EUROPEAN COOPERATION
IN THE FIELD OF SCIENTIFIC
AND TECHNICAL RESEARCH

EURO-COST

COST 2100 TD(08) 538
Trondheim, Norway
2008/June/3-5

SOURCE: Mobile Communications Group,
iTEAM Research Institute,
Polytechnic University of Valencia,
Spain

MPE-iFEC Sliding RS Encoding for Streaming Services in DVB-H Systems

David Gozálvez, David Gómez-Barquero, Narcís Cardona
Mobile Communications Group
iTEAM Research Institute
Polytechnic University of Valencia
Camino de Vera, s/n, Edificio 8G
46022 Valencia
SPAIN
Email: {dagoser, dagobar, ncardona}@iteam.upv.es

MPE-iFEC Sliding RS Encoding for Streaming Services in DVB-H Systems

David Gozálvez, David Gómez-Barquero, and Narcís Cardona

Mobile Communications Group – iTEAM Research Institute, Polytechnic University of Valencia, Spain

Abstract: *In this paper we describe the new DVB-SH (Digital Video Broadcasting – Satellite Services to Handheld Devices) link layer protection mechanism called MPE-iFEC (Multi Protocol Encapsulation – inter Burst Forward Error Protection), and investigate the potential gain that can be achieved for DVB-H (Digital Video Broadcasting – Handheld) streaming services. The MPE-iFEC framework has been recently defined within the DVB-SH standard in order to improve the signal robustness against the long signal blockages present in satellite reception. It is fully compatible with the DVB-H link layer and it can be incorporated in already deployed DVB-H networks. Contrary to the legacy intra-burst link layer FEC MPE-FEC (Multi Protocol Encapsulation –Forward Error Protection), MPE-iFEC is a multi-burst FEC mechanism capable of providing protection across several time-sliced bursts and recovering the loss of complete bursts. However, multi-burst protection implies an increase in the network latency that can seriously degrade the user experience. The same Reed Solomon (RS) algorithm employed in MPE-FEC can be used in MPE-iFEC by performing a sliding coding (SRSE, Sliding RS Encoding). In this paper the operation of SRSE is explained in detail, and its performance for DVB-H streaming services compared to the legacy MPE-FEC is evaluated using DVB-H field measurements and link margin gain simulations.*

I. INTRODUCTION AND MOTIVATION

The most representative technology of digital broadcasting networks in Europe is DVB-H (Digital Video Broadcasting – Handheld), an extension of the European terrestrial digital TV standard, DVB-T (Digital Video Broadcasting – Terrestrial) designed to reach handheld terminals [1]. 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 reutilizes the physical layer of DVB-T, all the content is delivered in the form of IP packets. DVB-H introduces two main features at the link layer; a discontinuous transmission technique where data is periodically sent in bursts known as time-slicing, which reduces the power consumption of terminals, and an optional protection mechanism called MPE-FEC (Multi Protocol Encapsulation – Forward Error Correction), which provides a more robust transmission in the presence of mobility and impulse interference.

MPE-FEC is an intra-burst link layer protection mechanism in which the information is encoded burst by burst with a Reed Solomon (RS) (255,191) encoder. It was mainly introduced to cope with the fast fading in covered areas where static reception is possible, increasing the robustness of reception for mobile terminals such that the signal strength requirement becomes practically independent of the speed. MPE-FEC is a mandatory feature of DVB-H that requires the implementation of dedicated hardware in the user terminals. However, some commercial networks currently do not employ MPE-FEC as the protection offered does not compensate the overhead introduced by MPE-FEC in the form of parity data. Instead of relying in MPE-FEC, the networks operators have to deploy new transmitters to provide enough coverage in the service area.

MPE-FEC is only capable of correcting partial errors and can not repair the complete loss of time sliced bursts. In order to recover entire bursts, it is necessary to encode information from different bursts in a jointly manner. This technique is called multi-burst encoding and is capable of achieving a protection that extends across several bursts. Multi-burst encoding has been standardized in DVB-H for the provision of file delivery services by means of AL-FEC (Application Layer – Forward Error Correction) [2]. However, AL-FEC can also bring an important improvement in the provision of streaming services as it has been shown in [3], where the idea of delivering streaming content as a succession of time-constrained files is developed.

Multi-burst encoding increases the robustness of streaming services at the expense of additional latency in the system. The receiver must wait until all the information encoded together has been received before decoding it and passing it to the upper layers. This situation increases the zapping time, which normally does not represent any constraint for file delivery services, but it is usually considered as one crucial parameter for DVB-H usability [4] in the case of streaming services. In order to not compromise the user experience it is necessary to establish an adequate trade-off between the protection provided by multi-burst encoding and the latency introduced in the system.

The new DVB-SH (Digital Video Broadcasting – Satellite Services to Handheld Devices) standard designed for the provision of mobile TV services through hybrid terrestrial-satellite networks [5], defines a new link layer protection mechanism called MPE-iFEC (Multi Protocol Encapsulation – inter burst Forward Error Correction) [6]. Although the DVB-SH standard supports both MPE-FEC and MPE-iFEC, the simultaneous operation of both mechanisms is not allowed, and the usage of MPE-iFEC is recommended. MPE-iFEC defines a multi-burst generic framework which is fully compatible with the link layer of DVB-H and thus can be employed in DVB-H networks to improve the reception of streaming services. Currently deployed DVB-H networks can be easily updated in order to provide MPE-iFEC protection to the transmitted services. However, current DVB-H terminals can not support MPE-iFEC operation as they do not count with the necessary hardware to handle the memory requirements. To date, two MPE-iFEC configurations have been presented for DVB-SH networks; one based on Raptor codes and one based on Reed Solomon codes. At present, only the alternative based on Reed Solomon codes called Sliding RS Encoding (SRSE) has been incorporated into the DVB-SH implementation guidelines.

The SRSE mechanism employs a sliding window approach in order to interleave the information from different bursts and generate the parity data by means of the same RS (255,191) encoding algorithm as MPE-FEC. Because of this, the SRSE allows the reuse of several components of the MPE-FEC implementation, which facilitates the incorporation of MPE-iFEC into the user terminals. Although MPE-iFEC was originally designed for satellite transmissions, the implementation of the SRSE in DVB-H networks can increase the robustness of streaming services, improving the user experience and extending the service area without the need of new network infrastructure.

The rest of the paper is organized as follows. First, the advantages of multi-burst protection over intra-bursts mechanisms like MPE-FEC will be explained in Section II. In Section III the operation of the SRSE is described in detail. Section IV is dedicated to the explanation of the performance evaluation methodology. In Section V, the results of the SRSE performance and the gain achieved over MPE-FEC for different configurations are presented.

II. Multi-burst Encoding for DVB-H Streaming Services

In DVB-H, the link layer receives the IP information from upper layers in the form of datagram bursts. The legacy MPE-FEC encodes the IP information from each datagram burst separately by means of a Reed Solomon (255,191) encoding algorithm. The IP information and the generated parity data are transmitted together in the same time sliced burst in the form of information and parity sections. At the terminals, the received sections are considered either completely erroneous or completely correct based on a CRC (Cyclic Redundancy Check) field carried by every section such that the MPE-FEC decoder sees a virtual erasure channel. Reed Solomon codes are perfect codes in the sense that they are capable of correcting as many lost packets as the number of parity packets transmitted. Assuming the same size for information and parity sections, the amount of sections that can be corrected in MPE-FEC for each time sliced burst is equal to the number of parity sections transmitted. Therefore, it is not possible for MPE-FEC to correct time sliced bursts in which all the information and parity sections are lost.

The physical layer of DVB-H is characterized by the rapid transition between perfect and null reception of information in terms of Carrier to Noise Ratio (CNR). When passing through shadowed areas, the received terminal experiments signal outages that can corrupt the majority or even all the information from several time sliced bursts. MPE-FEC can not cope with the loss of entire time sliced bursts, and becomes insufficient in outage situations.

Contrary to intra-burst mechanisms as MPE-FEC, multi-burst techniques compute the parity data across several successive datagram bursts instead of across one single datagram burst. Even if an entire time sliced burst is lost, the information can be retrieved if enough IP information and parity data are received in consecutive time sliced burst. The protection achieved by multi-burst encoding spans across a number of time sliced bursts which is usually refer to as interleaving depth. Multi-burst encoding takes advantage of the time-spatial diversity derived from the user mobility and the time slicing transmission of DVB-H. The diversity gain that can be achieved by multi-burst encoding techniques increases with the interleaving depth, although it heavily depends on the error distribution over time (which in turn depends on the user velocity and shadowing characteristics). As an example, if multi-burst protection is used to encode 10 bursts in a jointly manner and a code rate of 1/2 is employed, it would be capable of correcting the entire loss of 5 bursts of information provided that the rest of the IP information and all the parity data are received without errors.

The set of IP information that is encoded together constitutes one source block. In order to encode information from different bursts in a jointly manner, portions of information can be interleaved into one small source block, or several entire bursts can be encoded together in one source block of greater size. Multi-bursts mechanisms with greater source blocks achieve a better performance as they have higher probabilities of decoding the erroneous information contained in the source blocks. Reed Solomon codes have a limitation in the amount of information that can encode and so, MPE-iFEC configurations based on encoding algorithms capable of encoding larger source blocks can potentially achieve better performance than SRSE. However, as the SRSE scheme decodes a single datagram burst each time a time sliced burst is received, it avoids the computational peaks of other mechanisms where bigger matrixes are decoded every certain number of time sliced bursts.

III. MPE-iFEC Sliding RS Encoding

A. SRSE Operation

The MPE-iFEC sliding RS encoding scheme incorporated by DVB-SH employs the same RS (255,191) encoder than MPE-FEC. However, a sliding window approach is used to distribute the columns of each datagram burst between several encoding matrixes. This way, IP information from different datagram bursts will be encoded together in the same encoding matrix. The process is depicted in Fig. 1. At first, the sliding window encloses several encoding matrixes. Each time a new datagram burst is received from upper layers, the sliding window advances one matrix forward and the information of the datagram burst is distributed between the encoding matrixes now enclosed by the sliding window. Every time this process takes place, one encoding matrix is filled with IP information, from which the parity data will be generated by means of the RS (255,191) coder. Each datagram burst will be transmitted in one time sliced burst along with parity data. Although the datagram bursts are transmitted in the same order as they were received from upper layers, the parity data generated in each encoding matrix is interleaved and transmitted across several time sliced bursts. The number of time sliced bursts in which the IP information and parity data from one encoding matrix are interleaved and transmitted corresponds to the interleaving depth.

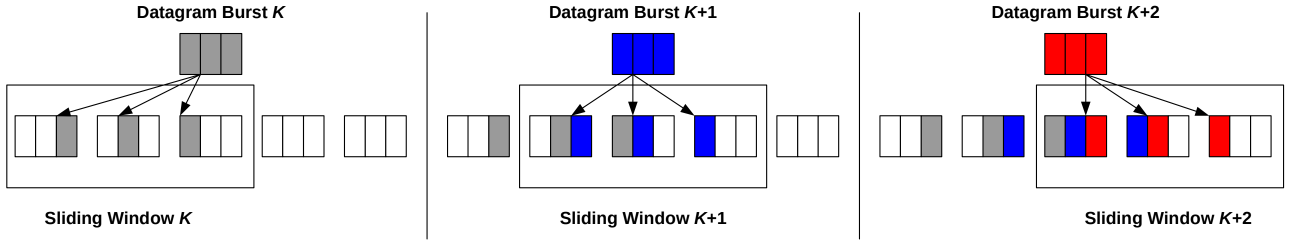

Fig. 1: Interleaving of the datagram bursts among the encoding matrixes performed by the SRSE.

The decoding process is basically the reverse of the encoding operation. A sliding window is used in this case to distribute the IP information and the parity data across their corresponding encoding matrixes. Each time a time sliced burst is received, one encoding matrix is decoded and the IP information is passed to upper layers.

There are three main parameters that regulate the operation of the SRSE:

- The Encoding Parallelization (*B*) is the number of encoding matrixes enclosed by the sliding window, or what is the same, the number of encoding matrixes along which the information from each datagram burst is distributed. Due to the fact that in the SRSE scheme one encoding matrix can contain an amount of IP information equal to one datagram burst, each encoding matrix is filled with portions of information from *B* different datagram bursts. As seen if Fig. 1, each datagram burst will be interleaved over *B* encoding matrixes with the previous *B*-1 bursts and the next *B*-1 bursts. Higher values of *B* increment the duration of the interleaving performed by SRSE along with the latency added to the system, as the receiver must wait until the arrival of *B* bursts in order to retrieve the IP information of one encoding matrix.
- The FEC Spreading (*S*) is the number of time sliced bursts along which the parity information generated from one encoding matrix will be interleaved in transmission. Therefore, each time sliced burst will carry one datagram burst and parity data from *S* encoding matrixes. Similarly to the *B* parameter, higher values of *S* allow better protection but also increments the latency added to the system.
- The Transmission Delay (*D*) refers to the time between when a datagram burst is received from upper layers and when is finally transmitted in one time sliced burst. Normally, the datagram bursts received from upper layers are transmitted in a time sliced burst without further delay. The parity data generated from the encoding matrixes will be transmitted in the subsequent time sliced bursts. This way, the IP information and the parity data of one encoding matrix are transmitted in different time sliced bursts. By delaying the transmission of the datagram bursts, it is possible to transmit the datagram bursts and the corresponding parity data in the same time sliced bursts, which effectively reduces the interleaving depth without affecting the values of *B* and *S*.

As it has been explained, the values of *B* and *S* regulate the interleaving performed to the IP information and the parity data respectively. Although many combinations between these two parameters are possible, in [6], it is recommended to set *B* and *S* proportionally to the amount of information and parity present in each time sliced burst in order to achieve the best performance. Although the multi-burst encoding performed by SRSE is capable of repairing complete burst losses, it is possible that partial burst losses can not be repaired. The SRSE employs the same RS encoding algorithm as MPE-FEC, and each one of the encoding matrixes can only decode one datagram burst. If enough information from one encoding matrix gets corrupted along the time sliced bursts, it will not be possible to repair the information even if the information from the other matrixes are received without errors.

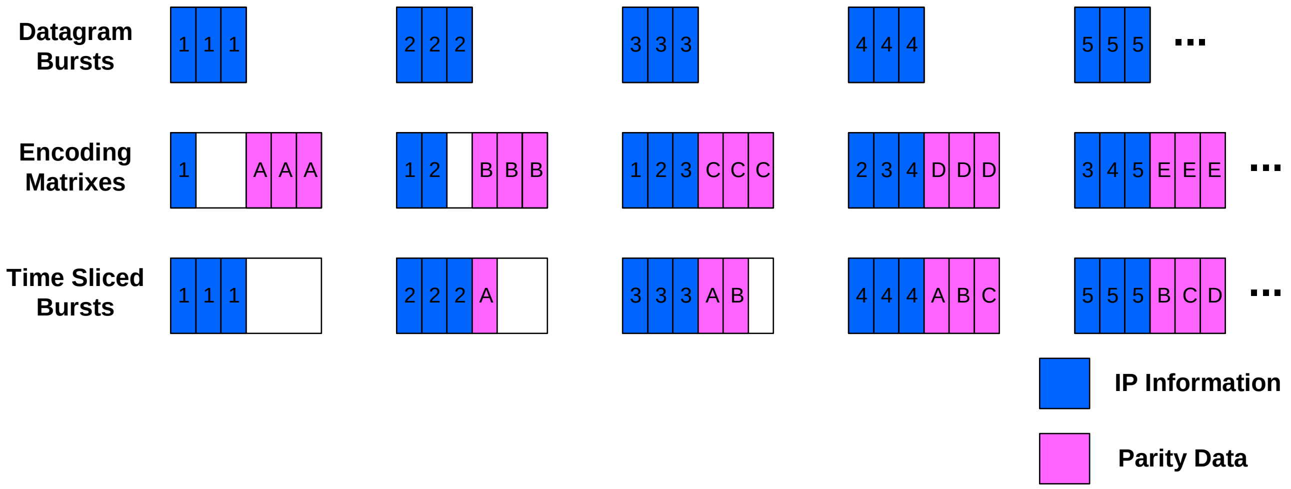

Fig. 2: Encoding and transmission operation of the SRSE configured to $B = 3$, $S = 3$ and $D = 0$.

B. Memory and Latency issues

In order to successfully decode each encoding matrix in MPE-iFEC, the terminal must store all the bursts along which the IP information and parity data from the matrix have been distributed. The memory requirements derived from the MPE-iFEC usage are proportional to the interleaving depth, meaning that if the interleaving depth has been configured to 10 bursts, the terminal must incorporate enough memory to store 10 bursts containing IP information and parity data. The interleaving depth performed by SRSE can be computed as it is shown in equation 1:

$$Interleaving\_Depth = B + \max(0, S - D) + \max(0, D - B) \qquad (1)$$

The multi-burst encoding applied by MPE-iFEC brings also an increment in the zapping time. The zapping time is defined as the amount of time that passes between when the user switches to a new channel and when the service from that channel is being displayed in the terminal. MPE-FEC only needs the information from one time sliced burst in order to perform the decoding process. When the user switches to a new channel, the terminal only has to wait until it has received the first burst containing information of the new service. Once the burst has been received, it will be decoded and the correct sections will be forwarded to upper layers. Assuming that each burst contains a random access acquisition point and the buffer size of the codec is well configured, the information will be reproduced without further delay. The worst case scenario in MPE-FEC takes place when the user has switched to the new channel just after the start of a burst from the new service. In that case it must wait during a cycle time until the arrival of the next burst. The best case scenario takes place when the user switches to the new channel just before the arrival of a burst of the new service, in which case the terminal can begin the reproduction of the service after receiving and decoding the information of the burst.

When using MPE-iFEC, the terminal must wait until the arrival of all the time sliced bursts carrying information from the same encoding matrix, including IP information and parity data. As it can be seen in equation 1, the transmission delay parameter D can be configured to reduce the interleaving depth while maintaining the B and S values. When D is set to 0, the terminal must wait a total of B+S bursts to retrieve all the IP information and the parity data from the encoding matrixes and achieve full protection. However, if D is set to a value between B and S, the parity data is now transmitted along the IP information from which was generated. This way, both the memory requirements and the zapping time are reduced to B bursts. However, since the IP information is being interleaved with the parity data from the same encoding matrix in the same time sliced bursts, the performance of the MPE-iFEC mechanism is compromised.

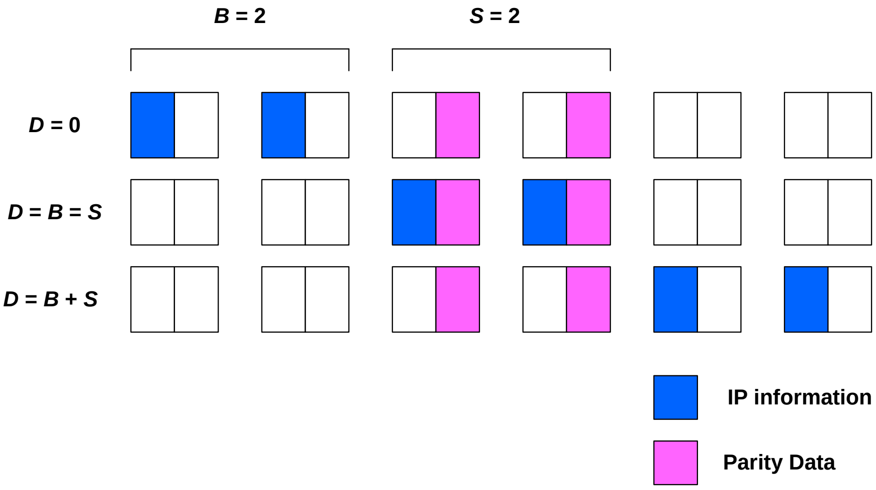

Fig. 3: Distribution of the IP information and parity data along the time sliced bursts for different *D* values.

If *D* is set beyond *B*+*S*, the parity is again not interleaved with the IP information. Although the zapping time is maintained to *B* bursts, the memory requirements and end to end latency are incremented. Hence, it is necessary to configure the transmission delay in an adequate manner in order to decrease the zapping time and memory requirements without compromising the protection offered by the system. Although high zapping times usually affect the user experience, they can be compensated by different techniques such as slowing the service playback [7] or employing a zapping channel [4].

Several fast zapping techniques can be also used in order to reduce the zapping time. If the terminal is in good reception conditions and the bursts are received without errors, it is not necessary to receive the rest of the IP information from the same encoding matrix or the parity data. Because of the fact that the IP information is transmitted in time slicing bursts without being interleaved, the terminals can reproduce the new channel from the moment it receives the first burst containing information of the service without performing the decoding process. The main drawback of this approach is the fact that when an error occurs, the terminal must stop the reproduction of the service in order to buffer all the information from the encoding matrix. This interruption would only take place the first time an error is encountered in the received information, as from that moment the terminal would have buffered all the needed data for decoding.

IV. Performance Evaluation

The performance of the SRSE has been evaluated by means of field measurements. Dynamic simulations have been also performed in order to quantify the potential link margin gain over MPE-FEC. In the evaluations, a 6 minute streaming service with a constant data rate of 384 kb/s has been considered. A constant IP packet size of 512 bytes and 512 rows per burst have been assumed for both MPE-FEC and SRSE, while the number of IP information and parity columns has been set accordingly to the coding rate. The results have been averaged over 1 minute in order to achieve a statistical smoothing of the results. In streaming services, the actual Quality of Service (QoS) perceived by the user is not only affected by the amount of errors in the received information but also by their time distribution. The Erroneous Second Rate (ESR) and the ESR5(20) quality indicators take into account the distribution of the errors over time, and are being used in the implementation guidelines of DVB-SH to evaluate the performance of physical and link layer protection mechanisms.

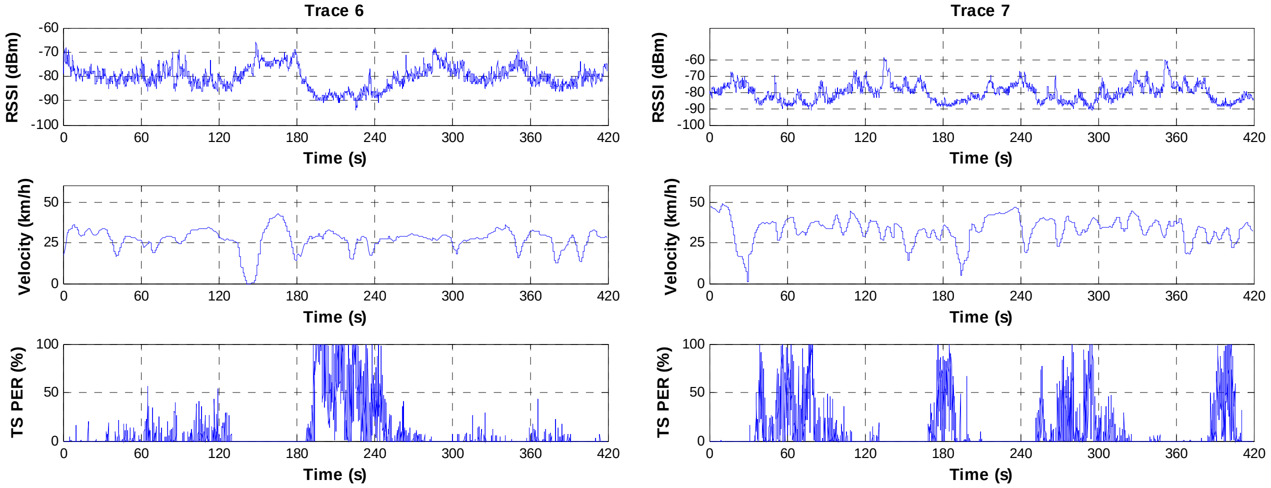

Fig. 4: Example data of vehicular DVB-H field measurement.

Table 1: Average MPEG-2 TS packet error rate, MPE section error rate and error burst length of the measured traces analyzed in this paper.

Trace ID	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
TS PER	1.8%	3.0%	3.9%	7.9%	8.7%	10.1%	10.5%	13.1%	15.9%
MPE SER	2.1%	3.3%	4.1%	8.6%	9.2%	10.7%	11.3%	14.1%	16.5%
AEBL	14.0	20.1	37.2	27.4	33.1	39.9	28.9	28.3	56.0
Trace ID	**10**	**11**	**12**	**13**	**14**	**15**	**16**	**17**	**18**
TS PER	16.3%	17.8%	19.6%	21.3%	27.1%	29.9%	31.1%	36.3%	36.7%
MPE SER	17.5%	18.7%	20.4%	22.4%	28.4%	30.7%	31.8%	38.4%	38.8%
AEBL	28.5	38.1	55.4	40.0	43.6	73.8	96.0	37.9	35.7

However, for the sake of simplicity, the IP Packet Error Ratio (IP PER), which is the percentage of erroneously retrieved IP packets, has been employed in this paper. The IP PER indicator only accounts for the total amount of erroneous information that could not be repaired after the decoding process, and represents an easy, objective and fair indicator to establish comparisons between the correction capabilities of MPE-FEC and the SRSE.

A. Field Measurement Setup

By recording the MPEG-2 TS packet error trace at the physical layer and emulating the upper layers it is possible to reproduce the actual QoS experienced by the measuring terminal across the measured trajectories for any type of service. It is possible to evaluate and compare the performance of MPE-FEC and the SRSE for different parameters and configurations. However, the measured error traces are determined by the physical transmission mode employed during the measurements and so, the physical parameters are fixed.

The TS packet error traces were obtained in the test network of the University of Turku (Finland). The network consists on two transmitters at 610 MHz that were configured to operate in single frequency mode. Although the network was dimensioned for providing service to pedestrian users, the measurements were taken in vehicular conditions. The DVB-H 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. The measurement system was formed by two professional receivers with a common external antenna and a GPS receiver that registered synchronized reception information with a sampling interval of 100 ms. The measurements consisted of synchronized RSSI (Received Signal Strength Indicator), terminal position and speed, and MPEG-2 TS packet error information at the DVB-H physical layer. TS packet error traces are possible to record because of the MPEG-2 TS packet header called TEI (Transport Error Indicator), which indicates if each packet is erroneous or not when passed to the link layer. An example of the data recorded during the filed measurement campaign can be observed in Fig. 4.

18 different measurements of 7 minutes each were recorded from the field measurement campaign performed in Turku. The main statistics of the 18 TS packet error traces including the TS Packet Error Rate (TS PER), the MPE Section Error Rate (MPE SER) and the Average Error Burst Length (AEBL), are shown in Table 1. As it can be seen in this table, the 18 traces recorded have a wide range of transmission errors in order to evaluate the performance of the SRSE in a great variety of reception conditions.

A. Link Margin Gain Simulations

Dynamic simulations have been performed in order to investigate the potential gain that can be achieved by SRSE over MPE-FEC in the link budget. Link margin gains can be easily interpreted not only as an improvement in the reception quality for high data rates, but also as a coverage extension that allows the network to reach further areas. Because of this, link margin gains are of paramount importance for network operators as they allow a reduction of the high costs involved in the deployment of DVB-H networks.

The simulation scenario is similar to the one employed in the standardization work of AL-FEC for filecasting services. Basically, it assumes a user moving at a constant velocity, defined by the Doppler frequency f_d and RF frequency f_{RF}, across a lognormal CNR map defined by its average value and its standard deviation σ and correlation distance d_{corr} (resembling shadowing characteristics). The same physical layer transmission mode used during the field measurement campaign has been considered, and the physical layer performance model proposed in [8] has been employed. This performance model was developed from laboratory measurements for the TU6 channel model using the same DVB-H receivers employed in the filed measurements. This channel model was proven to be representative for DVB-H mobile reception for Doppler frequencies above 10 Hz (i.e., vehicular reception). The link margin gain has been computed in the simulations as the reduction in the CNR requirement to achieve an IP PER criteria of 1 %.

V. Results and Discussions

A. Field Measurements Results

In Fig. 5, the IP PER obtained by the SRSE scheme for the traces shown in Fig. 4 is represented as a function of the latency for the five coding rates defined for MPE-FEC usage. The latency can be computed as the result of multiplying the interleaving depth by the cycle time. In the two traces, a decreasing tendency of the IP PER as a function of the latency, is observed. However, the performance of SRSE is significantly higher in trace 7, where all five coding rates experiment a considerable gain as the latency is incremented. On the contrary, although all coding rates benefit in some way from higher interleaving depths in trace 6, the improvement brought by the added latency is much more discrete. The cause behind the difference in performance between the two traces lies in the error distribution. Both traces have a MPE section rate around 11 %; however, the errors are clearly more concatenated in trace 6 as can be seen in Fig. 4. In this case, the majority of the errors are grouped around one long outage period situated between 180 and 240 s.

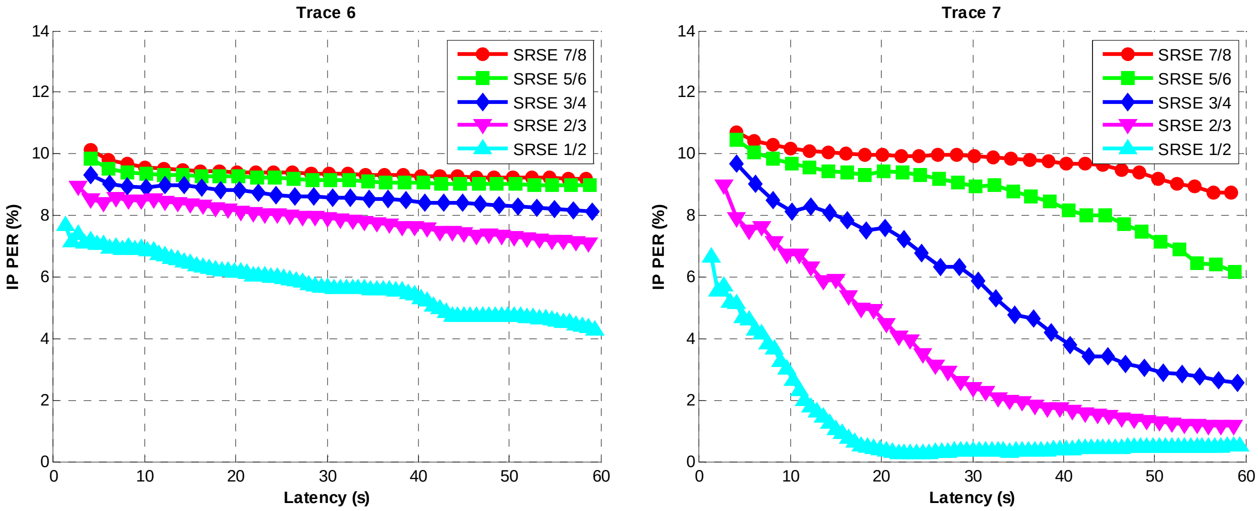

Fig. 5: IP packet error rate vs. Latency for the measured vehicular urban traces shown in Fig.2. Streaming service 6 minutes at 384 kb/s.

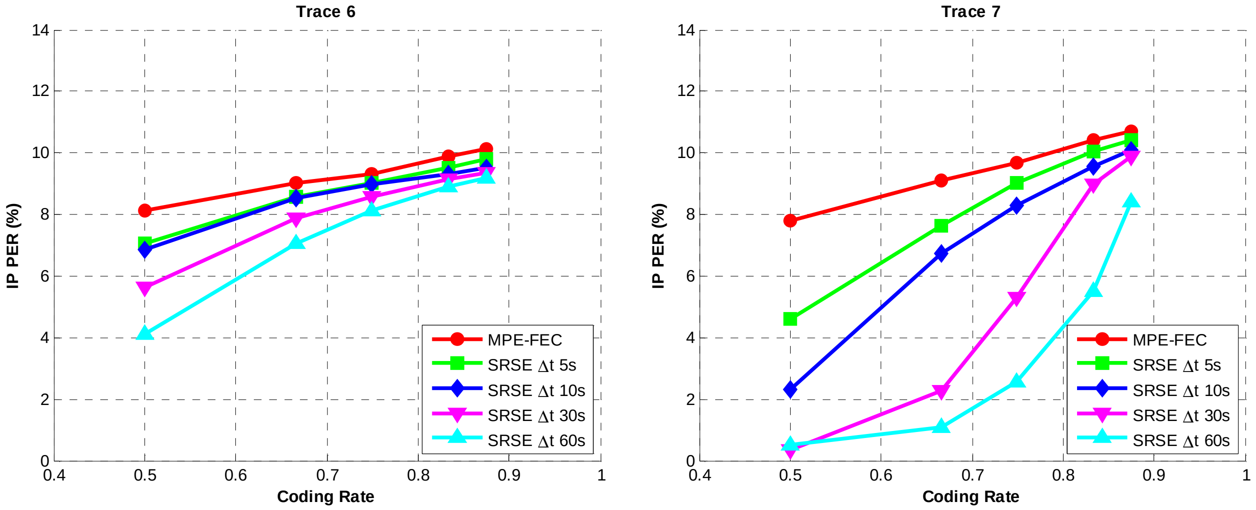

Fig. 6: IP packet error rate vs. Coding rate for the measured vehicular urban traces shown in Fig.2. Streaming service 6 minutes at 384 kb/s.

Trace 7 on the other hand, has a more uniform error distribution over time in the form of four error bursts. This statement is reinforced by the AEBL values shown in Table 1, which in the case of trace 6, usually denotes a concentrated error distribution. Longer error bursts occasioned by extended outage periods require greater interleaving depths to spread the errors across the encoding matrixes. The number of errors in each encoding matrix must be low enough with respect to the coding rate in order to repair the erroneous information. Larger interleaving values can distribute the errors across a higher number of different encoding matrixes and increase the probabilities of recovering the lost information.

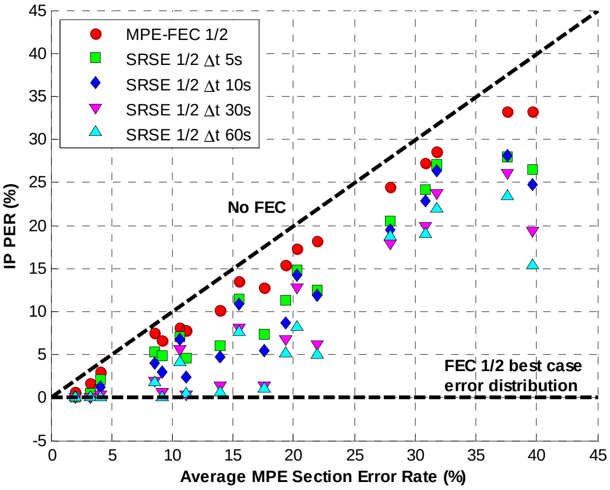

Fig. 7: IP packet error rate vs. Average MPE section error rate for all measured vehicular urban trajectories. Streaming service 6 minutes at 384 kb/s.

The improvement achieved by SRSE over the legacy MPE-FEC in terms of IP PER is better appreciated in Fig. 6, where both mechanisms are compared for five different values of latency and coding rate. Again, the difference in the performance achieved by the SRSE in the two traces is evident. In spite of having a similar amount of errors, the improvement over MPE-FEC is much more important in trace 7. If a coding rate of 1/2 is employed in trace 6, it is necessary a latency of about 60 s in order to reach an IP PER value of 4 %. On the contrary, if the same coding rate is used in trace 7, an IP PER value close to 4 % can be achieved with only 5 s of latency.

Fig. 7 compares the results obtained by MPE-FEC 1/2 and SRSE 1/2 for different values of latency as a function of the average MPE section error rate in all 18 measures trajectories. A coding rate of 1/2 has been employed as it is the most robust configuration defined for MPE-FEC. Also, the results that would have been obtained by a FEC 1/2 with a completely uniform error distribution and a configuration with no FEC are represented. The percentage of errors in the 18 traces is below the 50 % and so, a coding rate of 1/2 should be low enough to repair all the errors if these were to be distributed uniformly along time. On the other hand, if no FEC protection is applied, it is not possible to recover any lost portion of information, and the obtained IP PER is equal to the section error rate. MPE-FEC is not capable to cope with the erroneous information contained in the traces, and its results are very close to the curve with no FEC at all. On the other hand, the SRSE scheme reduces the IP PER considerably in all 18 traces, and in some cases is capable of repairing all the errors if the latency is high enough. The improvement accomplished by SRSE with respect to MPE-FEC varies from trace to trace, as it depends on the error distribution. Again, this statement can be corroborated with the AEBL results of Table1. Traces with lower values of AEBL achieve better performance and require lower latencies in order to approximate the best case error distribution. This is the case of trace 18 which, in spite of having the higher percentage of errors, achieve lower IP PER values than traces 17, 16, 15 and 14 if the latency is configured to 60 s.

B. Link Margin Gain Results

In the link margin simulations, a reference scenario has been defined with the following parameters: f_d 20 Hz, σ 5.5 dB, d_{corr} 20 m and f_{RF} 600 MHz (i.e., user velocity 36 km/h). Fig. 8 shows the IP PER achieved in this scenario by MPE-FEC and the SRSE for a wide range of CNR values. On the other hand, four different latencies have been investigated in the SRSE scheme (5, 10, 30 and 60 s).

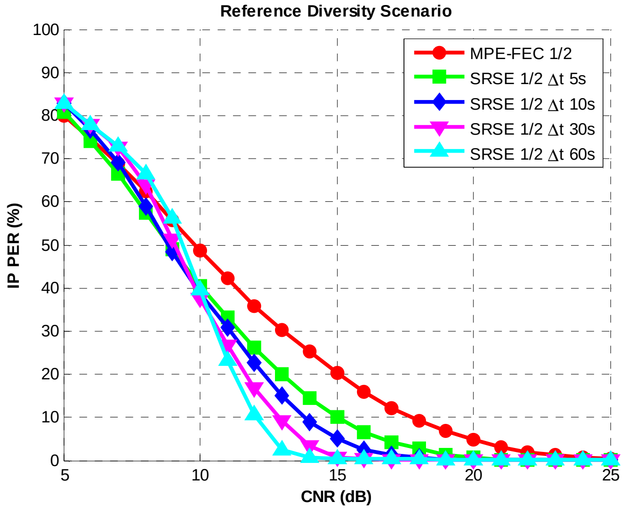

Fig. 8: IP packet error rate vs. CNR. Streaming service 6 minutes at 384 kb/s. Reference diversity scenario: f_d 20 Hz, σ 5.5 dB, d_{corr} 20 m, f_{RF} 600 MHz.

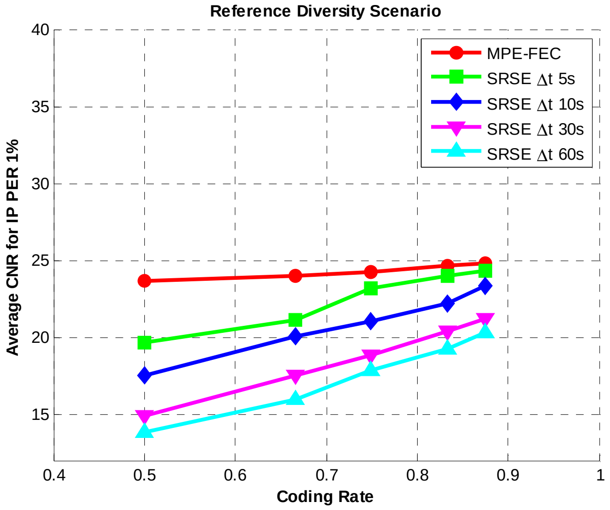

Fig. 9: Average CNR for IP PER 1% vs. Coding Rate. Streaming service 6 minutes at 384 kb/s. Reference diversity scenario: f_d 20 Hz, σ 5.5 dB, d_{corr} 20 m, f_{RF} 600 MHz.

As it can be seen, the multi-burst encoding performed by SRSE achieves considerable better results than MPE-FEC even for low latencies values. However, the effect of increasing the interleaving is not the same in all the CNR range. Increasing the latency in the SRSE actually degrades the performance for low values of CNR instead of improving the protection. I

In Fig. 9, the average CNR required to achieve an IP PER value of 1 % is represented for both MPE-FEC and SRSE in the reference diversity scenario. The results have been obtained for four different SRSE latencies. As it can be seen in the figure, SRSE achieves important link margin gains that are greater for larger latencies and more robust coding rates. Smaller latencies need lower coding rates in order to significantly improve the link margin over MPE-FEC. In the SRSE, a latency of only 5 seconds does not bring a considerable advantage with coding rates over 3/4; however, it can achieve a reduction of about 4 dB in CNR when the coding rate is set to 1/2.

Fig 10 shows the same results but in this case for two different scenarios. A high diversity scenario (f_d 40 Hz, σ 8 dB, d_{corr} 20 m and f_{RF} 600 MHz) and a low diversity scenario (f_d 10 Hz, σ 5.5 dB, d_{corr} 100 m and f_{RF} 600 MHz) have been defined to test the performance of the SRSE. In the high diversity scenario the user is moving at 72 km/h in an environment where the shadowing is characterized by a higher standard deviation. In the low diversity scenario the user is moving at 18 km/h and the shadowing presents a higher correlation distance. This scenario has an f_d/d_{corr} ratio of 1/10, which may correspond with pedestrian reception conditions. As it was expected, the multi-burst encoding performed by SRSE benefits of the added diversity and offers bigger improvements in the high diversity scenario. On the other hand greater latencies are needed in the low diversity scenario, as interleaving durations of 5 and 10 seconds are not capable of significantly improving the link margin even for the lower code rates.

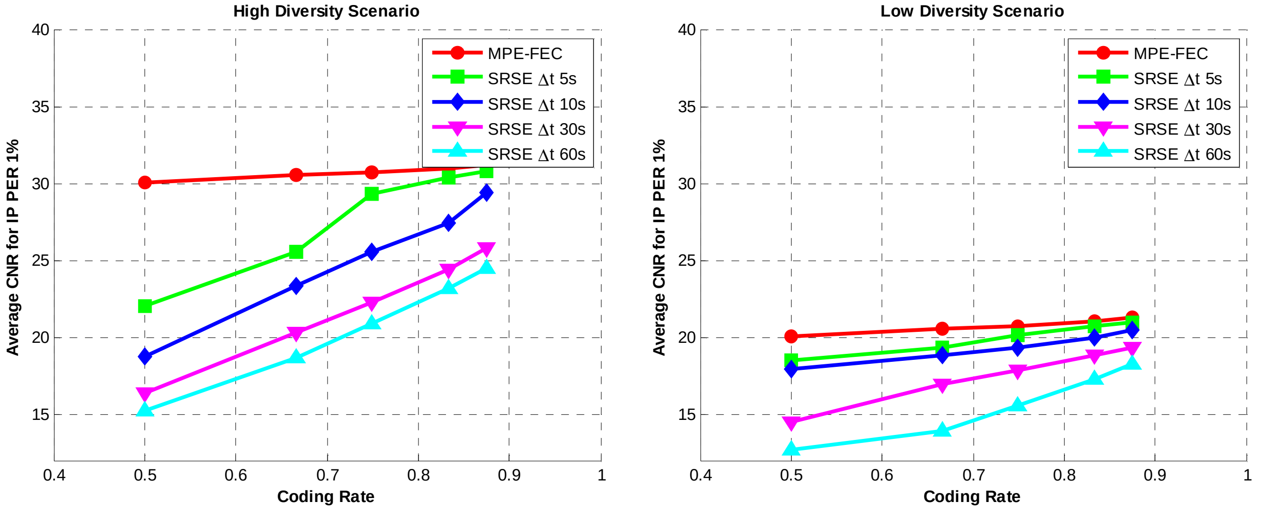

Fig. 10: Average CNR for IP PER 1% vs. Coding Rate. Streaming service 6 minutes at 384 kb/s. Left, high diversity scenario: f_d 40 Hz, σ 8 dB, d_{corr} 20 m, f_{RF} 600 MHz (v 72 km/h). Right, low diversity scenario: f_d 10 Hz, σ 5.5 dB, d_{corr} 100 m, f_{RF} 600 MHz (v 18 km/h)

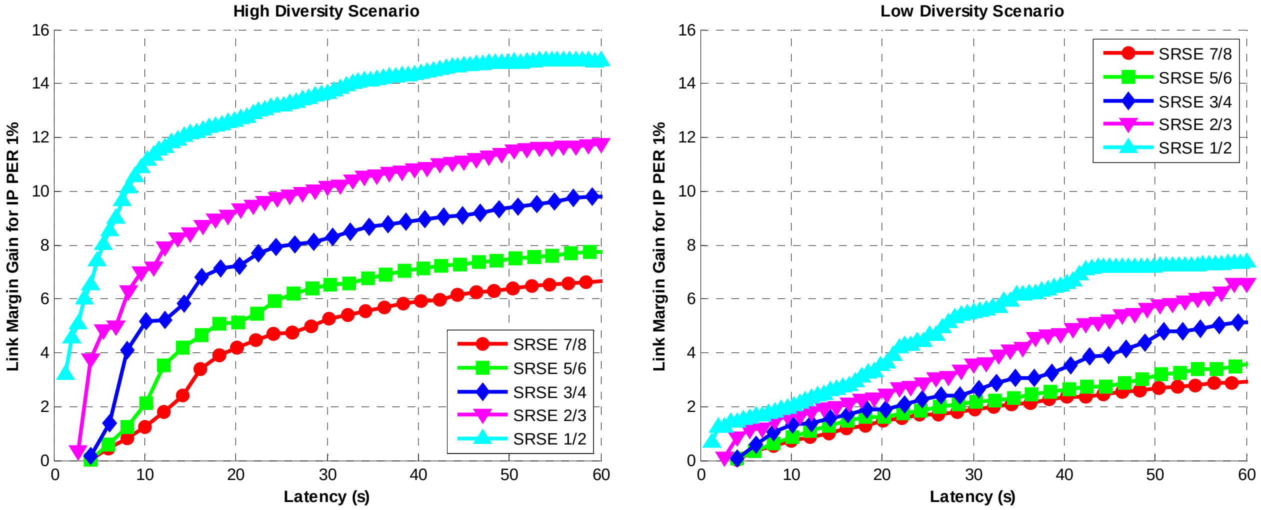

Fig. 11: Link margin gain for IP PER 1% vs. Latency. Streaming service 6 minutes at 384 kb/s. Left, high diversity scenario: f_d 40 Hz, σ 8 dB, d_{corr} 20 m, f_{RF} 600 MHz (v 72 km/h). Right, low diversity scenario: f_d 10 Hz, σ 5.5 dB, d_{corr} 100 m, f_{RF} 600 MHz (v 18 km/h)

The link margin gains can be appreciated better in Fig. 11, where the link margin gain over MPE-FEC is represented for five coding rates when the latency is incremented. Again, the simulations have been performed for the high and low diversity scenarios. As it can be seen the SRSE achieves important gains in high diversity conditions even for low latencies. If a coding rate of 1/2 is employed, it is possible to obtain a gain of 11 dB with only 10 s of latency. If the latency is incremented to 60 s, this value increases to 15 dB. Although the bigger gains correspond to the more robust coding rates, higher coding rates are also capable of providing significant gains if the latency is increased enough. This is the case of coding rates 7/8 and 5/6, which only achieve a gain of 1 and 2 dB respectively with 10 s of latency, but can improve the link margin in more than 6 dB if the latency is incremented to 60 s.

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] ETSI, TS 102 591 v1.1.1, “IP Datacast over DVB-H: Content Delivery Protocols Implementation Guidelines,” October 2007.

[3] D. Gómez-Barquero and J. Poikonen, “Filecasting for Streaming Content Delivery in IP Datacast over DVB-H Systems,” *Proc. IEEE Broadband Multimedia*, Las Vegas, USA, 2008.

[4] ETSI, TR 102 377 v1.3.1, “Digital Video Broadcasting (DVB); DVB-H Implementation Guidelines,” May 2007.

[5] ETSI, EN 302 583 v1.1.1, “Digital Video Broadcasting (DVB); Framing Structure, Channel Coding and Modulation for Satellite Services to Handheld devices (SH) below 3G,” July 2007.

[6] ETSI v0, “Digital Video Broadcasting (DVB); DVB-SH Implementation Guidelines,” May 2007.

[7] M. Kalman, E. Steinbach and B. Girod, “Adaptive Media Play out for Low-Delay Video Streaming over Error-Prone Channels,” *IEEE Trans. On Circuits and Systems*, vol. 14, pp. 841-851, June 2004.

[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.