Filecasting for Streaming Content Delivery in IP Datacast over DVB-H Systems

David Gómez-Barquero[1]
Mobile Communications Group, iTEAM Research Institute
Polytechnic University of Valencia, Spain
Email: dagobar@iteam.upv.es

Jussi Poikonen
DTV Group, Department of Information Technology
University of Turku, Finland
Email: juspoi@utu.fi

***Abstract*— In this paper we investigate the potential gain that can be obtained in DVB-H by delivering streaming content as a succession of time-constrained files using Application Layer - Forward Error Correction (AL-FEC) for improving the reception for mobile terminals. Compared to the conventional approach with link layer MPE-FEC, this technique allows to increase the robustness of the DVB-H transmission not only as a function of the capacity devoted for error repair (FEC overhead), but also as a function of the number of data bursts coded jointly ("file" or, source data block, size). The main drawback of this approach is an increase of the network latency proportional to the source block size, that can be translated into a larger service access time, and, in the case of mobile TV, a larger zapping time between channels, which is currently seen as a crucial parameter for DVB-H usability. In this paper the performance of the proposed approach is evaluated using vehicular urban DVB-H field measurements. We evaluate the gain compared to MPE-FEC in terms of reduced Erroneous Second Ratio (ESR) of the streaming service as a function of the FEC overhead and the system latency introduced. Moreover, simulations have been performed to quantify feasible link margin gains.**

I. Introduction

Mobile multimedia broadcasting is a fast emerging area with a potential economic and societal impact. The most representative mass mobile multimedia service today is mobile TV, which is expected to become a key application in next generation wireless systems.

The highest potential for providing mass multimedia services is presented by digital broadcast networks specially designed for mobile services, DVB-H (Digital Video Broadcast - Handheld) being the most representative technology in Europe [1]. DVB-H is an extension of the European terrestrial digital TV standard, DVB-T (Digital Video Broadcast - Terrestrial), designed to reach handheld terminals. The main technical features introduced are a discontinuous transmission technique where data is periodically sent in bursts known as *time-slicing*, which reduces the power consumption of terminals, an optional intra-burst Forward Error Correction (FEC) mechanism at the link layer called *MPE-FEC*, which ensures more robust transmissions, especially under mobility and impulsive interference conditions, and the use of IP (Internet Protocol). In contrast to DVB-T, where content is delivered in the form of MPEG-2 packets, DVB-H is IP-based, and all content is delivered in the form of IP data packets.

DVB-H is a transmission standard that specifies the physical and link layers, but it does not define transport protocols, audio and video coding formats, etc. It adopts the same physical layer than DVB-T, implementing both time-slicing and MPE-FEC at the link layer, in such a way that is fully backwards compatible with DVB-T. MPE (Multi Protocol Encapsulation) is the adaptation protocol used to encapsulate DVB-H services into the MPEG-2 DVB-T transport stream.

The end-to-end system is known as *IP Datacast* (IPDC) [2]. The set of IPDC specifications contribute with the higher layer protocols to build a complete end-to-end system, and it specifies, among others, the content delivery protocols, the electronic service guide for service discovery, and mechanisms for service purchase and protection. In IPDC systems, multimedia content is delivered either as a *streaming service* or as a *file delivery service* to the end user, in a time-constrained or unconstrained manner [3].

For streaming services generally a continuous data flow of audio, video and subtitling is transmitted to the terminals using RTP (Real-time Transport Protocol), which is directly consumed by the users. The most representative service is mobile TV. For streaming services DVB-H terminals play the information received in the last data burst until the next burst is received, in such a way that users do not notice the discontinuous transmission. If one burst is lost, the media stream is interrupted until the next burst is received. Occasional data errors may be tolerated if the quality of the audio and video is enough for providing a satisfactory user experience. When MPE-FEC is employed, it is possible to recover from IP packet losses within bursts partially received [1]. However, MPE-FEC cannot recover from complete lost bursts (e.g., when passing through outage areas).

For file delivery services (also called *filecasting*), a finite amount of data is delivered and stored into the terminals as a file using FLUTE (File Delivery over Unidirectional Transport Protocol). Both single file transfers and data carousel sessions are supported. Applications that fall within this category are: video clips, digital newspapers, software download, etc. For file delivery services DVB-H terminals first store the information correctly received in each burst associated to the

[1]The work presented in this paper was performed while the author was a guest researcher at the University of Turku. This work has been partially supported by the Spanish Ministry of Industry, Tourism and Commerce under the project FURIA (Futura Red Integrada Audiovisual).

file until the complete file is available at the receiver, before being accessed by applications. On the contrary to streaming services, filecasting typically requires an error-free reception of the files, as even a single bit error can corrupt the whole file and make it useless for the receiver. In order to increase the robustness of the DVB-H file delivery, an additional FEC mechanism at the application layer using *Raptor coding* [4] has been adopted to be used instead of MPE-FEC.

The key with Application Layer - FEC (*AL-FEC*) is that it can provide *protection across several bursts*, rather than across a single burst as with MPE-FEC [5]. As a consequence, AL-FEC outperforms MPE-FEC when the file is spread over several bursts, being even possible to correct complete lost bursts [6]. The larger the file, the higher the gain obtained with AL-FEC. However their performance is almost identical for small files that fit within a single burst. Note that this is the case for streaming services with MPE-FEC, where each data burst can be seen as a unique file.

As AL-FEC is not currently standardized for streaming services in IPDC, in this paper we investigate the potential gain that can be obtained by delivering streaming content with FLUTE as a succession of time-constrained source data blocks using AL-FEC for improving the mobile reception of streaming services. In this way, the robustness of the DVB-H transmission can be increased not only as a function of the capacity devoted for error repair (i.e., *FEC overhead*), but also as a function of the number of data bursts coded jointly ("file" or, *source data block*, size). In other words, the transmission robustness can be improved keeping the proportion of data employed for error correction (i.e., *coding rate*) by delivering the streaming content as a succession of larger source blocks spanning over more bursts.

The main drawback of this approach is an increase of the network latency proportional to the source block size, that can be translated into a larger service access time, and, in the case of mobile TV, a larger zapping time between channels, which is currently seen as a crucial parameter for DVB-H usability [7]. In this sense, it is interesting to evaluate the gain compared to the conventional streaming delivery approach with MPE-FEC and RTP as a function of the AL-FEC overhead and the network latency introduced. In this paper we evaluate the mentioned trade-off using vehicular urban DVB-H field measurements, and we provide simulation results with feasible link margin gains that can be achieved.

The rest of the paper is organized as follows. First we describe in detail and illustrate the concept of delivering streaming content as a succession of files using AL-FEC in Section II. Then in Section III we explain the performance evaluation methodology, including the performance measures, the measurement set-up, and the field measurements and the simulations performed. In Section IV we provide some illustrative results of the trade-off with AL-FEC between transmission robustness, FEC overhead, and latency, and compare it with the conventional approach with MPE-FEC and provide indicative link margin gains. The paper is concluded in Section V.

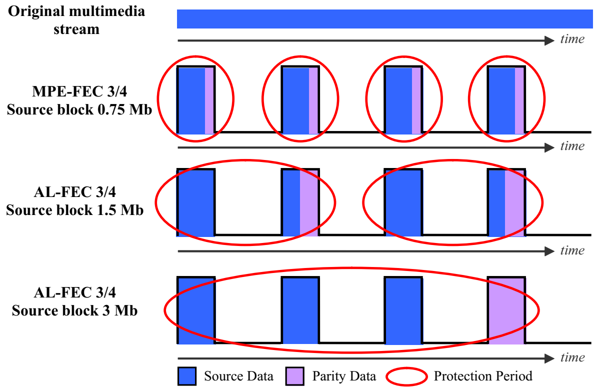

Fig. 1. Example of delivery of 3 Mb streaming content (12 s at 256 kb/s) using MPE-FEC 3/4 and AL-FEC 3/4 with source block size of 1.5 Mb and 3 Mb. Burst size is 1 Mb. Cycle time is 3 s.

II. Application Layer FEC for Improved Mobile Reception of DVB-H Streaming Services

A. Concept

The basic idea of delivering streaming content as a succession of time-constrained files in IPDC over DVB-H systems is to be able to employ application layer FEC for streaming services in a way compliant with the current IPDC specifications using FLUTE as the content delivery protocol.

The key with AL-FEC is that it can provide a *multi-burst protection* of the transmission. This way, thanks to the discontinuous transmission pattern of DVB-H, it is possible to benefit of the spatial diversity introduced by users' mobility. This diversity gain is difficult to quantify in real life, as it depends on several factors, such as the coverage level, the FEC overhead employed, and the statistical correlation between reception conditions of consecutive bursts (which in turn depends on the user velocity, the cycle time between bursts, and the shadowing standard deviation and correlation distance) [5]. However, the larger the number of data bursts coded jointly, the higher the spatial diversity gain, enhancing the coding efficiency to protect against transmission errors. This property can be used to improve the transmission robustness for streaming services by delivering the content as a succession of larger source blocks spanning more data bursts.

As an illustrative example, Fig. 1 shows three different ways of transmitting the same 3 Mb streaming content (which corresponds to approximately 12 seconds at 256 kb/s) using the conventional approach with MPE-FEC, and using AL-FEC with the same coding rate but different source block sizes. The coding rate considered is 3/4, and the burst size 1 Mb, meaning that for both MPE-FEC and AL-FEC the content is divided into 4 bursts with a cycle time of 3 s. For AL-FEC we have considered source block sizes of 1.5 Mb and 3 Mb, which correspond to 2 and 4 bursts coded jointly respectively, and we have assumed a systematic code, where the first packets transmitted are the source packets, and the rest consist of additional parity packets. Note that MPE-FEC can be considered as a particular case of AL-FEC where each burst contains one source data block.

Hereafter we explain first how transmission errors appear in DVB-H, and the error correction capability of MPE-FEC and AL-FEC. Then we will use Fig. 1 to illustrate the improvement in transmission robustness that can be achieved.

Generally speaking, a mobile user will experience two types of errors when moving around a DVB-H network with imperfect coverage of the service area. The terminals will miss the bursts completely due to slow fading (shadowing) when passing through outage areas. Moreover, they may also receive the bursts partially, while being in a covered location due to fast fading or impulse noise. This happens because the physical layer does not provide any time interleaving between consecutive OFDM symbols (maximum duration is 1.12 ms).

The DVB-H standard works with MPEG-2 packets at the physical layer , and IP packets at the link layer. Each burst consists of a number of MPE sections, and each IP packet is encapsulated into a section. At the receiver, the physical layer FEC corrects bit errors within MPEG-2 packets, whereas the link and application layer FEC recover from IP packet losses performing erasure decoding, considering each section either completely received or completely lost based on a CRC (Cyclic Redundancy Check) field. Basically, the error correction capability of MPE-FEC and AL-FEC can be expressed in terms of the maximum number of erroneous sections that can be corrected (assuming the same size for data and parity sections). MPE-FEC can cope with a maximum percentage of erroneous sections per burst equal to the proportion of parity data employed for error repair. For example, the coding rate 3/4 can cope with up to 25% erroneous sections per burst. This is due to the fact that the MPE-FEC scheme consists of a Reed-Solomon (RS) code, which is capable of correcting as many lost packets as the number of parity packets. AL-FEC provides the same section error correction capability that MPE-FEC (assuming an ideal code like RS), but instead than within a single burst, across several bursts.

Coming back to Fig. 1, for the conventional case with MPE-FEC 3/4, if the percentage of erroneous sections in a burst exceeds 25%, the MPE-FEC decoder will fail and the stream will be interrupted. In this case only correctly received data sections containing IP packets will be available for playback. For the cases with AL-FEC 3/4 shown in the figure, it is possible to compensate the same percentage of erroneous sections, 25%, but across 2 and 4 bursts; being possible to compensate for more errors in a particular burst if the other bursts of the source block are received with less than 25% erroneous sections. In particular it would be possible to recover from a burst received with up to 50% erroneous sections and from a complete erroneous burst respectively, if all other bursts of the source block are received without any errors.

The improvement of the FEC coding efficiency as a function of the number of bursts coded jointly is especially evident for low coding rates, where a significant amount of parity data is transmitted. For example, for a coding rate 1/2 and 6 bursts coded jointly, it would be possible to recover from up to 3 complete erroneous bursts (assuming that the other 3 are received without errors).

B. Network Latency and Zapping Time

The main drawback of the proposed delivery technique is an increase of the network latency. As the terminals process the received data as blocks which are treated and decoded independently, they must in general wait to receive all bursts corresponding to the first data block with source and parity information. This latency affects the user experience by delaying the initial reproduction of the services. For most cases, a larger service access time will not be an issue, as the channels with non-real time content will probably be dominant. However, for mobile TV this delay is translated into a larger zapping time between channels, which is currently seen as a crucial parameter for DVB-H usability [7].

The zapping time can be defined as the maximum time (worst case) that a user has to wait to start watching the chosen TV channel in a covered area without transmission errors. It should be noted that the actual zapping time perceived by the users will depend on the transmission errors suffered and the time of switching channels.

With MPE-FEC the zapping time equals to the cycle time between bursts, which depends on the amount of IP data transmitted in the burst and the data rate of the multimedia stream. On the other side, the calculation of the zapping time with AL-FEC is particularly cumbersome, as users may be able to decode the first source block even if they do not receive all its corresponding bursts. This depends on several aspects, such as the source block size, the effective coding rate and the amount of the transmission errors experienced. Assuming that each terminal must buffer the first data block in order to avoid any interruptions of the stream presented for playback, the zapping time with AL-FEC can be considered equal to the latency introduced, which corresponds to the amount of data seconds of the original multimedia stream that are coded jointly into a source block.

C. Application Layer FEC Codes

The standardized AL-FEC code in IPDC over DVB-H systems is systematic Raptor coding [3]. Raptor codes are a computationally efficient implementation of *fountain codes* that achieve close to ideal performance [4]. They can be implemented on software without the need of dedicated hardware, which, in turn, allows to efficiently support a large range of file sizes. At the receivers, only slightly more data than the original source block is needed for reliable reconstruction compared to an ideal code (less than 1% reception overhead in average [6]).

Although the advantages of Raptor codes are evident, as they outperform other FEC solutions in terms of reliability, spectrum efficiency and flexibility, they are subject to intellectual property rights. However, it should not be mixed the benefits obtained by performing multi-burst FEC instead of the legacy intra-burst MPE-FEC in DVB-H (that could be achieved by any code, and its evaluation is the main topic of this paper), and the benefits brought by the Raptor implementation (performance close to an ideal fountain code that allows for a software implementation able to handle large source data blocks).

III. Performance Evaluation

A. *Performance Measures and Assumptions*

In our evaluations we employ the Erroneous Second Ratio (ESR) as the performance indicator to compare the robustness of different transmission configuration schemes. The ESR represents the percentage of erroneous seconds during the service time. We compute it as the percentage of correctly received IP packets, assuming that each IP packet can be played successfully without the need of any previous IP packet. Note that this is not generally true, but it allows an easy and fair comparison between MPE-FEC and AL-FEC.

We consider a 6 minutes streaming service of 256 kb/s, and we assume a constant IP packet size equal to 512 bytes and 512 number of rows per burst for both MPE-FEC and AL-FEC. The number of columns with MPE-FEC depends on the coding rate [1], but for AL-FEC we have considered 255 columns for all cases (constant burst size of 1 Mb).

To account for a practical implementation of an AL-FEC code, a constant 1% reception overhead has been assumed. In the case of Raptor coding, this will generally allows recovery of the source blocks in most of the cases [6].

In the next section we present results on the gain achieved using AL-FEC compared to MPE-FEC in terms of reduced ESR for a given measured/simulated trace as a function of the coding rate employed and the latency introduced. Shown results are average results over the different traces (6 minutes service over 7 minutes trace with a 0.1 s sampling interval), and for a maximum latency equal to one minute.

B. *Field Measurement Set-up*

Field measurements were performed in the DVB-H Single Frequency Network (SFN) test-bed of the University of Turku (Finland) for vehicular urban reception conditions, with speeds ranging approximately from 0 to 60 km/h. The network has two transmitters operating at 610 MHz covering the city center, and it is dimensioned for pedestrian outdoor reception. The DVB-H physical layer transmission mode employed was: FFT size 8K, Guard Interval (GI) 1/4, modulation 16QAM, and coding rate 1/2, which provides a channel data rate of 10 Mb/s at the physical layer.

Fig. 2 shows an example of the data recorded during the measurement campaign. Two DVB-H professional receivers with a common external antenna placed inside a vehicle and a GPS receiver were used to record synchronized reception information (sampling interval 100 ms). The measurements consisted of synchronized RSSI (Received Signal Strength Indicator), terminal position and speed, and MPEG-2 Transport Stream (TS) packet error information at the DVB-H physical layer. The total measurement time was over 2 hours, divided into 18 measurements of 7 minutes each. We considered traces with a wide range of transmission errors to investigate the maximum error correction capability of AL-FEC.

By recording the MPEG-2 TS packet error trace at the physical layer it is possible to reproduce the actual Quality of Service (QoS) experienced by the measuring terminals across

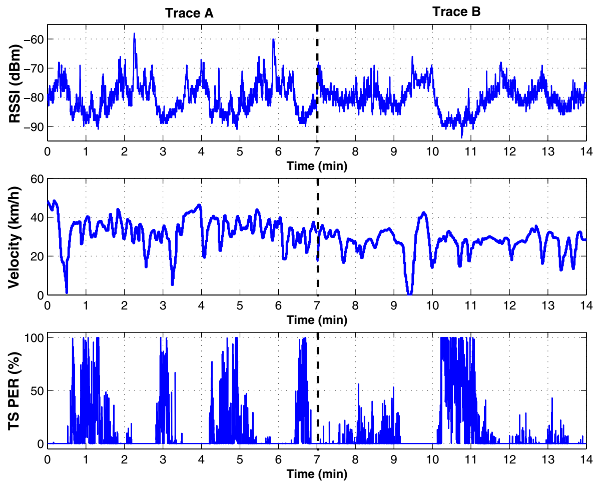

Fig. 2. Example data of vehicular DVB-H field measurement. Average packet error rates for trace A (TS PER 10.5%, IP PER 11.3%), and for trace B (TS PER 10.1%, IP PER 10.7%). IP packet size 512 bytes.

the measured trajectories for any type of service emulating the upper layers, being possible to investigate the effect of different DVB-H transmission configurations at the link and application layer, such as the MPE-FEC and AL-FEC configuration parameters. Note that the measured error traces depend on the physical layer transmission mode employed, and thus the physical layer parameters are fixed.

C. *Link Margin Gain Simulations*

To provide further insight into the potential gain that can be achieved with AL-FEC for streaming services, we perform dynamic simulations to quantify the link margin gain.

The simulation scenario is similar to the one employed in the standardization work of AL-FEC for filecasting services in DVB-H [6]. Simulations parameters include the average Carrier-to-Noise Ratio (CNR), shadowing characteristics (standard deviation σ and correlation distance d_{corr}), Doppler frequency f_d, and RF frequency f_{rf}. We consider the same DVB-H physical layer transmission mode employed during the field measurement campaign, and use the physical layer performance model proposed in [8] developed from laboratory measurements for the TU6 channel model using the same DVB-H receivers employed in the field measurements. In the simulations we compute the link margin gain as the reduction in the average CNR required to achieve a given ESR criteria.

IV. Results and Discussions

A. *Field Measurements Results*

As an example of the results obtained, Fig. 3 shows the ESR as a function of the network latency for the measured traces shown in Fig. 2 for different AL-FEC coding rates. The markers shown correspond to an entire number of bursts coded jointly (the first marker to the left is the reference case with only one burst per source block). It should be pointed out that the latency depends on the number of data bursts coded jointly

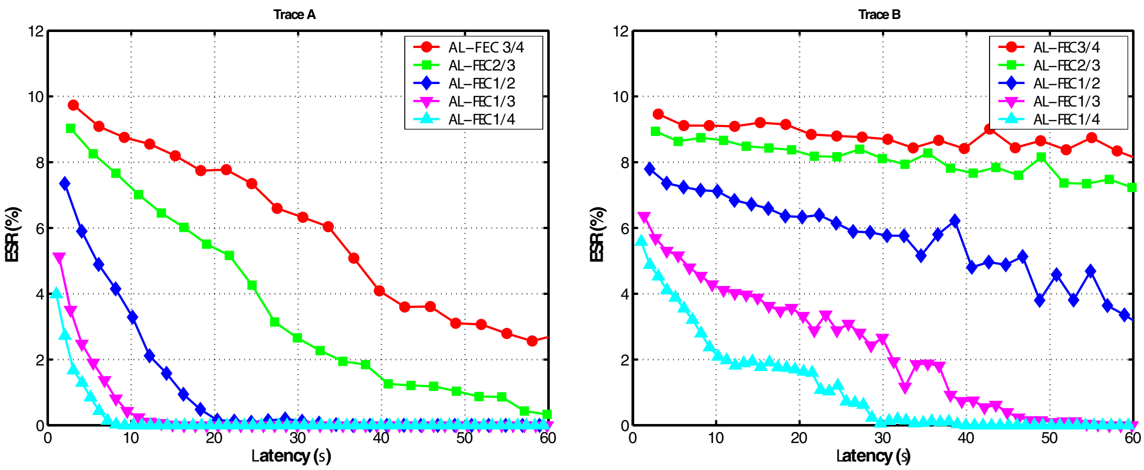

Fig. 3. Erroneous second ratio vs. Latency for the measured traces shown in Fig. 2. Streaming service 6 minutes at 256 kb/s.

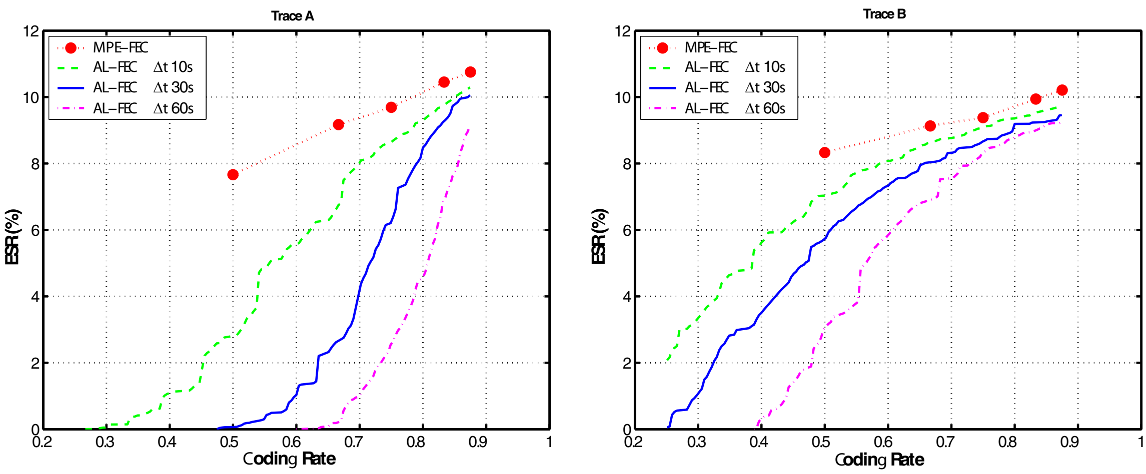

Fig. 4. Erroneous second ratio vs. Coding rate for the measured traces shown in Fig. 2. Streaming service 6 minutes at 256 kb/s.

and the coding rate, as it determines the amount of source data per burst, and hence the source block size.

In the figure we can see the decreasing tendency of the ESR as a function of the latency. However, we can note very different slopes in the two traces despite of having similar error statistics. This is due to the different distribution of the transmission errors over time. In Fig 2 we can see that for trace B most errors are concentrated in one minute, whereas for trace A the errors are basically grouped in four instants more or less uniformly distributed over time. If we take into account that all coding rates considered can potentially cope with the packet error rates of the traces under study[2], the importance of the errors distribution becomes apparent.

Long outage periods where most of the packets are erroneously received imply that very large interleaving durations (latencies) are required to cope with the errors (keeping the coding rate constant). Otherwise, more robust coding rates are necessary. The trade-off between latency and coding rate is illustrated in Fig. 4, where the minimum ESR as a function of the coding rate for latencies of 10, 30 and 60 seconds with AL-FEC is depicted for the measured traces shown in Fig. 2. The case with MPE-FEC it is also included for comparison. We can see that the gain obtained with AL-FEC is very significant, especially for trace A, where the coding efficiency improvement is evident even for 10 s latency. Note that the gain is higher for larger latencies and more robust coding rates.

Therefore, it can be concluded that the error correction capability depends heavily on the errors time distribution. Ideally, if the errors are uniformly distributed, it is possible to cope with IP packet error rates proportional to the coding rate. However in practice the DVB-H channel is very bursty as we have seen in Fig. 2, and long error bursts are common. As a consequence, the coding efficiency is reduced, yielding erroneous IP packets. The more grouped the errors are, the higher is the number of erroneous packets. This degradation of the coding efficiency can be ideally solved increasing the interleaving duration. Indeed if a sufficiently large time interleaving is employed, it is possible to cope with as many erroneous packets as in the best case.

[2]Recall that under our assumptions the error correction capability can be expressed in terms of the maximum percentage of erroneous MPE sections, or IP packets, that can be corrected, and thus a coding rate 3/4 can ideally cope with a 25% IP PER.

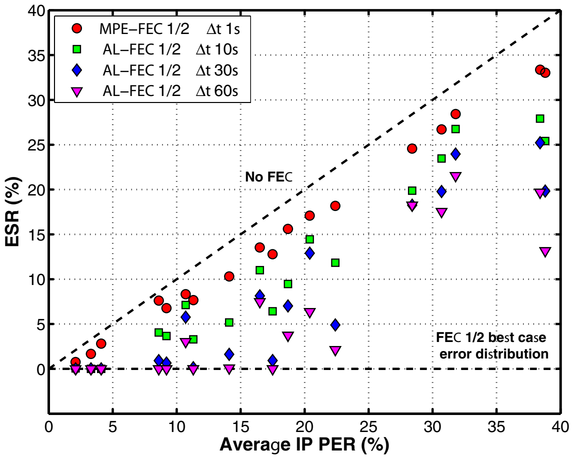

Fig. 5. Erroneous second ratio vs. Average IP PER for all measured trajectories. Streaming service 6 minutes at 256 kb/s.

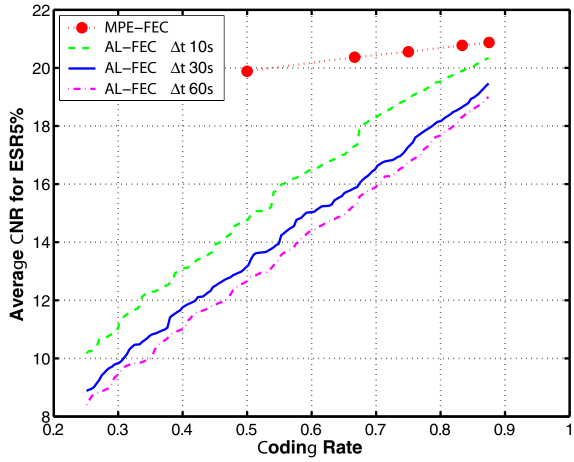

Fig. 6. Average CNR for ESR5% vs. Coding rate. Streaming service 6 minutes at 256 kb/s. f_d 20 Hz, f_{rf} 600 MHz, d_{corr} 20 m, σ 5.5 dB.

Fig 5 compares the ESR achieved with AL-FEC 1/2 for different latencies with MPE-FEC 1/2 as a function of the IP PER for all the trajectories measured. We consider MPE-FEC 1/2 as it provides both the most robust transmission and the minimum latency (1 second in our case) with MPE-FEC. We also depict the reference case without FEC, and the best error distribution case for a coding rate 1/2.

In the figure we can see that the performance of MPE-FEC is relatively close to the case without any FEC at all, and that it can only compensate small error rates (up to 5% IP PER) to provide a good service quality. As it provides protection only within bursts, the interleaving depth equals to the burst duration (typical values 0.2-0.4 s), and it cannot cope efficiently with longer error patterns characteristic of the DVB-H channel in the field. With AL-FEC, the interleaving depth equals to the latency introduced, and it can last up to several minutes. The larger the latency, the lower is the ESR, although the gain compared to MPE-FEC varies from trace to trace as it depends on the error distribution. We can note that a very significant gain is achieved with a 30 s latency, being able to provide good quality for traces with up to 15%-20% IP PER, and that interesting gains are achieved with only 10 s latency.

B. Link Margin Gain Results

Finally, Fig 6 shows simulation results with the required average CNR to achieve an ESR of 5% for MPE-FEC and for AL-FEC for latencies of 10, 30, and 60 s. We can note very significant link margin gains of up to 7 dB for a coding rate 1/2. It should be pointed out that the gain increases for more demanding ESR results, and for example link margin gains for an ESR1% criteria are about 2 dB higher than for ESR5%.

V. Conclusions

In this paper we have evaluated the gain of performing multi-burst forward error correction for DVB-H streaming services compared to the conventional approach with MPE-FEC that provides protection across a single burst. With a multi-burst protection of the transmission it is possible to improve the transmission robustness trading both system capacity and network latency, especially for robust coding rates (1/2 or lower) and for large numbers of bursts coded jointly. This technique can be directly implemented with the current IPDC specifications by delivering the multimedia content as a succession of time-constrained source data blocks with FLUTE using AL-FEC.

Our numerical results with vehicular urban field measurements show that the gain depends not only on the coding rate and the total amount of transmission errors, but also on the errors time distribution. The potential gain is very high, but relatively large latencies are required (30 s or more), although interesting gains are feasible with small latencies of 10 s. Our simulation results show that very significant link margin gains are feasible (around 7 dB for a coding rate 1/2). These gains can be used in existing DVB-H networks to drastically enlarge the area coverage and improve the QoS perceived by the users with the same infrastructure, and to alleviate the required investment in network infrastructure to start providing DVB-H services.

References

[1] G. Faria, J. A. Henriksson, E. Stare, and P. Talmola, "DVB-H: Digital Broadcast Services to Handheld Devices," *Proc. of the IEEE*, vol. 94, no. 1, Jan. 2006, pp. 194-209.

[2] M. Kornfeld and G. May, "DVB-H and IP Datacast - Broadcast to Handheld Devices," *IEEE Trans. on Broadcasting*, vol. 53, no. 1, pp. 161-170, March 2007.

[3] ETSI, TS 102 472 v1.2.1, "*IP Datacast over DVB-H: Content Delivery Protocols*," Dec. 2006.

[4] M. Luby, *et al.*, "Raptor codes for Reliable Download Delivery in Wireless Broadcast Systems," *Proc. IEEE CCNC*, Las Vegas, USA, 2006.

[5] D. Gómez-Barquero and A. Bria, "Forward Error Correction for File Delivery in DVB-H," *Proc. IEEE VTC Spring*, Dublin, Ireland, 2007.

[6] ETSI, TS 102 591 v1.1.1, "*IP Datacast over DVB-H: Content Delivery Protocols Implementation Guidelines*," Oct. 2007.

[7] ETSI, TR 102 377 v1.3.1, "*Digital Video Broadcasting (DVB); DVB-H Implementation Guidelines*," May 2007.

[8] J. Poikonen and J. Paavola, "Error Models for the Transport Stream Packet Channel in the DVB-H Link Layer," *Proc. IEEE ICC*, Istanbul, Turkey, 2006.