Transmission of Mobile Multimedia Services in DVB-SH Networks

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Resumen — La presente Tesina estudia la transmisión de contenidos y servicios multimedia a través de redes de difusión basadas en el sistema DVB-SH (*Digital Video Broadcasting – Satellite Services to Handheld Devices*), el cual permitirá conseguir una cobertura nacional en toda Europa de servicios de TV móvil y cuya puesta en funcionamiento está prevista para el 2008. DVB-SH es una evolución tecnológica del estándar europeo de TDT móvil DVB-H (*Digital Video Broadcasting - Handhelds*). Sus principales características son una arquitectura híbrida terrestre-satelital y la transmisión en banda S en torno a los 2 GHz. la cual está disponible en Europa para este tipo de servicios. Además, debido a las peculiaridades de la recepción vía satélite, DVB-SH incorpora nuevos mecanismos de protección entre los que destacan un entrelazador de larga duración de nivel físico y un nuevo esquema de codificación de nivel de enlace. Debido a que en la actualidad el sistema DVB-SH todavía se encuentra en una fase muy temprana de validación, el trabajo realizado en esta tesina está encaminado a evaluar los nuevos mecanismos de nivel de enlace DVB-SH a partir de trazas de errores DVB-H. Se analizará la calidad de usuario para diferentes mecanismos de protección y finalmente se identificarán aquellos esquemas y configuraciones más adecuados para las situaciones de recepción más importantes.

Abstract — This Master's thesis studies the transmission of multimedia digital contents in broadcasting networks based in the DVB-SH (Digital Video Broadcasting – Satellite Services to Handheld Devices) standard. This standard will make possible a network capable of providing mobile TV services in all Europe, which is due at 2008. DVB-SH is the technological evolution of the European mobile DTT standard DVB-H (Digital Video Broadcasting - Handhelds). Its main characteristics are a hybrid terrestrial – satellite architecture and the transmission at 2 GHz in the S-band, which is at disposal in Europe for this kind of services. Because of the peculiarities of satellite reception, DVB-SH incorporates new protection mechanisms. Among these mechanisms, a long duration interleaver and a new link layer codification scheme stand out. However, DVB-SH technology is still in an early phase of validation. Because of this, the work fulfilled in this Master's thesis is aimed at evaluating these codification schemes by means of DVB-H error traces. The user experience quality will be analyzed for different protection mechanisms and the schemes and configurations with better performance will be highlighted.

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I. INTRODUCTION AND OBJECTIVES

Nowadays mobile TV has become a very important issue in multimedia mobile communications. Mobile phones have already incorporated radio, music player, digital camera, video recorder, and mobile TV will be soon another feature. As recent market studies reveal, mobile TV interest among consumers is rapidly increasing. The demand on this kind of services is expected to rise very fast, reaching more than 500 millions of users around the globe in 2011 [1]. Although it will be necessary to adapt the services to the mobile environment, it will be possible to offer a more personal service than the offered by conventional TV. This way, a wider range of new services will be available for the consumers, along with new business opportunities for all the agents from the telecommunications and audio-visual industry. Due to the fact that these services allow a universal access to multimedia content anywhere and anytime, they are also a key element in the development of the Information Society.

Thanks to the arrival of the third generation (3G), cellular operators have recently begun to offer mobile TV services. However, their offers are considerably limited, from both technical and economical points of view [2]. In order to offer mobile TV services, the requirement of employing broadcasting networks as a complement of cellular networks, is commonly accepted. Broadcasting networks are the only ones able to cope with the capacity needed by these services. These kinds of networks are capable of distributing high speed multimedia services around wide extensions without any limitation in the number of users that can receive the service.

The European Mobile Digital Terrestrial Television (DTT) standard is known as DVB-H (*Digital Video Broadcasting - Handhelds*), which is a technological evolution of the Digital Terrestrial Television (*Digital Video Broadcasting - Terrestrial*) standard, adapted to mobile reception [3]. DVB-H was originally designed to work in the UHF band between 470 and 862 MHz, and is capable of providing a capacity between 5 and 10 Mb/s in an 8 MHz channel. Although DVB-H maintains the physical layer of DVB-T, it adds new link layer elements, making possible a reutilization of the network infrastructure (transmitters, multiplexors etc.). His main feature is a non continual transmission technique in which the information is transmitted within bursts. This technique is capable of decreasing the average power consumption of the user terminals. DVB-H also adds an additional error correction mechanism, which provides a more robust transmission, especially in a mobile environment where mobility and interference are common. Today, commercial mobile TV services are being offered in the main cities of Italy, Finland and Albany inside Europe, and of Vietnam, India and Filipinas inside Asia, and it is expected that these services will ramp up in the 2008.

Although DVB-H represents an important advance in mobile multimedia broadcasting, there are two important obstacles in order to achieve a mobile TV European market. These are the high cost of the network deployment and the frequency shortage in Europe. One of the greatest concerns about the viability of DVB-H is the large amount of infrastructure needed in order to provide an acceptable level of coverage. First market studies about mobile TV revealed that the level of coverage expected by mobile TV users is about the same as the cellular networks [4]. Reception conditions are more severe in DVB-H than in DVB-T though, especially in the case of indoor or vehicular reception. Because of this, DVB-H requires much more infrastructure than DVB-T [5], including more transmission power and specially, a larger number of transmitters and repeaters (gap-fillers) [6]. This is even more evident for high coverage levels (over the 90%) [7]. Therefore, a mobile TV deployment only seems to be viable in urban areas with a large number of clients capable of recovering the inversion in infrastructure.

On the other side, the UHF frequency range in where the system is to operate is already occupied by the traditional analogue TV and the DTT. This frequency scarce will go on until the so called *analogue switch off*, which will not occur until 2010 in the majority of European countries, Spain included.

In this context, DVB-SH (*Digital Video Broadcasting – Satellite Services to Handheld Devices*), a technological evolution of DVB-H, is gathering momentum within the telecommunication world thanks to its capacity to generate a mobile TV European market [8]. The main technical characteristics of DVB-SH are two: a satellite – terrestrial hybrid architecture, and the transmission in the 2 GHz S-band, which is right now available in Europe for this type of services.

The legacy DVB-H incorporates the MPE-FEC, a protection mechanism based in a Reed Solomon (255,191) code capable of interleaving the information inside one burst. This mechanism is aimed to protect the information from the fast variations in the received signal strength caused by fast fading. The fast fading is a known phenomenon in mobile communications that is a consequence to the user mobility. MPE-FEC can successfully correct the errors derived from the fast fading in a wide range of user velocities. However, an intra burst mechanism such as MPE-FEC can not mitigate the errors caused by shadowing. Contrary to the fast fading, the shadowing is a phenomenon responsible for the slow variations in the received signal strength. When the user passes under a large obstacle such as a tree or a building, the line of sight with the transmitter is lost. As long as the terminal is in a shadowed zone, the majority of the received information will be lost. MPE-FEC is only capable of correcting a burst of information when the number of errors is below the amount of parity received in that same burst. It can not employ parity information from other bursts. Because of this, in a satellite reception environment, interleaver mechanisms aimed to spread the errors along a large portion of time are of most importance.

DVB-SH is going to incorporate on one hand, a physical layer interleaver of long duration, and on the other hand, a link layer multi burst protection mechanism. While the physical layer interleaver has been completely defined in [8] with several available profiles, the link layer mechanism is still open to debate inside de DVB standardization organism. At this moment, a sliding mechanism based on the same RS code as MPE-FEC is the only option supported by the standard [9]. However, another scheme based in Raptor codes could be added as an alternative. The RS sliding mechanism can benefit from the reuse of a well known code such as the RS (255,191), whereas the Raptor based mechanism can benefit from a full software implementation of a fountain code. However, Raptor codes are under the protection of the intellectual property, which complicate their incorporation into the standard. Both approaches are aimed to cope with the long errors bursts caused by shadowing.

This Master Thesis, will evaluate the performance of the link layer mechanism proposed for DVB-SH by means of DVB-H error traces. These traces have been obtained in a measurement campaign that took place in Turku (Finland). Although the information stored is not from a DVB-SH system, but from a DVB-H network, it will be useful to evaluate the performance of the new multi bursts scheme in comparison with the legacy MPE-FEC. Different settings of this mechanism will be analyze in order to identify those configurations better suited for the most relevant reception conditions.

In Chapter II the DVB-SH standard will be explained, describing the main benefits of the hybrid satellite terrestrial approach. The main elements of DVB-SH from both the physical and link layers will be described next. In Chapter III, the motivation of FEC codes and interleaving mechanisms will be explained. In Chapter IV, the link layer scheme proposed for DVB-SH will be exposed, especially the sliding RS mechanism, which is presently the only one accepted by DVB-SH.

II. THE DVB-SH STANDARD

II.1.*Introduction*

The most convenient medium when distributing a TV signal among a large number of users in a wide area is a satellite. One geostationary satellite is capable of providing coverage to all Europe since the very moment it becomes operative. When the number of users breaks a certain threshold, satellite transmission becomes the most profitable system since the infrastructure cost is constant and independent from the number of users. However, the transmitted signal by the satellite can be much deteriorated in situations where there is no line of vision, which is quite common in the case of handheld reception in urban zones. The solution adopted in DVB-SH consists in reinforcing the signal transmitted by the satellite by means of a complementary terrestrial broadcasting network, as long as its deployment is fully justified. The complementary terrestrial network is suited for areas with great number of potential users where the cost of deployment can be recovered. This hybrid architecture, capable of combining the advantages of terrestrial and satellite philosophies, becomes a very efficient method for the provision of digital mobile TV services.

All terrestrial broadcasting television systems work in the UHF band (from 470 to 862 MHz) such as in the case of analogue TV, DTT and mobile TV. This band is characterized by low propagation loss (which is proportional to the carrier frequency) and by the fact that allows the usage of large bandwidths. It is also characterized by the absence of human noise which is common to low frequencies. However, the majority of UHF channels are today occupied by analogue and digital TV, which is jeopardising the roll out of the mobile TV market in Europe. On the other hand, hybrid satellite-terrestrial systems work in the S-band, which goes from 2 to 4 GHz. The frequency band reserved for satellite systems situated between 2170 MHz and 2200 MHz has been recently assigned by the European Union to these systems [10].

Figure 1 shows the IMT2000 systems frequency bands, in which there are 30 MHz at disposal for mobile satellite systems next to the band reserved for 3G cellular systems. This fact makes possible the reuse of the equipment, antennas and sites already employed by 3G systems for the terrestrial component, which represents a considerable saving in the cost of the terrestrial network deployment. Several studies as in [11] describe a reduction of about 50% in the cost of S-band transmitters over the cost of UHF-band transmitters thanks to the cellular sites reuse.

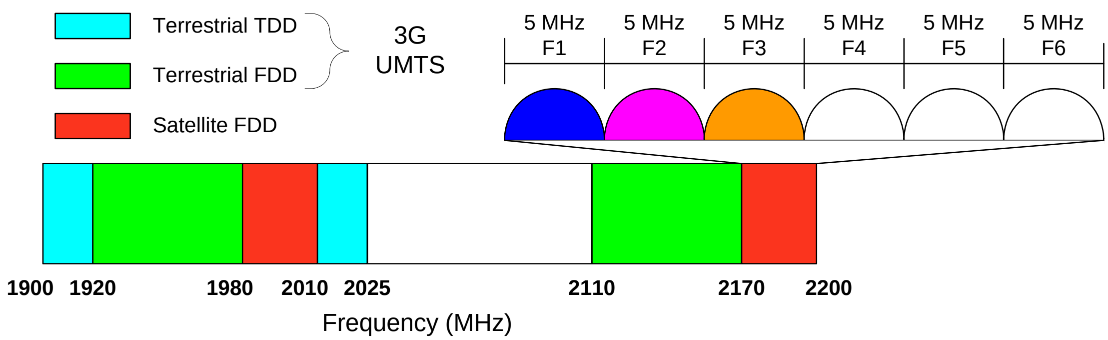

Fig. 1. Frequency band reserved for IMT2000 systems

Higher transmission frequencies suffer higher signal propagation losses. This fact involves a denser terrestrial network and an increase in the cost of the network infrastructure. This penalization is partly compensated thanks to the fact that higher frequencies also mean higher gains in the small antennas of handhelds devices. For instance, the typical gain of a DVB-H terminal is around -7 dBi at 700 MHz, whereas it would be possible to achieve a gain of about 0 dBi working at S-band [12]. The S-band also allows for diversity techniques based in a two antennas reception, which is not possible when working in the UHF-band due to the big size of the antennas. An antenna diversity scheme is capable to achieve gains in the order of 6 dB (assuming the same physical layer) in environments with a strong multipath component such as in indoors [12].

Today there are already two hybrid satellite-terrestrial mobile TV systems up and running, one of them in Korea and the other one in Japan. The Korean system is based in S-DMB (*Satellite – Digital Multimedia Broadcasting*) [13], while the Japanese employs ISDB technology (*Integrated Services Digital Broadcasting*), although it is true that both systems are quite similar.

Due to the similitude between the Korean and Japanese systems, only the former will be described. Korean S-DMB system became operative in May of 2005, and today has more than 890.000 subscribers. This system offers a service of 15 TV channels, 19 audio channels, 3 data channels and bidirectional interactive services as well. The network architecture includes the Hanbyul geostationary satellite together with about 10.000 terrestrial repeaters in charge of reinforcing the satellite signal. Satellite receives two signals in Ku-band (12-18 GHz) from the terrestrial station, one of them being a CDM (*Code Division Multiplexing*) signal and the other one a TDM signal (*Time Division Multiplexing*). The satellite retransmits the TDM signal in Ku-band to the terrestrial repeaters, while down-converts the CDM signal in Ku-band to S-band (in the S-DMB case at 2.6 GHz) and transmit it to the user terminals. Terrestrial repeaters on the other hand convert the Ku-band TDM signal transmitted by the satellite to S-band CDM signal. This way, user terminals receive the information within an S-band CDM carrier, whether it comes from the terrestrial component or from the satellite component. S-band signals are combined in reception by means of a spatial diversity scheme based in two antennas, with a Rake receiver of six fingers in each antenna. CDM spreading is accomplished by a Walsh code at 16.384 MHz which provides a spreading factor of 64. Today, 31 256 kbps CDM channels are transmitted, achieving a total velocity of 8 Mbps in a 25 MHz channel. In the same manner as DVB-H, S-DMB employs a Reed Solomon (RS) (204,188) code together with convolutional coding in order to protect the information at a physical level. The information goes through a byte wise convolutional interleaver between the two coders and through a bit wise interleaver after the convolutional coding.

II.2. *System Overview*

DVB-SH (*Digital Video Broadcasting – Satellite services for Handheld devices*) is a hybrid satellite-terrestrial solution aimed to the mobile TV broadcasting which operates in the S-band. Figure 2 shows the reference architecture. Similarly to S-DMB, DVB-SH employs a geostationary satellite together with a terrestrial network which provides service in those areas where the satellite signal is seriously degraded. The terrestrial network also allows for an increase in the system capacity.

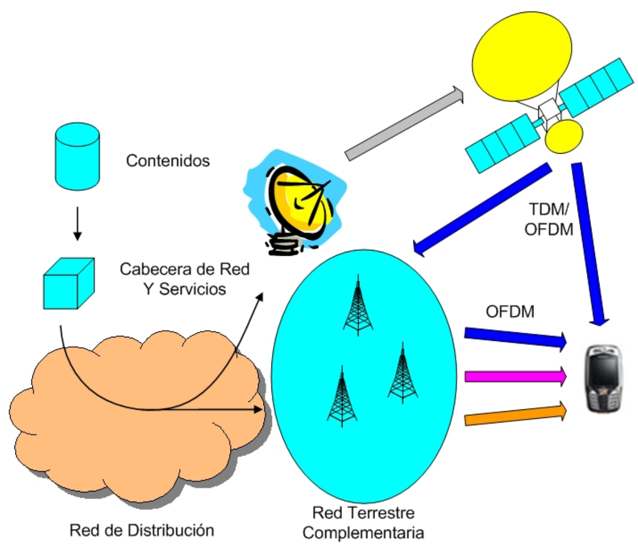

Fig. 2. Hybrid reference architecture

Contrary to S-DMB, in which the user receives a CDM signal, DVB-SH employs an OFDM (*Orthogonal Frequency Division Multiplexing*) waveform in the complementary terrestrial network and an OFDM or a TDM waveform in satellite transmissions. OFDM waveform allows the deployment of Single Frequency Networks (SFN) in which receivers can combine the signals incoming from all the transmitters, satellite included, as long as the delay between signals does not overcome the guard interval of the OFDM signal. DVB-SH also allows for the TDM waveform in the satellite component, which is capable of support higher satellite transmission power. This is due to the fact that multicarriers signals as OFDM are characterized by a high peak power, forcing the on board high power amplifiers to work far from the saturation point (for which the transmitted power is maximised) in order to avoid non linear distortions. TDM waveform allows the on board transmitters to work closer to the saturation point, which represents a power advantage of 1-1.5 dB with respect to OFDM [9]. This gain involves an increase in the coverage provided by the satellite component.

The possibility of employing two different waveforms increases the flexibility of the system and increments the number of possible configurations. Two different architectures are derived from this:

- SH-A architecture: both the satellite and terrestrial components employ the OFDM waveform.
- SH-B: while the satellite component employs the TDM waveform, the terrestrial component employs the OFDM waveform.

SH-A architecture makes possible the set up of SFN networks in which both the satellite component and the terrestrial component transmit the same contents at the same frequency. SFN networks allow for a high spectral efficiency, but force the signal transmitted by the terrestrial component to be identical to the signal transmitted by the satellite in all the territory. Because of this, the possibility of deploy multi frequency networks (MFM) is allowed. In a MFM network, although both the satellite and terrestrial component transmit with an OFDM waveform, they employ separated frequency channels. Although a MFM network reduces the spectral efficiency of the system, it brings some benefits. An MFM network makes possible the transmission of local contents in the same frequency channel as the one employed for the transmission of national contents as long as enough capacity is available. Also, in a MFM network, it is possible to tune the transmission parameters of each region or city in order to achieve an optimal planning. The same can not be done in a SFN network where the same signal must be transmitted in all the territory. In terminals equipped with two different radio interfaces, it is possible to improve the reception quality by means of soft-combining, a technique that combines radio signals from the satellite and the terrestrial network in base band. Moreover, if complementary encoding ratios between the terrestrial and satellite components are employed, it is possible to achieve an additional gain due to code-combining.

In SH-B architectures, as each of the components that constitute the DVB-SH network transmits in a different waveform, it is necessary that they do so in different frequency channels in order to not interfere with each other. Although the spectral efficiency of the network is reduced with respect to a SFN SH-A architecture, satellite transmission achieves a better performance thanks to the TDM waveform. Also, as in the MFM SH-A case, it is possible to optimize the terrestrial component configuration in the different regions under the coverage of the satellite. In the same way as in a MFM SH-A network, it is possible to employ soft-combining and/or code-combining techniques to improve the signal reception.

While a SH-A architecture is the best suited to spectrum limited scenarios, a SH-B architecture is better suited when satellite transmission becomes power limited. The existence of two architectures implies the necessity of two different terminal categories too, one for each architecture. While SH-B terminals are compatibles with a SH-A architecture and capable of working under a SFN or a MFM network, SH-A terminals cannot work in SH-B architectures because of the fact that they are not equipped with a TDM receiver.

Satellite-terrestrial hybrid architecture involves a lot of possibilities, turning DVB-SH planning in a more complex task than the DVB-H case, because the satellite and terrestrial component must be taken into account. The problem is even more complex when multibeam satellites are employed. This kind of satellite featured antennas aggregations and feeding systems capable of generating advanced antenna patterns. By means of those antenna patterns it is possible to achieve coverage areas overlapped and totally independent in the same satellite. This kind of satellites is suited to areas with a great linguistic and cultural diversity as Europe, because the fact that they allow for a particularisation of the content transmitted for each of the beams. This feature, together with the frequency reuse among the different beams similarly to how it is done in cellular planning, allows for a spectrum optimization, increasing the system capacity. In the case of Europe, a three frequency scheme is good enough to prevent the interference among two adjacent beams as it is shown in Figure 3.

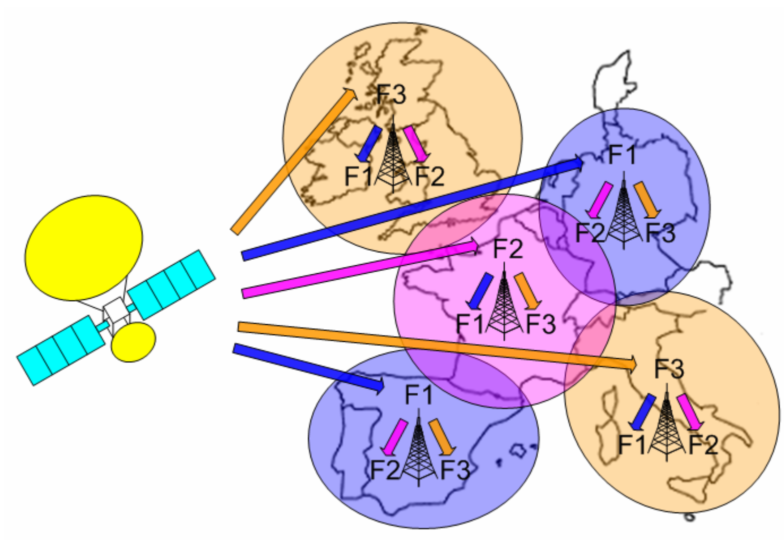

Fig. 3. Example of frequency planning in Europe for both de satellite component and the complementary ground component

The terrestrial component not only can be used to complement the coverage of the satellite component in those areas where the satellite signal is weak, but also can be used to transmit additional content particularized to each region. This capability increases the offer that is available to the users. In order to do this, the terrestrial network does not operate in the same frequency channels that are employed by the satellite transmission in its country, but in the channels employed by the satellite in the neighbour countries. However, the interferences between the terrestrial network of a country and the satellite beams from nearby countries must be taken into account.

As an example of a DVB-SH network operating in Europe, a three 5 MHz channel SFN network is taken into account. This is the case of a SFN SH-A network and it is showed in Figure 3. A usual transmission mode in DVB-SH. consists in QPSK modulation, 2K FFT, guard interval of 1/4 and a turbocoding ratio of 1/3. This configuration provides 2.3 Mbps in a 5 MHz channel, making possible the transmission of up to 9 TV channels of 256 kbps each in one channel and up to 27 TV channels if all the bandwidth at disposal is employed. This way, the same 9 channels are transmitted in all the country thanks to the satellite, while in those regions within terrestrial network coverage 18 additional channels can be received.

II.3.*Physical Layer*

The physical layer of DVB-SH incorporates new protection mechanisms with respect to DVB-H such as a long duration interleaver and a turbocoder that replaces the convolutional coder found in DVB-H. The interleaver is introduced in order to mitigate the effects of the shadowing present in satellite reception. On the other hand, the turbocoder increases the robustness of the transmission thanks to its proven effectiveness in mobile environments.

Similarly to DVB-H, DVB-SH is based in the transmission of MPEG2 transport streams (TS) carrying information bursts compatible with the time slicing technique of DVB-H. An MPEG2 TS consists of MPEG2 packets of 188 bytes that are transmitted one after another. A CRC-16 is added to each one of these packets, so the receiver can known if any of these packets is erroneous. As it is shown in Figure 4, an MPEG2 TS can follow two different paths before its radio transmission according to the waveform is going to be transmitted. The two possible waveforms, OFDM and TDM, share common subsystems that process the information no matter what the waveform is. These subsystems are the turbocoder and the physical layer interleaver. The turbocoder incorporated in DVB-SH is the one standardized by the 3GPP2 and also employed by 3G systems. However, the turbocoder implementation in DVB-SH features new code rates in order to increase the flexibility in the level of protection. The turbocoder input consist of blocks of 12282 bits and for each one of these blocks, the turbocoder will output (12282+6)/(Code rate) bits of encoded information. This turbocoder at a code rate of 1/3 is capable of providing a gain up to 3 dB in a Gaussian channel compared to a convolutional coder at a rate of 1/2 plus a Reed Solomon coder (204,188) and a MPE-FEC coding of 3/4 (same effective 1/3 code rate) [12].

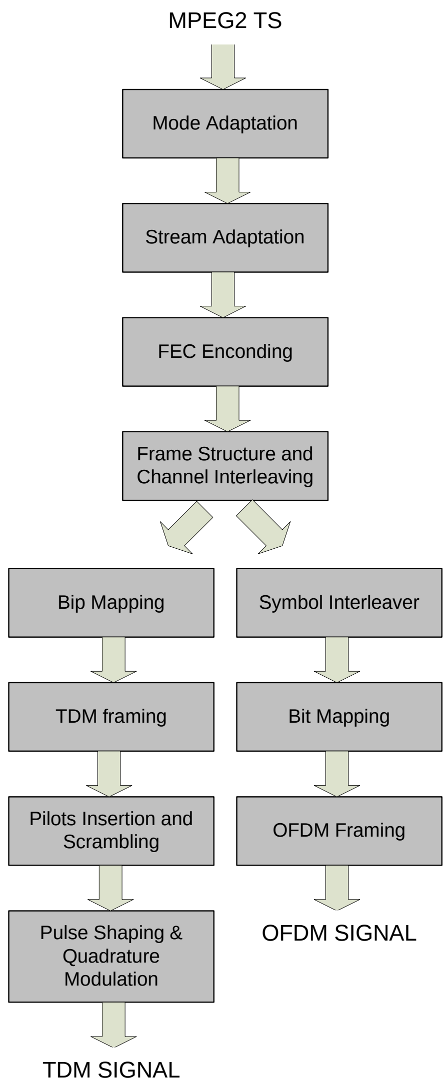

Fig. 4. Physical layer scheme of DVB-SH

The channel interlaver incorporated in DVB-SH is situated right after the turbocoder output and is made of two cascade interleavers, a bit interleaver and a time interleaver. The bit interleaver is in charge of rearrange the bits inside each block word that comes from the turbocoder. After this, a rate adaptation process punctures the block words so they can be multiples of 126 bits, which is the size of one interleaving unit (IU). Following the rate adaptation process, the time interleaver takes as input the sequence of IUs of 126 bits cells and interleaves them by means of a convolutional interleaver. There are two different physical layer interleaving durations defined in DVB-SH: one of about 200 ms that can work with a number of IUs up to 6528, and other one of about 10 s that can work with a number of IUs up to 417792. This possibility brings out two different terminal classes: a class 1 terminal, with short physical layer interleaving (200 ms), and a class 2 terminal with long physical layer interleaving (10 s). In order to stored and process the information interleaved, class 1 terminals need at least 4 Mb whereas class 2 terminals need at least 256 Mb [9].

II.4. *Link Layer*

The link layer of DVB-SH has two main characteristics; the support of the time slicing technique and the support of link layer FECs. The time slicing technique supported by DVB-SH is identical to the DVB-H one and it is based in the transmission of information by means of bursts. The data therefore is transmitted at a higher data rate during a short time in a periodical way achieving an equivalent constant data rate. This is shown if Figure 5.

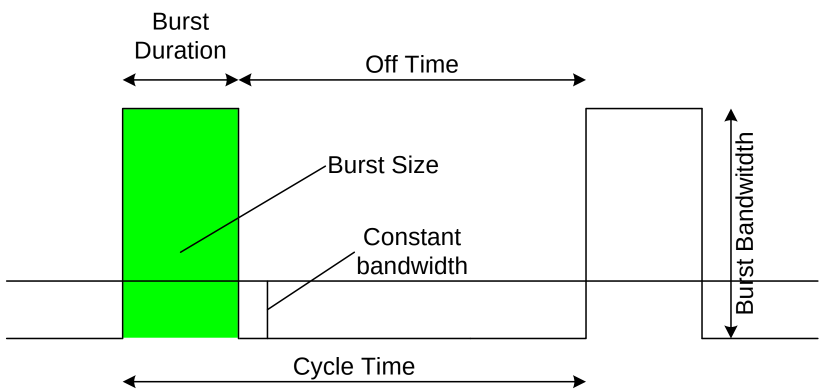

Fig. 5. Burst transmission mechanism

Time slicing is capable of saving a large amount of power and it also allows seamless handovers. The receiver turn off the RF front end during the intervals of time between bursts and it wakes up just before each one of the bursts carrying the desired service. Therefore, the terminal is only receiving information during a portion of time instead of having its RF front end always switched on. The terminal needs to know the time of arrival for each one of the bursts in order to awake in time. This is achieved thanks to the introduction of the delta-t parameter which is carried in every burst and indicates the remaining time until the arrival of the next burst. As the terminal does not need to be receiving information of one stream all the time but only during the on-time, it can switch to other streams on the off-time and retrieve useful information. Also, it is possible to switch from the service carried in one stream to the same service in other stream during the off time without any kind of delay or interruption, achieving a seamless handover. Moreover, as the burst size does not have to be constant in size along different burst cycles, a variable bit rate service is easily achieved by means of a variable burst size scheme. This way, the burst size changes every burst cycle in order to accommodate with the amount of IP information.

The link layer protection offered by the legacy DVB-H, MPE-FEC (*Multi Protocol Encapsulation – Forward Error Correction*), is aimed to mitigate the impulsive noise and the impairments in a mobile environment such as the fast fading. MPE-FEC is capable of providing a constant reception quality in terms of burst error rate between a wide range of velocities as well as improving the reception of handheld terminals which are equipped with small gain antennas.

A datagram burst is a collection of one or several continuous OSI layer 3 (Network layer) datagrams (e.g. IP datagrams) that will be sent in a time slicing burst. All IP datagrams from a datagram burst are placed in a matrix of 255 columns and a number of rows (256, 512, 768, 1024) that depends on the burst. The matrix, called MPE frame, is showed in Figure 6. The MPE frame is made of an Application Data Table (ADT) and a RS Data table. Each position in the ADT corresponds to a byte of IP information. Each datagram is positioned in the frame starting in the upper left corner and going down. An IP datagram does not have to fit exactly in one column of the table, and can continue in the upper part of the next column. When an IP datagram has been fully inserted in the frame, the next one is inserted right after. This is shown if Figure 6. MPE-FEC technique is based on a Reed Solomon coder (255,191). When all the 191 columns have been filled with information, the RS table is calculated by means of the RS (255,191) coder which outputs 64 bytes of parity for each row in the ADT. Eventually all the rows will be coded and the RS table will be filled with parity information.

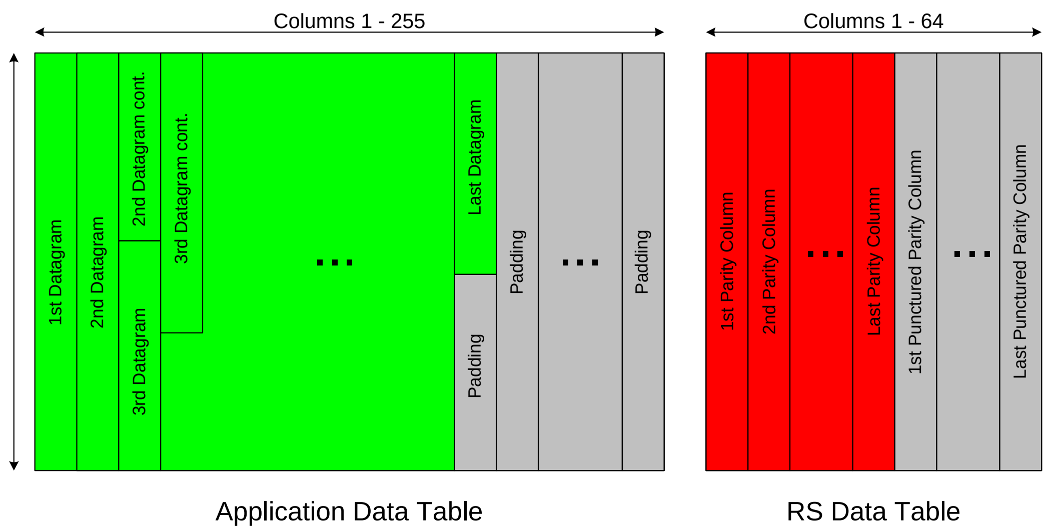

Fig. 6. MPE-FEC frame

When the entire MPE-FEC frame has been filled, the IP datagrams will be encapsulated into MPE sections and will be sent in a time slicing burst. Each MPE section carries exactly one IP datagram. On the other hand, the parity information in the RS table will be encapsulated into MPE-FEC sections and will be sent in the same time slicing burst as the IP datagrams after all the MPE sections had been transmitted. One MPE-FEC section carries one column of the RS table. Both MPE and MPE-FEC sections incorporate a header where relevant data such as the delta parameter is carried. They also add a CRC field that is used by the RS decoding to identify erroneous bytes in the MPE-FEC frame.

The RS (255,191) provides a code rate of 3/4, but other rates are possible by means of padding and puncturing. In order to decrease the code rate and increase the protection offered by MPE-FEC, the 255 rows of the ADT might not be filled with IP information. A portion of the table can be filled by padding information which will not be transmitted. The padding information will always be placed in the right columns of the ADT. On the contrary, if it is desired to increase the code rate, a number of the RS columns can be punctured and will not be transmitted. The number of total parity sections is a parameter transported in the MPE-FEC header, and therefore, the receiver knows that the number of padding columns in the RS table is 64 minus the number of sections transmitted. Only the columns at the right of the table can be punctured and filled with padding in reception. The amount of data and parity for each code rate can be seen in Table 1.

In case a section is erroneous, the CRC incorporated in each section will fail, and the bytes corresponding to that section will be marked as erroneous in the MPE-FEC frame. Erroneous bytes which positions are known are called erasures. RS codes are perfect codes because of the fact that the amount of erased information that can be corrected is equal to the amount of parity information received. The reception overhead is zero, and therefore, MPE-FEC coding is capable of correcting as many erased columns as the number of parity columns transmitted. However, one erroneous TS packet is enough to make a section erroneous, and all the information contained in the section will be marked as an erasure. If none of the sections are erroneous, the RS decoding does not have to take place and the IP information can be forwarded to upper layers. Instead of using the CRC field of the sections to mark the bytes as erased, the CRC from the TS packets can be used. This technique increases the resolution when marking each byte of the table as an erasure and improves the RS decoding [16].

	1/2	2/3	3/4	5/6	7/8
Data Columns	64	128	191	190	189
Parity Columns	64	64	64	38	27

Table 1. Number of data and parity columns corresponding to each MPE-FEC code rate

III. FEC AND INTERLEAVING MECHANISMS

III.1.*FEC mechanism*

Forward Error Correction (FEC) mechanisms are based in the transmission of parity information which allows for a reconstruction of the original information in the presence of errors and without a return channel. Although FEC results in an increase of the amount of information transmitted, it can increment the system efficiency, saving both transmission time and bandwidth in comparison with the retransmissions that will be needed otherwise.

A FEC encoder generates parity data from a set of information that must be protected (source data). This set of information can be a file or a portion of a streaming service. For example, a set of information can be divided into *k* data packets, and a FEC encoder generates $n \geq k$ packets from the original *k* packets. The code rate is therefore n/k, because although the original set of information is only made of *k* packets, *n* packets are being transmitted. If a systematic code is employed, the first k packets transmitted are precisely the same *k* packets that originally formed the encoded information. The rest *n*-*k* packets are the additional parity packets. In the receiver, the decoder can retrieve the original data packets if it receives at least *r* packets, where $k \leq r \leq k$. The value of *r* depends on the encoding method, and it does not matter which *r* packets are received. A perfect code is capable of retrieving the original data packets with only *k* packets from the *n* packets transmitted.

The reception overhead refers to the amount of encoding packets over the original information size are necessary for the receiver to retrieve the file. The failure probability is the probability of not being able to decode a set of information for a certain amount of overhead. For instance, the reception overhead can be of about 2% with a failure probability of 10^{-3}, or can be of about 4% with a failure probability of 10^{-6}. Some FEC codes have intrinsically a reception overhead that is a consequence of its probabilistic nature.

Reed Salomon (RS) codes are a classic example of a FEC that is able to reconstruct all the original data packets from any set of *k* received packets. However, because of its high encoding and decoding complexity (proportional to the encoded block, *n*), this codes are only use for small blocks (e.g., *n*=255). RS encoders are usually implemented on hardware.

III.2.*Interleaving mechanisms*

Mobile satellite reception is characterized by long and deep fades in the received signal. Those fades are occasioned when the terminal is passing through a shadowed area where the signal from the satellite is blocked by trees or buildings. This signal blocking causes error bursts in which all the information received is erroneous. Due to the fact that the error correction mechanisms are much more effective when the errors are distributed along the time in a uniform manner, these erroneous bursts difficult the correct reception of information. An interleaver rearranges the information to be transmitted so that portions of information that are close to each other end up being transmitted separated in time. The interleaving duration is the separation in time between the transmissions of two portions of information that were next to each other before the interleaving.

The signal fading duration depends on the reception environment and the user velocity. The blockage derived from trees causes shorter fading durations than the blockage caused by large obstacles as buildings. On the other hand, if the user is moving at high velocities, he will go faster through shadowed areas and the fades will be shorter than if he is moving at low velocities. The interleaving time needed to assure a correct reception of the information is directly related to the length of the erroneous bursts and therefore is also related to the length of the fades. Durations of about 10 seconds have been estimated to be long enough to mitigate the shadowing effects caused by trees, as long as the velocity is not inferior to 10 km/h. The same duration is also long enough to mitigate the shadowing effects caused by large obstacles such as buildings, as long as the velocity is not inferior to 60 km/h [9]. In case reception is taking place at inferior velocities, as in the typical pedestrian velocity of 3 km/h, is advisable to assure service continuity by means of a larger deployment of the terrestrial component instead of increasing the interleaving duration. As the interleaving duration increases, also does the protection against long fades. However, long interleaving durations require more memory available in the terminal and bring along an increase in the delay of the information, which increments the zapping time and might degrade the user experience [13].

The portion of time between the moment when one user switches to one service and the moment when the information of this service is being displayed is referred to as zapping time. In the case of streaming services, when the user switches from one stream to another, the terminal must wait until the arrival of all the interleaved information before it can display any content. File delivery services are not affected by zapping time; the user has to wait until the file has been completely downloaded. File delivery services are usually background services because of the fact that the user is not aware of the service until it has been completed. However, streaming services are consumed by the user during its transmission, and it is important that the user has a fast feed-back of the service. Because of this, several fast zapping mechanisms can be used in order to decrease the zapping time in the presence of long interleavers.

IV. MPE-iFEC

IV.1.*Overview*

As it has been explained, mobile satellite reception is bound to long fades that can corrupt complete bursts of information. This is beyond MPE-FEC correction capabilities, which is an intra burst protection technique that is only able to cope with a limited number of erroneous sections per burst. A mechanism capable of coding information from several bursts jointly is thus needed. DVB-SH introduces a new inter burst protection technique at the link layer called MPE-iFEC (*Multi Protocol Encapsulation – Inter burst Forward Error Correction*) [9]. MPE-iFEC is a generic multi burst framework that can be configured in multiple ways and presents enough flexibility for a variety of applications. In DVB-SH its parameters are restricted to some specific values via the “framework mapping”. The joint codification of information from different bursts can be done in two different ways. Either several portions of different bursts are interleaved and fill an encoding matrix of a similar size to the MPE-FEC frame from DVB-H, or an encoding matrix bigger enough to code several burst together is used instead. The combination of the two previously mentioned mechanisms is also possible in MPE-iFEC. For instance, if 10 bursts are jointly coded and a 30% of parity is transmitted, the terminal will be able to overcome the complete loss of 3 bursts for every 10 bursts, assuming the rest of the information has been received correctly. This would be impossible in MPE-FEC because of the fact that the parity received for one burst cannot be used to correct errors in other bursts.

At the encoder side, each one of the datagrams bursts received from upper layers passes through the Application Data Sub Table (ADST). This function distributes the data columns of each burst between the encoding matrixes. Each one of these matrixes is made of an Application Data Table (ADT) where the IP datagrams are placed and an iFEC Data Table (iFDT) that hosts the parity data. Each time the ADST fills one ADT with IP information (plus padding) the encoding process takes place and the parity data correspondent to that ADT is generated and placed in an iFDT.

Although the IP datagram bursts are interleaved before the encoding process by the ADST, they are sent without being interleaved and in the same order they were generated. The IP datagrams are encapsulated into MPE sections and are transmitted in bursts in the same way as in MPE-FEC. The parity columns from the iFDTs are encapsulated in MPE-iFEC sections and are also sent in time slicing bursts along with the data information. Contrary to MPE-FEC, the parity information of one time slicing burst does not have to correspond with the IP information transported in the same time slicing burst. Also, although the IP information is sent without being interleaved, the parity information of one iFDT can be sent interleaved with parity information from other iFDTs. This way the parity information of one iFDT will be transmitted over several time slicing bursts. Contrary to MPE-FEC where all MPE-FEC sections have to be transmitted after the MPE sections, in MPE-iFEC, MPE sections and iFEC sections can be freely interleaved as long as their respective lengths are respected.

The whole MPE-iFEC encoding process is described in Figure 7. Although MPE-FEC and MPE-iFEC can be used at the same time, it does not bring any benefit compared to AL-FEC alone, and thus in practice MPE-iFEC is recommended to be employed instead of MPE-FEC

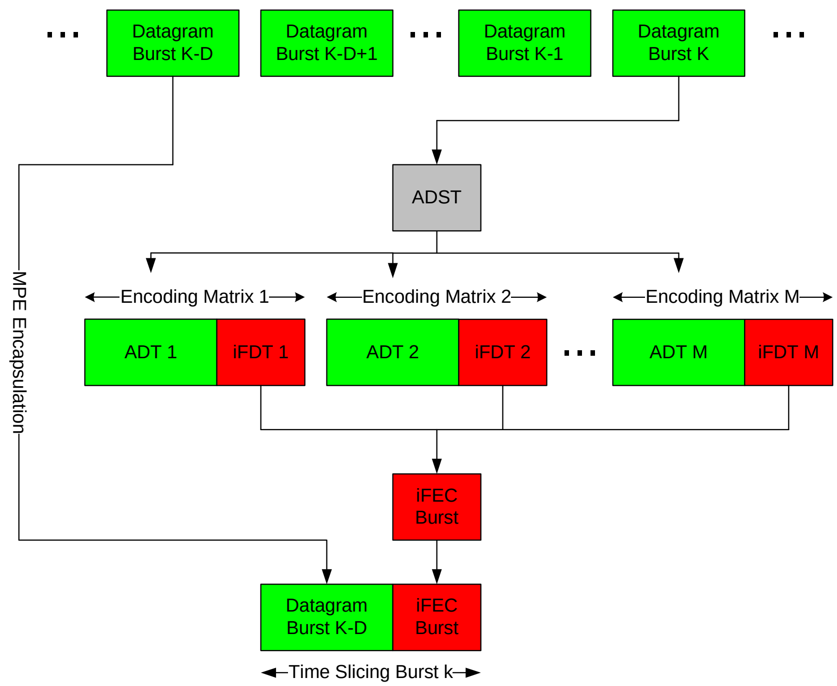

Fig. 7. MPE-iFEC encoding process

The main parameters that control the MPE-iFEC operation are:

- Encoding Period *EP*, period of the encoding process expressed in burst units. An encoding matrix will be entirely filled with IP information (plus padding) after a certain number of bursts. When this occurs, the encoding of the matrix will take place and the parity information will be generated. The value of the *EP* is determined by the size of the encoding matrixes. As the matrixes size gets bigger, the number of bursts needed to fill the matrixes increases and so does the *EP* parameter.
- Encoding parallelization *B*, also expressed in burst units. The ADST will distribute the columns of a datagram bursts between al the available encoding matrixes. The number of bursts needed to fill each one of these matrixes is the *B* parameter.
- FEC spreading factor, *S*. Although the datagrams bursts must be sent unaltered, the parity placed in one iFDT can be sent interleaved with parity from other iFDTs over a number of time slicing bursts. The number of bursts that will carry the parity of one iFDT is established by the *S* parameter.
- Delay applied to the data, *D*. The datagram burst do not have to be sent immediately after its retrieving from upper layers, it can be stored and be sent a number of bursts later. The number of bursts after which a datagram burst will be sent in a time slicing burst is called the *D* parameter.

IV.2.*Sliding RS encoding*

MPE-iFEC is a framework in which a generic mechanism is defined, without specifying the code that must be employed or the values that each of the parameters must adopt. However, at this moment, two different mappings are defined as suitable for DVB-SH transmission. One based in Reed Solomon coding and the other one based in a Raptor code similarly to the scheme selected in DVB-H for file delivery. These mappings define more specifically the encoding and decoding mechanism. The specification of the DVB-SH link layer is not finalized, and the link layer protection mechanism has not been completed. At the moment, the mapping based in RS coding is the only one officially supported by DVB-SH. This mapping is based on a sliding window that encodes the information by means of the same Reed Solomon (255,191) coder as DVB-H. This means that the maximum number of information that can be jointly coded is the same as in DVB-H which is 191 columns of IP information. Since the size of ADT is the same as one datagram burst, the EP of the sliding encoding based in RS has a value of 1. This means that a matrix of the same size as one datagram burst is encoded each time a time slicing burst is generated. The number of bursts interleaved B, the spreading factor S, and can the delay applied to the data D, can be up to 255. The value of these parameters will be determined by other factor such as information delay or available memory.

Each time a datagram burst is retrieved from upper layers its columns are distributed between B ADTs. One of these ADTs will be filled with IP information (plus padding) from B different bursts and will be encoded by means of the RS (255,191) coder. The generated parity data will be placed in one iFDT. If the D parameter is more than 0, the datagram burst will be stored for later transmission. The datagram burst received D bursts ago will be placed in the time slicing burst completely unaltered. Parity columns from S different iFDT matrixes will be inserted in MPE-iFEC sections and will be transmitted in the same time slicing burst as the datagram burst. The process is depicted in Figure 8 for the particular case of C=6, F=4, B=3, D=0 and S=2, where C is the number of columns of the ADT and F is the number of columns of the iFDT.

Fig. 8. Example of the RS sliding coding mechanism for *C*=6, *F*=4, *B*=3, *D*=0 and *S*=2

In the example showed in Figure 8, as each one of the time slicing bursts carries 6 columns of information data and 4 columns of parity data, the code rate achieved by the sliding RS mechanism is 3/5. This means that, in this particular case, the receiver will be able to correct up to 2 erroneous bursts every five burst assuming the rest of the information has been received correctly. This particular example is shown in Figure 9.

Fig. 9. Erroneous reception of 2 bursts

As it is depicted in Figure 9, time slicing bursts 2 and 3 have not been received correctly, and none of their sections can be used in the decoding process. The datagram bursts carried in the time slicing bursts 2 and 3 were encoded in ADTs 2, 3 and 4 and in ADTs 3, 4 and 5 respectively. Thanks to the interleaving made between information and parity data, the iFEC information generated from those ADTs are transported by different time slicing bursts that the ones carrying the datagram bursts, and therefore were correctly received. As it can be seen, in iFDT 2 there are 2 correct parity columns which are just enough to correct the two erroneous columns in ADT 2. The same situation is presented in ADT 0. In iFDTs 3 and 4, there are 4 correct parity columns which are able to correct the 4 erroneous data columns localized in the respective ADTs. The result is that all the erroneous information can be corrected and the datagrams will be passed to the upper layer without errors.

On the contrary, in Figure 10 is depicted the erroneous reception of 3 bursts which represents an error rate of 3/5. The 3/5 code rate use in this example can only correct an error rate up to 2/5, and therefore all the information can not be corrected. Time slicing bursts 2, 3 and 4 are received erroneously, and none of their information, data or parity, can be used. As it can be seen in Figure 10, encoding matrixes 2, 3 and 4 do not receive enough parity data to correct all the erroneous columns. Only four columns of the datagram 4 and two columns of the datagram 3 can be corrected whereas the rest will remain erroneous.

Fig. 10. Erroneous reception of 2 bursts

IV.3. *Encoding and Interleaving Configuration*

MPE-iFEC is a more complex scheme than the legacy DVB-H MPE-FEC and because of this, there are quite more parameters to consider. In MPE-FEC, the code rate and the frame size are the main parameters that determine the level of protection of the information. Better protection could be achieved by means of decreasing the code rate, which can be achieved by transmitting more parity data and/or less information data per encoding matrix. On the contrary, the capacity of the system is reduced because of the fact that more bandwidth is dedicated to the transmission of FEC information.

The same situation also occurs in the MPE-iFEC scheme and more specifically in the sliding RS encoding mechanism. As it has been explained, in the sliding RS encoding mapping presented inside the MPE-iFEC framework, the encoding matrixes are of a size that corresponds to the MPE-FEC frame. This is due to the fact that the same RS (255,191) coder is being used to generate the parity data. Therefore, in order to obtain other code rate than the standard 3/4 it is needed to add padding information in the ADTs and/or puncture some of the columns in the iFDTs. When the code rate increases the protection offered by the code decreases and vice versa. However, in the MPE-iFEC framework, the other parameters such as *EP*, *B*, *S* and *D* have not only a very important impact in the performance of the encoding mechanism, but also in the receiver latency and memory requirements. This is clearly represented by the equations 1 and 2:

$$receiver\_latency = burs\_repetition\_interval * sizing(B, S, D) \tag{1}$$

$$memory = (ADT\_size + FDT\_size) * sizing(B, S, D) \tag{2}$$

$$sizing(B, S, D) = B + \max(0, S - D) + \max(0, D - B) \tag{3}$$

In the particular case of the sliding RS mapping, the encoding period is always 1, which is being fixed by the fact that the encoding matrixes can only contain the information of one datagram burst plus parity. Other codes such as Raptor are able to jointly encode much bigger slices of information and, since the encoding gain is related to the source block size, in certain environments they could achieve a greater performance. However, thanks to the unitary encoding period of the sliding RS mechanism, the encoding and decoding processes occur one time per burst. Every burst cycle, the same amount of information is being encoded in the transmitter and it is being decoded in the receiver. This continual decoding does not cause the computational peaks that take place in mappings with greater encoding periods and can have a considerable impact in the power consumed by the user terminal. Moreover, the encoding period is a very decisive parameter in the memory required by the user terminal as it determinates the size of the ADTs and iFDTs. As it can be clearly seen in equation 2, the memory required is proportional to the encoding matrixes size, and therefore, mappings with bigger EPs will have as a consequence greater memory requirements.

The encoding parallelization and FEC spreading factor have a direct impact in the interleaving of the data and parity information along time. Greater values increments the duration of the interleaving which increases the protection offered against long signal fading. However, as the interleaving increments, the amount of information that must be stored is greater, and so is the memory required in the user terminal. High *B* and *S* values have also a big impact in the receiving latency, the time between the moment the data information arrives at the encoder and the moment when the information is available to the user. The latency in receiving the information is not very relevant in file delivery services, but can become critical in streaming or interactive ones. The zapping time, which is the period of time that passes between when a user switches to a service and when the service is finally displayed, it is also very dependant on the *B* and *S* values. In following sections of this Master Thesis, the performance of the sliding RS encoding as the values of *B* and *S* increase will be evaluated.

Finally, the delay applied to the datagram bursts in the transmission side is very relevant to both memory requirements and receiver latency as it can be seen in equations 1, 2 and 3. The sizing function showed in equation 3 that determinates the memory and latency is conditioned by the value of *D*. Because of this, the *D* parameter can be very critical in tweaking the configuration of the encoding scheme in order to achieve the best results. The sizing function adopts a minimum value when the delay is situated between the encoding parallelization and the FEC spreading factor as it can be seen in Figure 11. However, as the delay increases, the performance of the system decreases due to the fact that the FEC information is being transmitted in the same time slicing bursts as the data information from which it has been generated. If one time slicing burst is received erroneous, information from ADTs and iFDTs pertaining to the same encoding matrix will be lost, and therefore, the correction of the erasures will become impossible. On the contrary, if the *D* value increases up to a *B*+*S* value, the FEC information will be again not interleaved alongside its correspondent data information, and so, the performance will not be degraded. This latter case is only possible if the memory in the receiver is large enough and only if the delay of the received information is not a critical parameter. The impact of the *D* value in the performance of the sliding RS encoding will be also evaluated in following sections of this Master Thesis.

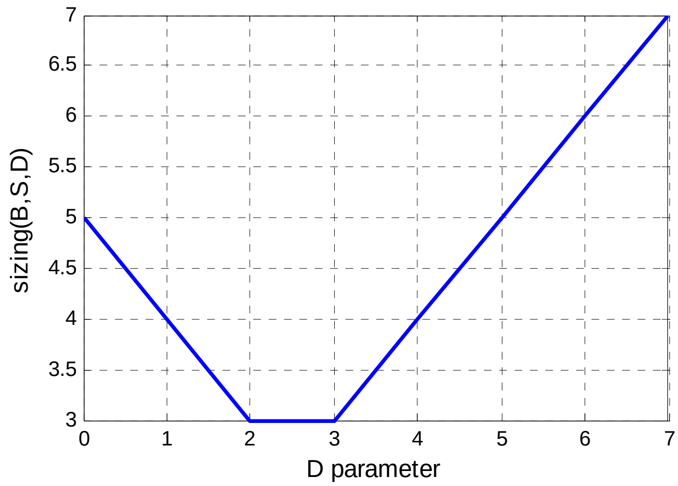

Fig. 11. Link layer sizing for B=3 and S=2

V. STUDY BASED ON ERROR TRACES

V.1.*Justification for error traces*

In every communications system quality indicators are fundamental in order to evaluate its performance. By means of these parameters it is possible to compare the quality of the received signals in different propagations environments and for different transmission modes. This is a critical point in the deployment of any network and so, many parameters had been used along the years to establish the threshold of what can be considered as an adequate reception and what not. Besides physical layer parameters such as the CNR (Carrier to Noise Ratio), the MER (Modulation Error Rate) and the END (Equivalent Noise Degradation), it is interesting the use of link layer parameters such as the FER (Frame Error Ratio) and the MFER (MPE Frame Error Ratio).

Physical layer parameters are not enough to evaluate the quality experimented by the user when receiving a streaming or a file delivery service. These parameters are obtained before the chain of correctors and interleaving mechanisms of DVB-SH, which have an important impact in the final number of errors. It is necessary to evaluate the parameters situated in a higher level which capture more faithfully the final quality of a service.

However, in a field measurement campaign, for each one of the measurement routes, it is possible to evaluate the user quality only for the service being transmitted and only for the configuration employed. If the gain obtained when decreasing the code rate or when using other codification scheme as Raptor must be known, there is no other way but to make one measurement for each one of the configurations. Also, application of novel protection mechanisms not implemented in any receptor cannot be evaluated.

Every TS packet in DVB-SH is transmitted along with a CRC field in order to detect any erroneous packet. In the same way, MPE and MPE-FEC sections can be signalled as correct or erroneous thanks to the CRC field incorporated in each section. The user terminal has information about the correct or incorrect status of each one of the sections and TS packets. Therefore, the possibility of store the status of all TS packets and/or all sections during the reception of a service is possible.

If information about the errors at a section level or at a TS packet level is available, it is possible to reproduce the quality of any kind of service, whether it is the same transmitted during the measurements or not. It is possible to apply to the same trace different streaming and file delivery services. Also, it is possible to compare different protection schemes, from the legacy MPE-FEC with different code rates, to more advances mechanisms as MPE-iFEC

V.2.*Work Method*

This Master Thesis is aimed to the analysis and evaluation of different link layer protection mechanisms by means of error traces. Due to the fact that the DVB-SH standard is still in development and it is right now at a very early validation phase, it is very difficult to dispose of actual DVB-SH measurements. There is actually no satellite in orbit capable of transmitting DVB-SH signal, and no test network has been deployed anywhere in the world. Because of this, the work presented here is based on DVB-H traces. Although DVB-SH has a physical layer very different from DVB-H, these error traces can be used to compare the performance of the new proposed link layer mechanism to the performance of the legacy MPE-FEC. The legacy DVB-H incorporates a multi burst protection mechanism based in Raptor codes only for file delivery services. The multi burst scheme defined in MPE-iFEC can also become useful for streaming services and can improve the reception in DVB-H as well. This Master thesis will analyze the impact of MPE-iFEC in DVB-H and will investigate its performance in terrestrial transmissions.

The error traces were obtained by means of measurements taken in the DVB-H test network of the University of Turku, Finland. The test network has three transmitters. Two, Pääskyvuori and Vanha Studio, are operating in a SFN-configuration at 610 MHz and one Kuusisto at 690 MHz. For the traces analyzed in this Master Thesis, only the two SFN transmitters were used. The transmitters have the following characteristics:

Transmitter	ERP	Antenna Height
Pääskyvuori	800 W	93 m
Vanha Studio	1 kW	40 m

Table 2. Turku Network Transmitters

An error trace consists in a file where each one of the received TS packets is represented by a character. This character is ‘0’ if the TS packet was received erroneous or a ‘1’ if the TS packet was received without errors. These characters were recorded every 100 ms along with the RSSI, the terminal velocity, and the geographical position of the terminal. A great number of traces were recorded during the field trials in Turku, although only 6 of them were selected to be evaluated in this work. These traces represent a wide range in signal quality and reception environments. All 6 six traces were obtained in vehicular reception conditions at a velocity between 0 and 30 km/h. An example of these traces is shown in Figure 12.

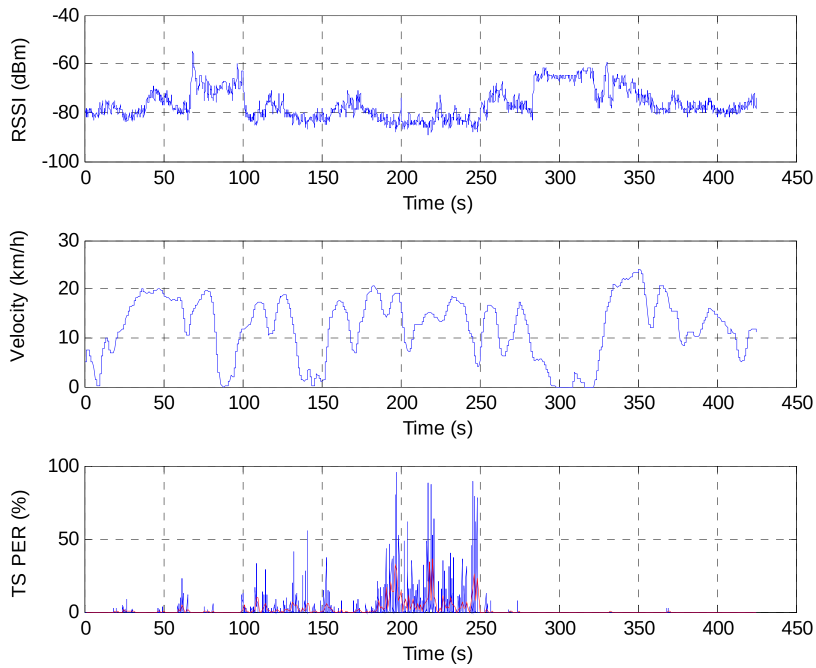

Fig. 12. Example of a trace obtained in Turku

The link layer mechanisms described in this Master Thesis such as Time Slicing, MPE-FEC and MPE-iFEC, have been implemented in .m Matlab files. Several Matlab programs had also been written in order to analyzed the TS packet traces and accommodate them with the programs where the link layer mechanisms are implemented.

Only streaming services will be evaluated in this Master Thesis as they are the most relevant services in broadcast networks. File delivery services are left for future work. The main parameter used to evaluate a streaming service quality is the Erroneous Second Ratio (ESR), defined as the percentage of time that the displayed information contains an error. Assuming that every IP datagram can be played independently, the ESR can be calculated as the percentage of sections that are received incorrectly. Another important parameter is the MFER, which is the percentage of datagram burst that could not be corrected by MPE-FEC.

V.3.*Traces Analyzed*

The 6 traces selected are shown in Table 3. They were taken in two different scenarios. The student village scenario can be considered as a rural environment, with low height obstacles that rarely block the signal from the transmitters. Figure 13 shows the error distribution along time for the three traces taken in the student village scenario. As can be seen the signal rarely gets blocked and when it does, the blockage condition does not last for more than 15 seconds. On the contrary, the suburban scenario presents high buildings that block the line of sight with the transmitter. Figure 14 shows the error distribution along time for the three traces taken in the sub urban scenario. In this case the signal is frequently blocked by big obstacles that cause error bursts of long duration. In trace 6 these error bursts can reach a length of one minute, which is over the capabilities of any reasonable interleaving. As it can be seen also in Table 3, the traces go from very good reception conditions with less than 2 % of TS PER (Packet Error Rate) to bad reception condition with more than 25 % of TS PER.

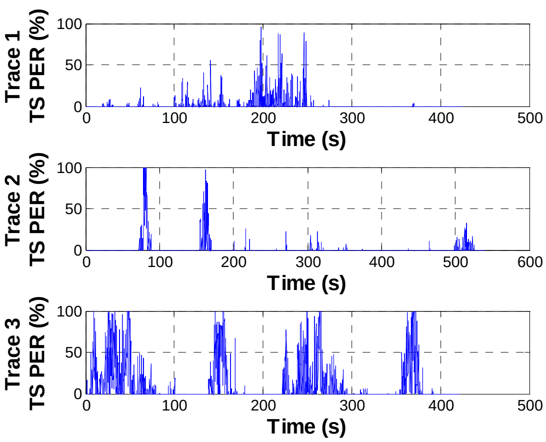

Fig. 13. TS PER distribution along time for traces 1-3

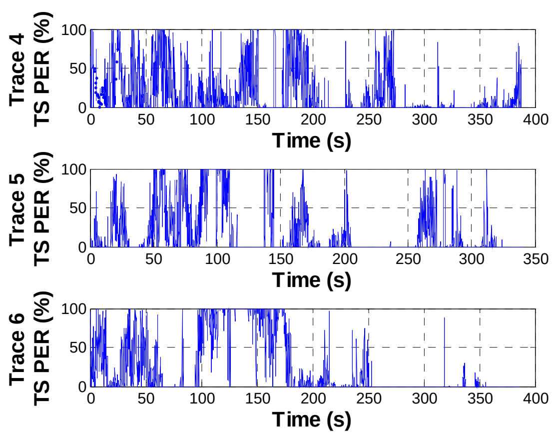

Fig. 14. TS PER distribution along time for traces 4-6

Trace name	File name	Scenario	Duration	TS PER	Avg RSSI
Trace 1	hmittaus10.mat	Student Village	7.1 min	1.5 %	-76.5 dBm
Trace 2	hmittaus9.mat	Student Village	9.0 min	1.6 %	-76.7 dBm
Trace 3	hmittaus7.mat	Student Village	7.0 min	10.5 %	-80.0 dBm
Trace 4	hmittaus20.mat	Sub Urban	6.4 min	16.1 %	-83.3 dBm
Trace 5	hmittaus23.mat	Sub Urban	5.7 min	18.1 %	-83.4 dBm
Trace 6	hmittaus21.mat	Sub Urban	6.4 min	26.0 %	-83.7 dBm

Table 3. Measurements selected for analysis

VI. RESULTS AND DISCUSSIONS

VI.1.*MPE-FEC*

First, the performance of MPE-FEC based on the error traces will be evaluated. In Tables 3 and 4, the MFER and ESR obtained after MPE-FEC decoding for five different encoding ratios. The MFER is always higher than the correspondent ESR due to the fact that bursts that can not be fully corrected can still carry correct sections. An ESR of 5 % is the value that is often considered as the threshold of good quality reception in a streaming service. As it can be seen in Table 4, only traces 1 and 2 are able of providing an ESR inferior to the 5 %. These traces have a TS PER of less than the 2 %, which is a very low value and only achievable in very good reception conditions. The rest of the traces have ESR values over the 5 %. This situation exposes the limited correction capacity of MPE-FEC, which is only capable of assuring a good streaming quality in scenarios with a large deployment of transmitters and repeaters.

	File name	1/2	2/3	3/4	5/6	7/8
Trace 1	hmittaus10.mat	1.03	1.78	2.44	4.32	5.67
Trace 2	hmittaus9.mat	1.19	1.64	2.01	2.44	2.94
Trace 3	hmittaus7.mat	8.73	12.24	14.53	17.30	19.10
Trace 4	hmittaus20.mat	14.95	20.12	23.92	28.22	30.68
Trace 5	hmittaus23.mat	18.01	22.36	25.90	29.34	31.35
Trace 6	hmittaus21.mat	28.09	30.85	32.67	34.55	36.03

Table 3. MFER (%) of all the traces analyzed

	File name	1/2	2/3	3/4	5/6	7/8
Trace 1	hmittaus10.mat	0.75	1.01	1.14	1.56	1.78
Trace 2	hmittaus9.mat	0.95	1.11	1.19	1.29	1.37
Trace 3	hmittaus7.mat	6.87	8.08	8.59	9.21	9.50
Trace 4	hmittaus20.mat	12.31	14.08	14.85	15.85	16.26
Trace 5	hmittaus23.mat	15.71	17.21	18.12	18.88	19.20
Trace 6	hmittaus21.mat	26.23	27.25	27.63	28.08	28.35

Table 4. ESR (%) of all the traces analyzed

VI.2.*RS Sliding Encoding*

As the RS sliding encoding method employs the same RS (255,191) code as MPE-FEC, the amount of data and parity information carried in each time slicing burst for each code ratio is also the same. The two main parameters that regulate the operation of the mechanism are the encoding parallelization, *B*, and the FEC spreading factor *S*. As it can be seen in Figures 15-20 when the value of these parameters increases, the obtained ESR is reduced. However, this interleaving gain is more relevant for the lower code rates. When the employed code rate is too high for the number of errors in the information, the RS sliding mechanism is not enough to improve the reception quality, no matter how much the interleaving is increased. On the other hand, the interleaving gain is remarkable for all traces except for trace 6. At first, this could be attributed to the fact that the number of errors in this trace is higher than in the rest of the traces. However, even a code rate of 1/2 can not obtain a significant gain when applying the longest interleaving. This is due to the way the errors are distributed along time. As can be seen in Figure 14, the error bursts in this trace can reach a length of more than one minute. An interleaving scheme capable of correction an error burst of such duration is not feasible for memory and delay reasons. On the contrary, traces 1-5 contain error bursts of duration no longer than 30 seconds that can be corrected applying the adequate code rate and interleaving. This is the reason why the biggest gains are obtain with traces 3-5. In these traces, MPE-iFEC mechanism can cope with error burst that are beyond the correction capabilities of MPE-FEC, so the results are greatly improved thanks to interleaving.

As it has been previously said, all 6 traces have been taken in vehicular reception conditions with a velocity between 0 and 30 km/h. However, in pedestrian reception conditions, the user velocity is of about 3 km/h. The lower the velocity, the longer the user remains in shadowed areas. Therefore, in a pedestrian scenario, the duration of the error bursts will be even longer, limiting the corrections capabilities of both MPE-FEC and MPE-iFEC. Longer interleaving durations will be needed, and the ESR values presented would be higher for a same interleaving configuration.

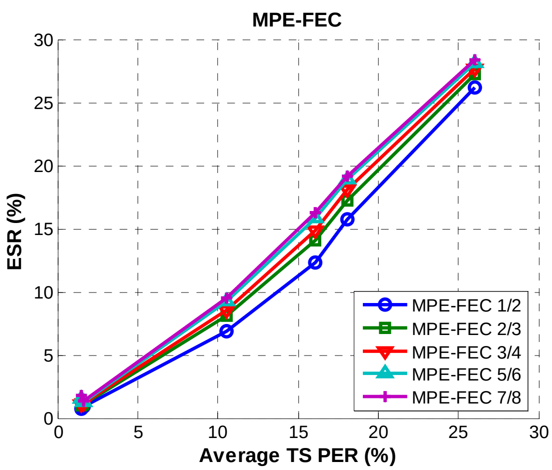

Fig. 15. ESR (%) vs. average TS PER (%) for each code rate with MPE-FEC

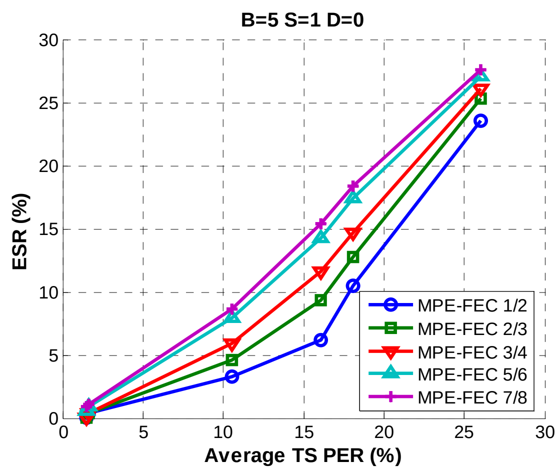

Fig. 16. ESR (%) vs. average TS PER (%) for each code rate with B=5 S=1 and D=0

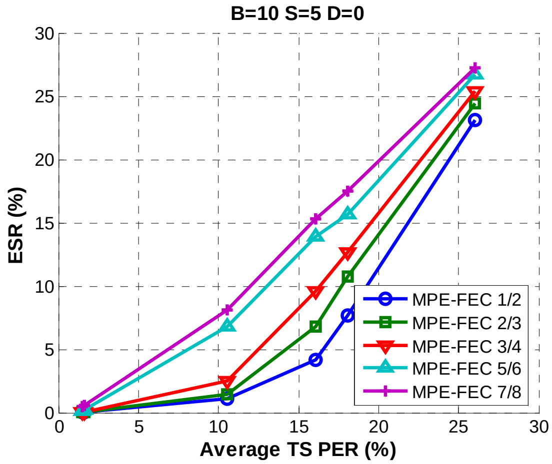

Fig. 17. ESR (%) vs. average TS PER (%) for each code rate with B=10 S=5 and D=0

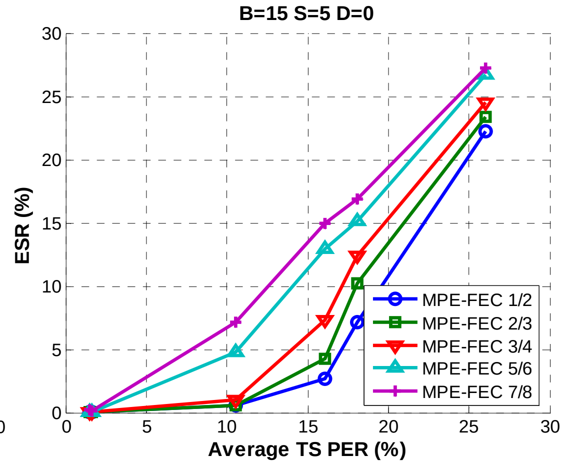

Fig. 18. ESR (%) vs. average TS PER (%) for each code rate with B=15 S=5 and D=0

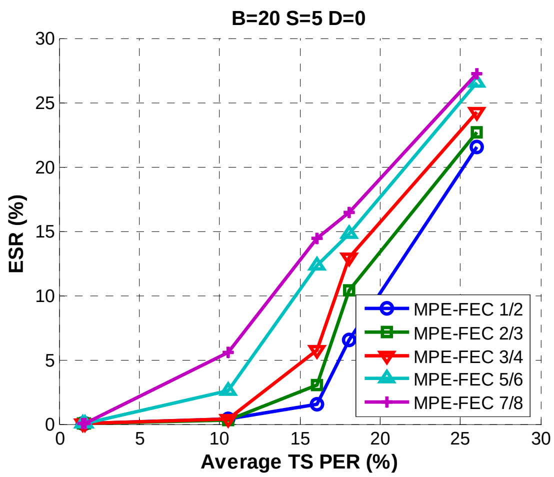

Fig. 19. ESR (%) vs. average TS PER (%) for each code rate with B=20 S=5 and D=0

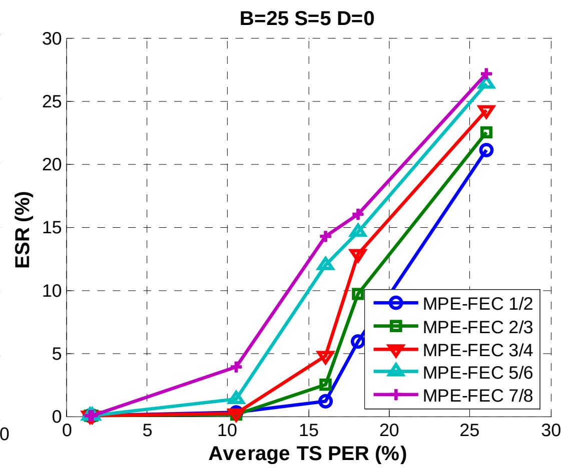

Fig. 20. ESR (%) vs. average TS PER (%) for each code rate with B=25 S=5 and D=0

In Figures 21-26 the ESR obtained after MPE-iFEC with respect to the B+S parameter is represented for each of the six traces. The horizontal black line represents the ESR5% threshold. Traces 1 and 2 already have a low amount of errors, so MPE-iFEC represents a little gain with respect to MPE-FEC, although it manages to keep the ESR at 0% which involves a perfect reception quality. It is however in traces 3 and 4 where the interleaving mechanism has the most impact. In this traces, MPE-iFEC is capable of reaching the ESR5% value for low B+S configurations in case code rates 1/2 or 2/3 are used. On the other hand, assuming larger B+S configurations are available, it is also capable of passing under the ESR = 5% threshold for a code rate up to 7/8 in trace 3 and 3/4 in trace 4.

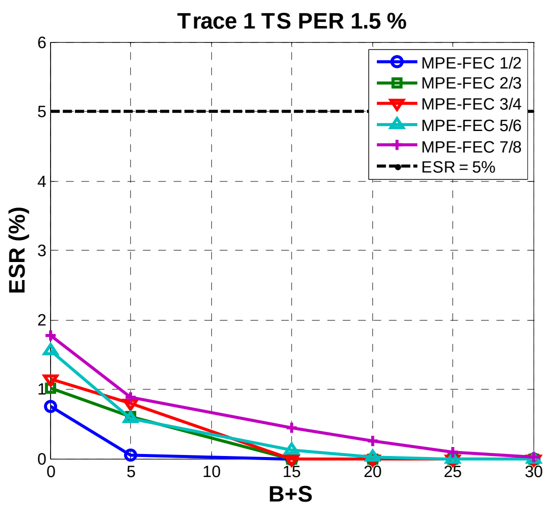

Fig. 21. ESR (%) vs. B+S (D=0) for each code rate in trace 1

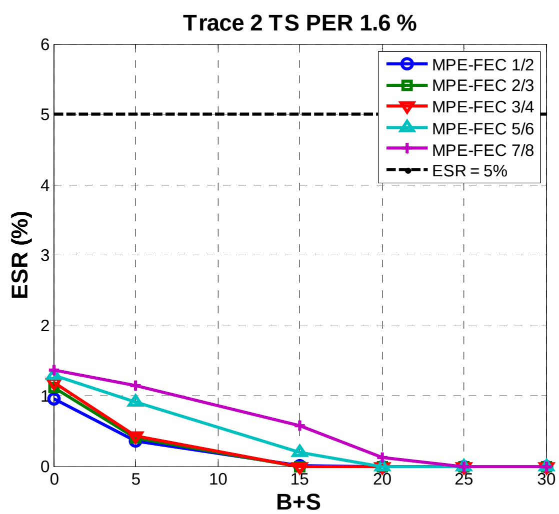

Fig. 22. ESR (%) vs. B+S (D=0)for each code rate in trace 2

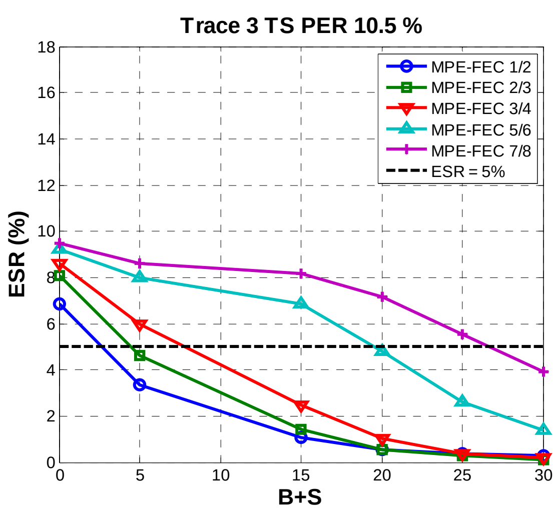

Fig. 23. ESR (%) vs. B+S (D=0)for each code rate in trace 3

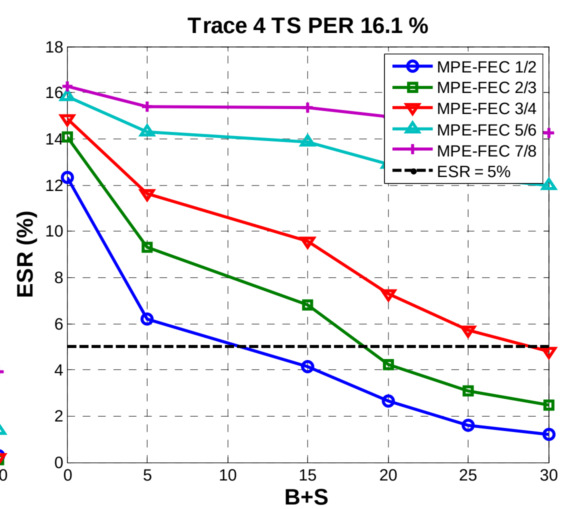

Fig. 24. ESR (%) vs. B+S (D=0)for each code rate in trace 4

However, in spite of achieving better results than MPE-FEC, MPE-iFEC is barely able to reach a value of ESR = 5% in trace 5 and not capable of reaching it at all in trace 6. As it has previously discussed, the duration of the errors burst in trace 6 is too much for any degree of interleaving and can not be corrected. This explains the little effect that interleaving has in the number of errors that can be corrected in this trace, even for a code rate of 1/2.

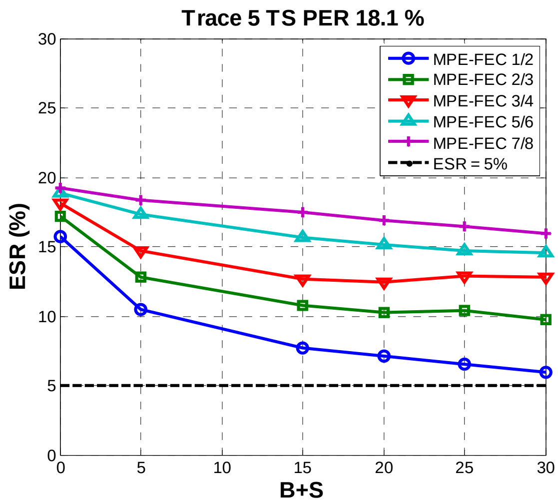

Fig. 25. ESR (%) vs. B+S (D=0) for each code rate in trace 5

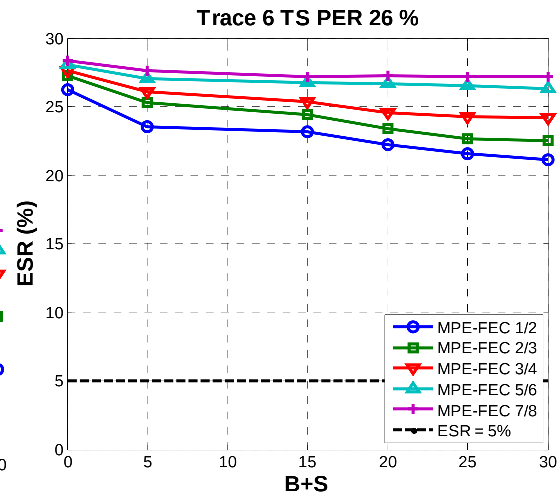

Fig. 26. ESR (%) vs. B+S (D=0) for each code rate in trace 6

Finally, the effect of the *D* parameter in the correction capability of the sliding RS encoding will be studied. As it was explained in section III, the delay parameter can be useful in order to reduce the zapping time and memory requirements in the user terminal. However, it can also affect the performance of the interleaving mechanism. Figures 27 and 28 show the impact that incrementing the delay in transmitting the data has in the ESR for two different configurations. As can be seen, the *D* parameter has little effect in the ESR for high code rates that are not able to correct a great number of errors. This is the case of code rates 5/6 and 7/8 in both configurations. However, the same can not be said about lower code rates. As it is shown, the Delay parameter has an important impact in the corrections capabilities. At first, when increasing the *D* value, the performance decreases because of the fact that the parity information is being sent in the same time slicing bursts as the data from which was generated. The ESR reaches a maximum value when the delay is situated at (*B*+*S*)/2, which is the point of maximum overlap between parity and data. Passing this point, the trend reverses and the ESR begins to decrease. When the delay is situated over the *B*+*S* value, there is again no overlap, and the performance increases as *D* is incremented.

In Figures 29 and 30, the sizing function according to the *D* value is represented for both configurations. As can be seen in equations 1 and 2, sizing function is directly proportional to the receiving latency and terminal memory requirements. Increasing the *D* value over *B*+*S* can achieve better performance, but also involves increasing the latency in the received information and the memory that the terminal must implement. It is only recommended in case there is enough memory available and the service transmitted is not sensitive to delay. On the other hand, setting the *D* parameter to a value between *S* and *B*, reduces memory and latency, but worsens the performance of the interleaving mechanism. Operators must configure the *D* parameter according to each service and expected reception conditions.

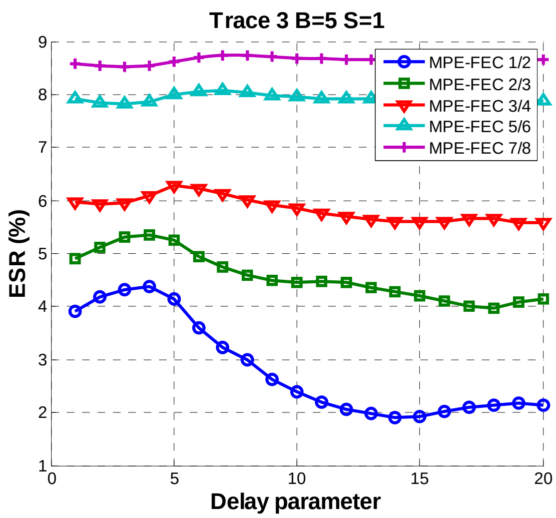

Fig. 27. ESR (%) vs. *D* with *B*=5 and *S*=1 for each code rate in trace 3

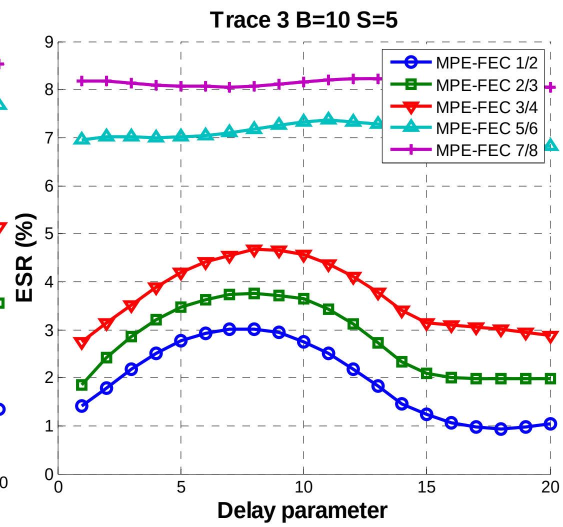

Fig. 28. ESR (%) vs. *D* with *B*=10 and *S*=5 for each code rate in trace 3

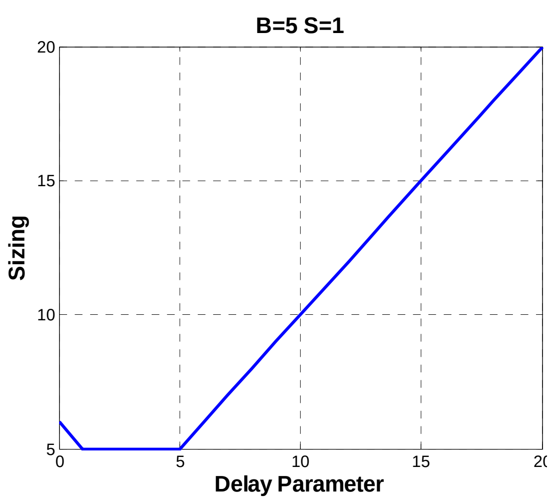

Fig. 29. Sizing function for B=5 and S=1

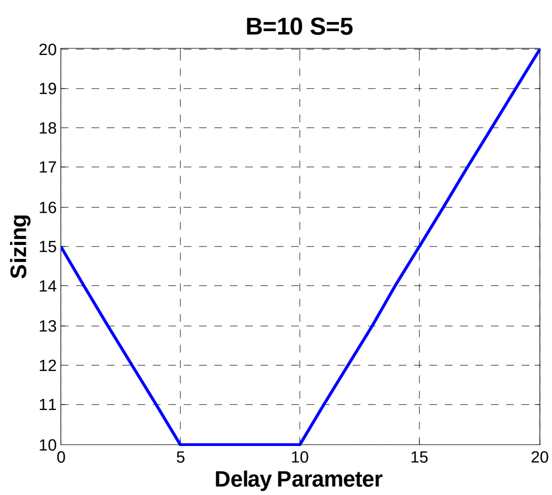

Fig. 30. Sizing function for B=10 and S=5

Regarding the zapping time, any *D* value over *S* will limit the maximum zapping time to a *B* value. The maximum zapping time takes place when the first burst received from a recently switched service is erroneous, and the information has to be corrected. The terminal needs to wait enough bursts in order to receive all the information from one encoding matrix. If *D*=0, the maximum zapping time is equal to *B*+*S*. However, when $D > S$ and all *B* burst are received, the terminal does not need to wait anymore until the arrival of parity because of the fact that it has been already received. Although increasing the *D* parameter effectively reduces the zapping time, it also reduces the encoding performance and therefore must be setting accordingly.

VII. CONCLUSIONS AND FUTURE WORK

In this Master Thesis, the new link layer protection mechanism incorporated in the latest European mobile TV standard, DVB-SH, the RS sliding encoding has been studied. Its performance has been emulated by means of Matlab simulations based on DVB-H MPEG-2 packet error traces obtained from vehicular field measurements in the DVB-H test network of the University of Turku (Finland). The satellite reception characteristic of DVB-SH has to cope with the impairments of the satellite channel. In satellite reception, the signal is frequently blocked by buildings or other obstacles and so, long and deep fades in the received signal strength due to shadowing are to be expected. DVB-H link layer protection mechanism MPE-FEC is not suited to combat this type of degradation as is designed to cope with the fast fading variations in the signal. In order to mitigate the impairments of shadowing new multi burst encoding mechanisms are needed.

MPE-iFEC is a multi burst protection framework designed to have enough flexibility for a great number of applications. Multi burst schemes are based in the interleaving of information from different bursts. One MPE-iFEC mapping, the RS sliding encoding have been incorporated into the DVB-SH standard. This new mechanism has more parameters to configure than MPE-FEC, and therefore it is very important to analyze the performance for each setting.

MPE-FEC has been demonstrated to not be capable of providing a good streaming quality in bad reception conditions. In this Master Thesis it has been shown that MPE-FEC needs a large deployment of transmitters and repeaters capable of assuring a low number of errors in the received information. However, RS sliding encoding is able of considerably increasing the quality of a streaming service at expense of some delay in the displayed information. The delay introduced by the RS sliding encoding increments the zapping time, a parameter very important from the point of view of the user. In order of decreasing the zapping time, fast zapping techniques can be used. These techniques are aimed to shorten the period of time between when the user switches to a service and when the terminal begins to display the service.

The results show how the RS sliding encoding is capable of improving the reception of streaming services even in DVB-H reception conditions. The ESR value normally used to measure the quality of streaming services is greatly reduced by using this mechanism. Increasing the number of bursts interleaved and FEC spreading achieves progressively better results; however, the reception delay is also increased as consequence. In case it is needed, the delay parameter can be used to reduce the reception delay and zapping time at expense of some performance.

The future lines of work after this Master Thesis are the analysis of MPE-iFEC for file delivery services. Multi burst encoding techniques are even more advantageous in file delivery because of the fact that this kind of services is not as sensitive to delay as streaming services. In file delivery services, B and S can be configured to high values and a greater interleaving gain can be achieved. Other aspect that will be studied after this work is the analysis of the MPE-iFEC with DVB-SH error traces. The validation process of the standard will take place in 2008, in a test trial that is going to take place in Barcelona in the context of the European project Celtic B21C (Broadcast for the 21st Century) and the Spanish project FURIA (Futura Red Integrada Audiovisual). This Master Thesis has been developed inside of FURIA and therefore, it is expected to have access to the results of these trials. The measurements and traces will be used in future investigations of the work presented in this Master Thesis.

However, at present, error traces from field trials are very difficult to obtain due to the early state of the standard. Until the validation process take place, error traces from physical layer simulators can be used instead. The performance of the RS sliding encoding mechanism will be evaluated in actual DVB-SH transmissions where the entire chain of physical layer elements such as the interleaver or the turbocoder, is implemented.

Also, other multi burst encoding mechanisms based in MPE-iFEC apart from the RS sliding encoding are of interest. This is specially the case for mappings based in Raptor codes. These codes can be implemented in software and are able to encode a large amount of information in a jointly manner. Therefore, Raptor codes are able to obtain greater interleaving gains than the RS (255,191) used in the RS sliding mechanism.

ACKNOWLEDGEMENTS

This work has been founded by the Ministry of Industry, Tourism and Commerce within the project FURIA (Futura Red Integrada Audiovisual) in the Avanz@ framework. FURIA is a strategic singular project in the area of Network audio-visual technologies of the e-NEM platform (www.furiapse.com).

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ANNEX A

TRANSMISIÓN DE SERVICIOS DE TELEVISIÓN DIGITAL MÓVIL EN REDES DVB-SH

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Abstract — Este artículo presenta el futuro estándar europeo de radiodifusión de TV digital móvil DVB-SH (*Digital Video Broadcasting – Satellite Services to Handheld Devices*), el cual permitirá conseguir una cobertura nacional en toda Europa de servicios de TV móvil, y cuya puesta en funcionamiento está prevista para el 2008. DVB-SH es la evolución tecnológica del estándar europeo de TDT móvil DVB-H (*Digital Video Broadcast – Handhelds*). Sus principales características son una arquitectura híbrida terrestre-satelital y la transmisión en banda S en torno a los 2 GHz, la cual está disponible a nivel europeo para este tipo de servicios. En este artículo se presentan las principales características técnicas de DVB-SH, prestando especial atención a las mejoras que se han introducido en el estándar con respecto a DVB-H, las cuales están principalmente destinadas a reforzar la transmisión vía satélite. En el artículo también se describen y se comparan las diferentes arquitecturas de red posibles en DVB-SH, y se exponen las principales líneas de investigación sobre esta tecnología.

I. INTRODUCCIÓN

Dentro de las comunicaciones móviles multimedia, el máximo exponente a día de hoy es la TV móvil. Los teléfonos móviles actuales ya disponen de radio, reproductor de música, cámara digital y grabador de video, y la TV móvil será pronto una nueva característica. Recientes estudios de mercado han revelado un gran interés por parte de los consumidores, y se prevé que la demanda de estos servicios explotará en el 2011, con más de 500 millones de usuarios en todo el mundo [1]. Aunque será necesario adaptar los contenidos al entorno móvil, será posible ofrecer un servicio mucho más rico y personalizado que la TV convencional, abriendo un abanico de nuevos servicios para los consumidores, y proporcionando nuevas vías de negocio para todos los agentes de la industria audiovisual y de las telecomunicaciones. Estos servicios son además clave para el desarrollo de la Sociedad de la Información, ya que permiten el acceso universal a contenidos multimedia en cualquier momento y lugar.

Con la llegada de la tercera generación (3G), los operadores de telefonía móvil han empezado recientemente a ofrecer servicios de TV móvil, sin embargo, sus ofertas son considerablemente limitadas, tanto desde un punto de vista técnico como de coste de servicio [2]. A la hora de prestar servicios de TV móvil, hoy en día está comúnmente aceptada la necesidad de emplear redes de radiodifusión específicamente diseñas para servicios móviles como complemento a las redes celulares. Sólo estas redes tienen la capacidad necesaria para soportar un consumo a gran escala de este tipo de servicios, ya que pueden distribuir servicios multimedia de banda ancha en áreas extensas sin limitación alguna en el número de usuarios que pueden recibir el servicio.

El estándar europeo de Televisión Digital Terrestre (TDT) móvil es conocido como DVB-H (*Digital Video Broadcating – Handhelds*), el cual es una evolución tecnológica adaptada para terminales móviles del estándar de TDT, DVB-T (*Digital Video Broadcasting – Terrestrial*) [3]. DVB-H fue diseñado originariamente para trabajar en la banda UHF entre 470 y 862 MHz, y proporciona capacidades de 5 a 10 Mb/s en canales de 8 MHz. Mantiene la capa física de DVB-T, y añade nuevos elementos en la capa de enlace, siendo posible reutilizar la infraestructura de red (transmisores, multiplexores etc.). Su principal característica es el empleo de una técnica de transmisión discontinua en la que la información se transmite a ráfagas (*bursts*), lo cual reduce considerablemente el consumo de potencia medio de los terminales. Además, añade un mecanismo de corrección de errores adicional, el cual proporciona una mayor robustez a la transmisión, sobre todo en situaciones de movilidad e interferencias. Aunque DVB-H supone un avance

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significativo en el mundo de la radiodifusión multimedia móvil, presenta dos serias barreras para conseguir un mercado europeo de servicios de TV móvil: el elevado coste del despliegue de la red, y la indisponibilidad de frecuencias a nivel europeo. Actualmente ya se ofrecen servicios comerciales en los principales núcleos urbanos de Italia, Finlandia y Albania en Europa, y en Vietnam, India y las Filipinas en Asia, y se espera que el 2008 sea el inicio del despegue de estos servicios.

Una de las mayores preocupaciones sobre DVB-H es la gran cantidad de infraestructura de red necesaria para proporcionar niveles aceptables de cobertura. Los primeros estudios de mercado sobre TV móvil revelaron que los usuarios esperan unos niveles de cobertura similares a los ofrecidos por la telefonía celular [4]. Sin embargo como las condiciones de recepción de DVB-H son mucho más severas que las de DVB-T, sobre todo para situaciones de recepción en interiores (*indoor*) y en automóviles (*vehicular*), DVB-H requiere mucha más infraestructura de red que la existente para DVB-T [5]. Esto es, mayores potencias de transmisión y, especialmente, un considerable mayor número de transmisores y repetidores (*gap-fillers*), formando redes de frecuencia única SFN (*Single Frequency Networks*) con una alta densidad de emplazamientos [6]. Esta penalización es particularmente evidente para niveles de cobertura muy elevados (mayores que el 90%) [7]. Por este motivo, el despliegue de una red de TDT móvil tan sólo parece viable en zonas urbanas con un gran número de potenciales clientes capaces de amortizar la inversión necesaria en infraestructura.

Por otro lado, el rango de frecuencias UHF en el que en principio ha de operar el sistema está también ocupado por la televisión analógica tradicional y la TDT. Esta escasez de frecuencias continuará hasta que se produzca el llamado “apagón analógico”, lo cual no sucederá hasta el 2010 en la mayoría de países europeos incluido España.

Bajo este contexto, la evolución tecnológica de DVB-H, llamada DVB-SH (*Digital Video Broadcasting – Satellite Services to Handheld Devices*) está generando un gran interés en el mundo de las telecomunicaciones para crear un mercado europeo de servicios de TV móvil [8]. Las principales características de DVB-SH son dos: una arquitectura híbrida terrestre satelital, y la transmisión en la banda S situada en torno a los 2 GHz, la cual está disponible a nivel europeo para este tipo de servicios.

En este artículo se presentan las principales características técnicas de DVB-SH. La estructura del artículo es la siguiente: en la sección 2 se introduce el concepto de red híbrida terrestre-satelital. En la sección 3 se detalla el funcionamiento del sistema DVB-SH, describiendo y comparando las diferentes arquitecturas de red posibles. En las secciones 4 y 5 se describen la capa física y la capa de enlace de DVB-SH. Finalmente, en la sección 6 se exponen las principales líneas de investigación dentro de DVB-SH.

II. REDES HÍBRIDAS TERRESTRES SATELITALES

El satélite es probablemente el medio ideal para distribuir una señal de TV a grandes audiencias cubriendo áreas extensas. Un único satélite geoestacionario es capaz proporcionar cobertura en todo el territorio europeo desde el mismo momento en el que el satélite está operativo. Puesto que el coste de la infraestructura de red es fijo, e independiente del número de usuarios, a partir de cierto umbral, el satélite se convierte en el medio más económico de transmisión. Sin embargo, la señal proveniente de un satélite puede verse muy deteriorada en situaciones donde no existe visión directa, como es el caso de zonas urbanas o en interiores, especialmente si se trata de terminales de mano. La solución consiste en reforzar la señal procedente del satélite mediante una red terrestre de radiodifusión complementaria, siempre y cuando su inversión esté económicamente justificada. Esta arquitectura híbrida, capaz de combinar las ventajas de ambas filosofías, se presenta por tanto como un método muy eficiente para la provisión de servicios de TV digital móvil.

Los sistemas terrestres de difusión de televisión, como es el caso de la TV analógica, la TDT y la TDT móvil, emplean la banda UHF (entre 470 y 862 MHz). Esta banda se caracteriza por unas pérdidas de propagación de las señales radioeléctricas reducidas (son directamente proporcionales a la frecuencia de portadora), y posibilitar anchos de banda considerables, así como la ausencia del ruido humano característico a bajas frecuencias. Sin embargo, la mayoría de los canales de la banda UHF están ocupados a día de hoy por la TV analógica y digital, lo cual está dificultando la puesta en marcha del mercado de la TV móvil en Europa. Los sistemas híbridos, por el contrario, trabajan en la banda S, la cual va desde los 2 hasta los 4 GHz. La banda de frecuencias reservada para los sistemas por satélite, y que está situada entre los 2170 MHz y los 2200 MHz, ha sido recientemente asignada por la Unión Europea para los sistemas híbridos terrestres satelitales [9].

La Figura 1 muestra las bandas de frecuencia para los sistemas IMT2000, dentro de la cual se encuentran los 30 MHz disponibles para los sistemas móviles por satélite, situados justo al lado de la banda reservada para la telefonía móvil celular 3G. Esto posibilita la reutilización de los equipos, antenas y ubicaciones empleadas por los sistemas 3G, reduciendo considerablemente el coste del despliegue de la red terrestre. Diversos estudios como en [10] hablan de un abaratamiento en torno al 50% en el coste de los transmisores en la banda S comparado con el coste de los transmisores en la banda UHF en el caso de reutilizar emplazamientos celulares.

Trabajar a frecuencias más elevadas supone mayores pérdidas de propagación de la señal radioeléctrica, lo que en principio conlleva una red terrestre más densa que en el caso de UHF, y por tanto un aumento de la inversión en infraestructura de red. Esta penalización se compensa en cierta medida por el hecho de que frecuencias más elevadas también implican mayores ganancias en antenas de reducido tamaño, como las que se implementan en los terminales de mano. Por ejemplo, la ganancia típica de antena de un terminal DVB-H a 700 MHz es de -7 dBi, mientras que en la banda S se podrían conseguir ganancias del orden de 0 dBi. Además, trabajar en banda S también permite recurrir a técnicas de diversidad basadas en un receptor de dos antenas, lo cual no es posible cuando se trabaja en la banda UHF. Un esquema de diversidad de dos antenas en recepción puede conseguir ganancias del orden de 6 dB (asumiendo la misma capa física) en entornos con una fuerte componente multicamino, como es el caso de interiores [11].

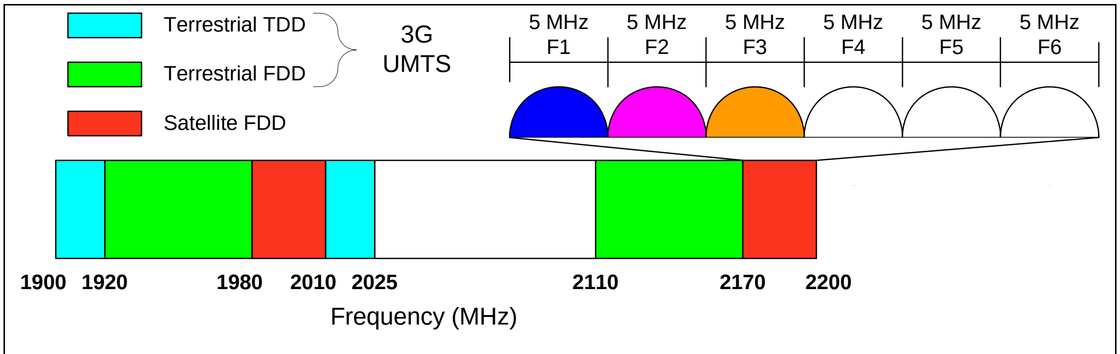

Fig. 1. Banda de frecuencia de los sistemas IMT 2000.

En la actualidad existen dos sistemas de difusión de TV móvil híbridos terrestre satelitales en funcionamiento, uno en Corea del Sur y otro en Japón.

El sistema coreano está basado en S-DMB (*Satellite – Digital Multimedia Broadcasting*) [12], mientras que el sistema japonés utiliza tecnología ISDB (*Integrated Services Digital Broadcasting*), si bien es cierto que ambos sistemas son muy similares. El sistema S-DMB coreano entró en funcionamiento en Mayo del 2005, y actualmente cuenta con más de 890.000 subscriptores. Este sistema ofrece un servicio compuesto por 15 canales de TV, 19 canales de audio y 3 canales de datos, así como servicios interactivos bidireccionales. La arquitectura de red está compuesta por el satélite geoestacionario Hanbyul junto con cerca de 10.000 repetidores terrestres encargados de reforzar la señal proveniente del satélite. El satélite recibe dos señales en banda Ku (12-18 GHz) procedentes de la estación terrena, una señal CDM (*Code Division Multiplexing*) y otra señal TDM (*Time Division Multiplexing*). El satélite retransmite la señal TDM en banda Ku a los repetidores terrestres, mientras que convierte la señal CDM en banda Ku a banda S (en el caso de S-DMB a 2.6 GHz), y la transmite directamente a los terminales de usuario. Los repetidores terrestres por su parte convierten la señal TDM en banda Ku a señal CDM en banda S antes de su transmisión. De esta forma los terminales de usuario reciben la información modulada en CDM en banda S, tanto si proviene de la componente terrestre como de la componente satelital. Las señales en banda S se combinan en recepción mediante un esquema de diversidad espacial basado en dos antenas con un receptor Rake de seis dedos por antena. El ensanchado de la modulación CDM se realiza mediante un código Walsh a 16.384 MHz lo cual proporciona una ganancia de procesado por un factor 64. Actualmente se transmiten 31 canales CDM a 256 kb/s cada uno, obteniendo una velocidad total de 8 Mb/s en un canal de 25 MHz. Al igual que DVB-H, S-DMB emplea un codificador Reed Solomon (RS) (204,188) junto con un codificador convolucional para proteger la información en la capa física. Entre ambos codificadores se emplea un entrelazador convolucional a nivel de byte, y tras el codificador convolucional se emplea un entrelazador a nivel de bit.

III. EL ESTÁNDAR EUROPEO DVB-SH

DVB-SH (*Digital Video Broadcasting – Satellitel services for Handheld devices*) es una solución híbrida terrestre satelital para la difusión de televisión digital móvil que opera en la banda S. La Figura 2 muestra la arquitectura de referencia. De manera semejante a S-DMB, DVB-SH emplea un satélite geoestacionario junto con una red terrestre que proporciona el servicio en aquellas zonas donde la señal del satélite se ve seriamente degradada. Además la red terrestre permite aumentar la capacidad del sistema. Al contrario que S-DMB, donde la señal que reciben los usuarios es de tipo CDM, DVB-SH emplea señales OFDM (*Orthogonal Frequency Division Multiplexing*) en la red terrestre complementaria y señales OFDM o TDM en la transmisión vía satélite. La utilización de OFDM permite desplegar redes de frecuencia única SFN, en las que los receptores pueden combinar todas las señales provenientes de los transmisores a su alcance (incluido el satélite), siempre y cuando el retardo entre señales no supere el intervalo de guarda de la señal OFDM. DVB-SH permite además el empleo de la forma de onda TDM para la componente satelital, la cual permite mayores potencias de transmisión en el satélite. Esto es debido a que las señales multiportadora como OFDM se caracterizan por una elevada potencia de pico, que impide que los amplificadores de alta potencia trabajen cerca del punto de saturación (en el cual la potencia transmitida es máxima) para evitar la aparición de distorsiones no lineales. Por otro lado transmitir con señales TDM permite que los amplificadores puedan trabajar más cerca del punto de saturación, lo cual representa una ganancia en potencia con respecto a OFDM y por tanto, un aumento en la cobertura de la componente satelital. Esta ganancia es significativa en aquellas configuraciones de carga para las cuales cada amplificador trabaja con una única señal DVB-SH, ya que en el caso de tener que trabajar con un múltiplex de señales DVB-SH moduladas en TDM es necesario bajar también el punto de trabajo.

La presencia de dos capas físicas aumenta las posibilidades de configuración del sistema, y da lugar a dos arquitecturas diferentes:

- Arquitectura SH-A: tanto la componente satelital como la componente terrestre emplean la forma de onda OFDM.
- Arquitectura SH-B: la componente satelital emplea la forma de onda TDM mientras que la componente terrestre emplea OFDM.

La arquitectura SH-A posibilita la creación de redes de frecuencia única SFN, en las que tanto la componente satelital como la componente terrestre transmiten los mismos contenidos a la misma frecuencia. Las redes SFN permiten una elevada eficiencia espectral, pero imponen que la señal transmitida por la componente terrestre sea idéntica a la transmitida por la componente satelital en todo el territorio bajo la huella del satélite. Por este motivo se permite la posibilidad de emplear redes MFN (*Multi Frequency Networks*), en las que, aunque tanto la componente satelital como la terrestre emplean OFDM, transmiten en canales de frecuencia diferentes. Aunque en una red MFN se reduce la eficiencia espectral, es posible la transmisión de contenidos locales en el mismo canal de frecuencia empleado por la red terrestre para la transmisión de contenidos a nivel nacional. Además, en una red MFN es posible ajustar los parámetros de transmisión en cada región o ciudad con el fin de conseguir una planificación óptima, ya que en una SFN la red terrestre debe transmitir con la misma configuración en toda la geografía en la que se desea prestar servicio. En aquellos terminales con dos interfaces radio es posible mejorar la calidad de la recepción mediante técnicas de *soft-combining*, en las que las señales radioeléctricas provenientes del satélite y de la red terrestre se combinan en banda base. En el caso de emplear tasas de codificación complementarias en la componente terrestre y satelital, es posible obtener una ganancia adicional por diversidad gracias al uso de *code-combining*.

En una arquitectura de tipo SH-B, cada una de las dos componentes que constituyen la red DVB-SH emplea una forma de onda diferente, por lo que es necesario que transmitan en diferentes frecuencias para no interferirse entre sí. Aunque la eficiencia espectral de la red se reduce con respecto a una arquitectura SH-A SFN, la transmisión desde satélite alcanza un mayor rendimiento gracias al empleo de TDM. Además, al igual que en el caso SH-A MFM, es posible optimizar la configuración de la componente terrestre en diferentes partes de la región cubierta por el satélite. Al igual que en las redes SH-A MFN, y dado que en este caso los terminales disponen obligatoriamente de un interfaz radio para cada forma de onda, es posible recurrir a técnicas *soft-combining* y/o *code-combining* para mejorar la recepción de la señal.

Mientras que una arquitectura SH-A es más adecuada en aquellas situaciones limitadas en espectro, una arquitectura SH-B es más adecuada cuando la transmisión desde satélite queda limitada en potencia. La existencia de dos arquitecturas implica la presencia de dos categorías de terminales disponibles, uno para cada arquitectura. Mientras que los terminales de tipo SH-B son compatibles con una arquitectura SH-A, pudiendo funcionar tanto en una red SFN como MFN, los terminales de tipo SH-A no pueden funcionar en arquitecturas SH-B, dado que no disponen del receptor TDM.

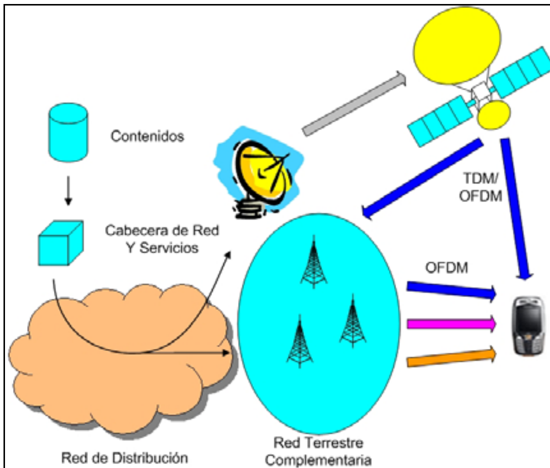

Fig. 2. Arquitectura híbrida terrestre satelital de referencia.

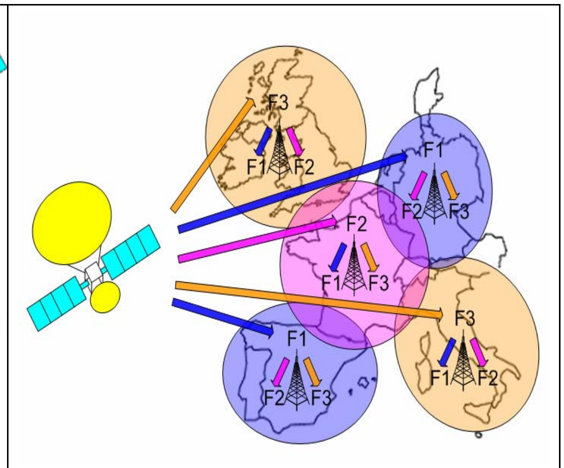

Fig. 3. Ejemplo de asignación de frecuencias en Europa tanto para los haces del satélite como para la red terrestre complementaria.

La arquitectura híbrida terrestre satelital plantea una gran cantidad de posibilidades, convirtiendo la planificación DVB-SH en una tarea mucho más flexible y compleja que en DVB-H, especialmente si se empelan satélites con capacidad multihaz. Dichos satélites están equipados con agrupaciones de antenas y sistemas de alimentación capaces de generar complejos diagramas de radiación. Gracias a dichos diagramas es posible conseguir zonas de cobertura independientes y solapadas entre sí con un único satélite. Este tipo de satélites son aconsejables en zonas de gran diversidad lingüística y cultural como es el caso de Europa, ya que permiten particularizar los contenidos que se transmiten por cada uno de los haces. Esta característica, junto con una reutilización de frecuencias entre los distintos haces de manera similar a como se realiza en planificación celular, permite optimizar el uso del espectro, aumentando la capacidad del sistema. En el caso de Europa, un esquema de reutilización de tres frecuencias es suficiente para evitar que dos haces adyacentes empleen la misma frecuencia y no se interfieran entre sí. Cada país queda cubierto por un haz del satélite en el que se transmiten contenidos nacionales a una frecuencia diferente que la empleada en los haces adyacentes. La componente terrestre no sólo puede emplearse para complementar la cobertura de la componente satelital en aquellas zonas donde la señal del satélite llegue muy debilitada, también puede emplearse para transmitir contenidos adicionales particularizados a la región en la que se presta servicio, y aumentar así la oferta de servicios de la que dispone el usuario. En este último caso, la red terrestre complementaria no opera en aquellos canales de frecuencia que se emplean para la transmisión vía satélite en su país, sino en aquellos canales que se emplean en los países colindantes. Sin embargo, es preciso una adecuada configuración de los parámetros de transmisión en las zonas de solapamiento entre haces adyacentes, puesto que pueden existir problemas de interferencias entre la componente terrestre de un país y el haz del satélite de otro país que en este caso se solapa.

A modo de ejemplo, para ilustrar las posibilidades de transmisión de una red DVB-SH operando en Europa, se puede considerar una arquitectura SH-A SFN operando con tres canales de 5 MHz, tal y como se muestra en la Figura 3. Un modo de transmisión típico de DVB-SH consiste en una modulación QPSK, un tamaño de FFT de 2K, 1/4 de intervalo de guarda OFDM y un ratio de turbocodificación de 1/3, lo cual proporciona una capacidad de 2.3 Mb/s en un canal de 5 MHz. Con esta configuración sería posible transmitir hasta 9 canales de TV de 256 kb/s en un único canal de 5 MHz, y hasta 27 canales si se emplea todo el ancho de banda disponible. De esta forma, en toda la superficie cubierta por un mismo haz del satélite se reciben los mismos 9 canales, mientras que en aquellas zonas al alcance de la red terrestre pueden recibirse hasta 18 canales adicionales particularizados a la región de emisión.

IV. DESCRIPCIÓN DE LA CAPA FÍSICA DVB-SH

DVB-SH incorpora nuevos mecanismos de protección en la capa física en comparación con DVB-H, como son la inclusión de un entrelazador de larga duración con el objetivo de combatir los desvanecimientos característicos del canal satélite y la sustitución del codificador convolucional por un turbocodificador para incrementar la robustez de la transmisión.

Al igual que DVB-H, DVB-SH se basa en la transmisión de flujos de información MPEG-2, los cuales transportan *bursts* de información compatibles con la transmisión discontinua característica de DVB-H.

Como se observa en la Figura 4, el flujo de información compuesto por paquetes MPEG-2 puede seguir dos caminos diferentes antes de su transmisión vía radio según vaya a transmitirse mediante TDM o mediante OFDM. Ambas propuestas comparten subsistemas comunes por los que la información deberá pasar sea cual sea la forma de onda empleada. Entre estos subsistemas comunes figuran la turbocodificación y el entrelazado de capa física. El turbocodificador incorporado en DVB-SH es el estandarizado por el 3GPP2, el cual emplean también los sistemas 3G, pero al que se le han añadido nuevos ratios de codificación con el fin de aumentar la flexibilidad en el grado de protección de la información. Este turbocodificador trabajando a una tasa de 1/3 es capaz de ofrecer una ganancia por encima de 3 dB en un canal Gaussiano con respecto a un codificador convolucional de tasa 1/2 más un codificador Reed-Solomon (204,188) complementado con codificación MPE-FEC 3/4 [11].

La recepción móvil vía satélite se caracteriza por experimentar largos y profundos desvanecimientos de la señal recibida. Dichos desvanecimientos se producen cuando el terminal atraviesa una zona de sombra y la señal proveniente del satélite es bloqueada por árboles o edificios, lo cual provoca que los errores que aparecen en la información recibida se agrupen en ráfagas de muy larga duración. Puesto que los mecanismos correctores de errores son mucho más eficaces en aquellas situaciones donde los errores se encuentran distribuidos uniformemente a lo largo del tiempo, estas ráfagas de errores dificultan la recuperación de la información transmitida, y por este motivo, se necesitan entrelazadores de larga duración.

La duración de los desvanecimientos que se producen en la señal depende del entorno en el que se encuentre el usuario así como de la velocidad a la que se desplace. Las situaciones de sombra en las que la señal se ve obstaculizada por árboles y otros objetos de reducido tamaño provocan desvanecimientos de menor duración que las situaciones de bloqueo debidas a la presencia de edificios. Por otro lado, si el usuario se mueve a grandes velocidades atravesará rápidamente las zonas de sombra y bloqueo, por lo que los desvanecimientos que experimenta la señal serán más cortos. El tiempo de entrelazado necesario para asegurar la correcta recepción de la información depende directamente de la longitud de las ráfagas de errores, y por tanto de la duración de los desvanecimientos. Se ha estimado que una duración de 10 segundos es suficiente para contrarrestar los efectos de sombra provocados por árboles, a velocidades no inferiores a 10 km/h, así como los efectos del bloqueo ocasionado por grandes obstáculos, a velocidades no inferiores a 60 km/h. En caso de que la recepción se produzca a menores velocidades, como es el caso de la velocidad característica de peatón de 3 km/h, es preferible asegurar la continuidad de servicio mediante un despliegue más denso de la componente terrestre en vez de aumentar aún más la duración del entrelazado. A pesar de la protección que ofrece un entrelazado temporal de 10 segundos, su implementación es bastante compleja, motivo por el que la duración de entrelazado de nivel físico definido por el estándar es configurable. Hay definidas dos longitudes diferentes de entrelazado, una en torno a los 200 ms y otra alrededor de los 10 s. Esta posibilidad da lugar a la aparición de dos clases de terminales: los de clase 1, con entrelazado corto de nivel físico (200 ms), y terminales de clase 2, con entrelazado largo de nivel físico (10 s). Una mayor longitud de entrelazado permite proteger la información frente a desvanecimientos de mayor duración, sin embargo, también supone unos mayores requisitos de memoria en el terminal. Para almacenar y procesar la información, en los terminales de clase 1 se necesita una memoria de 4 Mb y 8 Mb para QPSK y 16-QAM respectivamente, mientras en los terminales de clase 2 se necesita una memoria de 256 Mb y 512 Mb para QPSK y 16-QAM respectivamente. Además, un mayor entrelazado implica un aumento en el retardo de la información recibida, lo cual aumenta el tiempo de zapping y puede degradar la calidad de usuario [13].

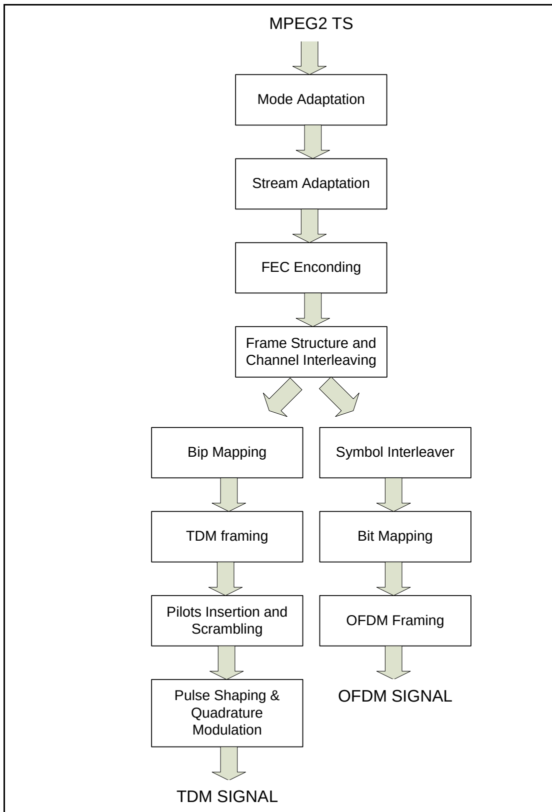

Fig. 4. Esquemas de la capa física DVB-SH en el que se muestran cada uno de los bloques que ha de atravesar la señal

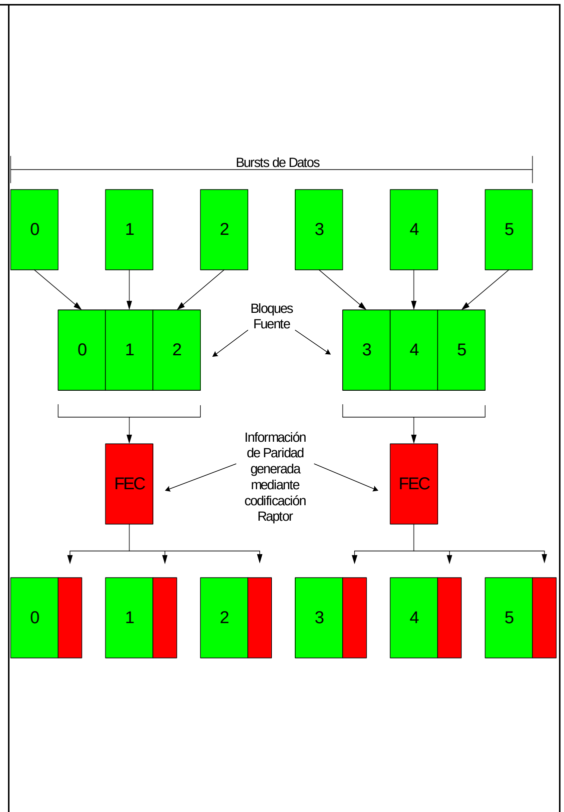

Fig. 5. Codificación en bloque basada en códigos Raptor propuesta para la capa de enlace.

V. DESCRIPCIÓN DE LA CAPA DE ENLACE DVB-SH

Debido a la complejidad que supone la implementación de entrelazadores de capa física de muy larga duración, en DVB-SH se incorporan nuevos mecanismos entrelazadores de nivel de enlace. A pesar de las diferencias en la capa física con respecto a DVB-H, la capa de enlace en DVB-SH soporta tanto la encapsulación MPE (*Multi Protocol Encapsulation*) como la transmisión discontinua empleada en DVB-H. Así mismo también es compatible con la codificación MPE-FEC (*Multi Protocol Encapsulation – Forward Error Correction*) utilizada por DVB-H para mejorar la recepción en situaciones de movimiento. Esta codificación, basada en un código Reed-Solomon (RS), lleva a cabo un entrelazado de la información a nivel de *burst* con el fin de contrarrestar los desvanecimientos rápidos de la señal ocasionados por la movilidad de los usuarios. Sin embargo, dicho entrelazado no posee la suficiente capacidad como para hacer frente a los desvanecimientos de larga duración característicos del canal satélite. Este es el motivo por el que DVB-SH soporta extensiones a la codificación MPE-FEC. Estas técnicas de codificación llevan a cabo un entrelazado capaz de englobar a varios *bursts* de información, y son por tanto mucho más aconsejables para la recepción vía satélite que los mecanismos de codificación y entrelazado intra-*burst* como MPE-FEC. A día de hoy, el mecanismo codificador de la capa de enlace en DVB-SH sigue siendo objeto de estudio y discusión dentro del proceso de estandarización. Actualmente existen dos propuestas: una basada en codificación *multi-burst* RS de ventana deslizante, y otra basada en codificación Raptor de bloque. Ambos mecanismos pueden incorporarse a las dos clases de receptores existentes en DVB-SH, si bien su utilización es más recomendable en los terminales de clase 1. La menor duración de entrelazado de capa física en estos terminales se complementa mediante los mecanismos de capa de enlace para ofrecer una buena protección sin elevar la complejidad hardware en el receptor. Es muy probable que las dos alternativas se incluyan en el estándar para su uso opcional.

Los códigos Raptor son una implementación computacionalmente eficiente de un código fuente (*fountain code*) con unas prestaciones cercanas a las de un código FEC ideal, los cuales pueden ser implementados en software sin la necesidad de un hardware específico. Esto a su vez permite soportar eficientemente un amplio rango de tamaños de ficheros. Los códigos fuente son una clase especial de códigos FEC que pueden generar una cantidad infinita de información de paridad. Fueron originariamente diseñados para transmitir datos eficientemente en canales *multicast* asíncronos. Sin embargo, su utilización en la capa de aplicación para aplicaciones *mulitcast*/*broadcast* en sistemas de comunicaciones inalámbricas ha resultado ser muy ventajosa superando en prestaciones a otros tipos de códigos FEC en términos de fiabilidad, eficiencia espectral y flexibilidad [14].

En la práctica los códigos Raptor son capaces de recuperar la información original si se reciben correctamente un número total de paquetes (de datos o de paridad) ligeramente superior al tamaño del bloque de datos codificado (en media un 1-5% más). La eficacia de un algoritmo codificador depende del tamaño de bloque, es decir, de la cantidad de información que se codifica de manera conjunta. Cuanto mayor es esta cantidad, la capacidad para corregir los errores presentes en la información aumenta. Dado que la codificación Raptor es capaz de trabajar con grandes tamaños de bloque fuente, la protección ofrecida por este algoritmo de codificación es muy elevada. La propuesta de codificación de bloque basada en códigos Raptor para DVB-SH consiste en codificar conjuntamente un número determinado de *bursts*, de forma que todos y cada uno de estos *bursts* han de recibirse antes de poder proceder a su decodificación. Por ello, se introduce un retardo en la recepción de la información que incrementa el tiempo de zapping, por lo que hay un claro compromiso entre robustez de la señal y tiempo de zapping. El esquema de funcionamiento de la propuesta basada en codificación Raptor puede verse en la Figura 5.

La propuesta de ventana deslizante basada en códigos RS consiste en codificar de manera conjunta información perteneciente a *bursts* diferentes mediante un código RS. La información de paridad se transmite junto con los datos en un orden distinto al orden en que se generó. Aunque es posible emplear cualquier código RS, la implementación del algoritmo estándar empleado en la capa física de DVB-T/H, RS (255,191), es bien conocida, lo cual favorece su utilización. Los códigos RS pertenecen a la categoría de códigos perfectos, por lo que tan sólo es necesario recibir sin errores una cantidad de información igual a la original sin ningún *overhead* añadido como es el caso de los códigos Raptor. Sin embargo, al trabajar con un tamaño de bloque más reducido, ofrecen un menor rendimiento que los códigos fuente. Al contrario que la codificación de bloque, donde la codificación y decodificación se lleva a cabo cada cierto número de *bursts*, en la codificación de ventana deslizante, la codificación y decodificación se produce *burst* a *burst*, de manera que la información de paridad se genera de manera continua. El consumo y la carga computacional se mantienen constantes evitando los picos presentes en la codificación de bloque. Además, como no existe ningún retardo en la transmisión de la información, los usuarios de terminales en buenas condiciones de recepción pueden empezar a utilizar la información de inmediato. En el caso de que existan errores en los datos, los terminales sí que han de esperar a recibir la información de paridad correspondiente para intentar corregir los errores.

El rendimiento de las dos propuestas es bastante semejante en la mayoría de los escenarios de simulación, aunque es cierto que la codificación de bloque basada en códigos Raptor obtiene una ligera ventaja. Sin embargo, el mayor inconveniente de la propuesta basada en códigos RS es la necesidad del hardware añadido que requieren los procesos de codificación y decodificación.

VI. FUTURAS LÍNEAS DE INVESTIGACIÓN

Se espera que el estándar DVB-SH se complete en otoño de 2007. El plan de implantación previsto contempla que el despliegue de la parte terrestre se lleve a cabo a lo largo de 2008, mientras que la puesta en marcha del satélite se produzca en el 2009. Durante el verano de 2007, Alcatel-Lucent ha realizado el primer piloto DVB-SH en el sur de Francia, utilizando un helicóptero para emular la señal procedente del satélite. El proceso de validación del estándar tendrá lugar en 2008, en un piloto que se va a realizar en Barcelona dentro del proyecto europeo Celtic B21C (*Broadcast for the 21st Century*) y del proyecto español FURIA (Futura Red Integrada Audiovisual).

Las principales líneas de investigación sobre DVB-SH se pueden agrupar en tres áreas, las cuales se enumeran a continuación:

- Planificación y despliegue de la red terrestre. Para ello es necesario estudiar cómo se puede reforzar la señal principal del satélite con las señales procedentes de la red terrestre (tanto si se emplea OFDM como TDM). Son necesarios modelos de propagación basados en cartografía 3D, que tengan en cuenta las características de la recepción vía satélite, y que sean capaces de reproducir la calidad de la señal recibida según las condiciones de propagación. En muy importante poder evaluar las diferentes contribuciones que alcanzan al receptor, sobre todo en el caso de trabajar bajo una red SFN. En las redes DVB-SH SFN, la combinación de las diferentes contribuciones OFDM que se reciben, provienen tanto de la red terrestre como del satélite, por lo que es necesario estudiar bajo qué condiciones dicha combinación es beneficiosa, e identificar las situaciones en las que puede resultar perjudicial.

- Configuración de la transmisión; configuración óptima (*cross-layer*) de los distintos mecanismos de corrección de errores FEC y de entrelazado en la capa física y la capa de enlace en función del tipo de servicio considerado. Para este tipo de estudios son necesarios modelos de rendimiento de la capa física capaces de emular el comportamiento de la capa física sin tener que simular paso por paso todos los subsistemas que comprende. Dichos modelos de rendimiento tratan de reproducir las propiedades estadísticas de los errores que el canal radio provoca en la información recibida. Mediante una adecuada parametrización, dichos modelos pueden particularizarse para cada una de las diferentes configuraciones de transmisión y entornos de recepción. Estos modelos de rendimiento son también necesarios para desarrollar simuladores dinámicos de sistema capaces de determinar la calidad de servicio experimentada por los usuarios, fundamentales para estudiar el rendimiento de los diferentes entrelazados.
- Gestión de recursos radio. Dentro de este campo se encuadran por ejemplo los algoritmos encargados de decidir qué contenidos transmitir a través del satélite y cuáles a través de la red terrestre. Es importante optimizar el funcionamiento de la red terrestre tanto a la hora de complementar la cobertura ofrecida por el satélite como a la hora de transmitir contenidos adicionales. Las redes 3G pueden transmitir contenidos en aquellas zonas donde la tecnología DVB-SH todavía no se ha puesto en funcionamiento o donde el número de clientes todavía es muy reducido. Las redes 3G también pueden corregir errores en la información recibida en aquellas zonas donde sí existe servicio DVB-SH pero el despliegue de la red terrestre todavía no es el suficiente. Para evaluar todas estas posibilidades, de nuevo es de gran utilidad contar con simuladores dinámicos de sistema que implementen tanto las redes 3G como DVB-SH.

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