The birth of Digital Terrestrial Broadcasting in Europe: a brief history of the creation and the standardization phases of Digital Audio Broadcasting (DAB) and Digital Terrestrial TV broadcasting (DVB-T)

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Abstract — **Research work on Digital Audio Broadcasting (DAB) began in Europe in the middle 1980's within the European Broadcasting Union. The CCETT (Centre Commun d'Etudes de Télédiffusion et Télécommunications) contributed to these studies and proposed to analyze the performance of a multi-carrier transmission scheme as the PHY layer for DAB. The first concepts studied in 1986 were further refined and, in 1987, the principles of a "coded OFDM" transmission scheme for digital audio broadcasting to mobile receivers were proposed. Later on, this scheme was also adopted to form the European Digital Terrestrial TV broadcasting standard (DVB-T).**

Index Terms — **Broadcasting, DAB, DVB-T, multi-carrier, OFDM.**

I. Introduction

Major European broadcasters and Consumer Electronic Industry companies decided in the late 80's to combine their efforts to develop a new fully digital technology for audio broadcasting, so that the broadcast segment of the audio chain become digital, as well as the production and storage (CD) segments. In 1987, these actors formed together the EU 147 DAB consortium, under the banner of the Eureka projects.

In the meantime, CCETT (Joint Research Centre for Broadcasting and Telecommunications), part of the R&D branch of France Télécom and Télédiffusion de France, had invested research means into modulation systems able to efficiently mitigate multipath in the mobile radio channel [1] [2], and was investigating the benefits of multicarrier transmission. The studied transmission scheme was called COFDM (Coded Orthogonal Frequency Division Multiplex), where the "C" underlines the necessary conjunction of multi-carrier transmission, channel Coding and interleaving, in order to maximize the system performance.

Audio coding had made at that time sufficient advances to reach a CD-like quality for stereophony at a bit rate lower than 2x128 kbps. The sound coding system, developed by IRT (Institüt für Rundfunk Technik, research centre of the German public broadcasters), Philips and CCETT, was called MUSICAM. The first hardware developments of the so-called COFDM-MUSICAM system were made at CCETT for the COFDM part and at IRT for the MUSICAM part, between fall 1987 and summer 1988. This first implementation was used during the ITU WARC-ORB 88 (World Administrative Radio Conference), to provide the very first DAB on-air demonstration in September 1988 in Geneva [3], using a terrestrial transmitter located on Mount Salève, in the vicinity of Geneva.

In summer 1990, after a study period of possible alternatives, the EU 147 Steering Committee decided to select the COFDM scheme. In the meantime, the MUSICAM sound coding scheme became an ISO standard (MPEG1 Layer II), so that the two basic DAB subsystems were ready to be standardized to form, together with the service information layer, a complete broadcasting system.

In 1992, it was decided to establish within ETSI (European Telecommunication Standard Institute) a project team with the goal to elaborate the DAB specification. The ETS 300 401 standard, entitled « Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers », was first issued in February 1995.

The experience gained on COFDM technology within the DAB project had a clear influence on the definition of the European standard for digital TV broadcasting. The dTTb (digital Terrestrial Television broadcasting) project, funded by the European Commission, began in 1992 and produced stable outputs for standardization by beginning of 1995, which were then approved by the DVB Forum (Digital Video Broadcasting) in November of the same year. The first issue of the DVB-T ETSI standard was released in February 1996 (EN 300 744). This standard has opened the way to the well-known successful deployment of Digital Terrestrial Television in a number of European countries and in other parts of the world.

This paper intends to recall the history of the first steps of digital terrestrial broadcasting in Europe, and at the same time to describe the technical benefits of the transmission technologies which compose these broadcasting standards.

II. The First Step of the Technology

A. The principles

Broadcasting high throughputs to mobile receivers was certainly a bet in the late 80's when considering the mobile channel constraints:

- signals are generally suffering from distortions due to multipath: multiple replica (echoes) of the transmitted

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signal reach the receiver antenna, as a consequence of reflections on surrounding obstacles;

- signals characteristics are time varying, as a consequence of the receiver motion: the various echoes combine differently as the receiver moves, leading to a time variation of the amplitude/phase channel response at a given frequency.

However, these constraints can be turned into benefits when considering the diversity in frequency and time brought by these propagation phenomena:

- signal "elements" conveyed at different frequencies will suffer from uncorrelated attenuations (assuming that these elements are conveyed at a frequency distance larger than the inverse of the echo spread – also called the channel impulse response),
- similarly, when the receiver moves, signal "elements" conveyed at different times suffer from uncorrelated attenuations (assuming that these elements are conveyed at a time distance larger than the inverse of the Doppler spread).

This frequency/time diversity can be efficiently exploited by a system which links signal "elements" transmitted at different frequencies/instants (conjunction of channel coding and frequency/time interleaving), so that if a given element is strongly attenuated by a destructive echoes combination, it will hopefully be reconstructed in the receiver thanks to the link which relates it to other (hopefully well received) elements.

This principle is shown in Fig. 1, which has been published in a number of articles [4] [5] to give a simple and comprehensive explanation of the fundamentals of coded multicarrier transmission.

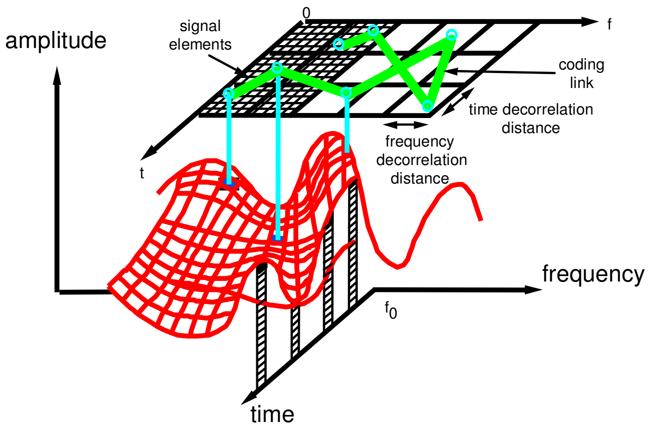


Fig. 1: propagation channel and coded multicarrier scheme representations

The surface in red is the 2-dimensional (frequency-time) representation of the channel amplitude response. The black grid shows the transmitted signal composed of juxtaposed signal "elements" (segments of amplitude and/or phase modulated sinewaves, as it will be explained below). The link between distant elements, represented in green, symbolizes the conjunction of channel coding applied on data modulating the signal elements, and time/frequency interleaving.

Starting from these principles, it was then necessary to design the signals elements and their grouping to provide the definition of the complete signal, under several constraints:

- each signal elements should not individually suffer from distortion (these elements are simply attenuated and phase-rotated by the channel effect, the channel response being stable over their individual duration/bandwidth),
- these elements should be compactly tightened together, in order to maximize the spectrum efficiency.

The scheme presented in Fig. 1 can be achieved by juxtaposing in frequency a set of modulated carriers with separate spectra (a Coded Frequency Division Multiplex), showing therefore suboptimal spectrum efficiency. The last step to design a spectrum efficient solution is to consider that carrier can spectrally overlap while keeping orthogonality (data modulating each carrier can be recovered without interference from neighboring carriers). This is achieved by the so-called ODFM scheme (Orthogonal Frequency Division Multiplex) [6]. The construction of an OFDM signal can be explained in two steps:

- carriers are modulated in amplitude/phase using a rectangular window function of duration T_u, and the carrier separation is set to $1/T_u$. This provides, in absence of multipath and Doppler, a set of orthogonal signal elements and insures an optimal spectral efficiency, as the signal density is maximized in time and frequency. The orthogonality in frequency comes from the fact that the filter matched to the detection of each individual carrier shows a null contribution from the other carriers, although they overlap. In the discrete domain, the multicarrier signal over the time duration T_u (called the OFDM symbol) results from the inverse discrete Fourier transform of the set of data modulating the carriers; these data can then be recovered in the receiver applying a discrete Fourier transform. Obviously, the orthogonality holds also in time, since successive OFDM symbols do not overlap.
- in the presence of multipath, this orthogonality is impaired because OFDM symbols time-overlap. This can be avoided by inserting a guard interval Δ, longer than the delay spread, between successive OFDM symbols. This guard interval is constructed as a cyclic prefix to ensure orthogonality over any time slice of duration T_u non-affected by multipath in the period $T_u + \Delta$.

This construction [4] is illustrated in Fig. 2, which displays the *sinc* shape of the carriers' spectrum and the rectangular time shape of the successive symbols.

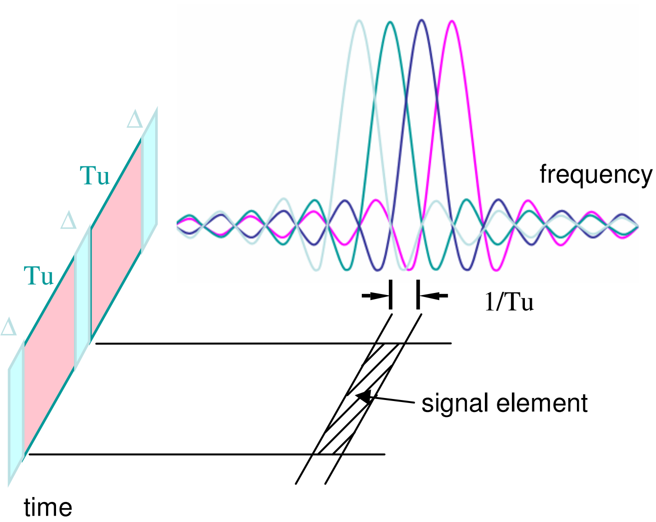


Fig. 2: time-frequency representation on an OFDM signal with guard interval

B. From theory to application

Starting from the above principles, in July 1987, the CCETT team began to assemble a hardware prototype (transmitter and receiver), convinced by the fact that the best way to promote the solution was to demonstrate real time on-air transmission. In the context of audio broadcasting, the challenge was to be ready to demonstrate a full running system one year later, during the ITU WARC-ORB conference in September 1988.

The system parameters had to be defined to show at best the system performance, while keeping the hardware development feasible. An important constraint was the choice of the Fast Fourier Transform (FFT) size (the tool to implement the discrete Fourier transform mentioned above), as very few components were available on the market at that time.

The main system parameters were chosen as follows:

useful symbol duration:	$T_u = 64$ µs
guard interval duration:	$\Delta = 16$ µs
number of transmitted carriers:	N = 448
signal bandwidth:	$B_S = 7$ MHz
FFT size:	512
channel coding:	R = 1/2, K = 7 (133, 171) convolutional code
interleaving:	Frequency: over 7 MHz Time: over 384ms
modulation:	Differential QPSK
useful bit rate:	5.6 Mbps

Table 1: Characteristics of the first OFDM/DAB implementation

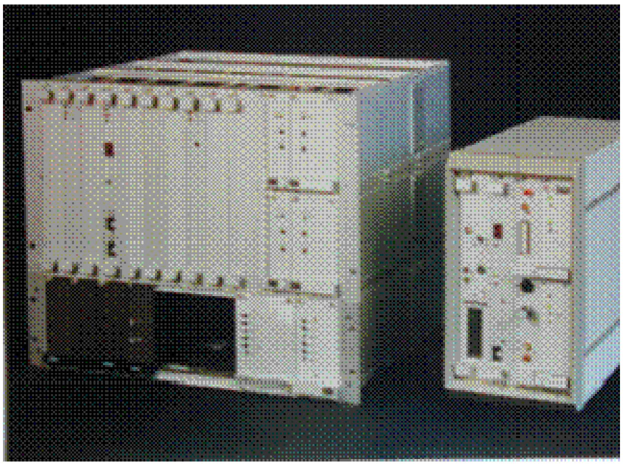
A 16 µs guard interval allowed the system to cope with echoes as long as 4.8 Km. Keeping this guard interval as ¼ of T_u led to a spectral efficiency reduction of 20%, giving a useful bit rate of 5.6 Mbps in a 7 MHz bandwidth, when using a rate ½ code and a 4-state modulation, which was considered appropriate to achieve good performance under limited signal-to-noise ratio. The total bandwidth of 7 MHz provided a fair frequency diversity (as soon as the echoes spread exceeds some hundreds nanoseconds). The carrier spacing ($1/T_u$) of 15625 Hz was large enough to avoid intra-symbol distortion due to the Doppler effect at a speed as high as 200 Km/h at 1GHz carrier frequency (in other words, the channel remained quasi-stable over the duration T_u).

The hardware built between mid-87 and mid-88 relied mostly on discrete components, except for the specific operations of FFT and Viterbi decoding of the convolutional code. The FFT was implemented by a Zoran processor, and the Viterbi decoder was designed by CCETT and developed by the SOREP company. This harware was a powerful tool to prove the feasibility and to assess the performance of a COFDM system able to broadcast a useful bit rate of 5.6 Mbps in a 7 MHz bandwidth.

C. First test and demonstrations

The hardware was properly working in CCETT lab from May 1988. Time had come to go for an on-air test. This was made possible using a UHF channel at 794 MHz, thanks to the availability in the city of Rennes of a broadcast tower owned by Télédiffusion de France (TDF). The very first transmission, in June 1988, has been a quite interesting experience. The signal spectrum could be observed in real time in the minivan where the receiver would display the amplitude channel response and its time variation, thus showing the extremely strong local fadings affecting the signal. For this first test, the time synchronization was not fully implemented, so that the FFT window had to be manually adjusted with an external synthesizer. However, the experience was quite convincing, as the Bit Error Rate was maintained to 0 during almost the whole ride. A few weeks later, the pseudo-random binary sequence was replaced by a true encoded MUSICAM sound signal, providing a crystal clear CD quality sound on the move.

Reliability tests and integration with the MUSICAM sound coding/decoding equipments of the Institüt für Rundfunk Technik were achieved during summer, and the demonstration was ready for the WARC-ORB 88 conference in September, in the city of Geneva. The transmitting equipments were installed on a TDF site on the Mount Salève, a mountain on the French side in the vicinity of Geneva. The demonstration consisted in a ½ hour tour starting from the "Centre International de Conférence de Genève", where WARC delegates were invited to listen, with headphones, to the sound coding quality and to appreciate the error-free transmission provided by the coded OFDM transmission scheme. This very first on-air public demonstration certainly played a key role in the selection of OFDM for Digital Audio Broadcasting, because it provided the real proof that a very efficient system could be industrially implemented, as it relied on the conjunction of functions easily implementable on silicon.



One of the first COFDM/DAB implementation

D. From prototype to standard

As written in the introduction, this first event and fruitful discussions with the Eureka 147 DAB project partners led to OFDM adoption by the project Steering Committee during a meeting held in Paris CDG airport in July 1990.

The period from 1990 to 1992 was used to perform experiments and field trials, in Europe, Canada and USA, to gather technical data in order to decide on the final choice of the system parameters. The main point in discussion was to determine the signal bandwidth, under opposing constraints:

- better system performance is obtained by using a large signal bandwidth, because of the benefit of frequency diversity brought by coding and frequency interleaving;
- easier frequency planning can be achieved with a smaller bandwidth;
- easier multiplexing of different source programs can be achieved when the total bit rate remains moderate, and therefore when the bandwidth is smaller.

As the VHF spectrum was targeted for DAB in several European countries, the compromise was to select a signal bandwidth of 1.5 MHz, which can accommodate 4 DAB signals within a 7 MHz VHF channel with sufficient frequency separation to achieve a good adjacent channel rejection in the receiver.

This period of system optimization was also used to test very interesting properties of coded OFDM: as the system is able to take benefit from multipath, averaging local fadings over the signal bandwidth and the time interleaving depth, it is possible to generate "artificial" echoes to fill gaps or to extend the coverage area. This can be done by locally re-amplifying the on-air signal, or by feeding a set of transmitters able to transmit the same signal at the same time on the same frequency. The first technique was called "gap filling", as it relates to local mitigation of shadowing, and the second was called "Single Frequency Network" (SFN), because the concept relies in using the same frequency over a large area to save spectrum and to ease frequency planning.

These two properties certainly contributed to the system adoption, as they opened new perspectives for broadcast network operators to construct their networks.

The ETSI project team in charge of drafting the DAB specification, from the work performed within the EU 147 « system definition » working group, was set up in 1992. The ETS 300 401 standard, entitled « Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers », was first issued in February 1995.

III. Second step: from audio to video broadcasting

Starting from the DAB experience, it was quite clear that a sound broadcasting system able to transmit simultaneously a multiplex of audio channels, would also be able to transmit a video channel using the full signal capacity. A first implementation and experimentation of video broadcasting using DAB was performed through cooperation between CCETT and Philips in 1991. Although this scheme has been selected a few years later to form the T-DMB system (Terrestrial – Digital Multimedia Broadcasting), quite popular for mobile TV in Korea, the requirements for a digital system able to replace analogue TV broadcasting would request some important technical evolutions ahead from the DAB technology.

The objective for digital TV broadcasting was to define a system able to co-exist and then replace analogue transmission, re-using the existing broadcasters' infrastructures (towers, antennas) and users installations (roof-top antenna, cables). It was also clear that the existing TV channelling had to be maintained, thus leading to the definition of a digital signal with a bandwidth compatible with 8 MHz and 7 MHz channels. Obviously, such signal had to carry a certain number of TV channels to improve the spectrum use and provide more programmes to the customers, and should be noise-resistant because, during the period of analogue and digital coexistence, digital signals had to be transmitted on adjacent analogue channels a few dBs below those channels (-10 dB to -20 dB) to avoid interfering analogue reception. Finally, the digital signal needed to be resistant to multipath, always present in terrestrial transmission.

These constraints required the design of a power and spectrum efficient system, resistant to multipath. The two factors which determine the spectrum efficiency in COFDM are:

- the choice of the coding rate and modulation
- the ratio τ between the guard interval and the total symbol duration ($\tau=\Delta/(\Delta+T_u)$)

A system using a rate ½ code and a 4-state modulation (as DAB) reaches a spectrum efficiency of 1 bit/s/Hz, further decreased by the ratio τ. Under the hypothesis prevailing in the mid 90's that a standard definition MPEG2 channel would require a rate of ~ 4-6 Mbps, a spectrum efficiency of ~ 3 bit/s/Hz was necessary to accommodate 4 to 6 TV programmes in an 8 MHz channel. This requirement led to the following choices:

- coding: punctured convolutional code with rate 1/2, 2/3, 3/4, 5/6 and 7/8
- modulation: 4-PSK, 16-QAM, 64-QAM
- Δ/T_u : 1/4, 1/8, 1/16, 1/32

On top of the selection of these parameters, a major issue was the choice of the symbol length T_u: for a given protection against multipath, determined by the value of Δ, the longer T_u is chosen, the lower is the lost of spectrum efficiency (τ). Hence, it is in theory preferable to select a large symbol length, the physical limitation coming from the phase noise impairing the performance, and the practical limitation coming from the FFT length. This discussion resulted in the choice of 2 FFT sizes: 2048 points and 8192 points, known as the 2K and 8K modes. DVB-T decoders using a 2K FFT were first on the market, allowing some broadcasters to early launch terrestrial TV broadcasting. Fig. 3 gives the available bit rates versus echoes resistance for the 2K and 8K modes.

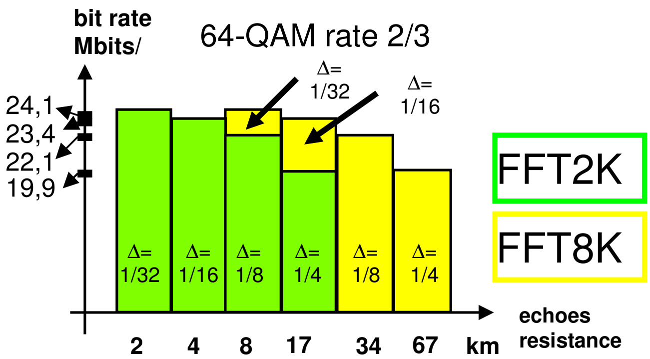


Fig. 3: available DVB-T bit rates versus echoes resistance for the 2K and 8K modes

After a long period of parameters' selection within the dTTb project, and further optimizations in a broader forum within DVB, the Technical Module of the DVB project decided upon the main DVB-T parameters during its November 1995 meeting in Geneva, paving the way to a tremendous effort of industrial companies, broadcasters and regulators in the following years, which gave birth to the nowadays well-know and largely deployed DVB-T technology.

IV. Some Technology Evolutions

In the period going from the mid-nineties, when DAB and DVB-T were standardized, to the present days, the communication technology has known some other major breakthroughs. Among them, one can mention a widespread introduction of the Multiple Input Multiple Output (MIMO) technology and, the generalization of the use of advanced coding schemes (Turbo or LDPC). In the meantime the broadcasting community also introduced some innovative techniques, firstly in the DVB-H and DVB-SH standards, and more recently in the DVB-T2 standard.

A. DVB-T2

While a couple of countries have adopted the use of MPEG4 on DVB-T to deliver HDTV, it was agreed inside DVB Forum in June 2006 that the second generation of terrestrial broadcasting had to be defined in order to improve performance and increase throughput. After about two years of intensive work, DVB-T2 specifications were finalized in June 2008, and the related standard EN 302 755 was published by ETSI in September 2009.

As DVB-T, DVB-T2 uses a COFDM modulation, with a FFT size up to 32K in order to increase SFN size without increasing overhead. An advanced coding scheme, LDPC (Low Density Parity Check) code inherited from DVB-S2 and adapted to DVB-T2 context, has been chosen in order to improve performance. An innovative T2 technology, called "rotated constellation", leads to more robustness thanks to a clever exploitation of diversity. Unlike DVB-T, a time interleaver has also been implemented in the standard. Multiple scattered pilot patterns, used for channel estimation, have been designed to minimise overhead for a given guard interval. A specific per service robustness can be defined thanks to the concept of physical layer pipe (PLP) inherited from DVB-S2. SFN coverage can also be increased thanks to the use of distributed MIMO Alamouti coding scheme; this transmit diversity scheme is applied over two distant antennas of an SFN area.

Thanks to these state-of-the-art technologies [7], a 50% increase in data throughput can be observed as shown below.

	DVB-T (current UK mode)	DVB-T2
Modulation	64QAM	256QAM
FFT size	2K	32K
Guard Interval	1/32	1/128
FEC	2/3 CC+RS (8%)	3/5 LDPC+BCH (0.3%)
Scattered Pilots	8%	1%
Continual Pilots	2.6%	0.35%
Frame structure overhead	1%	0.7%
Bandwidth	Normal	Extended
Capacity	24.1Mbit/s	35.9Mbit/s

Table 2: parameter sets of DVB-T and DVB-T2

B. From OFDM to FilterBank Multi-Carrier modulation

Further significant improvements can even be expected, and with the aim of increasing throughput and improving robustness, a novel multicarrier modulation known as

ODFM/OQAM (for OFDM Offset QAM) is currently being studied.

Actually, a common feature of the aforementioned broadcast standards is the use of OFDM scheme for modulation. With OFDM, the signal element is a rectangular window in time and *sinc* function in the frequency domain. This is the only practical choice to simultaneously get, with a base of Fourier functions, complex orthogonality together with a maximum spectral efficiency. In the end, a part of the spectral efficiency is lost when adding a cyclic prefix. Some other possibilities can be obtained, if instead, as shown in [8], we transmit, alternately in time and frequency domains, the real and imaginary part of a QAM symbol, applying the Offset-QAM (OQAM) rule. This idea of relaxing orthogonality to the real field, i.e. with respect to OQAM symbols instead of QAM ones, is analyzed in details in [5]. In this last reference, in order to optimize the transmission over time-frequency dispersive channels, the aim is to build an OFDM/OQAM scheme where time and frequency play a perfectly symmetrical role, while maintaining orthogonality. The Isotropic Orthogonal Transform Algorithm (IOTA) corresponds to an orthogonalization in time and frequency of the Gaussian function [5]. It leads to a signal element identical in time and frequency domains and orthogonal to all is neighbours in spite of their overlapping. This signal element being well concentrated in time and frequency, even without cyclic prefix, the orthogonality can nearly be preserved when transmitting over doubly dispersive channels. IOTA has been already retained in a Telecommunications Industry Association (TIA) standard [9]. In the recent years, it has also been shown that IOTA, and more generally speaking OFDM/OQAM, had some strong connections with filter bank systems and it appears that, for the next generation of wireless communication, Filter Bank Multi-Carrier (FBMC) modulation attracts a widespread interest [10].

V. Conclusion

Although the first developments of the coded OFDM technology in the field of broadcasting were targeting audio, the most popular application is currently television broadcasting. The reason is probably that FM broadcasters were not massively ready to move to digital, for various reasons, among which probably the fact that spectrum in the lower bands was not available in many countries, and Digital Audio Broadcasting at frequencies as high as L-Band would require a rather dense transmitters network to serve large coverage areas. Nevertheless, DAB has been successful in several countries, in particular UK.

Digital Television Broadcasting was answering a demand of the public and the national authorities to access a larger choice of channels. Another important objective for the public authorities was to use the spectrum more efficiently: an 8 MHz digital TV signal can transmit about 6 TV channels (depending on the selected compression), and achieves a similar coverage as former analogue channels with a much lower transmit power. This allowed freeing spectrum in the upper UHF band (the digital dividend), in order to better developing mobile internet access in this very propagation-efficient portion of the spectrum.

Finally, it is worth mentioning that the telecommunication industry has also been convinced by these technologies, since coded OFDM has been selected for several radio mobile standards such as 3GPP/LTE, IEEE802.11 and 802.16.

Acknowledgement

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