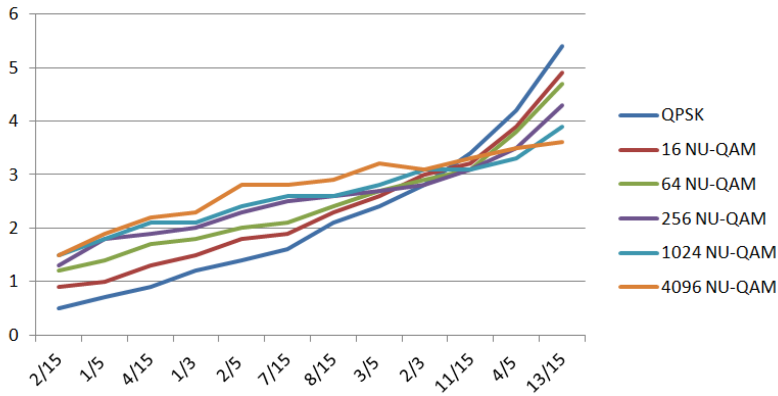


Figure 57. Eb/N_0 increment for change in the target scenario

3.2.1.3 Conclusions of this study

This research work includes a spectral efficiency and robustness study of the ATSC 3.0 BICM module. The results show that that the BICM module provides not only the highest spectral efficiency compared to any DTT system today, but it also provides a significant increase in the maximum transmission robustness and capacity, as it was later shown in [182].

When various BICM combinations satisfy the capacity requirements, some considerations must be taken into account in order to choose the most robust modulation and code-rate combination.

On the one hand, the increment in the constellation order degrades the Eb/N_0 lower for low code-rates. Equally, the increment in the code-rate degrades the Eb/N_0 lower for low constellation sizes. On the other hand, the change in the LDPC length from 64800 to 16200 bits degrades the Eb/N_0 lower for high code-rates while the change in the target scenario to a more challenging one degrades the Eb/N_0 lower for low code-rates and low constellation orders.

3.2.2 Study I: ATSC 3.0 Interleavers Influence in Reception Performance

The main objective of this research work is to evaluate the influence of each type of ATSC 3.0 interleaver in the system performance for different scenarios so as to check for which target scenario they are useful. As a matter of fact, the most advantageous interleaver and the related configuration can be identified for each scenario based on these results.

3.2.2.1 Interleavers Utility

ATSC 3.0 FEC codes are the most efficient existing LDPC codes in the approach to the Shannon capacity limit in transmissions over memoryless channels with randomly distributed and statistically independent errors [183]. However, when the signal suffers from impulsive noise or selective fading, the performance highly degrades [197]. This is because severe burst errors can occur in the same FEC block and its high robustness is not enough to correct so many erroneous bits.

One possible solution to this problem is based on the distribution of burst error patterns in different FEC blocks [198]. By this way, the severe burst errors are divided in several FEC blocks and LDPC codes could probably effectively correct the erroneous bits. Therefore, different channel interleavers have been included in the ATSC 3.0 standard in order to uniformly distribute codewords in time and frequency. Consequently, the transmitted symbols subject to impulsive noise and selective fading do not end up in the same coded frame, taking advance of the available time and frequency diversities from the channel.

3.2.2.2 ATSC 3.0 Interleavers

ATSC 3.0 includes several channel interleavers: Time Interleaver [186] in two different ways: Convolutional Time Interleaver [199] and Hybrid Time Interleaver, which includes an optional Cell Interleaver, and Frequency Interleaver [200] interleavers is to ensure an uncorrelated error distribution inside the FEC blocks, over time and frequency selective propagation channels.

Once the coded and bit interleaved bits have been mapped in constellations, they can be time interleaved. The use of the ATSC 3.0 Time Interleaver (TI) [186] is defined as optional. The TI can be configured enabling different trade-offs in terms of time diversity and transmission robustness, latency and power saving. Figure 58 shows the TI main blocks depending on the number of transmitted PLP [200].

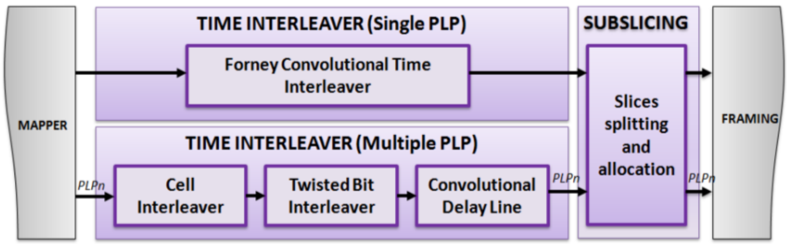


Figure 58. ATSC 3.0 Time Interleaver Blocks

On the one hand, single PLP modes use a conventional Forney convolutional TI (CTI) [199], with interleaving depths of up 200 ms depending on its number of rows. Besides, for low order constellations (QPSK), these values can be increases up to 400 ms using the extended interleaving option. On the other hand, for multiple PLP, a hybrid TI (HTI) is used. This consists of a cell interleaver, a twisted BI and a convolutional delay line based on First-In-First-Out (FIFO) registers.

The Cell Interleaver (CI) is also optional in ATSC 3.0 so as to randomize residual burst errors within a FEC block. To this end, the CI uniformly spreads the cells in the FEC block, to ensure in the receiver an uncorrelated distribution of channel distortions and interference along the FEC blocks. For this purpose, the CI permutes each FEC block according to a pseudo-random sequence that varies every FEC block.

Furthermore, the time interleaving depth can be highly increased by means of subslicing. That means that the PLP data cells from the TI output can be divided into two or more subslices. Each subslice shall occupy a set of contiguous data cells, but the highest data cell index of a subslice shall be non-contiguous to the lowest data cell index of the following subslice of the same PLP.

After being time interleaved and organized in OFDM symbols, the data symbols can be frequency interleaved by changing their cell index, as it is shown in Figure 59. By this way, consecutive data cells are uniformly spread over the available spectrum. The use of the Frequency Interleaver (FI) is defined as optional in ATSC 3.0 [200].

Figure 59. ATSC 3.0 Frequency Interleaver Blocks

Each FI consists on a change of the cell index inside an OFDM symbol. For this purpose, each FI address generator consists of three generation blocks: a toggle block, an interleaving sequence generator with a wire permutation, and a symbol offset generator.

3.2.2.3 ATSC 3.0 configuration

The high number of configuration options defined in ATSC 3.0 makes the system suitable for different uses [201]. Although fixed and mobile scenarios are considered in this study, the influence of some of the interleavers emphasizes in mobility. For this reason, the tested ATSC 3.0 configuration is robust enough to be correctly received in mobile scenarios at high speeds.

A single and a multiple PLP modes will be tested to evaluate the influence of all the interleaving options in the reception performance. Table 43 describes the main configuration parameters of each ATSC 3.0 mode.

Table 43. ATSC 3.0 Tested Configurations main parameters

Common configuration parameters		
Bandwidth	LDPC Length	FFT / Pilot Pattern / GI
6 MHz	64800	8k / SP3,4 / 1024
Specific configuration parameters		
Mode	Constellation	Code-rate
Single PLP	QPSK	5/15
Multiple PLP	QPSK	5/15
	64 NU-QAM	9/15

On the one hand, both configurations have been defined for 6 MHz bandwidth and 64800 LDPC length to obtain lowest bound reception thresholds. The FFT size is 8k, as it is a study based on mobile reception. Pilot pattern and guard interval length has no real influence on the performance as ideal channel estimation is considered in the ATSC 3.0 emulation platform.

On the other hand, for Single PLP mode a QPSK constellation with a 5/15 Code-rate (CR) has been selected aiming at mobile scenarios. As matter of fact, this is very robust, which should be enough for being correctly received in mobility environments. Its BICM theoretical threshold in AWGN is -1.7 dB SNR or 0 dB Eb/N_0 as determined in Study H: ATSC 3.0 BICM Analysis, and the requirements for mobile indoor and outdoor reception are at least -1.5 dB SNR in AWGN [158]. Additionally, it must be noted that its capacity of more than 3 Mbps, obtained with equation (7), is enough for transmitting HD services with HEVC [101] [102].

Finally, the multiple PLP mode includes the single PLP configuration as well as a 64 NU-QAM with 9/15 code-rate, which presents a good trade-off between capacity and robustness. However, only the common PLP is tested because the second PLP configuration does not target mobile reception and the influence of the interleavers cannot be correctly analyzed. The number of codewords considered for the multiple PLP mode is set to 15 (486000 cells with QPSK constellation) as ATSC 3.0 limits the time deinterleaving memory in a receiver to a maximum of 219 cells.

3.2.2.4 Evaluation Trials

These simulations are conducted with the purpose of evaluating the influence of each ATSC 3.0 interleaver in the reception performance in different scenarios. In order to establish the performance reception, the SNR threshold for correct reception is measured. The signal reception is considered error free when the BER value at the outer coder output is lower than 10^{-6} [47]. This value is a reliable tradeoff between simulation time and system performance. In order to satisfy this criterion, AWGN is added at the receiver input with steps of 0.2 dB. The noise is added until the first errors appear in the receiver (BER higher than 10^{-6}). At this time, the SNR threshold can be obtained as the relation between the mean signal power level per ATSC 3.0 frame and the mean external AWGN power level per ATSC 3.0 frame.

The considered interleaving options are resumed in Table 44. The bit interleaver is not considered in the study as it is defined as compulsory in the ATSC 3.0 standard.

The CTI is optional (ON/OFF) and it can be configured depending on its number of rows and the possible use of the extended interleaving length (only for QPSK mode). The FI is also optional and its influence on the performance is also measured (ON/OFF). These two interleavers are tested with the single PLP mode defined in Table 43. The HTI and the CI for multiple PLP configurations are also optional (ON/OFF). Finally, the subslicing influence is tested with multiple PLPs considering different number of subslices.

Table 44. ATSC 3.0 Tested Interleaving options

Interleaver	Mode	Options
CTI Time Interleaver	Single PLP	ON/OFF Rows: 512, 724, 887, 1024 Extended: Yes/No
HTI Time Interleaver	Multiple PLP	Intra-subframe
Cell Interleaver (CI)	Multiple PLP	ON/OFF
Frequency Interleaver (FI)	Single PLP	ON/OFF
Subslicing	Multiple PLP	1, 50

As it has been stated before, the channel interleavers are used so as to improve the system performance in fading channels. The most representative example of this type of channels in broadcasting systems is mobile reception. For this reason, the TU6 [196], which is considered as the most representative channel model for mobility, is tested for speeds up to 200 kmph. Moreover, as TU6 channel model is not well adapted to slow speeds and it is generally more demanding than reality in portable scenarios [202] pedestrian more specific channel models are also considered, including PI at 3 kmph and PO at 3 kmph [76]. Finally, the AWGN is measured as reference as well as fixed reception in order to check that there is no real gain because of the use of channel interleavers. In this case, F1 and P1 channel models are considered [59].

3.2.2.5 Performance Results

The single PLP configuration SNR thresholds for the different combinations of ATSC 3.0 interleavers are gathered in Table 45 for stationary scenarios (AWGN, F1 and P1 channel).

As expected the interleavers show no gain for AWGN and F1 channel models. Nevertheless, for P1 channel models, the gain due to the FI or TI means a gain of only 0.2 dB as the selected configuration code-rate is very robust, and therefore, the multipath distortion could be overcame with the LDPC codes correction capability.

Table 45. SNR Threshold with Single PLP Interleaving options in stationary scenarios

Interleavers State	Channel model		
	AWGN	F1	P1
FI: OFF TI: OFF	-1.0	-0.8	0.2
FI: ON TI: OFF	-1.0	-0.8	0.0
FI: OFF TI: ON (Any rows, Extended: Yes/No)	-1.0	-0.8	0.2
FI: ON TI: 512, 724	-1.0	-0.8	0.2
FI: ON TI: 887, 1024 (Extended: Yes/No)	-1.0	-0.8	0.0

The same results for mobile scenarios are gathered in Figure 60. The SNR threshold without enabling any interleaving option is shown in black. The other results include the SNR threshold with FI and/or TI for all the possible interleaving lengths (from 50 up to 400 ms in case of extended TI).

The mobile scenarios performance is different to the fixed reception. As it can be seen in Figure 60, the gain due to FI is only up to 0.4 dB (in red). However, the CTI performance gain increases linearly with the number of rows (or interleaving length), especially for low speeds. It improves from 0.6 dB (50 ms TI length) up to 3.2 dB (400 ms TI length) when pedestrian channel models are considered. The main reason is the critical fading or “shadowing” that might appears. For higher speeds, this gain is reduced up to 1.4 dB as the time variability of the channel acts as a natural interleaver itself. In addition, it must be noted that the SNR reception thresholds are low, and therefore, the Doppler Noise is masked under the AWGN noise. Moreover, if both interleavers are considered, the simulations show that the SNR thresholds remain mainly unchanging with differences of between ±0.2 dB.

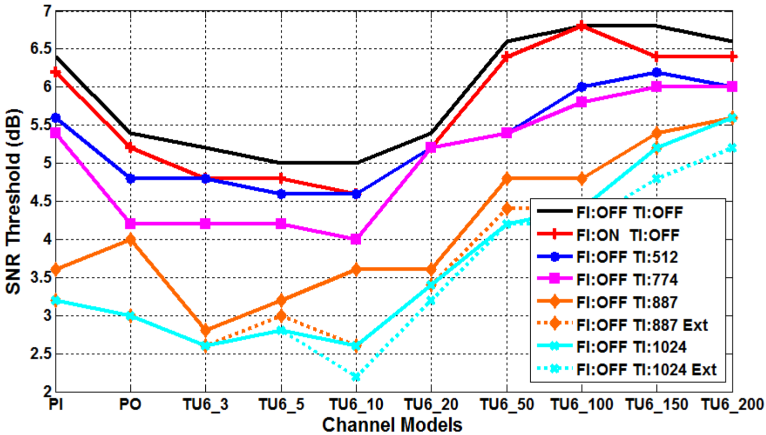


Figure 60. SNR Thresholds for single PLP interleaving options in mobile scenarios

The multiple PLP configuration SNR thresholds for the first PLP are gathered in Table 46 and Figure 61 for fixed and mobile scenarios, respectively.

As it can be seen, TI, CI and subslicing have no influence for AWGN and F1 channel models. In P1 channels, the gain due to these interleavers means only 0.2 dB gain as the highest robustness is provided by the robust measured code-rate.

Table 46. SNR Threshold with Multiple PLP Interleaving options in stationary scenarios

Interleavers State	Channel model		
	AWGN	F1	P1
TI: OFF CI: OFF Subslices: 1	-1.2	-0.8	0.2
TI: ON CI: OFF Subslices: 1	-1.2	-0.8	0.0
TI: ON CI: ON Subslices: 1	-1.2	-0.8	0.0
TI: ON CI: ON Subslices: 50	-1.2	-0.8	0.2

In mobility, as Figure 61 shows, the gain due to the HTI is up to 2.0 dB in pedestrian reception and 0.6 dB in high mobility (in red). These gains are slightly different if CI is considered (in blue) with a ±0.2 dB extra gain as the spread errors are mainly corrected by the bit interleaver. Besides, if subslicing is considered (in purple) an extra 0.6 dB gain can be reached because of the extra time interleaver.

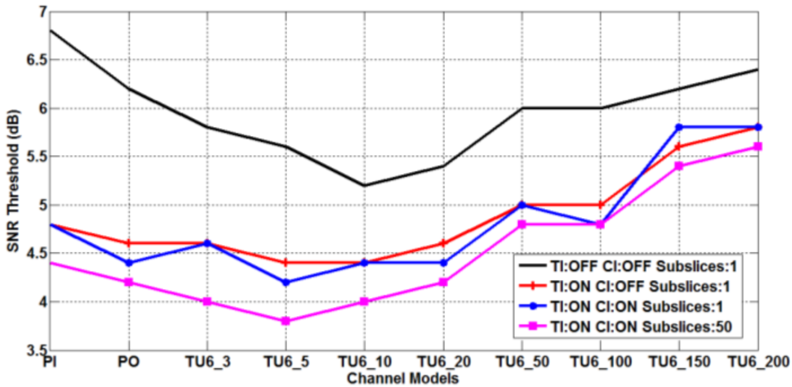


Figure 61. SNR Thresholds for Multiple PLP interleaving options in mobile scenarios

3.2.2.6 Conclusions of this study

This study determines the influences of each ATSC 3.0 interleaving option in the reception performance under different channel conditions.

Considering a very robust configuration intended for mobile reception, if all the interleaving options are compared, it can be stated that in fixed scenarios none of the considered interleavers improves the reception performance.

In mobile scenarios, the time interleaver is in general the interleaver that shows higher improvements. The convolutional time interleaver for single PLP configurations shows increasing improvements with the number of rows showing gains between 0.6 dB (for low number of rows and high speeds) and 3.2 dB (for the highest number of rows and pedestrian speeds). When multiple PLP are used, the hybrid time interleaver means up to 2.0 dB gain for the tested configuration. The gain is higher for the CTI as the interleaving length is longer than with HTI. In addition, the frequency and cell interleavers means almost no gain in all the studied cases as their effects are masked with the time interleaver, the bit interleaver and the high robustness of the tested configuration. Furthermore, the use of subslicing increases the time interleaving length and consequently, the performance improves in up to 0.6 dB.

These results show the high utility of the interleavers for mobile scenarios, but not for fixed reception, when it is not necessary to use them thus reducing the system latency. In mobile reception, the only interleaver that really improves the system performance for very robust ATSC 3.0 configurations is the time interleaver. Consequently, frequency and cell interleavers are not really useful.

3.2.3 Study J: LDM Core Services Performance in ATSC 3.0

A theoretical study is performed so as to shed some light on the possible receiving performance issues of the mobile services when LDM is used for some of the ATSC 3.0 signal parameters. However, these theoretical analyses only establish the lower bounds for the ATSC 3.0 performance, as the emulation platform cannot be considered as a real system.

For this reason, another objective of this study is to test on the laboratory the performance of the mobile service of an LDM system in ATSC 3.0 to deliver indoor or mobile services, obtaining results very close to a real scenario situation.

3.2.3.1 Theoretical Study

This study is based on obtaining the SNR thresholds for correct reception. These thresholds are results of computer simulations by means of the SW platform described in ATSC 3.0 Emulation Platform. Besides, perfect time and frequency synchronization are assumed while ideal channel and AWGN estimations are considered [47].

Injection Level penalty on LDM

LDM is based on splitting the available transmission power into two layers, and due to this power split the UL suffers from inter-layer interference. At the receiver, the LL acts as interference for the UL. As a result, the UL SNR threshold in LDM depends on the single layer SNR threshold and the defined injection range, as shown in equation (7). *SNR-UL* stands for the SNR threshold of the UL signal in the LDM system while *SNR-SL* is the SNR threshold value of the single layer configuration and IL is the injection level between both layers. All units are in decibels (dB).

Following (7), in Figure 62 the SNR-UL threshold is shown as a function of IL (Δ) and the SNR-SL. The lower the IL, the more power is shared with the LL. Therefore, the UL signal power is lower and, consequently, the SNRUL threshold increases. For instance, in Figure 62 vertical lines show that if the desired SNRUL threshold is kept constant at a value of 0 dB, the UL single layer configuration should guarantee an error free reception threshold of {-3, -2.5, -

2.1, -1.75} dB, for Δ={-3,-4,-5,-6} dB injection levels, respectively.

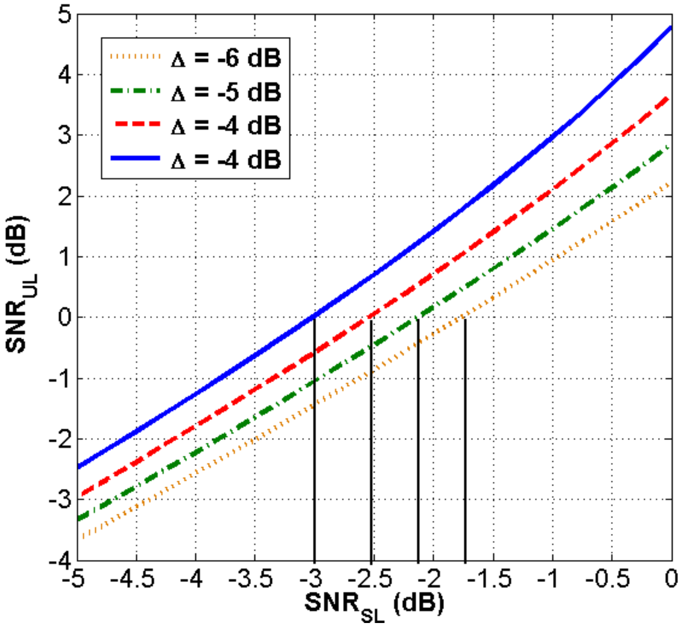


Figure 62. Upper layer minimum SNR depending on the selected injection level and selected configuration receiving threshold.

Inter-Carrier Interference (ICI)

One of the main novelties of ATSC 3.0 is the adoption of OFDM as the physical layer waveform with its consequent main weakness: the orthogonality loss that occurs in mobile environments. As explained in [203], the impact of ICI is usually measured through the relationship between the existing maximum Doppler frequency, fd, and the carrier frequency space, $\Delta f = 1/T_u$, which depends on the OFDM symbol duration, Tu. It has been demonstrated in [204] that this ICI leads to the presence of a Doppler noise. Doppler noise increases exponentially as the receiver speed goes up.

The overall shape of the ICI and its relevance on the final threshold can be seen in Figure 63. In this figure, the dashed blue line represents the ICI power obtained through experimental analysis from the behavior of the received

OFDM physical waveform considering a TU6 channel model with different normalized Doppler values, f_dTu.

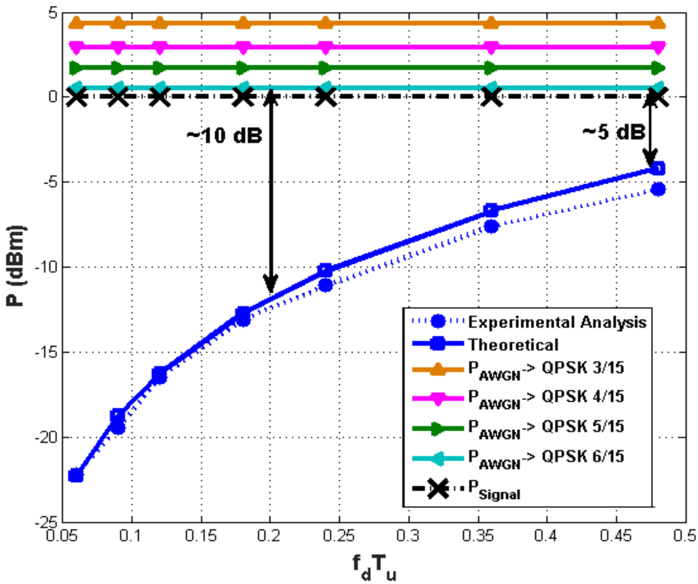


Figure 63. ICI influence in a TU6 channel and the tolerable AWGN power for different signal configurations.

The continuous blue line plots the theoretical upper bound for Doppler degradation described in [203], which aligns well with the practical results of the presented simulations. In addition, the dashed black line represents the total transmitted signal power (0 dBm) and the colored dashed lines show the allowed AWGN power for an error free reception according to the selected modulation scheme and code-rate. For instance, if QPSK modulation and 4/15 code-rate is selected the total tolerable AWGN power is 2.9 dBm (violet line).

When the tolerable AWGN power values are compared with the ICI power values in the analyzed cases, it can be seen that the difference is at least 5 dB for the worst case. In this case, there will be some degradation on the receiver performance, but for many of the rest cases, especially with differences higher than 10 dB, AWGN masks completely the impact of ICI. These results confirm the viability of using higher FFT sizes (16k, 32k) for mobile scenarios.

Besides, when the ICI is high (high receiver speeds), there is a Doppler noise contribution that cannot be neglected as it can be even similar to the AWGN contribution. For this reason, its power should be taken into account in the LDPC decoding process based on LLR. Figure 64 shows two performance curves in terms of SNR threshold for a QPSK 3/15 signal over a TU6 channel for different normalized Doppler values (fdTu).

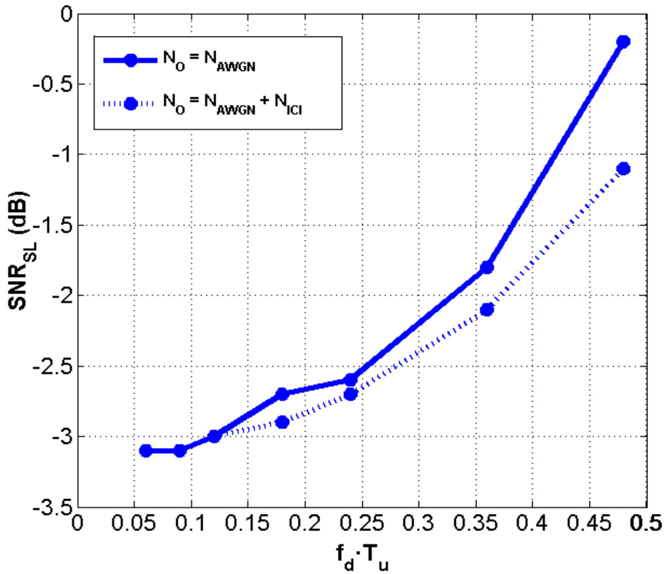


Figure 64. SNR thresholds for different mobile conditions and noise estimation algorithms.

The continuous line ($N_0 = N_{AWGN}$) represents the case where the ICI power is not considered for the overall noise calculation, whereas the dashed line ($N_0 = N_{AWGN} + N_{ICI}$) represents the case where the overall noise power, Gaussian plus Doppler, is taken into account.

For high Doppler scenarios, a SNR threshold gain of almost 1 dB can be achieved if the Doppler Noise contribution is considered. Nevertheless, for low speed scenarios, there is a small gain, always lower than 0.5 dB, or even no gain for pedestrian speeds. The obtained results are in line with the results obtained in Study G.

Time Interleaving in LDM

The main objective of this section is to confirm the impact of different TI lengths on the receiver performance in mobility. Figure 65 shows the results of the simulations carried out for evaluating the performance in terms of SNRUL threshold, using the four different TI depths described in the ATSC 3.0 standard (200, 150, 100 and 50 ms) which correspond to 1024, 887, 724 and 512 rows of a convolutional interleaver, respectively [186].

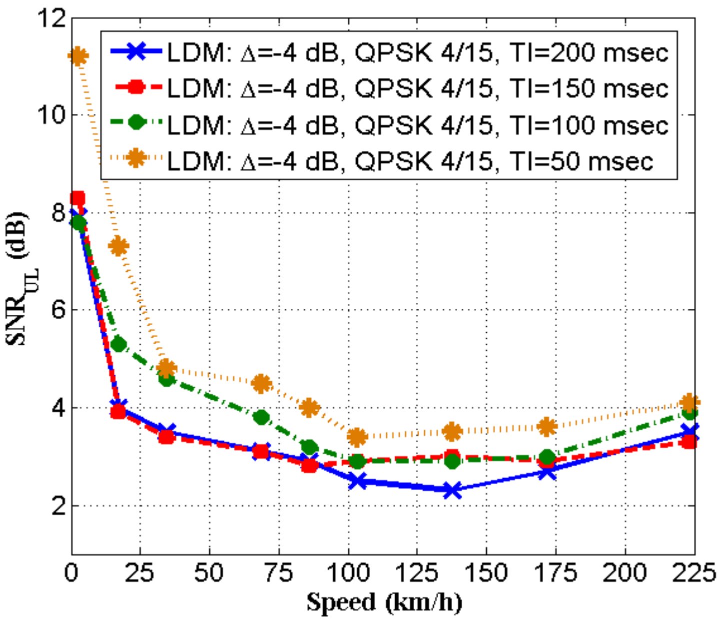


Figure 65. UL SNR thresholds for different receiving speeds and the four ATSC 3.0 time interleaving lengths.

Increasing the TI length provides higher gains at low speed scenarios (speed < 20 kmph), where critical fading appear (about 3 dB difference between the extreme TI depths). For high speed scenarios (speed > 20 kmph), the time variability of the channel acts as a natural interleaver itself, and therefore, the gain is lower (ranging from 0.5 to 1.5 dB for minimum to maximum TI depths respectively).

Cell and Frequency interleavers have not been tested because as it has been demonstrated in Study H, its influence on the reception performance is very low for very robust configurations as tested in LDM UL.

Upper Layer Code-rate

Figure 66 shows the SNR threshold values for the LDM UL with several code-rate configurations. The FFT size is 16k and the GI length is 150 ms. The modulation is QPSK and the code-rate ranges from 3/15 to 6/15, covering capacities ranging from about 2 Mbps to 4 Mbps respectively, enough for the delivery of HD services [101] [102]. Finally, it should be mentioned that in this case the LDM signal has a -4 dB injection level and the maximum TI length defined in ATSC 3.0 (200 ms) is used.

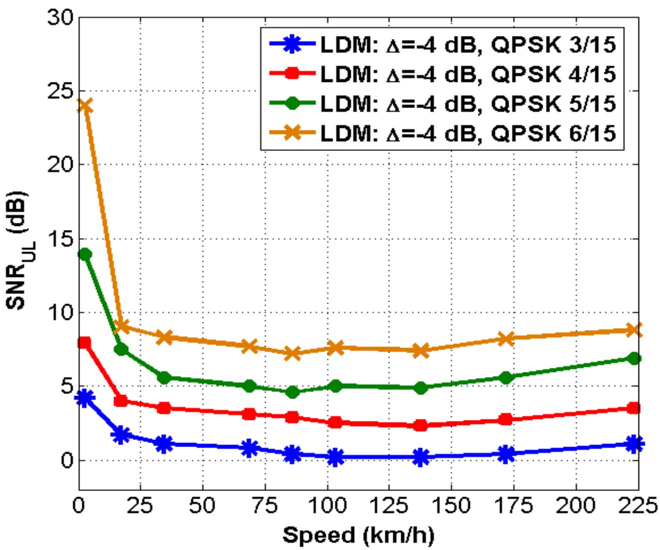


Figure 66. SNR thresholds for different code-rates in ATSC 3.0

On the one hand, the difference in SNR threshold for different code-rates depends on the speed of the receiver, with differences of about 2 dB between consecutive code-rates for high speed scenarios (speed > 20 kmph). However, the differences in SNR threshold for low speed scenarios (speed < 20 kmph) can be up to 10 dB considering consecutive code-rates.

On the other hand, the SNR thresholds at low speeds are higher because the biggest challenge is not the ICI, but the possible flat fading that may happen due to the channel slow time variability. Therefore, in order to overcome this drawback, the time interleaver should be increased. Thus, depending on the target use case, the time interleaving length is more important than the ICI on a specific FFT size associated to a certain receiver speed.

Finally, the performance curves are almost flat for speeds that range from 10 kmph to 175 kmph, meaning that the ICI degradation due is not significant and remains masked under the AWGN. However, for very high speeds (speed > 175 kmph) the ICI impact and exceeds the AWGN. In this case the SNR threshold is degraded accordingly, as explained in Inter-Carrier Interference (ICI).

3.2.3.2 Laboratory measurements

ATSC 3.0 Configuration

As laboratory measurements can be very time consuming, the number of configurations to be tested is highly reduced considering theoretical results. As a first approach, the delivery of about 3 Mbps in the mobile/indoor service is considered enough for two SD or one HD services when HEVC is used [101] [102]. The stationary service, on the contrary, can range from 8 Mbps to 25 Mbps, depending on the expected content quality.

As it has been already described in [147] in a low-profile complexity LDM receiver, both layers are added at the BICM output, and thus they share some configuration parameters for the OFDM physical waveform: TI, GI, and FFT size. For this study a 16k FFT has been selected with 1/16 GI, which is a good compromise between the Doppler resilience tolerance for the UL and the overhead due to the GI for the LL. The chosen PP is $PP_{6,2,}$ where Dx=6 and Dy=2, which offers a density strong enough to perform an accurate channel estimation under the worst multipath scenarios.

The other important configuration parameters can be found in Table 47 and have been chosen considering the theoretical studies. Three different UL configurations have been considered. The three of them are suitable for mobile and indoor reception ad they are very robust and have enough capacity for the delivery of HD services [101] [102].

Table 47. ATSC 3.0 LDM Signal Configuration for mobile and indoor reception

Main Changing Parameters			
TI Depth	**Injection Level**	**Upper Layer**	
		M-CR	**Capacity (Mbps)**
1024 512	-4 dB, -5 dB	QPSK 3/15	2.0
		QPSK 4/15	2.6
		QPSK 5/15	3.3

Main Common Parameters				
Bandwidth (MHz)	**FFT / IG / PP**	**Frame Length**	**Lower Layer**	
			M-CR	**Capacity (Mbps)**
6	16k|1/16| $PP_{6,2}$	200 ms	64 NU-QAM 7/15	13.7

The capacity shown in Table 47 is calculated based on (7).In this case, for a 6 MHz channel, an occupied bandwidth of 5.75 MHz is considered. In addition, two injection levels, IL={-4,-5} dB, have been selected. These values offer a very good balance between enhancing the mobile layer performance and maintaining a reasonable coverage for fixed services. Finally, the two extreme TI depths defined in ATSC 3.0 CTI mode (1024 and 512) have been selected to study the TI implication in a real system.

ATSC 3.0 SDR Receiver

Taking as a reference the SW platform defined in ATSC 3.0 Emulation Platform, a functional C/C++ based custom professional SDR receiver has been built by TSR research group in the UPV/EHU. This SW is a modified version of the previous DVB-T2 receiver. The main differences appear on the SW part as the HW components are the same than with DVB-T2 standard.

This receiver also has two operation modes: pseudo-real time and offline. Besides, the same quality and graphic information is provided.

Set-up

The implemented laboratory test bench is depicted in Figure 67. It is based in the two phases analysis methodology used in previous studies [105].

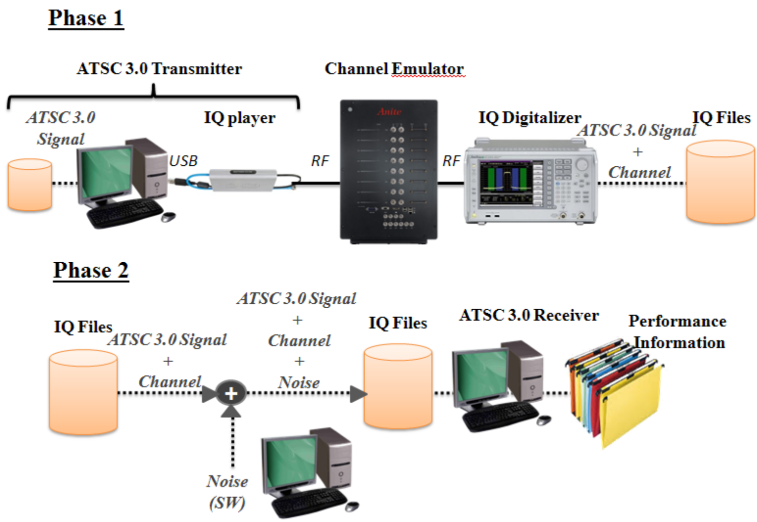


Figure 67. Laboratory measurements set-up

Firstly, the ATSC 3.0 signal must be generated, passed through the desired channel model and finally stored in a hard disk. The first half of this process is SW based, where the signals are generated, as IQ files, in a PC running the ATSC 3.0 baseline physical waveform SW implementation defined in Transmission Chain. The HW part consists of a general purpose Vector Signal Generator (VSG) with the capability of modulating the IQ files into the selected RF channel, which is defined in 590 MHz. The transmitter is connected to a RF channel emulator where the desired channel models are implemented. Finally, its output is directly recorded in a hard disk by a Vector Signal Analyzer (VSA), which digitalizes the signal fed into its RF input.

Mobility and indoor reception with handheld devices cases should be tested, as they are very important business models [44]. Even if in the literature there is a wide range of channel models, in this approach the analysis is going to be restricted to the TU6 [196] for mobility and PI and PO [76] for handheld reception in indoor and outdoor scenarios, respectively, as they are the most representative channel models to validate the use cases chosen. Besides, they are the most used channels in broadcasting and therefore, a direct comparison to a lot of previously presented mobile performance results is feasible.

In the second phase, which is based on SW, all the data stored in the hard disk has to be post-processed so as to obtain the system performance. For this purpose, increasing values of AWGN power are added by SW to the stored IQ file, in steps of 0.2 dB. As the tested channel models are mobile, the noise is injected symbol by symbol in the frequency domain, guaranteeing a controlled constant relation between the signal and noise powers. Afterwards, all the data is processed with the receiver defined in ATSC 3.0 SDR Receiver, in order to obtain the SNR thresholds. The implemented channel estimation and carrier recovery methods can be found in [104]. In these laboratory tests, it is considered that the reception is correct when the FBER is null [148] during a measured time established in 10 s.

3.2.3.3 Performance Results

This section describes the performance evaluation of the ATSC 3.0 LDM signal configurations defined for mobile and indoor reception. Only UL performance results are presented as LL influence over the UL is desired to be checked. Besides, the LL targets fixed reception and it does not contribute to mobile and indoor reception. The results have been divided in two different subsections: mobile and indoor reception performance.

Mobile Reception

In this subsection, the LDM configurations performance for mobile scenarios by means of laboratory measurements is analyzed. Figure 68 and Figure 69 show the SNR threshold for the UL configurations defined in Table 47 considering a 1024 TI depth (200 ms length) and 512 TI depth (50 ms), respectively, for different receiver speeds.

The results show the same tendency than the simulations the theoretical studies with exception of low speeds where the obtained SNR thresholds are much lower than expected. This may be because of the possible big differences between different realizations of the TU6 at 3 kmph in the simulations and in the laboratory measurements.

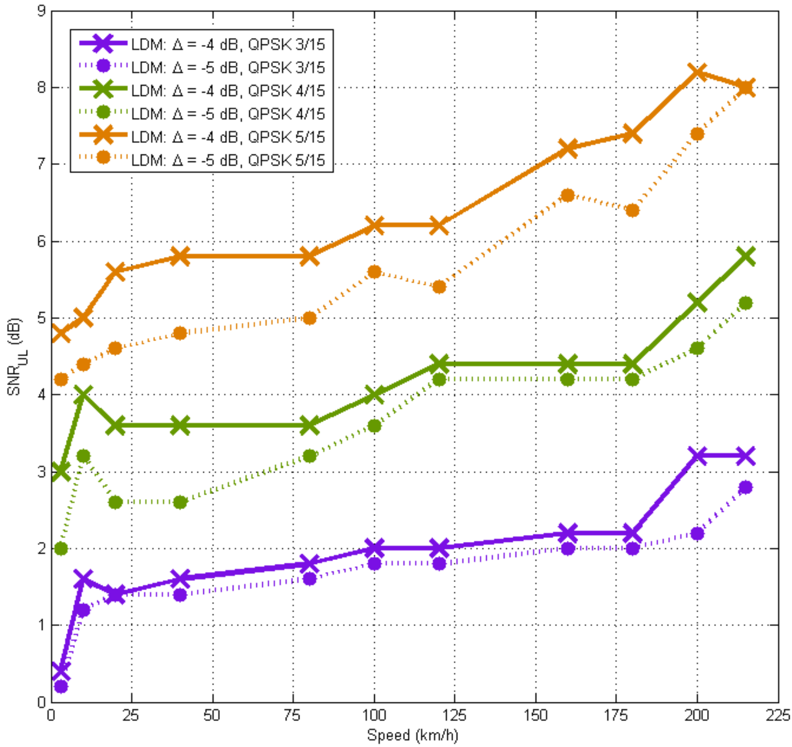


Figure 68. Performance evaluation of ATSC 3.0 for different code-rates and injection levels in mobile scenarios for 1024 TI depth.

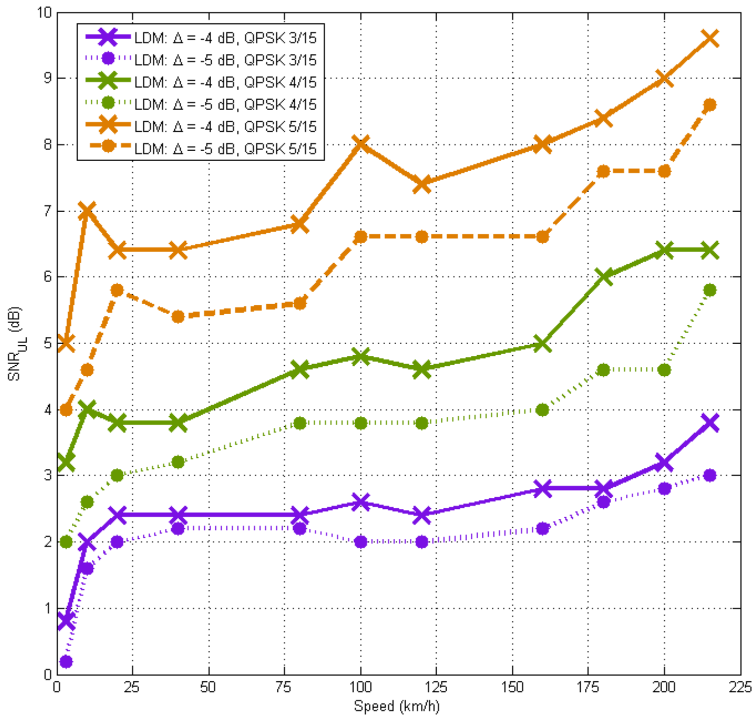


Figure 69. Performance evaluation of ATSC 3.0 for different code-rates and injection levels in mobile scenarios for 512 TI depth.

On the one hand, QPSK 3/15, which is the most robust tested configuration, shows an almost flat performance for speeds between 30 and 175 kmph with differences lower than 0.5 dB. However, a decrement in the robustness means an increment in the performance slope with differences of up to 1.5 dB for QPSK 4/15 and 2.0 dB for QPSK 5/15. This is because the SNR thresholds in these cases are closer to the ICI power due to the receiver movement and, thus, the degradation increases. Higher speeds (speed > 175 kmph) show always additional degradation as the receiver uses well-known channel estimation, interpolation and filtering algorithms that are not optimized for very high speeds and the performance could be improved for these challenging scenarios.

On the other hand, the change in the injection level between -5 and -4 dB means degradation in the performance of between 0 and 2.4 dB, with a median degradation value of 0.6 dB, which agrees with the simulation results in the theoretical studies.

Finally, the SNR threshold for different TI depths also follows the tendency from the simulations from the theoretical analysis for high speed reception (speed > 20 kmph) with a median gain value of 0.6 dB between the biggest (1024) and the smallest (512) TI depth values. In case of low speed reception (speed < 20 kmph), the gain due to the use of longer TI is not as high as expected based on the simulation results. It takes values up to 2 dB, but the median value is 0.3 dB. This is because there are not slow fadings in the majority of the tested realizations of the TU6 at low speed and, consequently, the gain due to the longer TI cannot be correctly appreciated.

Moreover, the performance under the PO channel model has been also tested. Table 48 shows the SNR threshold for the UL considering different IL and TI depths. On the one hand, the gain due to the increment in the IL from -4 to -5 dB ranges between 0 and 1.4 dB, with a median degradation value of 0.5 dB, which agrees with the simulation results in the theoretical analysis. On the other hand, the gain due to longer TI length is lower than 0.6 dB due to the absence of slow fadings in the tested realizations of the PO channel model.

Table 48. ATSC 3.0 Pedestrian Outdoor Performance SNR (dB).

Pedestrian Outdoor					
UL Configurations		**Time Interleaving Depth & Injection Level**			
		1024		**512**	
		-4 dB	**-5 dB**	**-4 dB**	**-5 dB**
QPSK	3/15	0.0.	0.0	0.6	0.0
QPSK	4/15	2.0	1.6	2.0	2.0
QSPK	5/15	4.6	3.2	4.6	3.4

All in all, considering the best results in terms of TI depth (1024) and IL (-5 dB), the UL SNR thresholds for speeds lower than 175 kmph are always lower than 2, 4.2 and 6.4 dB for code-rate 3/15, 4/15 and 5/15, respectively, which makes ATSC 3.0 a very promising standard for satisfying the handheld generation customers.

Indoor Reception

In this subsection, the LDM configurations performance are analyzed for indoor scenarios, which are supposed to be considered as one of the most promising business model for the broadcasting industry. For this purpose, the TU6 at 3 kmph and the PI channel models have been tested. The minimum SNR reception thresholds for indoor scenarios are gathered in Table 49.

Table 49. ATSC 3.0 Indoor Performance SNR (dB).

Pedestrian Indoor					
UL Configurations		**Time Interleaving Depth & Injection Level**			
		1024		**512**	
		-4 dB	**-5 dB**	**-4 dB**	**-5 dB**
QPSK	3/15	0.6	-0.4	0.6	0.0
QPSK	4/15	2.2	2.0	3.0	2.2
QSPK	5/15	4.6	4.0	5.6	4.0
Typical Urban 6 Paths 3 kmph					
UL Configurations		**Time Interleaving Depth & Injection Level (dB)**			
		1024		**512**	
		-4 dB	**-5 dB**	**-4 dB**	**-5 dB**
QPSK	3/15	0.4	0.2	0.8	0.2
QPSK	4/15	3.0	2.0	3.2	2.0
QSPK	5/15	4.8	4.2	5.0	4.0

The obtained results follow the same tendency than in the mobile study. On the one hand, he gain due to the increment in the IL ranges between 0.2 and 1.6 dB, with a median degradation value of 0.7 dB, while the gain due to longer TI length is lower than 1 dB. On the other hand, the TU6 at 3 kmph and PI channel models show similar performance results, with differences always lower than 0.8 dB, being in general the TU6 a slightly more SNR demanding channel model.

All in all, considering the best results in terms of TI depth (1024) and IL (-5 dB), the UL SNR thresholds indoor reception is always lower than 0.2, 2 and 4.2 dB for code-rate 3/15, 4/15 and 5/15, respectively.

3.2.3.4 Conclusions of this study

Some of the ATSC 3.0 configuration parameters have been studied in order to determine their optimal configuration for the simultaneous HD mobile and UHD fixed services delivery. It is important to find a trade-off between capacity and robustness when selecting the injection level in an ATSC 3.0 LDM signal. This is because the lower the injection level, the more power is shared with the lower layer. And the upper layer signal power is lower and, consequently, the SNRUL threshold increases. Besides, as the Upper layer signal will be usually dedicated to mobile and indoor portable services, it is important to take into account the effect of ICI on the receiver. The ICI acts as an additional noise that has to be taken into account, especially at high speeds. Moreover, an increment on the time interleaving length means a decrement on the SNR threshold for upper layer signals, especially for pedestrian speeds. This performance improvement is higher than in single layer systems as the upper layer robustness is degraded due to the presence of the lower layer. Finally, the effect of the upper layer code-rate is noticeable, especially for pedestrian speeds.

Once the LDM parameters theoretical influence has been studied in detail, practical laboratory measurements have been presented to prove the feasibility of the system testing the ATSC 3.0 performance with real equipment. The results are very close to the expected values, and therefore, they prove the ATSC 3.0 suitability to address the requirements of the new generation services offering simultaneous UHD services to fixed receivers (in the LL) and HD indoor or mobile services (in the UL) for SNR thresholds of up to 4.6 or 5.6 dB, in pedestrian outdoor and indoor scenarios, respectively. In addition, the SNR requirements for mobile reception increase up to 10 dB for very high speeds (200 kmph).

3.3 Improvement Study

One study has been carried out so as to improve the performance in ATSC 3.0 when LDM is considered.

- **Study K: Improving LDPC Decoding Performance for ATSC 3.0 LDM profiles (under revision)**

This study analyzes the gain in terms of performance of LDM over TDM when different LLR PDFs are considered: traditional, LDM optimized (from Study F) and new LDM semi-optimal approaches with lower implementation requirements.

3.3.1 Study K: Improving LDPC Decoding Performance for ATSC 3.0 LDM profiles

The main objective of this study is to analyze the gain in terms of performance of LDM over TDM when different LLR PDFs are considered (traditional, LDM optimized and LDM semi-optimal approaches). By this way, the best trade-off between robustness or capacity improvement and practical implementation is presented. Additionally, different channel conditions are analyzed by means of computer simulations under several channel model conditions.

3.3.1.1 Theoretical Study

Service Types and Bitrates requirements

In this study, UHD and HD quality services are considered. As it has been explained in “More Video Quality” section, the most recommended format for HD services is 1080p with 50/60 Hz [156]. In case of UHD quality, the preferred format is 2160p. In addition, an enhanced quality UHD can also be provided by means of improvements gathered in the BT.2020 recommendation [31]: HFR of up to 120 Hz, WCG covering 75% of CIE 1931 color space and HDR with more natural colors closer to those in real life. All these enhanced features can be applied in return to an increment in the necessary bitrate of about 8 Mbps [205] [206] [207] [208].

As ATSC 3.0 uses HEVC [35], in this study SHVC video coding algorithm is considered to reduce the necessary bitrates for the UHD services due to the fact that SHVC [36] reduces the necessary bitrate when the same information has to be transmitted in two different qualities, as often happens with simultaneous transmissions in TDM and LDM.

As it has been analyzed in Study E, the required output bitrate is still under discussion depending on the perceptual quality. However, based on more recent literature [208], Table 50 gathers some suggestions of the necessary bitrates for HD and UHD services with SHVC that slightly updates those in the Study E. The requirements for UHD stay almost unchanged whereas the HD quality necessary bitrate is reduced in about 0.5 Mbps in comparison to previous studies [156] [100] [101] [102]. Moreover, the specific considered bitrates in this study are also presented.

Table 50. HD and UHD Necessary Bitrates with HEVC and SHVC (Mbps)

Format	Suggested Bitrates with SHVC	Considered Bitrates in this study
HD (1080p)	2.0-3.5 (Base Layer)	2.7
UHD (2160p)	10.0-15.0 (Enhanced Layer)	12.0
Enhanced UHD	18.0-23.0 (Enhanced Layer)	20.0

Reception Scenarios and SNR Requirements

Current broadcasters should provide services in all kind of scenarios, including indoor and outdoor locations with static or portable/mobile receivers. For this reason, four different scenarios with different robustness requirements are considered [158] (the same as in Study E). Table 51 resumes the minimum SNR threshold ranges (for correct reception in AWGN channels) for the four target scenarios. Besides, the considered SNR threshold value in this study is also stated as an intermediate value in the suggested range. These exact values are slightly lower than those considered in Study E, so as to consider more demanding situations.

Table 51. SNR Requirements for ATSC 3.0 Reception Scenarios (dB)

Scenario	Required SNR	Considered maximum SNR
Portable Indoor (PI)	-6.0-1.5	0.5
Mobile Outdoor (MO)	1.5-8.0	4.5
Static Indoor (SI)	8.0-14.0	11.0
Static Outdoor (SO)	14.0-24.0	20.0

ATSC 3.0 Configuration Parameters

The studied ATSC 3.0 configurations are based on the bitrates and SNR threshold requirements from Table 50 and Table 51, respectively. All of them include the simultaneous delivery of UHDTV to rooftop antennas and HD services to mobile, or even indoor handheld devices.

Between all the configuration parameters in ATSC 3.0, only those which really mean a change in maximum offered bitrate and SNR threshold (modulation scheme (*M*), code-rates (*CR*), time division percentages (*%Time*) in TDM and injection level (*IL*) in LDM) are considered. The capacity calculation of each configuration is based on equation (7), with an occupied bandwidth (*BW*) of 5.7 MHz (6 MHz). A compromise 16k FFT length has been selected for LDM configurations as it is shared between mobile and enhanced layers. In TDM, on the contrary, there is no such limit, and thus, 8k has been considered for the mobile service and 32k for the stationary service. A guard interval (*GI*) of 148.15 μs has been considered to avoid ISI from transmitters located up to 45 km away. Regarding the pilot patterns, in LDM both services share $PP_{6,2}$, which is the most dense option and means an overhead around 8% (PP_{over}). In TDM each service has its own pattern: $PP_{6,2}$ for the mobile case and $PP_{12,2}$ for the stationary case, with a reduced overhead of around 4%.

The SNR threshold of each ATSC 3.0 BICM configuration, defined as SNR for the single layer system (*SNR-SL*), is gathered in Study H: ATSC 3.0 BICM Analysis. These are directly the theoretical SNR threshold values for each service in TDM, but these values must be corrected for LDM depending on the IL as defined in equations (8) and (9) for the mobile (UL) (*SNR-UL*) and enhanced services (LL) (*SNR-LL*), respectively.

After analyzing in terms of capacity and robustness of all the possible configurations in ATSC 3.0 to provide simultaneous UHDTV to rooftop antennas and HD services to portable and indoor receivers, some of the suitable ATSC 3.0 configurations have been defined in Table 52.

Table 52. Capacity (Mbps) and SNR (dB) Requirements for Measured Configurations

Configuration	Service	Capacity	SNR
TDM	Mobile: QPSK 11/15 40% Stationary: 256 NU-QAM 12/15 60%	2.8 18.8	3.5 20.4
LDM #1	Mobile & Indoor: QPSK 4/15 IL = -3 Stationary: 64 NU-QAM 10/15	2.6 19.6	0.2 17.6
LDM #2	Mobile & Indoor: QPSK 4/15 IL = -4 Stationary: 64 NU-QAM 10/15	2.6 19.6	-0.5 18.3
LDM #3	Mobile: QPSK 6/15 IL = -4 Stationary: 64 NU-QAM 10/15	3.9 19.6	2.9 18.3

As it has been previously demonstrated on the literature [132] [154], LDM improves TDM. However, in order to make some comparisons, a TDM configuration with the required capacity values for the UHD fixed service and HD mobile (but not portable indoor) has been also considered. In the case of LDM, three different configurations have been considered with different balanced energy sharing between fixed and mobile services.

LLR PDF Optimization for LDM

As it was widely demonstrated in Study F, the new LLR PDF formula described in (11) and (12) can be used in the LDPC decoding process in the reception of LDM UL signals to improve their performance. By this way, the gain of LDM configurations in comparison to TDM can be even higher.

The new multilayer optimized PDF formula for the UL decoding process depends on the specific LL constellation points. In fact, the receiver complexity gradually increases with the increment in the LL constellation order, which is usually high to deliver high capacity services. What is more, handheld receivers, which are very restrictive, especially in terms of size, weight and price [209], are supposed to receive only UL signals, which are generally very robust and use low order constellations. The need to store all ATSC 3.0 high order constellation points in order to adapt to all the possible LL signals means an increment in the receiver memory requirements. These two main issues are a practical implementation drawback in mobile receivers. For this reason, semi-optimal UL LLR PDF formulas will be tested in this study considering lower order LL constellation alternatives with independence of the specific LL constellation order as shown in Table 53, which resumes the modulation and code-rate considered for the semi-optimal PDF formulas in each combination.

Table 53. Combination and Associated Constellation for Semi-optimal LLR PDF

Comb.	MOD-COD.	Comb.	MOD-COD.
1	Classical PDF	14	16 NU-QAM 12/15
2	Optimized (64 NU-QAM 10/15)	15	16 NU-QAM 13/15
3	QPSK	16	64 NU-QAM 2/15
4	16 NU-QAM 2/15	17	64 NU-QAM 3/15
5	16 NU-QAM 3/15	18	64 NU-QAM 4/15
6	16 NU-QAM 4/15	19	64 NU-QAM 5/15
7	16 NU-QAM 5/15	20	64 NU-QAM 6/15
8	16 NU-QAM 6/15	21	64 NU-QAM 7/15
9	16 NU-QAM 7/15	22	64 NU-QAM 8/15
10	16 NU-QAM 8/15	23	64 NU-QAM 9/15
11	16 NU-QAM 9/15	24	64 NU-QAM 11/15
12	16 NU-QAM 10/15	25	64 NU-QAM 12/15
13	16 NU-QAM 11/15	26	64 NU-QAM 13/15

By this way, the possible degradation because of the consideration of lower order LL constellations can be measured so as to determine a good trade-off between capacity and implementation requirements.

3.3.1.2 Computer Simulations

Simulation Platform

The complete ATSC 3.0 emulation platform implemented in this thesis and described in "ATSC 3.0 Emulation Platform" section is used in order to evaluate the system performance under different scenarios. This tool allows any modification to the standard, enabling the possibility of carrying out performance tests of different ATSC 3.0 parameters implementing the traditional [172], the LDM optimized (Study F) and the suggested semi-optimized LDM LLR PDF (considering lower LL constellation orders) for the UL decoding process are implemented.

Simulation Procedure

These simulations are conducted with the purpose of evaluating the performance of the delivery of UHD services to rooftop antennas and HD services in indoor and mobile scenarios. For this purpose, the SNR threshold for correct reception is measured for several channel models to emulate all kind of reception conditions.

In addition to the traditionally considered stationary channel models (AWGN, F1 and P1) [59], the UHD service is tested in Ray6, Brazil-E & C, CRC-41, 2 & 3, while the robust HD service is also analyzed under indoor (PI, PA, PB) and mobile (PO, VA, VB, TU6) channel conditions. All the selected channel models are defined in the ATSC Proposed Recommended Practice document [195].

The reception is considered error free when the BER value at the outer coder output is lower than 10^{-6} [59] during 500 ms for stationary channels and during 2 s for portable and mobile channel conditions. In order to reach this situation, increasing AWGN in steps of 0.1 dB is added at the receiver input until the first errors appear in the receiver (BER higher than 10^{-6}). The SNR threshold is obtained as the relation between the mean signal power level and the mean external AWGN power level.

This procedure is repeated for the different ATSC 3.0 considered configurations and for all the considered channel models. Furthermore, different combinations are performed considering different LLR PDF formulas are in LDM cases to measure the performance differences.

3.3.1.3 Performance Results

LDM Improvement over TDM with optimized LLR PDF

On the one hand, the SNR threshold for the mobile & indoor service with TDM and LDM is gathered in Table 54 for different channel models. In LDM configurations, the optimized UL LLR PDF formula (Op.) (Study F) as well as the classical formula (Cl.) [172] are considered. In TDM, only the Cl. formula is considered. The LL performance is not included as the considered LLR is always the traditional one.

Table 54. SNR (dB) Thresholds for the mobile and indoor service in TDM and LDM with Classical and Optimized PDFs

Channel	TDM	LDM#1		LDM#2		LDM#3	
	Cl.	Cl.	Op.	Cl.	Op.	Cl.	Op.
AWGN	3.9	0.4	0.4	-0.2	-0.2	3.0	3.0
DVBT-F	4.4	0.8	0.7	0.2	0.2	3.5	3.4
DVBT-P	7.4	2.4	2.1	1.4	1.3	6.1	5.6
RAY6	6.4	1.9	1.7	1.1	1.0	5.3	4.9
BRAZIL-E	7.8	2.7	2.4	1.7	1.6	6.7	5.8
BRAZIL-C	6.7	1.9	1.8	1.1	1.1	5.5	5.0
CRC-1	5.3	1.3	1.2	0.5	0.5	4.4	4.1
CRC-2	6.5	1.9	1.8	1.0	1.0	5.4	4.9
CRC-3	6.6	1.9	1.7	1.0	1.0	5.4	4.9
PA	9.1	10.4	8.3	8.0	7.4	10.2	9.1
PB	10.4	9.4	8.0	9.2	7.8	11.9	10.5
PI	8.3	5.5	4.8	4.0	3.6	7.5	6.6
PO	8.4	5.1	4.4	2.6	2.4	7.5	6.0
VA	8.9	5.1	4.5	3.1	2.9	7.8	6.7
VB	8.6	5.0	4.3	3.0	3.0	7.5	6.7
TU6 (129 Hz)	8.8	4.9	4.4	3.2	3.0	7.0	6.0

As it can be seen from Table 54, the three LDM measured configurations are always more robust than the TDM for the mobile and indoor service delivery, even with the classical PDF approach. The less robust LDM configuration (LDM #3) is between 0.9 and 1.3 dB better than TDM in stationary reception whereas this improvement is up to 1.8 dB in mobile channels. What is more, if the LDM optimized PDF formula is considered, the gain in terms of robustness of LDM over TDM is even higher with values up to 2 dB in stationary reception and 2.6 dB in mobility for LDM #3. There is an exception for PA and PB channel models as LDM show worse performance. This can be due to the huge differences between different realizations of these channels.

In general, the improvement due to the use of the LDM optimized PDF formula shows a similar tendency than in Study F, with higher gains for the less robust configurations (LDM #1 with lower IL and, especially LDM #3 with less robust code-rates). Under stationary channel models, the improvement ranges between 0-0.1 dB (LDM #2) up to 0.9 dB (LDM #3). The gain under indoor and mobile conditions is much higher with values up to 1.5 dB (or even higher for the PA/PB channel models).

On the other hand, the SNR threshold for the stationary service is gathered in Table 55 for TDM and LDM configurations for different channel models. In this case, LDM also improves TDM performance for all the tested channel models.

Table 55. SNR (dB) Thresholds for the stationary service in TDM and LDM

Channel	TDM	LDM#1	LDM#2	LDM#3
AWGN	20.7	18.1	19.0	18.9
DVBT-F	21.3	18.5	19.4	19.4
DVBT-P	24.7	21.9	22.7	22.9
RAY6	23.5	20.8	21.7	21.7
BRAZIL-E	24.3	21.8	22.4	22.6
BRAZIL-C	23.7	20.9	22.0	21.8
CRC-1	22.3	19.5	20.6	20.7
CRC-2	23.8	20.9	22.0	22.0
CRC-3	23.8	21.0	21.9	21.9

Optimized vs Semi-optimized UL LDM LLR PDF formulas

Figure 70 shows the SNR thresholds for LDM #3 configuration for different stationary channel models considering different decoding approaches.

Under stationary channel model conditions, combinations 3, 4 and 16 (and sometimes 13, 14 and 15) show a similar performance to the classical approach (combination 1) whereas the rest combinations show a similar performance than the optimized PDF (combination 2). However, the semi-optimized LLR PDF never means degradation in comparison to the classical approach. In addition, the most of the modulation and code-rate combinations have a similar performance to the optimized approach.

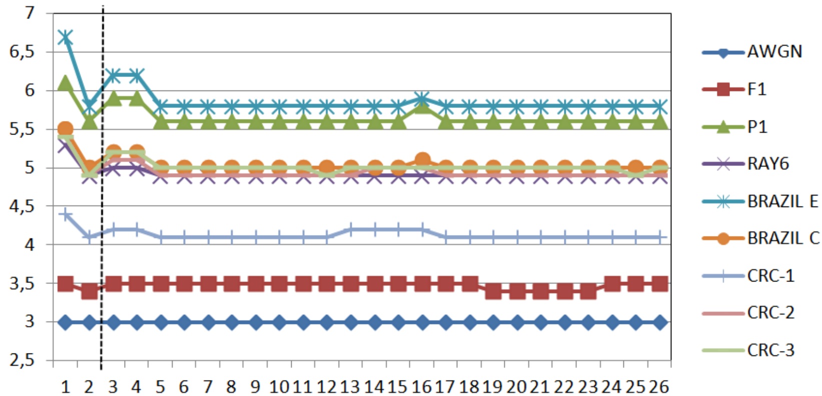


Figure 70. SNR Threshold for LDM #3 for different LLR PDF approaches under stationary channel model conditions

The situation follows the same tendency under indoor and mobile channel conditions for the three LDM measured configurations, as it can be seen in Figure 71 for LDM #3.

The use of combinations 3, 4 and 16 (and sometimes 13, 14, 15, 24 and 25) shows slightly worse performance than the rest of the combinations. However, in this case, the performance of the worst combinations is much better than the classical approach, with degradations of up to 0.4 dB in comparison to the optimized approach.

The small differences in performance for the different combinations can be due to the high robustness of the selected UL, which targets indoor and mobile reception. In these cases, the threshold situation is reached for high noise power levels and, consequently, the influence of the specific LLR PDF is lower.

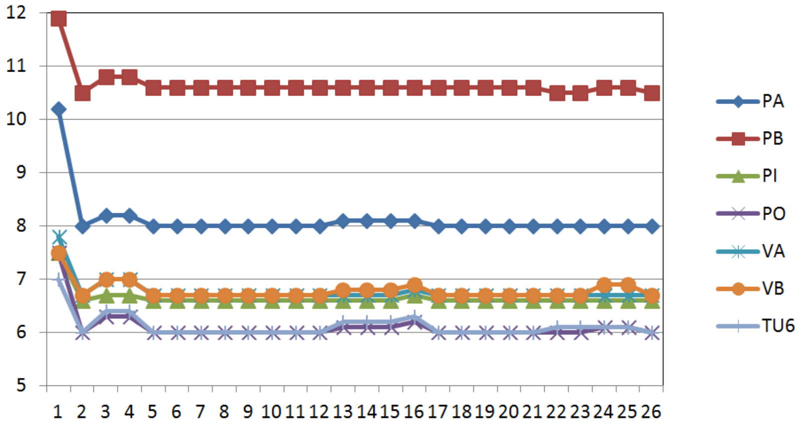


Figure 71. SNR Threshold for LDM #3 for different LLR PDF approaches under indoor and mobile channel model conditions

3.3.1.4 Conclusions of this study

This study includes a comparative analysis between TDM and LDM to offer HD indoor/mobile services as well as UHD stationary services. In addition, different LDPC decoding PDF have been considered: the classical approach, the LDM optimized approach described in Study F and several semi-optimal approaches in order to reduce the implementation complexity in the receiver.

Some computer simulations under stationary, indoor and mobile channel models have been presented showing that LDM is always more robust than TDM in at least 0.9 dB. However, if the optimized formula is considered, there is an extra gain of up to 0.9 dB in stationary reception and 1.5 dB in mobility.

The semi-optimal approaches, considering lower LL constellations than the existing one, show very promising results. The most of the options show a similar performance than when using the optimized approach. There are only a few cases with a degradation of up to 0.4 dB in comparison to the optimized approach. For this reason, it has been demonstrated that the consideration of a different LL than the existing one means mainly the same performance than considering the optimized one.

4. Summary

In this chapter, some of the first ATSC 3.0 performance results by means of computer simulations have been included so as to complete the performance evaluation first phase, needed to evaluate any DTT standard.

For this purpose a complete ATSC 3.0 emulation platform has been completely designed and implemented. This tool is useful because an ATSC 3.0 transmitter and receiver are totally implemented. By this way, performance simulations of any of the ATSC 3.0 configuration parameters can be carried out.

Study H obtained the performance and spectral efficiency of all the BICM existing combinations of modulation and code-rate for AWGN and P1 reference channel models. These values can be considered as reference in order to determine the most suitable configuration depending on the desired bitrate and robustness as these are the parameters with the major influence on the signal robustness. The spectral efficiency ranges between 0.3 and 10.4 bps/Hz, whereas the Eb/N_0 ranges between -0.6 and 22.6 dB in AWGN and -0.1 and 26.2 dB under P1 channel conditions, resulting in a BICM 1 dB closer to the Shannon limit than DVB-T2.

Moreover, the different options of interleaving performance have been theoretically measured in fixed, indoor and mobile scenarios, concluding, as determined in Study I, that their use does not mean real improvement in static reception. Furthermore, the time interleaving has been shown to be the interleaver with higher improvements (of up to 3.2 dB) in mobility with independence of the receiver speed. The frequency interleaving and the use of subslices improve the performance in about 0.4 and 0.6 dB, respectively.

Additionally, different multiplexing techniques to deliver several services have been analyzed. On the one hand, as LDM is a novelty in ATSC 3.0, a deep study of this technique has been carried out. First, the influence of some of the configuration parameters for LDM has been measured in Study J so as to determine the optimal configuration for different scenarios. For this purpose, the performance for different injection levels, code-rates and time interleaving lengths has been theoretically evaluated as well as the consideration of the ICI power as an additional source of interference. Based on these theoretical studies, suitable configurations for the simultaneous delivery of HD services in indoor and mobile scenarios, and UHD services in fixed scenarios have been defined. A LDM configuration is the most effective one with a very robust upper layer and high capacity lower layer. These configurations performance has been also tested in the laboratory for mobile and indoor reception and, consequently, with emphasis in the upper layer of the LDM signal which targets challenging scenarios. These results show the feasibility of using ATSC 3.0 LDM for the delivery mobile HD services in pedestrian outdoor and indoor scenarios with SNR of up to 4.6 and 5.6 dB, respectively. The SNR requirements for very high speed reception increase up to 10 dB.

On the other hand, Study K has proved that LDM is more efficient than TDM for the simultaneous transmission of HD mobile/indoor services and UHD stationary services. When the classical approach is considered for the UL decoding, LDM gains TDM in at least between 0.9 and 1.3 dB for stationary channel models and between 0.8 and 1.8 dB in indoor and mobile scenarios. The situation improves for LDM when a multilayer optimized decoding algorithm is considered, improving the gain over TDM up to 1.8 dB and 2.6 dB under stationary and indoor/mobile channel conditions, respectively. In addition, several semi-optimal decoding algorithms have been proposed to decrease the complexity and memory requirements of the optimized solution while improving the LDM system performance in a similar way than with the optimized algorithm. There are only a few cases with a degradation of about 0.4 dB.

Taking everything into consideration, the first step of the ATSC 3.0 performance evaluation process with computer simulations has been widely measured completing the existing system parameters performance studies. Additionally, some recommendable configurations have been defined and tested in the laboratory for new desired scenarios, such as indoor and mobile.

CHAPTER V: Contributions & Future Work

This chapter contains a summary of the main contributions of this thesis, as well as a brief description of the dissemination of those contributions. In addition, possible future research lines in the scope of the contributions of this thesis are suggested.

1. Contributions

The driving force for this research work is the willing to contribute to the existing new generation broadcasting standards performance evaluation and improving processes. In particular, this thesis is focused on checking the feasibility of recent DTT systems to give high quality services in very challenging scenarios in a spectrum efficient way.

More precisely, the main contributions of the research work carried out in this thesis can be divided in the same three major chapters the thesis is organized: DVB-T2 indoor studies, studies of new techniques for next generation DTT systems and ATSC 3.0 studies.

1.1 DVB-T2 Indoor Studies

The DVB-T2 performance evaluation process has been completed with studies about indoor reception in several laboratory measurements and field trials. In addition, the most suitable DVB-T2 configurations for indoor services delivery have been suggested obtaining a signal very robust to slow fading present in indoor scenarios, whose main parameters are presented in Table 56.

Table 56. Suggested DVB-T2 configuration parameters for DVB-T2 indoor reception

Configuration Parameter	Suggested Value (Optimal)
FFT Size	Small (8k)
Pilot Pattern	Dense (PP2)
Time Interleaver	Long (250 ms)
Modulation	Low order (QPSK)
Code-rate	Low (1/3, 1/2)

Furthermore, on the one hand, SNR thresholds have been obtained on the laboratory for several DVB-T2 configurations under different channel models intended for emulating indoor reception. Each measured channel model has each own features, showing differences in the performance of up to 9.2 dB. For this reason, it is very difficult to define a reference channel model. On the other hand, SNR thresholds have been also obtained on the field determining comparatively that the channel model Indoor Office A (IOA), defined by the ITU, is the most accurate to emulate real portable indoor reception in the laboratory, but with 3.3 dB of extra degradation because of the indoor too low time variability.

Moreover, a new detailed methodology for indoor performance study has been proposed, as it is shown in Figure 72. It is based on the calculation of time and location corrections factors (TCF and LCF, respectively) in order to obtain a SNR threshold for the 99% time and 95% locations.

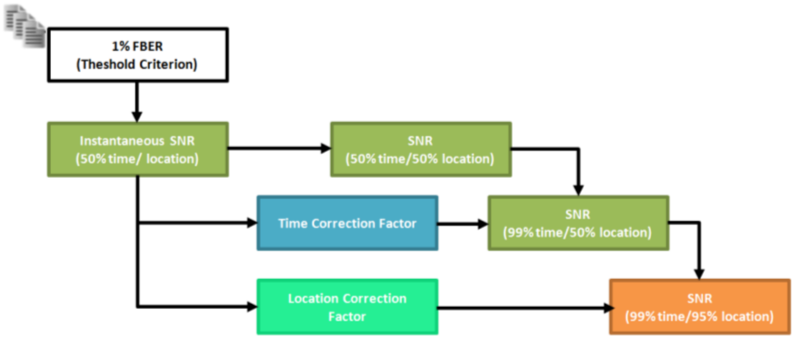


Figure 72. Suggested methodology for the study of DVB-T2 indoor performance.

As a result of all the indoor reception analyses, it can be concluded that DVB-T2 indoor reception of HD services is possible for SNR higher than about 8.9 dB and 15.4 dB for fixed and portable reception, respectively. By this way, the feasibility of the system to offer HD indoor services has been proved. The specific values following the methodology described in Figure 72 are gathered in Table 57.

Table 57. Suggested DVB-T2 SNR threshold, TCRF and LCF (dB) for indoor reception

	50% Time / 50% Location	**TCF**	**LCF**	**99% Time / 95% Location**
Fixed	3.0	3.5	2.4	8.9
Portable	5.7	8.1	1.6	15.4

As a result, the DVB-T2 coverage studies conducted in chapter II show that indoor reception is possible with current broadcasting networks. In fact, HD quality services can be correctly received with lower power levels than the existing in current broadcasting networks in Spain.

1.2 Studies of new techniques for the next generation DTT systems

New technologies have been applied to the traditional DVB-T2 standard so as to check the improvement in terms of robustness and efficiency. On the one hand, the suitability of recently suggested QC LDPC codes in order to increase the robustness of the DTT signals for new challenging scenarios have been tested by means of field trials in indoor locations. The indoor reception with these new codes is correct from 2.3 dB and 3 dB for fixed and portable reception, respectively. The robustness of the signal increases in about 0.7 dB in comparison to DVB-T2 most robust LDPC codes.

In addition, LDM technique has been proved to be efficient in the HD indoor services delivery in the UL or even UHD indoor services in the LL. For HD indoor reception, 4.7 dB are needed, whereas for UHD indoor services the SNR increases up to 19.5 dB. Anyway, these values show the feasibility of this new technology to provide DTT services indoors.

On the other hand, some new decoding algorithms have been defined and proved improving the reception performance, especially for multilayer signals and mobile scenarios.

First, a new LDPC decoding algorithm based on the adaptation of the LLR PDF to multilayer signals have been proposed showing performance gains of up to 4.5 dB when LDM technology is used in stationary reception. It is based on the IL value between the two layers in a LDM system as well as on the specific LL constellations as indicated in equations (22) and (23).

$$PDF_i\left(I_{MLr}, Q_{MLr} / I_{MLt}, Q_{MLt}\right) = \frac{1}{2\pi \times N_{AWGN}} e^{-\left(\frac{\left(I_{MLr}-\rho_r I_{MLt}\right)^2-\left(Q_{MLr}-\rho_r Q_{MLt}\right)^2}{2\times N_{AWGN}}\right)} \quad (22)$$

$$\begin{aligned}(I_{MLr}, Q_{MLr}) &= (I_{ULr}, Q_{ULr}) + 10^{-IL/20} \times (I_{LLr}, Q_{LLr}) \\ (I_{MLt}, Q_{MLt}) &= (I_{ULt}, Q_{ULt}) + \sum_k 10^{-IL/20} \times (I_{LL_k}, Q_{LLk})\end{aligned} \quad . (23)$$

Furthermore, different low complexity ICI power estimators have been tested in a DVB-T2 receiver so as to determine the most accurate in the estimation process of the ICI power present in portable and mobile reception that has to be considered as an additional noise source. The two most accurate ICI power estimators, which similarly depend on the Doppler spread and space between carriers, are presented in equations (24) and (25).

$$N_{ICI1} = P_s \frac{\pi^2}{3}\left(\frac{f_D}{\Delta f}\right)^2 \quad (24)$$

$$NIC_{ICI2} = P_s \frac{74\pi}{5}\left(\frac{f_D}{\Delta f}\right)^{2.88} . \quad (25)$$

With both ICI power estimators, performance improvements of about 0.5 dB (for medium receiver speeds) or up to more than 10 dB (for high speeds) have been measured.

1.3 ATSC 3.0 Studies

The first step of the ATSC 3.0 evaluation process has been performed in order to determine the system performance under different situations. By this way, the feasibility of this system to offer HD services in challenging scenarios in a more efficient way than previous existing standards have been demonstrated.

For this purpose, a complete ATSC 3.0 emulation platform has been designed and implemented to have a tool that allows the system performance simulation. With this tool, the required SNR for different challenging situations have been measured.

On the one hand, as a result of the studies carried out in Chapter IV with the ATSC 3.0 emulation platform developed during the thesis, the main features for all the possible combinations of the new NUCs and LDPC codes in the ATSC 3.0 BICM are resumed in Table 58.

Table 58. Measured ATSC 3.0 BICM Main features

	Minimum	Maximum
Spectral Efficiency (bps/Hz)	0.3	10.4
Eb/N$_0$ in AWGN (dB)	-0.6	22.6
Eb/N$_0$ in P1 (dB)	-0.1	26.2

On the other hand, the influence on the performance of all the interleavers defined in ATSC 3.0 has been checked for a very robust configuration. In fixed reception, all the interleavers influence on the performance is almost negligible. In mobility, on the contrary, the situation changes showing the performance improvements resumed in Table 59.

Table 59. Measured ATSC 3.0 interleaving performance gain

Interleaving Option	Performance Gain (dB)
Frequency Interleaver	0.4
Convolutional Time Interleaver	1.4 (mobile) / 3.2 (pedestrian)
Hybrid Time Interleaver	2.0
Hybrid Time Interleaver + Cell Interleaver	2.2
Subslicing	0.6

The interleaver that really improves the system performance in mobility is the time interleaver, masking the effect of the resting interleaving options.

Moreover, the LDM technique has been successfully tested in the ATSC 3.0 system for the simultaneous transmission of UHD contents to rooftop antennas and HD contents in mobile or indoor scenarios. Besides, the IL, BICM and time interleaver influence on the performance have been also tested with similar results to those obtained in single layer systems. Additionally, LDM has been proved to be more efficient than the traditional TDM approach in the simultaneous delivery of HD mobile/indoor and UHD stationary services. This gain is even higher if the multilayer optimized LLR PDF is considered. In addition, several semi-optimal LLR PDF approaches have been also tested reducing the receiver complexity and memory requirements and with a very similar performance to the optimized approach. The gains of LDM over TDM with different decoding algorithms (classical, multilayer optimized and semi-optimal approaches) are resumed in Table 60.

Table 60. Measured LDM over TDM gain (dB) in ATSC 3.0

Decoding Algorithm	Stationary	Indoor and Mobile
Classical approach	0.9-1.3	0.8-1.8
LDM Optimized approach	0.9-1.8	1.7-2.6
Best Semi-optimal approach	0.9-1.8	1.7-2.6

Finally, some of the first ATSC 3.0 laboratory measurements have been carried out, adapting a DVB-T2 custom professional SDR receiver to this new standard. In this context, the feasibility of ATSC 3.0 for the simultaneous transmission of mobile or indoor HD contents and UHD fixed contents by means of LDM has been also verified in the laboratory for SNR of up to 4.6 and 5.6 dB in pedestrian outdoor and indoor scenarios, respectively. The SNR requirements for very high speed reception increases up to 10 dB.

2. Dissemination

Equally to the thesis contributions, all the studies carried out during the thesis development can be organized in the same three major chapters the thesis is organized.

2.1 DVB-T2 Indoor Studies

DVB-T2 indoor evaluation studies are related to the studies described in Chapter II. They have been being disseminated as follows:

Title: "DVB-T2 field trials results for portable indoor reception using T2-Lite and multiple PLP".

Authors: **C. Regueiro**, G. Berjon-Eriz, I. Perez de Albeniz, I. Eizmendi, G. Prieto and M. Velez.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), London (England), pp. 1-5.

Date: June 2013.

DOI: 10.1109/BMSB.2013.6621781

Contribution: This paper includes the study and evaluation of the DVB-T2 feasibility to offer portable indoor reception by means of a coverage study with a real network testing different configuration parameters. This research is described in Study B.

Title: "Caracterización de la recepción portatil de señales de DVB-T2 en entornos indoor"

Title (in English): "Characterization of DVB-T2 portable reception in indoor environments".

Authors: **C. Regueiro**, X. Gomez, M. Velez and U. Gil.

Publication: XXIX Simposium Nacional de la Unión Científica Internacional de Radio (URSI), Valencia (Spain), pp. 1-4.

Date: September 2014.

Contribution: This paper includes the study and performance evaluation of the DVB-T2 portable indoor reception based on laboratory measurements under different channel model conditions. This research is described in Study A

Title: "Field Trials Based Planning Parameters for DVB-T2 Indoor Reception".

Authors: **C. Regueiro**, U. Gil, M. Velez, I. Eizmendi and P. Angueira.

Publication: IEEE Trans. on Broadcasting, vol. 61, no. 2, pp. 251-262.

Date: February 2015.

DOI: 10.1109/TBC.2015.2400814

JCR Impact Factor: 2.381. Q1 (12/82) .in Telecommunications.

Contribution: This paper includes the study and performance evaluation of the DVB-T2 indoor reception based on laboratory measurements and field trials. Besides, a new methodology for indoor performance analysis is also defined. Moreover, the most appropriate channel model for indoor reception is also studied. This research is described in Study C.

Title: "Field Trial Results of DVB-T2 Mobile Reception".

Authors: I. Eizmendi, **C. Regueiro**, I. Sobrón, I. Fernandez and M. Velez.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Nara (Japan), pp. 1-4.

Date: June 2016.

DOI: 10.1109/BMSB.2016.7522007

Contribution: This paper includes the study and performance evaluation of the DVB-T2 mobile reception based on field trials and different channel estimation methods. The field trials campaign definition and the analysis methodology considered in this research are derived from those considered in Study C.

2.2 Studies of new techniques for the next generation DTT systems

The studies related to new techniques for the next generation DTT systems have been defined in Chapter III. They have been being disseminated as follows:

Title: "Cloud Transmission System Performance in Portable Indoor Environments".

Authors: **C. Regueiro,** U. Gil, I. Fernández, I. Eizmendi, M. Velez, D. Guerra, I. Peña and D. de la Vega.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Beijing (China), pp. 1-5.

Date: June 2014.

DOI: 10.1109/BMSB.2014.6873504

Contribution: This paper includes the study and evaluation of the new QC LDPC codes in indoor scenarios by means of coverage studies in the field, checking their improvement in terms of robustness in comparison to previous existing LDPC codes. This research is described in Study D.

Title: "Cloud Transmission System Performance for Mobile Urban Scenarios in the Field".

Authors: U. Gil, **C. Regueiro**, I. Angulo, I. Eizmendi, M. Velez, D. de la Vega and J-L. Ordiales.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Beijing (China), pp. 1-5.

Date: June 2014.

DOI: 10.1109/BMSB.2014.6873505

Contribution: This paper includes the same research than the previous one but in mobile scenarios. The field trials campaign definition and the analysis methodology are very similar to those described in Study D.

Title: "Performance Study of Layered Division Multiplexing Based on SDR Platform".

Authors: J. Montalban, I. Angulo, **C. Regueiro**, Y. Wu, L. Zhang, S-I. Park, J-Y. Lee, H-M. Kim, M. Velez and P. Angueira.

Publication: IEEE Trans. on Broadcasting, vol. 61, no. 3, pp. 436-444.

Date: June 2015.

DOI: 10.1109/TBC.2015.2432463

JCR Impact Factor: 2.381. Q1 (12/82) .in Telecommunications.

Contribution: This paper includes the study and performance evaluation of LDM in fixed and mobile scenarios by means of computer simulations and laboratory measurements with a SDR LDM receiver. The considered simulation platform and the SDR LDM receiver are the same as the considered in Study D and Study E, respectively. Some work has been carried out helping in their definition, implementation and validation.

Title: "SHVC and LDM Techniques for HD/UHD TV Indoor Reception".

Authors: **C. Regueiro,** J. Barrueco, J. Montalbán, U. Gil, I. Angulo, I. Eizmendi, P. Angueira and M. Velez.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Ghant (Belgium), pp. 1-6.

Date: June 2015.

DOI: 10.1109/BMSB.2015.7177224

Contribution: This paper includes a performance evaluation and an analysis of the LDM suitability to efficiently offer HD and UHD indoor services by means of laboratory measurements. This research is described in Study E.

Title: "LDM and TDM Performance Evaluation for Next Generation Broadcasting".

Authors: J. Montalban, **C. Regueiro**, M. Velez, L. Zhang, Y. Wu, W. Li, H-M. Kim, S-I. Park and J-Y. Lee.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Ghant (Belgium), pp. 1-6.

Date: June 2015.

DOI: 10.1109/BMSB.2015.7360814.

Contribution: This paper includes a comparative study of LDM and TDM multiplexing technique to offer HD mobile and UHD fixed services by means of computer simulations. The considered simulation platform is the same considered in Study D. Some work has been carried out helping in their definition, implementation and validation.

Title: "Performance Evaluation of Different Doppler Noise Estimation Methods".

Authors: **C. Regueiro,** J. Barrueco, J. Montalban, I. Sobron, I. Eizmendi and M. Velez.

Publication: International Symposium on Advances in Wireless and Optical Communications (RTUWO), Riga (Latvia), pp. 125-128.

Date: November 2016.

DOI: 10.1109/RTUWO.2016.7821869

Contribution: This paper includes the comparative performance and tolerance to errors evaluation of different low complexity ICI power estimator algorithms, as well as the definition of a new one. This research is described in Study G.

Title: "LLR Reliability Improvement for Multilayer Signals".

Authors: **C. Regueiro,** J. Barrueco, J. Montalban, P. Angueira, J.L. Ordiales and M. Velez.

Publication: IEEE Trans. on Broadcasting, vol. 63, no. 1, pp. 275-281.

Date: March 2017.

DOI: 10.1109/TBC.2016.2617280

JCR Impact Factor: 2.381 (2015). Q1 (12/82) .in Telecommunications.

Contribution: This paper includes the definition and performance improvement evaluation of a new decoding algorithm for multilayer signals. This research is described in Study F.

2.3 ATSC 3.0 Studies

All the studies related to ATSC 3.0 are defined in Chapter IV. They have been being disseminated as follows:

Title: "LDM Core Services Performance in ATSC 3.0".

Authors: **C. Regueiro**, J. Montalban, J. Barrueco, M. Velez, P. Angueira, Y. Wu, L. Zhang, S-I. Park, J-Y. Lee and H-M. Kim.

Publication: IEEE Trans. on Broadcasting, vol. 62, no. 1, pp. 244-252.

Date: January 2016.

DOI: 10.1109/TBC.2015.2505411

JCR Impact Factor: 2.381 (2015). Q1 (12/82) .in Telecommunications.

Contribution: This paper includes a evaluation of different ATSC 3.0 configuration parameters related to LDM technique. In addition, a performance evaluation based on laboratory measurements have been carried out for fixed, indoor and mobile scenarios. This research is described in Study J.

Title: "ATSC 3.0 Interleavers Influence in Reception Performance".

Authors: **C. Regueiro**, J. Barrueco, J. Montalban, I. Eizmendi and M. Velez.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Nara (Japan), pp.1-4.

Date: June 2016.

DOI: 10.1109/BMSB.2016.7521940

Contribution: This paper includes a performance evaluation of all the interleaving options defined in ATSC 3.0 by means of computer simulations. This research is described in Study I.

Title: "Improving LDPC Decoding Performance for ATSC 3.0 LDM profiles".

Authors: **C. Regueiro**, J. Barrueco, J. Montalban, I. Eizmendi and M. Velez.

Publication: IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), Cagliari (Sardinia), pp. 1-5.

Date: June 2017 **(ACCEPTED).**

Contribution: This paper includes a performance comparison of TDM and LDM to offer HD mobile and UHD fixed services. Additionally, the performance with different decoding algorithms has been considered. This research is described in Study K.

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3. Future Work

Taking into account that this thesis describes the research activities carried out through a temporal evolution of DTT systems from DVB-T2 to ATSC 3.0, the following step should be directed to study the situation from ATSC 3.0 to the near future. In particular, DVB has recently started to research about new broadcasting techniques that could be considered in the future of broadcasting systems. In addition, there are still some pending research topics of the new techniques for the next generation DTT systems studied in the thesis.

More precisely, this research work could be continued in three different directions.

3.1 Additional Studies of new techniques for next generation DTT systems

In this context, some of the studies about new techniques for next generation DTT systems can be continued in the following way:

- Study the performance improvement in hardware of the new decoding algorithm optimized for different multilayer signals by means of laboratory measurements.
- Study the performance improvement in hardware of considering different ICI power estimator methods in the receiver by means of laboratory measurements.
- Study the performance improvement of combining the new multilayer and ICI power estimator algorithms in the receiver by means of computer simulations and laboratory measurements.

In addition, some research can be focused on the WiB technique, which makes use of some of the techniques studied in the thesis, in the following way:

- Study and performance evaluation of the multilayer signal generated in a WiB scenario with 1-frequency reuse.
- Study the application of the new multilayer optimized decoding algorithm

to a multilayer WiB environment.

- Carry out WiB one frequency reuse coverage studies in different reference cities so as to help the broadcasters in the network planning.

3.2 Additional ATSC 3.0 Performance Studies

In this context, as the ATSC 3.0 is still being widely evaluated, more research work is needed in order to complete the system performance evaluation process. More specifically:

- Study the influence on the performance of ATSC 3.0 advanced features (time aligned mode, segmentation options, multiple subframes, MISO, MIMO, PAPR, channel bonding) by means of computer simulations.
- Carry out field trials of ATSC 3.0 under different environments conditions so as to determine the system performance in real scenarios.
- Carry out ATSC 3.0 coverage studies to provide different kind of services in different reference cities so as to help the broadcasters in the network planning.

References & Glossary

This part of the thesis includes all the bibliography references in order of appearance. Besides, all the acronyms and abbreviations present in the document are alphabetically ordered.

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2. Glossary

1D	One Dimensional
2D	Two Dimensional
ALP	ATSC Link layer Protocol
ASI	Asynchronous Serial Interface
ATSC	Advanced Television System Committee
ARIB	Association of Radio Industries and Business
AWGN	Additive White Gaussian Noise
B21C	Broadcast for 21st Century
BB	Baseband
BBC	British Broadcasting Corporation
BCH	Bose, Ray-Chaudhuri and Hocquenghem
BER	Bit Error Rate
BICM	Bit-Interleaved Coded Modulation
BL	Base Layer
CDF	Cumulative Distribution Function
CI	Cell Interleaver
CIE	Comission Internationale de l´Eclairage
CL	Core Layer
COST	European COoperation in the field of Scientific and Technical Research
CR	Code-rate
CRC	Communications Research Centre
CRP	Correct Reception Percentage
CSP	Common Simulation Platform
CTI	Convolutional Time Interleaving
DTMB	Digital Terrestrial Multimedia Broadcasting
DTT	Digital Terrestrial Television
DVB	Digital Video Broadcasting
EBU	European Broadcasting Union
EC	European Commission
EL	Enhanced Layer
ENGINES	Enabling Next GeneratIon NEtworks for broadcast Services
ERP	Effective Radiated Power
ETSI	European Telecommunications Standards Institute
F1	Ricean Channel Model
FBER	FEC Blocks Error Rate

FD	Frequency Shift
FDM	Frequency Division Multiplexing
FEF	Future Extension Frame
FER	Frame Error Rate
FFT	Fast Fourier Transform
FI	Frequency Interleaving
FURIA	Futura Red Integrada Audiovisual
GBD	Greater Bit Depth
GI	Guard Interval
GPS	Global Positioning System
HD	High Definition
HDMI	High-Definition Multimedia Interface
HDR	Higher Dynamic Range
HEVC	High Efficiency Video Coding
HFR	Higher Frame Rate
HTI	Hybrid Time Interleaving
HW	Hardware
IBC	International Broadcasting Convention
ICI	Inter-Carrier Interference
IF	Intermediate Frequency
IL	Injection Level
IOA	Indoor Office A
IOB	Indoor Office B
IOPA	Indoor Outdoor and Pedestrian A
IOPB	Indoor Outdoor and Pedestrian B
IP	Internet Protocol
IQ	In-phase Quadrature
ISDB	Integrated Services Digital Broadcasting
ISI	Inter Symbol Interference
ITU	International Telecommunications Union
JCR	Journal Citation Report
L1	Layer 1
LCF	Location Correction Factor
LDM	Layered Division Multiplexing
LDPC	Low Density Parity Check
LFSR	Linear Feedback Shift Register
LL	Lower Layer
LLR	Log-likelihood ratio
LOS	Line-of-sight
MER	Modulation Error Rate

ML	Multilayer
MO	Mobile Outdoor
MPEG	Moving Picture Experts Group
MR	Motorway Rural
NAB	National Association of Broadcasters
NG-RADIATE	Nueva Generación de sistemas de Radiodifusión DigitAl Terrestre
NUC	Non-Uniform Constellations
OFDM	Orthogonal Frequency Division Multiplexing
P1	Rayleigh Channel Model
PAPR	Peak-to-Average Power Ratio
PDF	Probability Density Function
PI	Pedestrian Indoor
PLP	Physical Layer Pipe
PO	Pedestrian Outdoor
PP	Pilot Pattern
PRBS	PseudoRandom Binary Sequence
QAM	Quadrature Amplitude Modulation
QC	Quasi-Cyclic
QEF	Quasi Error Free
QPSK	Quadrature Phase-Shift Keying
RF	Radiofrequency
RS	Reed Solomon
RTP	Real-time Transport Protocol
RX	Receiver
SD	Standard Definition
SDR	Software Defined Radio//Standard Dynamic Range
SFN	Single Frequency Network
SHVC	Scalable High Efficiency Video Coding
SI	Static Indoor
SL	Single Layer
SMA	SubMiniature version A
SNR	Signal to Noise Ratio
SO	Static Outdoor
SSD	Signal Space Diversity
STB	Set-Top Box
SW	Software
TCF	Time Correction Factor
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing

TI	Time Interleaving
TS	Transport Stream
TU6	Typical Urban 6 paths
TV	Television
TX	Transmitter
Txn	Transmission
UC	Uniform Constellation
UDP	User Datagram Protocol
UHD	Ultra High Definition
UHF	Ultra High Frequency
UK	United Kingdom
UL	Upper Layer
UPV/EHU	University of the Basque Country
USA	The United States of America
USRP	Universal Software Radio Peripherical
VA	Vehicular A
VB	Vehicular B
VPN	Virtual Private Network
VSA	Vector Signal Analyzer
VSG	Vector Signal Generator
VU	Vehicular Urban
V&V	Verification and Validation
WCG	Wider Color Gamut
WiB	WideBand reuse-1
WING-TV	Services to wireless, integrated, nomadic, GPRS-UMTS & TV