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File Delivery for Streaming Content in IP Datacast over DVB-H Systems – an Overview

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File Delivery for Streaming Content in IP Datacast over DVB-H Systems – an Overview

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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. In particular, 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 AL-FEC overhead and the system latency introduced.*

I. INTRODUCTION AND MOTIVATION

Mobile multimedia broadcasting (i.e., delivering mass multimedia services to portable devices such as mobile phones or PDAs) 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 terrestrial 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. In contrast to DVB-T, where content is delivered in the form of MPEG-2 packets, DVB-H is IP-based (i.e., all content is delivered in the form of IP data packets). MPE (Multi Protocol Encapsulation) is the adaptation protocol used to encapsulate multiple IP streams (DVB-H services) into the MPEG-2 DVB-T transport stream. The main technical features introduced, compared to DVB-T, are 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 Forward Error Correction (FEC) mechanism at the link layer called MPE-FEC, which ensures more robust transmissions, especially under mobility and impulsive interference conditions.

DVB-H is a transmission standard that specifies the physical and link layer, but it does not define transport protocols, audio and video coding formats, etc [2]. The end-to-end system is known as *IP Datacast* (IPDC) [3]. 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 (CDP), the Electronic Service Guide (ESG) for service discovery, and mechanisms for service purchase and protection.

[†] The work presented in this paper was performed while the author was a guest researcher at the University of Turku.

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 [4].

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.

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 data carousel sessions and single file transfers 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 file until the complete file is available at the receiver, before being accessed by applications. On the contrary to streaming services, filecasting services typically require an error-free reception of the files (i.e., even a single bit error corrupts the whole file and makes 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 [5] has been adopted.

When MPE-FEC is employed, bursts contain IP packets and FEC parity information, which allows recovering from IP packet losses within bursts partially received, being able to cope with fast fading under mobility conditions. MPE-FEC increases the robustness of the transmission for mobile terminals, such that the service availability in terms of the burst error rate becomes practically independent of the speed [1]. However, MPE-FEC cannot recover from complete lost bursts (e.g., when passing through outage areas).

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 [6]. Hence, with AL-FEC it is possible to efficiently correct not only partially received bursts due to fast fading, but also complete lost bursts due to shadowing, taking advantage of the time and space diversity introduced by the bursty transmission pattern of DVB-H and the users' mobility. As a consequence, AL-FEC outperforms MPE-FEC when the file is spread over several bursts. 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. Although MPE-FEC and AL-FEC can be used at the same time, it does not bring any benefit compared to AL-FEC alone, and thus in practice AL-FEC should be employed alone [7].

As AL-FEC is not currently standardized for streaming services in DVB-H, in this paper we investigate the potential gain that can be obtained by delivering streaming content as a succession of time-constrained files using AL-FEC for improving the mobile reception of streaming services. This technique can be directly implemented with the current IPDC standard by delivering the multimedia content as a succession of time-constrained source data blocks using FLUTE. 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 that span over several 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 [8].

The performance of the proposed approach is evaluated using vehicular urban DVB-H field measurements. In particular 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 AL-FEC overhead and the system latency introduced. We investigate the optimum transmission configuration with AL-FEC for a given ESR or latency constraints, taking also into account terminal power consumption issues.

The rest of the paper is organized as follows. First we further describe and illustrate the concept of delivering streaming content as a succession of files using AL-FEC in Section 2. Then in Section 3 we explain the performance evaluation methodology, including performance measures, the measurement set-up and the field measurements performed. In Section 4 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.

II. FILECASTING FOR STREAMING CONTENT DELIVERY IN DVB-H

A. Concept

The basic idea of delivering streaming content in DVB-H as a succession of time-constrained files is to be able to employ application layer FEC for streaming services in a way compliant with the current IPDC specifications. It should be pointed out that it is possible to implement AL-FEC not only with FLUTE but also with RTP, like in the cellular 3G standard for MBMS (Multimedia Broadcast Multicast Services) streaming services [9]. However, this approach would require changes in the current standard.

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 area coverage, 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 characteristics (i.e., standard deviation and correlation distance) [6]. 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 enhance the transmission robustness for streaming content delivery 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 MPE-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. Note that for AL-FEC we have assumed a systematic code, where the first packets are the source packets, and the rest consist of additional parity packets.

In the figure we can note that although the amount of parity data (FEC overhead, coding rate) is the same in the three cases, the level of protection is different. The larger the source block, the higher the capability to protect against packet loss. We will use the example shown in Fig. 1 to make this apparent.

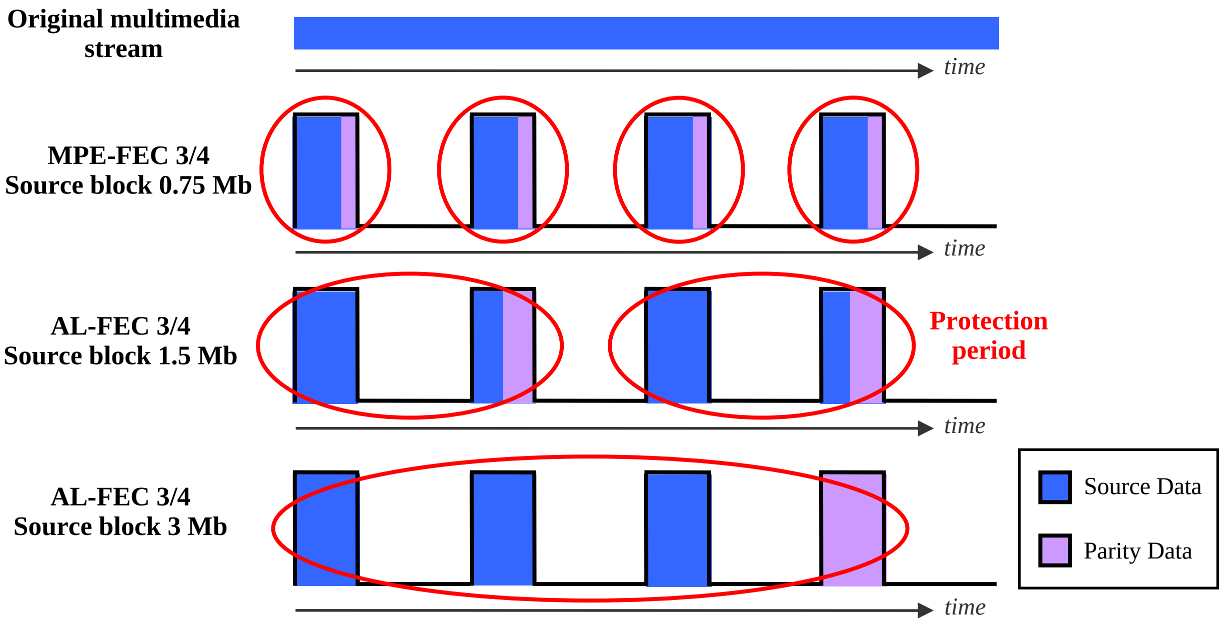

Fig. 1: Example of delivery of a 3 Mb streaming content (12 s at 256 kb/s) using MPE-FEC ¾ and AL-FEC ¾ with source block size of 1.5 Mb and 3 Mb. Burst size is 1 Mb. Cycle time is 3 s.

The DVB-H standard works with MPEG-2 packets at the physical layer (fixed size 188 bytes), and IP packets at the link layer (typical size 0.5-1.5 kB). 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, where each section is considered 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 then expressed in terms of the maximum number of erroneous sections that can be corrected. MPE-FEC can cope with a percentage of erroneous sections in a burst proportional to the coding rate. That is, the coding rate 3/4 can cope with a maximum percentage of erroneous sections per burst of 25%. 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.

For the conventional case with MPE-FEC ¾, if the percentage of erroneous sections in a burst exceeds 25%, the MPE-FEC decoder will fail and the streaming will be interrupted. In this case only correctly received data sections containing IP packets will be available for playback. For the other cases with AL-FEC ¾ shown in Fig. 1, it is possible to compensate the same percentage of erroneous sections, 25%, but across 2 and 4 bursts, being possible to compensate for more erroneous sections 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 ½ and 6 bursts coded jointly, it is 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 transmission technique for streaming services 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 [10]. 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 [4].

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 time of switching channels, being even possible to receive the channel of interest right away, and whether the first data burst of the new channel is successfully received.

With MPE-FEC the zapping time equals to the cycle time between bursts, T_C, which depends on the amount of IP data transmitted in the burst B_S (note that it does not include the amount of parity information transmitted in the burst), and the data rate of the multimedia stream R_b:

$$T_C = \frac{B_S}{R_b} \tag{1}$$

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 maximum latency introduced, which corresponds to the number of data seconds of the original multimedia stream that are coded jointly into a source block.

C. Application Layer FEC Codes in DVB-H

The standardized AL-FEC code in IPDC is Raptor coding [4]. Raptor codes are a computationally efficient implementation of *fountain codes* that achieve close to ideal performance [5]. They can be implemented on software without the need of dedicated hardware, which, in turn, allows to efficiently supporting a large range of file sizes.

Fountain codes are a special class of FEC codes that can generate an infinite amount of parity data on the fly (i.e., they are *rateless*). In practice, the standardized version of Raptor codes adopted in IPDC can generate up to 65536 encoding symbols on-the-fly from the source block. The adopted version is a systematic code, and the maximum size supported by the standard is 8 MB. At the receiver, 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 [7]).

Although the advantages of Raptor codes are evident, as they outperform other FEC solutions in terms of reliability, spectrum efficiency and flexibility, they are patented and subject to intellectual property rights. However, it should not be mixed the benefits obtained by performing error correction at the application instead of the link layer in DVB-H (that could be achieved by any code), and the benefits brought by the Raptor implementation (performance close to ideal that allows for a software implementation).

III. PERFORMANCE EVALUATION

A. 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. The network has two transmitters operating at 610 MHz covering the city center. The DVB-H transmission mode employed was: FFT size 8K, Guard Interval (GI) ¼, modulation 16QAM, and coding rate ½, 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 field measurement campaign. Two DVB-H professional receivers with a common external antenna 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. This is possible because MPEG-2 TS packets headers contain a field called TEI (Transport Error Indicator) that indicates whether the packet has been correctly received or not.

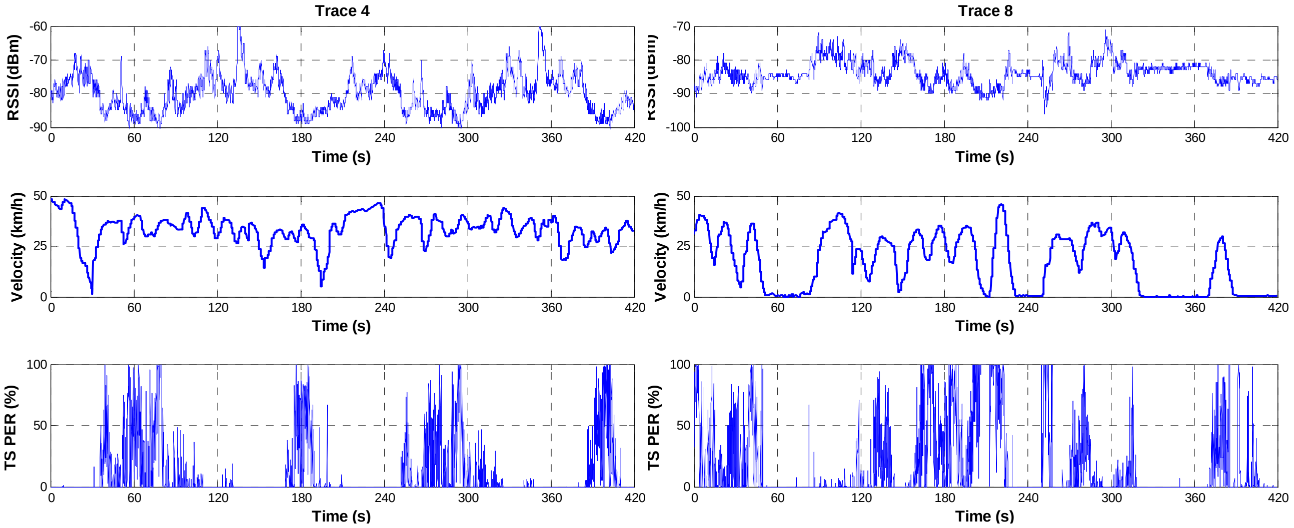

Fig. 2: Example data of vehicular DVB-H field measurement.

By recording the MPEG-2 TS packet error trace at the physical layer it is possible to reproduce the QoS experienced by the terminals across the measured trajectory for any type of service (streaming or file delivery). Moreover, it is possible to investigate the effect of different DVB-H transmission configurations at the link and application layer, such as the burst size, and 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).

In this paper, the measured packet error traces were used to emulate the quality experienced by the measuring terminals for a streaming service for different MPE-FEC and AL-FEC configuration settings, using the performance measures and assumptions described next. As an example, Fig. 3 shows the cumulative number of erroneous seconds over time for a streaming service at 256 kb/s for different MPE-FEC coding rates for the measured trajectories shown in Fig. 2. In the figure we can see the time evolution of the errors experienced by the users, as well as the actual improvement perceived by the user when varying the MPE-FEC coding rate.

Annex A contains figures with the MPEG-2 TS packet error distribution of the 15 traces analyzed in this paper. The duration of the traces is 7 minutes. Table 1 shows the average Packet Error Rate for the different traces.

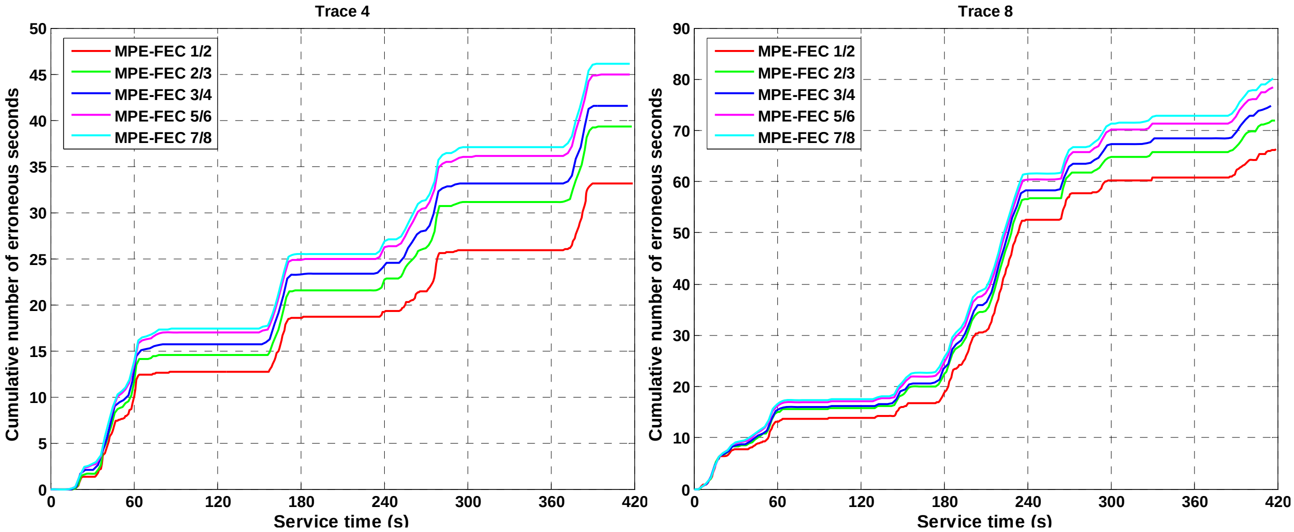

Fig. 3: Erroneous seconds for a 256 kb/s streaming service across the measured trajectories shown in Fig. 2. Maximum burst size is 1 Mb. IP packet size is 512 bytes.

Table 1: Average MPEG-2 TS packet error rate of the measured traces analyzed in this paper (%).

Trace ID	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**
TS PER	7.9%	8.6%	10.1%	10.5%	13.1%	15.9%	16.3%	17.8%
Trace ID	**9**	**10**	**11**	**12**	**13**	**14**	**15**	
TS PER	19.5%	21.2%	27.1%	29.8%	31.1%	36.2%	36.6%	

B. Performance Measures and Assumptions

Typically, in DVB-H a 5% burst error rate is considered as the degradation point for streaming services when using MPE-FEC. This criterion is known as MFER 5% (MPE-FEC frame error rate) [1]. However in our evaluations we employ the *Erroneous Second Ratio* (ESR) as the performance indicator to compare the robustness of different transmission configuration schemes. The more robust the transmission configuration is, the smaller the ESR perceived by the users.

The ESR represents the percentage of erroneous seconds during the service time, and it can be easily used to compare MPE-FEC and AL-FEC. Compared to the burst error rate, the ESR reflects more accurately the amount of correctly received information for streaming services, as it takes into account that it is possible to receive partially a burst with MPE-FEC. When a burst cannot be completely decoded with MPE-FEC, only correctly received IP packets will be available for playback. Note that this also applies for AL-FEC when the complete source block cannot be decoded. For the shake of simplicity, to address these cases we assume that each IP packet can be played successfully without the need of any previous IP packet.

Besides the ESR, we also consider the network latency, which is directly related with the zapping time, as QoS performance indicators. In particular in this paper we investigate the gain brought by AL-FEC compared to MPE-FEC in terms of reduced ESR for a given measured trace as a function of the latency introduced. We consider a 6 minutes streaming service of 256 kb/s, and the results presented in next sections are the average results over the different traces (7 minutes length). 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, but for AL-FEC we have considered 255 columns for all cases (constant burst size of 1 Mb).

For the sake of simplicity, 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 recovery of the source blocks in most of the cases [7].

IV. RESULTS AND DISCUSSIONS

As an example of the results obtained, Fig. 4 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 data burst per source block). It should be pointed out that the latency depends not only on the number of data bursts coded jointly, but also on the coding rate, as it determines the source block size. The latency is equal to the product of the number of bursts times the cycle time given by (1), which depends on the coding rate. As we are dealing with traces with a considerable amount of errors, we consider more robust coding rates than the commonly used with MPE-FE: 1/4, 1/3, 1/2, 2/3, and 3/4.

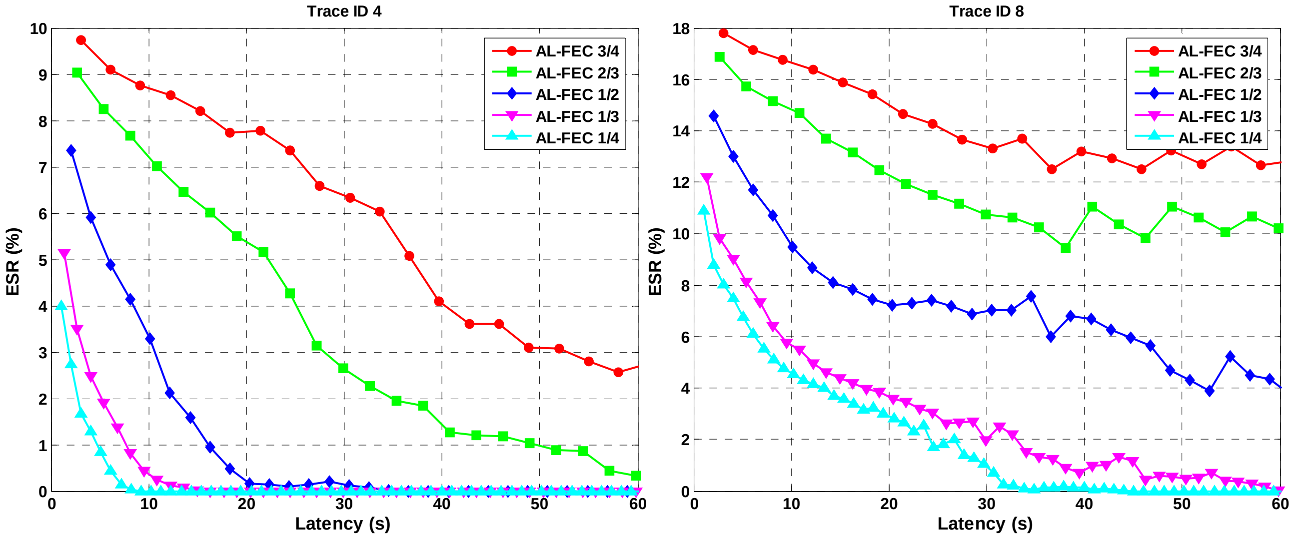

Fig. 4: Erroneous second ratio (%) vs. Latency (s) for the measured traces shown in Fig. 2. Streaming service 10 minutes 256 kb/s. Burst size 1 Mb.

In the figure we can see the decreasing tendency of the ESR as a function of the latency, achieving in some cases an ESR equal to zero (i.e., all streaming content is received correctly without errors), although we can note a saturation effect for coding rates 2/3 and 3/4 in trace 8. This saturation effect is due to the fact that the coding rate is not robust enough to compensate for the transmission errors, and thus, there is no gain by increasing the source block (interleaving duration). In the figure we can also note different slopes in the two measured traces. This difference is due not only to the different average TS packet error ratio of each trace (10.5% and 17.8% respectively), but also due to the different error distribution. In Fig. 2 we can note that for trace 8 there is an outage period of about 20 seconds (around the third minute and a half of measurement), where most of the packets are erroneously received, whereas for trace 4 the errors are distributed over time without outage periods. Outage periods imply that larger interleaving durations are necessary to cope with those transmission errors.

Therefore, it can be concluded that the required latency to achieve a given ESR requirement depends on the effective coding rate and the total amount of transmission errors experienced by the users and their time distribution. To illustrate the trade-off between latency and coding rate, we show in Fig. 5 the minimum coding rate required to achieve a target ESR of 1% and 5% as a function of the latency for the two traces considered.

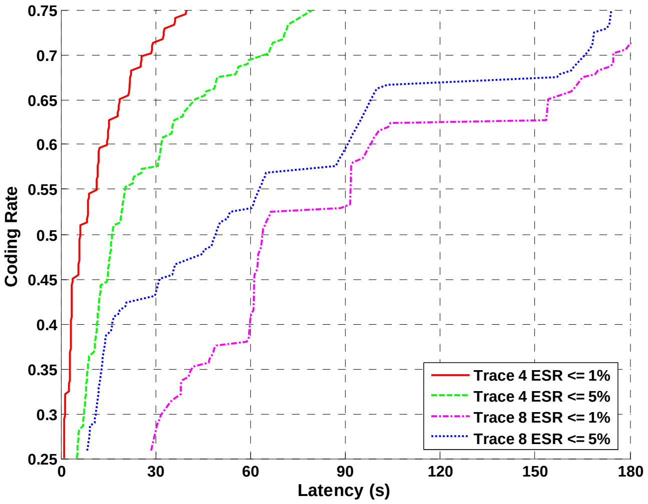

Fig. 5: Minim required coding rate vs. Latency (s) for erroneous second ratio targets of 1% and 5% for the measured trajectories shown in Fig. 2. Streaming service 10 minutes 256 kb/s. Burst size 1 Mb.

The larger the latency introduced, the higher the coding rate required and the smaller the amount of parity data required, and hence more services can be delivered with the same infrastructure. In Fig. 5 we can also see flat areas in the curves for trace 8. This is due to the outage periods discussed before.

The trade-off between latency and coding rate is further illustrated in Annex B, where figures with the minimum ESR as a function of the coding rate for latencies of 5, 10, 30 and 60 seconds are presented for the all the measured trajectories shown in Annex A. The case with MPE-FEC it is also included for comparison. MPE-FEC markers shown correspond to coding rates 1/2, 2/3, 3/4, 5/6 and 7/8. We can see that the gain obtained with AL-FEC is very significant, especially for latencies of 30 and 60 seconds. The improvement of the coding efficiency is apparent. We can see that the ESR reduction when reducing the MPE-FEC coding rate is significantly smaller compared to AL-FEC. For example for a coding rate 1/2, the gain in terms of ESR reduction ranges between 2% and 13% for a 30 s latency, and between 5% and 20% for a latency of 60 s. The minimum gain is obtained in trace 3, where most transmission errors are produced in the same minute, and the maximum gain is obtained in trace 15, the one with most errors of all measured. With MPE-FEC the mean ESR in a given trace is similar to its average MPEG-2 TS packet error ratio. We can verify this statement in Fig. 6, where the mean ESR it is plotted as a function of the TS PER for all the trajectories measured.

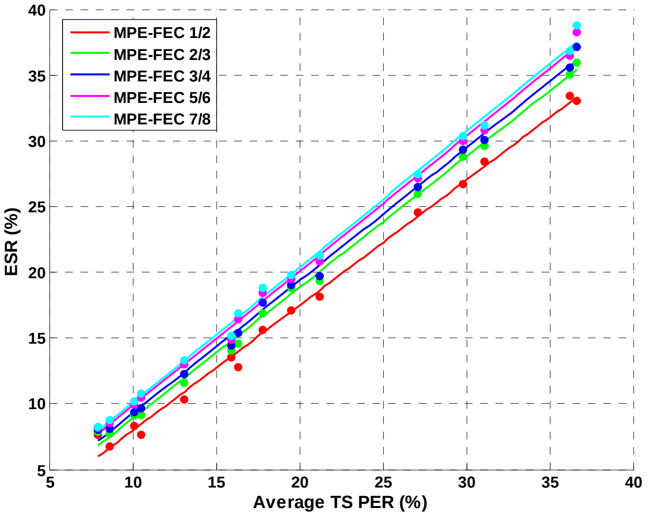

Fig. 6: Erroneous second ratio (%) vs. Average TS PER (%) of the measured trajectories shown in Annex A.

In the figure we can see that the ESR reduction between the maximum and minimum coding rate (7/8 and 1/2) is relatively small. Finally we compare the mean ESR over all measured trajectories with AL-FEC and MPE-FEC in Fig. 7. We consider the coding rate 1/2 with MPE-FEC as a reference, as it provides the most robust transmission and the minimum latency (1 second in our case). In the figure of the left we keep the coding rate fixed equal to 1/2, and we vary the latency. The gain in ESR reduction is approximately 2%, 4.5%, 7.5% and 10% for latencies of 5, 10, 30 and 60 seconds respectively. In the figure of the right we fix the latency, and we vary the AL-FEC coding rate. In this case we can see that with a coding rate of 2/3 we achieve the same performance than with MPE-FEC 1/2. Thus, an increase of the system capacity of 16.6% can be achieved by trading 9 seconds of latency.

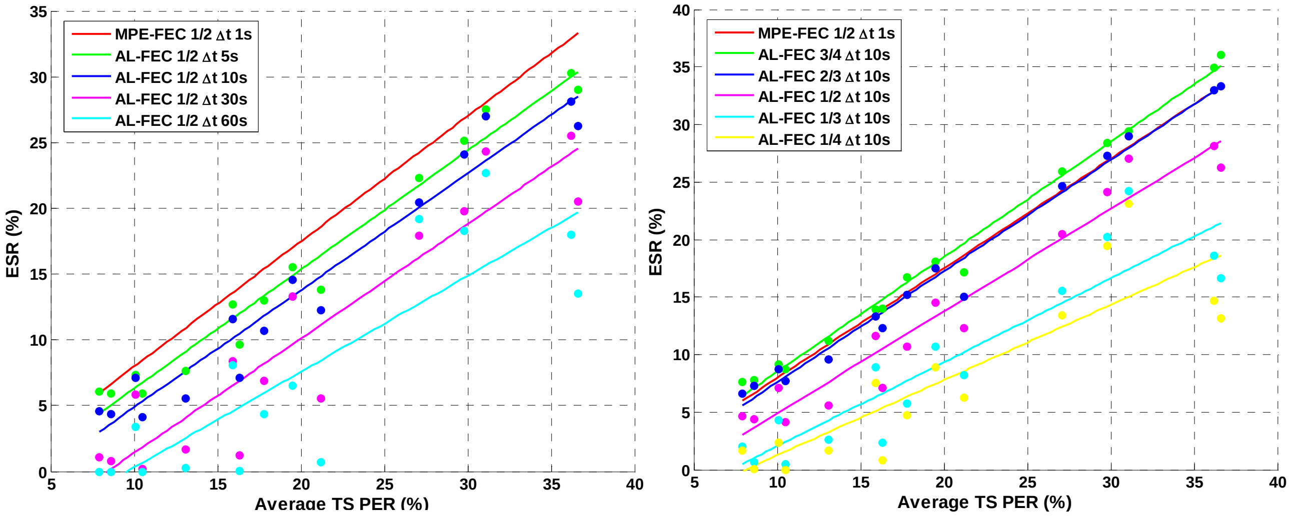

Fig. 7: Erroneous second ratio (%) vs. Average TS PER (%) of the measured trajectories shown in Annex A. Streaming service 10 minutes 256 kb/s. Burst size AL-FEC 1 Mb. Burst size MPE-FEC 0.5 Mb.

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ANNEX A: MEASURED ERROR TRACES

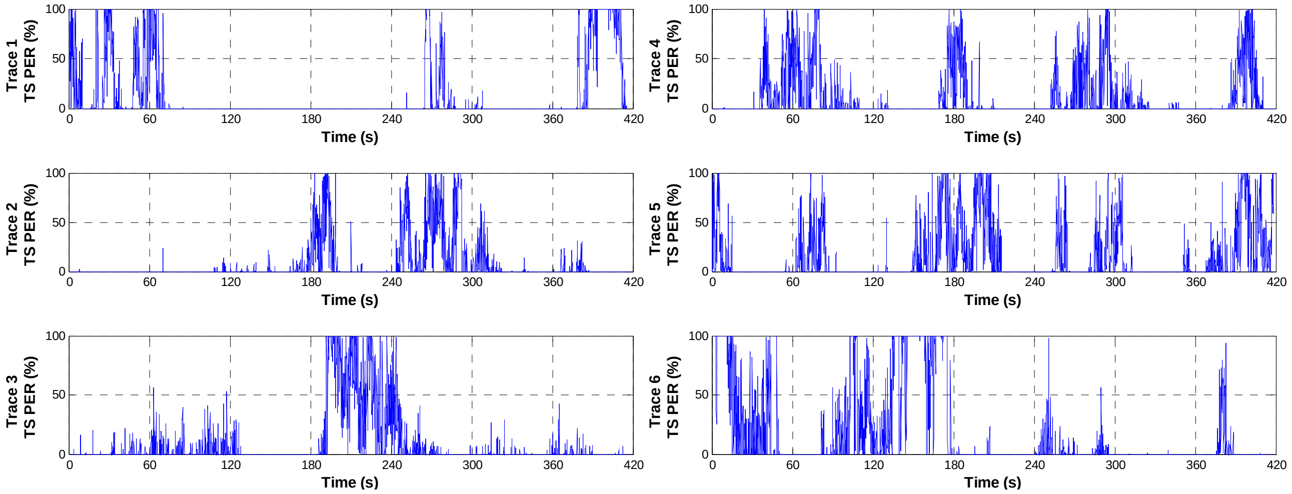

Fig. A.1: Measured MPEG-2 TS packet error rate over time. Traces 1-6.

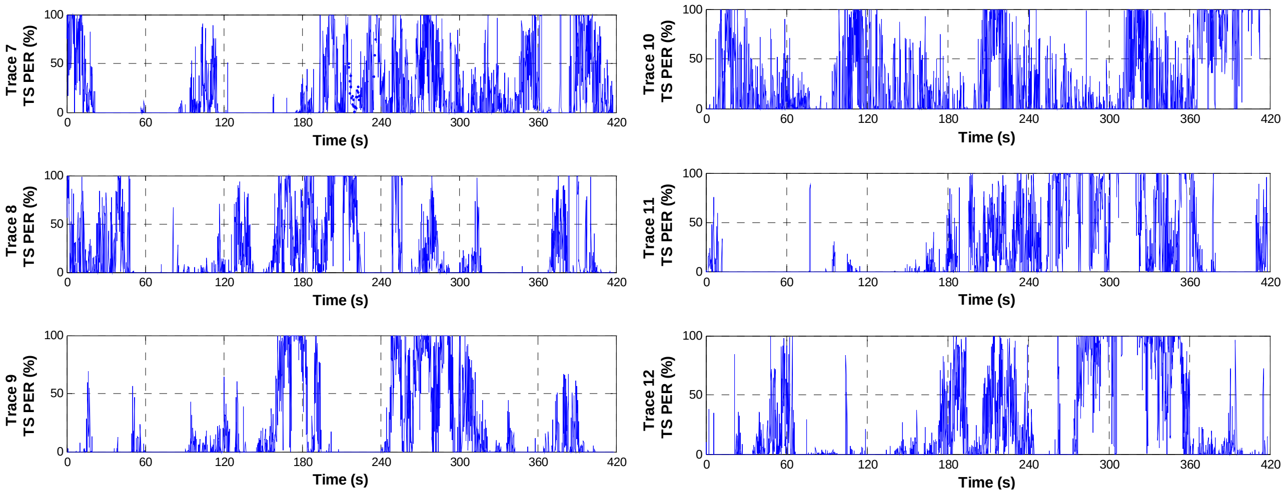

Fig. A.2: Measured MPEG-2 TS packet error rate over time. Traces 7-12.

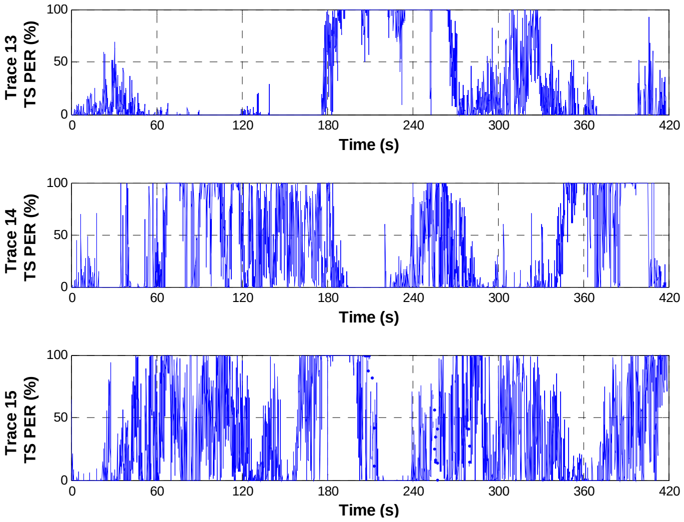

Fig. A.3: Measured MPEG-2 TS packet error rate over time. Traces 13-15

ANNEX B: DETAILED RESULTS

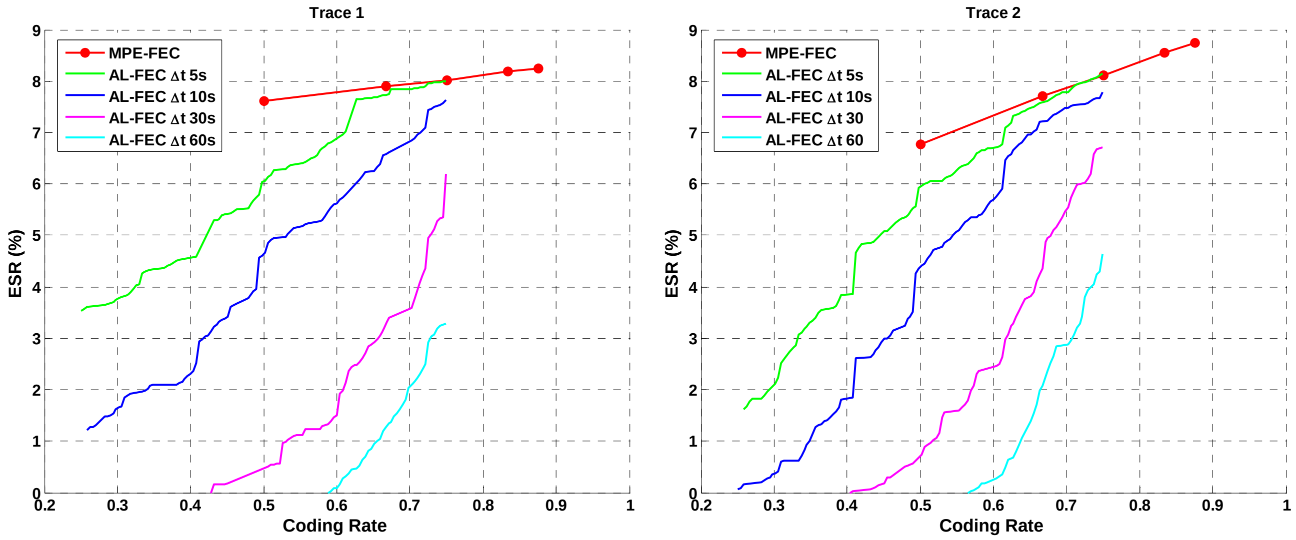

Fig. B.1: Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 1 and 2.

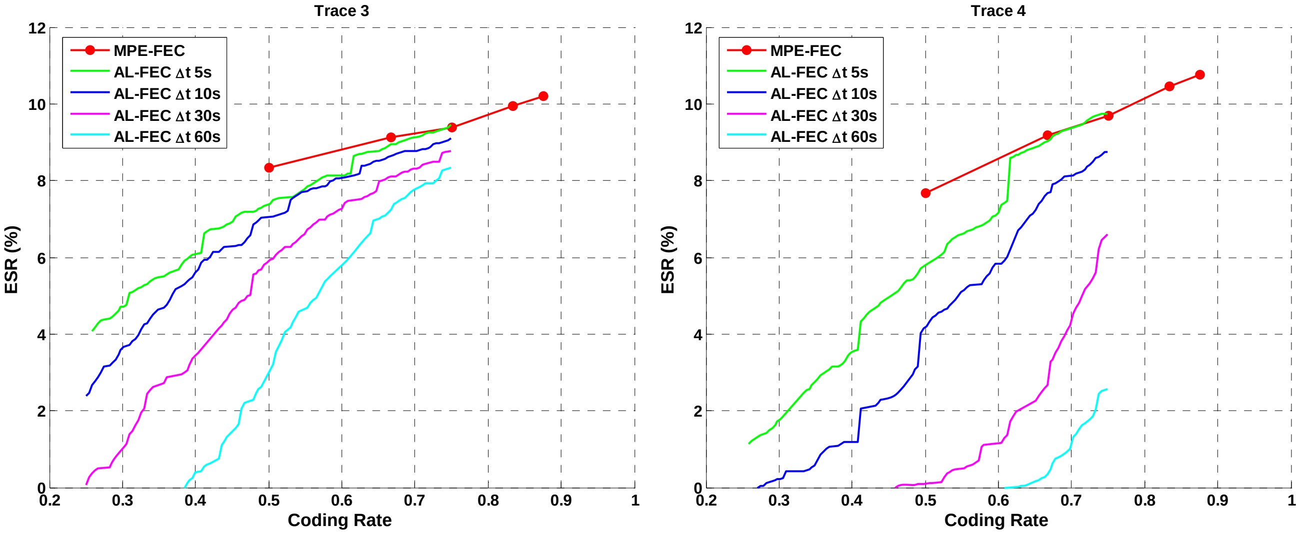

Fig. B.2: Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 2 and 3.

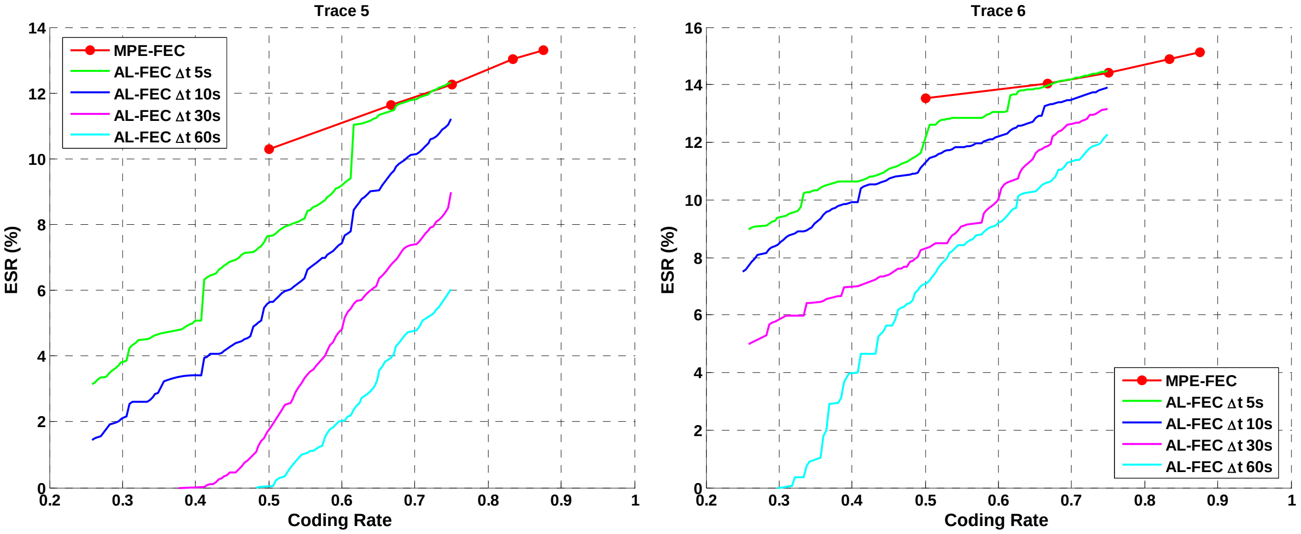

Fig. B.3: Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 5 and 6.

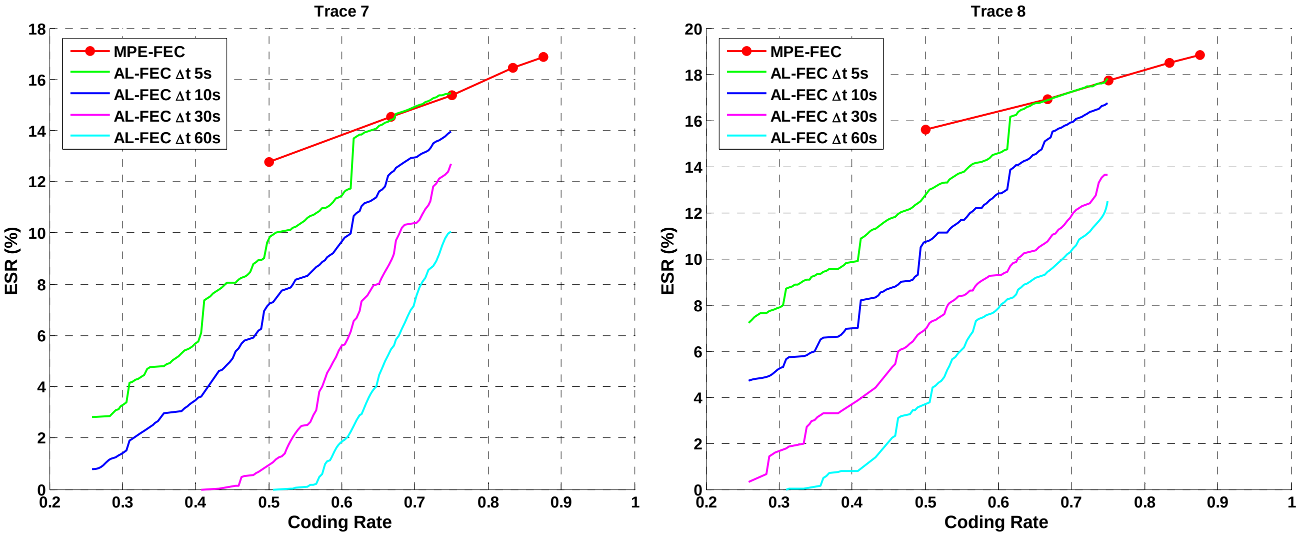

Fig. B.4: Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 7 and 8.

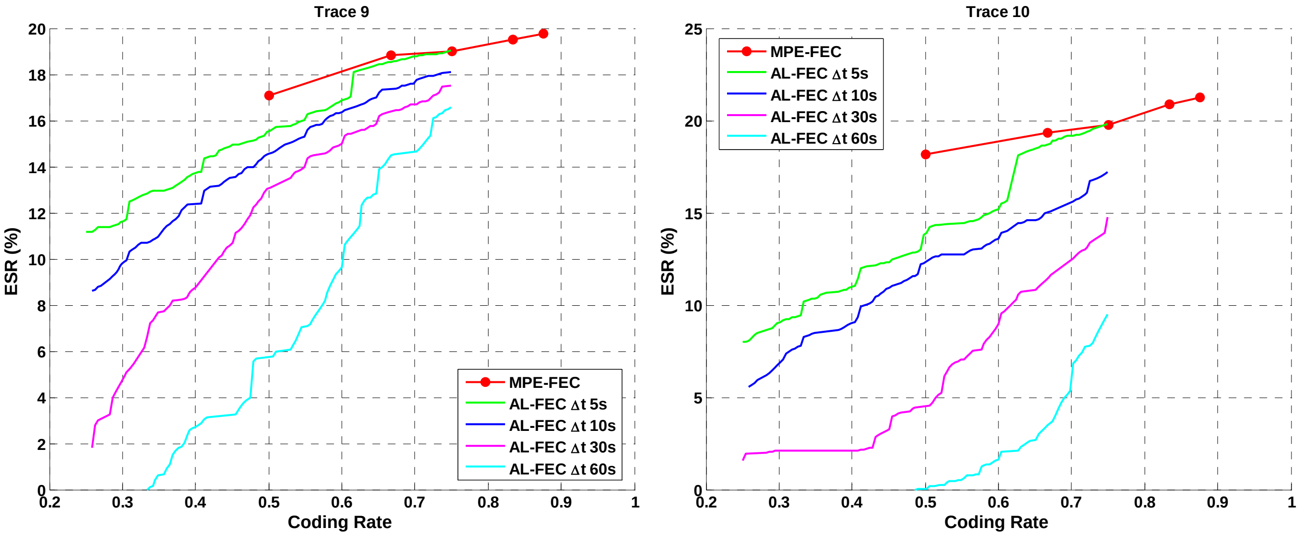

Fig. B.5: Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 9 and 10.

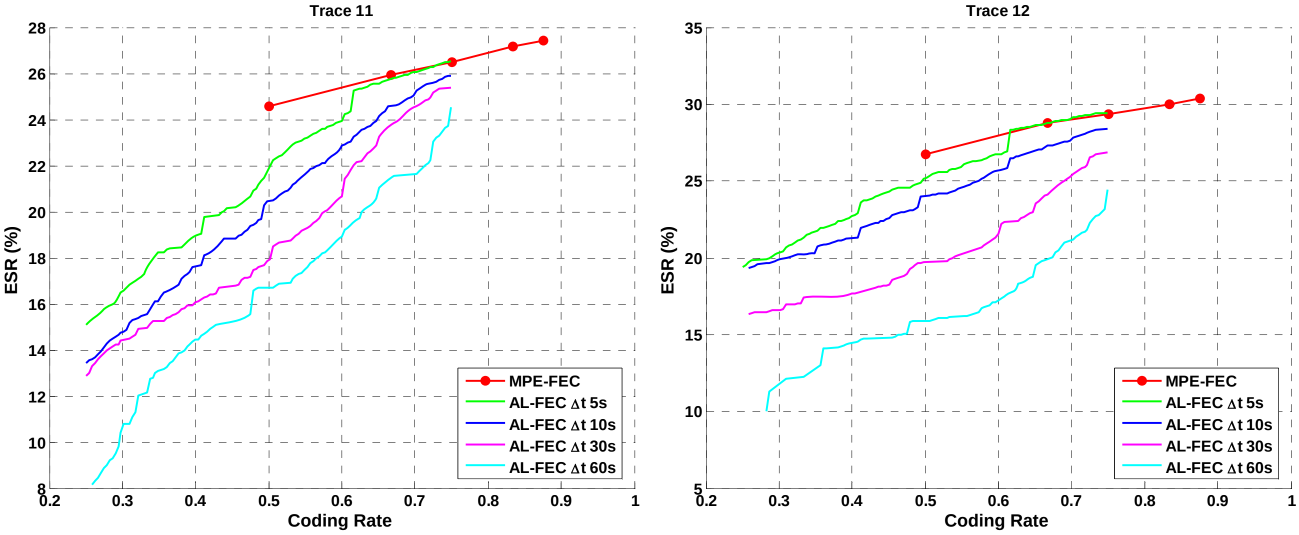

Fig. B.6: Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 11 and 12.

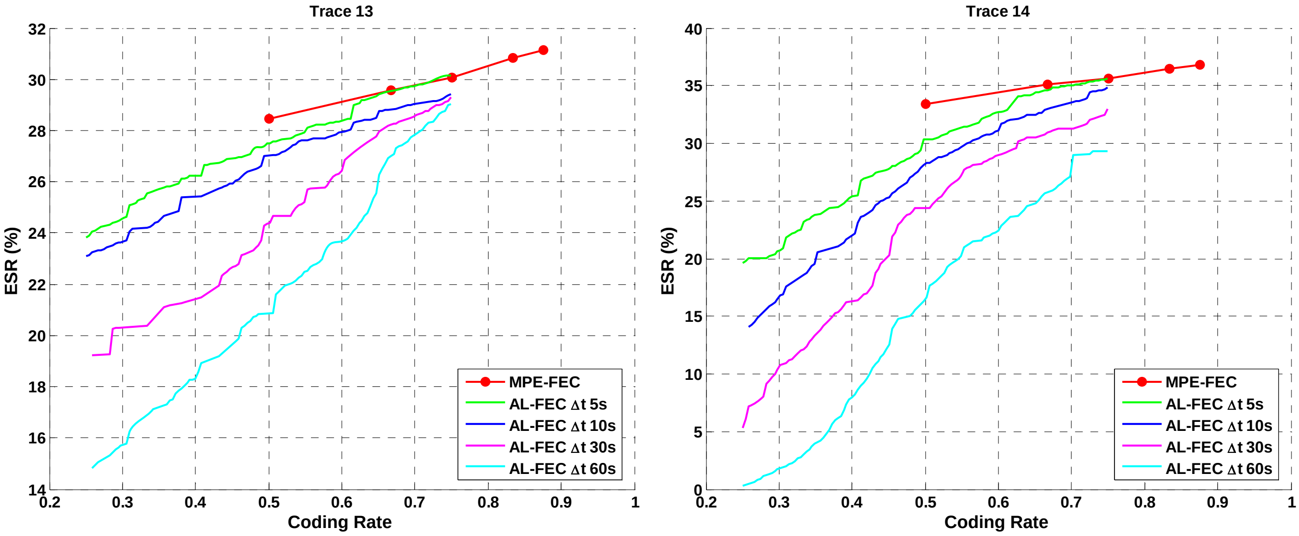

Fig. B.7 Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Traces 13 and 14.

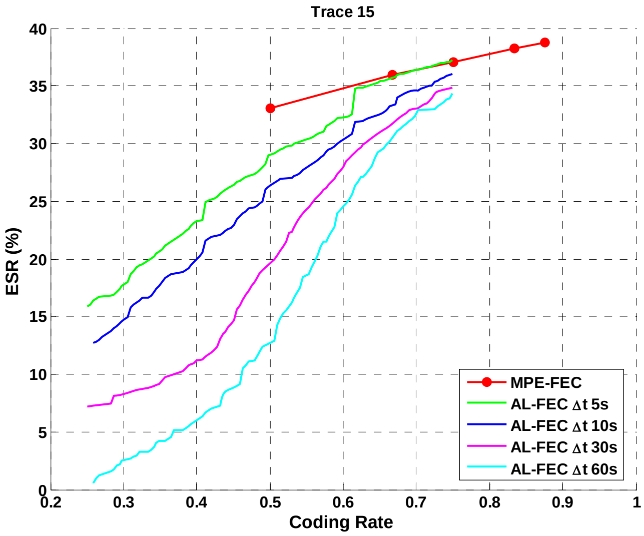

Fig. B.8 Erroneous second ratio vs. coding rate. 10 minutes streaming service at 256 kb/s. Trace 15.