2012 IEEE International Conference on Multimedia and Expo Workshops

Mobile TV with long Time Interleaving and Fast Zapping

Cornelius Hellge*†, Valentina Pullano‡, Manuel Hensel*, Giovanni E. Corazza‡, Thomas Schierl*, and Thomas Wiegand†
*Multimedia Communications Group, Image Processing Department
Fraunhofer HHI, 10587 Berlin, Germany
Email: (cornelius.hellge;manuel.hensel;thomas.schierl)@hhi.fraunhofer.de
‡Department of Electronics Computer Science and Systems, Faculty of Engineering
University of Bologna, Viale Risorgimento 2, 40136 Bologna, Italy
Email: (vpullano;gecorazza)@arces.unibo.it
†Image Communication Laboratory
Berlin Institute of Technology, 10623 Berlin, Germany
Email: twiegand@ieee.org

***Abstract*—The main challenge for provisioning of Mobile TV services is to overcome long burst errors as are often found in mobile reception conditions. Long time interleaving can be implemented by means of Application Layer FEC (AL-FEC) to increase the time diversity of the signal and thereby its robustness against burst errors. The main obstacle of long time interleaving for streaming services is the increase in service tune-in time, which significantly decreases the Quality of Experience (QoE) of end users. That is why today's Mobile TV systems are provisioned in a way to minimize the time interleaving length to provide an acceptable tune-in time, though the service robustness would significantly benefit from a longer interleaving length.**

This paper presents a new way of service provisioning that marries fast zapping and long time interleaving by combining Layer-Aware FEC and layered media codecs with unequal time interleaving and an appropriate transmission scheduling. The effect of the proposed scheme on the QoE as well as the service tune-in time is analyzed. Simulation results within a Gilbert-Elliot channel report the benefit of the proposed scheme, which for the first time enables broadcast services with fast tune-in and at the same time long time interleaving.

***Keywords*-Application layer FEC; SVC; Raptor; RaptorQ; Mobile TV; Time interleaving; Zapping time**

I. Introduction

Mobile TV services are affected by channel impairments characterized by long error bursts. Such long error burst are mainly caused by shadowing from obstacles that affect the wave propagation and can range from milliseconds to several seconds. Long time interleaving helps to overcome long error bursts by increasing the time diversity of the transmitted data. For that reason today's Mobile TV standards, like e.g. specified in 3GPP [1] or DVB [2], implement means for time interleaving on physical layer, link or application layer. The main obstacle of long time interleaving for streaming services is the increase in service tune-in time (zapping time). Long time interleaving requires the receiver to wait until all packets of that interleaving period have been received and filled into the deinterleaving buffer. That is why today's video transmission systems try to minimize the time interleaving length to provide a low tune-in time typically bellow two seconds (see 3GPP MBMS requirements [3]), though the service robustness would benefit from a longer interleaving length [4].

Today's standards for Mobile TV and IPTV contain approaches to marry a fast service tune-in with a long time interleaving. Two different solution are e.g. specified for multicast services in IPTV [5]. The usual receiver procedure to access an IPTV service is to join an IP multicast stream using the IGMP protocol. Without a solution for fast tune-in, the receiver needs to wait up to several seconds before it can start play-out the video stream until the play-out buffer is filled and a Random Access Point (RAP) has been received. With the so called server-based solution, a receiver simultaneously establishes a unicast connection via RTP to another server, which has cached several seconds of the multicast stream. This cached content can be transmitted in a much faster way than the normal streaming rate. The client can immediately play-out the cached content from the recent past while the play-out buffer is continuously filled with the multicast stream. Thereby, the server-based approach allows to provide services with fast tune-in and long time interleaving to the price of an increased end-2-end delay due to the required caching period. However, such an approach cannot be applied to a pure broadcast service since it requires a return channel to establish the RTP connection. The second solution relies on a companion stream. At start-up, a receiver joins two multicast streams of the same content but with different qualities. The stream with the lower quality has a higher RAP frequency and enables fast tune-in. After a transition time, which depends on the difference in RAP frequency, the receiver can jump on the higher quality stream. The companion stream solution could be applied to broadcast services but does not allow to combine fast tune-in with long time interleaving and wastes valuable bandwidth due to transmitting the same content twice.

Another solution is specified in DVB-SH by means of

978-0-7695-4729-9/12 $26.00 © 2012 IEEE
DOI 10.1109/ICMEW.2012.114

a Link-Layer FEC (LL-FEC) [6]. The LL-FEC scheme can be generated over large FEC source blocks that cover several seconds and thereby increases time diversity and robustness against long burst errors. Obviously, a large FEC source block increases the tune-in time to the service in the same way like long time interleaving, since the receiver has to wait until all FEC data of that source block has been received. In order to enable fast tune-in, DVB-SH proposes two options. One is to fast tune-in into the systematic part of the LL-FEC data. This alternative implies that users will experience an interruption of the service the first time an error is encountered, as terminals would need to buffer the remaining parity data. DVB-SH furthermore specifies an optional solution that allows the transition to the parity data without service interruption by use of adaptive media play-out codecs. With this solution, the initial play-out is slowed down in such a way that the buffer needed for the FEC data can be filled over time. However, this solution requires modifications to existing video and audio decoders due to the strict timing constraints of the decoder buffers.

This paper presents a new way of service provisioning that marries fast zapping and long time interleaving by combining Layer-Aware FEC [7] and Scalable Video Coding (SVC) [8] with unequal time interleaving and an appropriate transmission scheduling. The proposed solution does not require a feedback channel or any modifications on video or audio decoders.

II. Technology Background

A. Layered media codecs

Single layer (SL) media streams, as those encoded by H.264/AVC allow decoding of a single and predefined bit-rate and media quality. The difference between a single layer media stream and a layered media stream is illustrated in Fig. 1. A layered media stream contains multiple sub-bit-streams that allow to extract multiple bit-rates and media quality levels from the single bit-stream. These sub-bit-streams are referred to as media layers. Layered media bit-streams typically contain a hierarchy between the layers which results from encoding algorithm. That means, decoding the quality level of the first enhancement layer (EL) requires also the base layer (BL) like shown by the dependency in Fig. 1. H.264/AVC defines two profiles that enable encoding of layered media streams.

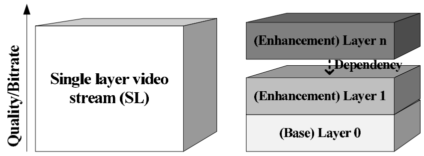

Figure 1. Single layer (SL) media stream and layered media stream

SVC[8] is a profile of H.264/AVC that allows up to three different scalability dimensions within one bit-stream: temporal, spatial, and quality scalability. The BL of SVC provides the lowest level of quality and is a H.264/AVC compliant bit-stream to ensure backwards-compatibility with existing receivers. Each additional EL improves the video quality in one of the three dimensions. In SVC, the BL is more important than the EL due to inter-layer prediction. Therefore, in case of missing BL information, the EL information becomes useless due to missing prediction information.

B. LA-FEC

Layer-Aware FEC (LA-FEC) has been already presented in [7]. The basic idea of LA-FEC is to extend the encoding process of the FEC algorithm across dependent media layers while the FEC processing of the BL remains untouched. Thereby, the BL can still be decoded independently and with the full correction capabilities of the original FEC algorithm. Due to the introduced connection from less important media layers within the FEC algorithm, the more important media layers are protected by additional repair data. LA-FEC can be applied to any linear FEC algorithm, such as e.g. Raptor or RaptorQ [9] like shown in [7].

Also in [7] it is shown that with the assumption of an ideal FEC code, the condition for decoding a media layer with ST-FEC requires the reception of $r > k$ symbols out of n symbols like shown by equation 1 with r_x being the number of received symbols of layer x:

$$r_x \geq k_x \tag{1}$$

Since for media decoding, layer 1 can only be decoded with a successfully received layer 0, the condition for a successful media decoding of layer 1 changes from equation 1 to equation 2:

$$(r_0 \geq k_0) \wedge (r_1 \geq k_1) \tag{2}$$

With LA-FEC, the symbols of layer 1 additionally protect the BL symbols. Following the discussion in [7], the condition for decoding layer 0 in equation 1 changes with LA-FEC to equation 3:

$$(r_0 \geq k_0) \vee (r_0 + r_1 \geq k_0 + k_1) \tag{3}$$

Since with LA-FEC, the EL FEC correction depends on the BL recovery as well, the condition for recovering the EL changes with LA-FEC to equation 4:

$$(r_1 \geq k_1) \wedge (r_0 + r_1 \geq k_0 + k_1) \tag{4}$$

C. Time interleaving

A time interleaver increases the time diversity of a message by spreading a portion of the data of a media layer over a longer period of time. This is exemplary shown in Fig. 2 for the k_0 source symbols and p_0 parity symbols related to a FEC source block SB_0(t=1s), which covers a protection period of $t = 1$ second of the media stream. For real-time transmission of the video data, the channel data rate needs to be $r \geq \frac{n_0 = k_0 + p_0}{seconds}$. With the assumed channel rate, the delay introduced by the FEC procedure is referred to as d_{FEC}, which is equal to one second. In the figure, the n_0=k_0+p_0 symbols are transmitted without interleaving at interleaving length IL=1 and with interleaving at IL=3.

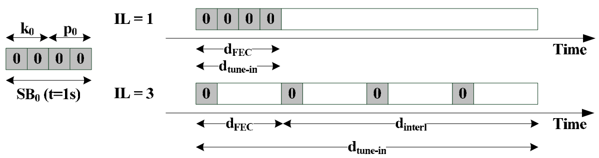

Figure 2. Different interleaving length (IL=1,3) of data of a FEC source block (SB(t=1s)) with k source and p parity symbols.

The delay d_{FEC} directly influences the service tune-in time $d_{tune-in}$. With an interleaving length of IL=1, no interleaving is applied and the tune-in delay is equal to the protection period d_{FEC} of the FEC source block. With IL=3, the tune-in delay increases to $d_{tune-in}$=d_{FEC}+d_{interl} with d_{interl}=IL$\cdot$ t=3 seconds.

III. LA-FEC UI or Long time interleaving with fast zapping

The combination of LA-FEC and SVC enables a new way of service provisioning that marries fast zapping and longtime interleaving. Fast zapping is given by a BL with lower quality and short time interleaving, which provides less robustness against burst errors but also introduces less delay. The robustness against burst errors is given by the EL long time interleaving for providing stronger robustness at the expense of an increased latency of the EL. Taking into account that the EL is LA-FEC coded across the BL symbols, the BL also benefits from the improved time diversity of the EL. This scheme is in the following referred to as unequal time interleaving with LA-FEC or LA-FEC UI. It is important to note that LA-FEC UI can not only be applied to SVC or any other kind of layered media, but also to any kind of time synchronized data. E.g. a possible application would also be to have an audio stream with short time interleaving and a video stream with long time interleaving. The combination of LA-FEC and SVC enables a new way of service provisioning that marries fast zapping and longtime interleaving. Fast zapping is given by a BL with lower quality and short time interleaving, which provides less robustness against burst errors but also introduces less delay. The robustness against burst errors is given by the EL long time interleaving for providing stronger robustness at the expense of an increased latency of the EL. Taking into account that the EL is LA-FEC coded across the BL symbols, the BL also benefits from the improved time diversity of the EL. This scheme is in the following referred to as unequal time interleaving with LA-FEC or LA-FEC UI. It is important to note that LA-FEC UI can not only be applied to SVC or any other kind of layered media, but also to any kind of time synchronized data. E.g. a possible application would also be to have an audio stream with short time interleaving and a video stream with long time interleaving.

The process of unequal stream interleaving with LA-FEC UI is shown in Fig. 3. The example uses an exemplary SVC stream with two layers, which are encoded with Layer-Aware Raptor FEC (see [7]). Each BL (white blocks) FEC source block consists of n_0=2 symbols with k_0=1 source and p_0=1 parity symbols. The EL (gray blocks) has a higher service bit-rate and consists of n_1=4 symbols with k_1=2 source symbols and p_1=2 parity symbols. The number within each symbol denotes the corresponding FEC source block. The protection period of each FEC source block covers one second of the media stream and the channel rate is assumed to be sufficient to transmit all base and EL symbols within $t_{channel}$=1 second. In the outlined example in Fig. 3, the BL is not interleaved with IL_0=1 while the EL is interleaved with other FEC source blocks with IL_1=4. The illustrated interleaving process works on FEC symbols of fixed size. It is important to note here that the interleaving process can also be performed on packet level like e.g. UDP packets containing the encoded FEC source and parity symbols.

Fig. 3 shows in the upper part a normal transmission order of the media stream, where FEC source blocks are transmitted in sequential order. First, the k source symbols of each layer are transmitted followed by the p parity symbols. For achieving the long time interleaving, the EL symbols are fed into a convolutional interleaver like shown in the figure. The symbols of a FEC source block are written column-wise into the interleaver memory. The data is read from the diagonal of the memory matrix, thereby the symbols of different FEC source blocks are interleaved. The interleaved symbols of the EL replace the original EL symbols in the new transmission order. After the writing process, the memory buffer is shifted column-wise towards the right side, thereby dropping the already processed data. Such an interleaving over multiple FEC source blocks requires buffering on the server side. This increases the end-2-end delay of the system by factor Δt_{interl}, which depends on the difference in time interleaving in both layers. While all symbols of FEC source block zero in the original transmission order can be transmitted at time instance t_0, the same FEC source block is transmitted after interleaving at time instance $t_0 + \Delta t_{interl}$. With the given transmission time for all data within the protection period of $t_{channel} = 1$ second, Δt_{interl} is simply calculated by $\Delta t_{interl} = (IL_1 - IL_0) \cdot t_{channel}$. Therefore, in the given example, the end-2-end delay of the transmission system increases by $\Delta t_{interl} = 3$ seconds. Such a transmission scheduling enables fast service tune-in with simultaneous long time interleaving with the price of an increased end-2-end delay. Note that the server-based IPTV

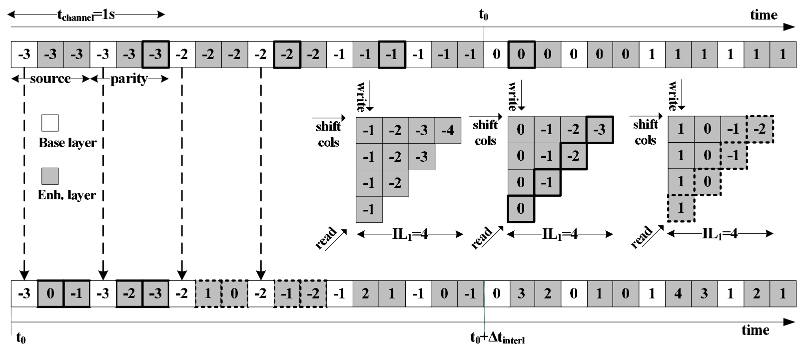

Figure 3. Unequal time interleaving of an exemplary SVC stream. The BL provides fast tune-in with interleaving length IL_0=1s and the EL provides long time interleaving with IL_1=4s.

solution increases the end-2-end delay in the same way.

Fig. 4 illustrates the user experience with such a stream within an exemplary transmission scenario over an error prone channel. The stream generated in Fig. 3 is affected by a burst erasure that affects four symbols. As a FEC code, we use the layer-aware extension for Raptor code [7], which approaches the performance of an ideal code very closely. Therefore, we can assume the related decoding conditions for LA-FEC like shown in Section II-B.

In the outlined example in Fig. 4, a user tunes into the service at time instance t_0. With a given $t_{channel}$=1 second, the client can start the play-out at BL quality after less than one second at time instance t_1. Because at this time instance, enough BL symbols of FEC source block SB=-3 have been received (i.e. r_0=1, which is sufficient to fulfill the condition for decoding the BL (cf.equation 1)). After less than two seconds, the client can switch to the higher video quality at time instance t_2, since there are enough symbols received for successful decoding of base and EL. In the following, all BL symbols of SB=-1 are lost due to a burst error. However, due to the LA-FEC protection and the longer time interleaving in the EL, there are r_1=3 EL symbols received, which is enough to fulfill the decoding probability of the LA-FEC EL (cf. equation 4) and to also recover the BL (cf. equation3).

As we can derive from the example, the tune-in time to the EL quality is function of the instantaneous channel conditions. Without errors, the tune-in time to the EL depends on the ratio of source to parity symbol, which is also referred to as code rate (CR). With the given value of CR_1=0.5, the error free tune-in time is $t_{tune-in}^{error_{free}}(EL)$=$IL_1 \cdot t_{channel} \cdot CR_1$=2 seconds. Note that at this point the full error correction capability is not yet reached because not all symbols of the decoded FEC source blocks have been received. The maximum error correction capabilities is reached at time instance t_4, which directly depends on the interleaving length of the EL. In case of an early tune-in in the error free case, the service robustness increases automatically over time.

The perceived quality of a user is illustrated in Fig. 5 in comparison with a state-of-the-art single layer transmission. In order to provide a comparable time diversity of both services, the single layer stream is provisioned with the same interleaving length like used in the SVC EL.

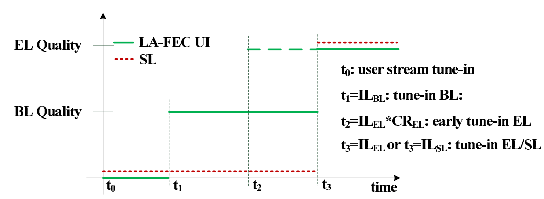

Figure 5. Experienced quality of users after service tune-in with LA-FEC UI single layer (SL) stream, both with the same interleaving length.

In case of a single layer stream, a user who tunes into the service at time instance t_0 needs to wait until t_3 to be able to start play-out. The waiting time depends on the interleaving length or FEC source block length that can be more than 10 seconds. In case of the LA-FEC UI scheme, the user can start play-out the BL quality earlier at t_1. In case of an error free channel, the user can even tune into the EL at time instance t_2. In error case, the user can tune into the highest quality at time instance t_3.

IV. Simulation

This section shows the performance of the LA-FEC UI (LA-FEC) scheme in comparison with standard FEC schemes (ST-FEC) and single layer transmission (SL). The simulation are based on real encoded video streams, which are encapsulated within RTP/UDP/IP protocols. As FEC scheme we assume the Layer-Aware Raptor code [7]. As a transmission channel we use a Gilbert-Elliot model with parameters taken from simulation conditions defined in MPEG [10]. The performance is shown with respect to the FEC performance in terms of FEC block error rate and with respect to the resulting video quality in terms of PSNR.

A. Media stream, FEC, and interleaving

The video streams have been encoded with the JSVM software [10]. The SL stream has been encoded at QVGA

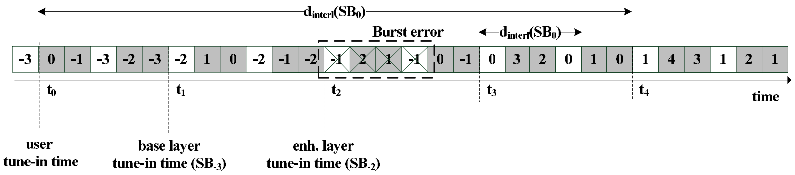

Figure 4. Exemplary transmission scenario over a burst erasure channel with a service tune-in at time instance t_0.

and 30fps using the High Profile at 570 kbps with GOPsize of eight frames and a RAP frequency of one per second. The SVC stream is encoded with coarse grain scalability (CGS) at the same resolution and frame-rate using the Scalable High Profile of H.264/AVC. The overall SVC stream shows a coding penalty compared to the SL stream of 10%, which results in an overall bit-rate of 627 kbps. The largest share with 70% of the overall bit-rate is allocated in the EL, while the remaining 30% is allocated in the BL stream. The average video quality of the SL and SVC stream is 34.5 dB in terms of PSNR and the SVC BL quality is 31.0 dB. The network abstraction units (NALUs) of the video stream are encapsulated into the RTP protocol with a target maximum transmission unit (MTU) size of 1500 byte. The RTP packets are encapsulated into UDP and IP protocol and the resulting IP stream is forwarded to the FEC Framework. The FEC generation is done in a way that each FEC source block covers one second of data and contains one RAP.

The FEC source block generation and the generation of repair data follows the IETF FecFramework specified in RFC 6363. In order to align the SL and the SVC flows, the total amount of overall service bit-rate should be the same for both of them. This means that SL will have more parity data than SVC. Therefore, we will consider a code rate of 0.5 for SL, $CR_{SL} = 0.5$, and a slightly higher code rate for the SVC, $CR_{SVC} \cong 0.56$. For the two considered SVC-FEC schemes, equal error protection (EEP) is applied with the same code rate for each layer. Note that unequal error protection (UEP) with a stronger protection for the BL does not improve the performance within the considered burst channel with an average burst error length equal to the FEC source block length (see Section IV-B). Simulations have shown a similar performance like the EEP setting. For the conducted simulation results, we assume the requirement of a service tune-in time of one second, like specified in the MBMS requirements [3]. Therefore, the FEC source block length and thereby the introduced delay of the FEC is set to d_{FEC}=1 second. This value leaves enough room for other system components to comply with the maximum channel switching time of two seconds like given by the 3GPP MBMS requirements.

For sake of simplicity, within this paper we consider two exemplary cases, without interleaving and with interleaving length of IL=5, for both single layer and SVC. In the SVC case, the interleaving is only applied to the EL, for assuring the defined tune-in time, following the LA-FEC UI approach. Note that the single layer case with IL=5 does not fulfill the fast tune-in condition and is only integrated as reference case.

B. Channel

For simulation of a mobile channel, we use a Gilbert-Elliot model, which parameters are derived from the evaluation criteria for AL-FEC in MPEG [10]. The model assumes a fixed averaged burst length ($ABEL$) and varying average loss probability, which is denoted by the erasure probability p_{er}. For each erasure probability we conducted 10000 repetitions of the simulation with different random seeds. For the conducted simulations we assumed an $ABEL$=1 second, which corresponds to the FEC source block length of 1 second.

C. Results

The graphical results in Fig. 6 compare the performance of the SL scheme with the SVC ST-FEC and the SVC LA-FEC approaches. The performance is evaluated in terms of FEC block error rate versus the error probability. For a given layer, the FEC block error rate is defined as the ratio between the number of FEC blocks, which can not be decoded and the total number of FEC blocks per media layer.

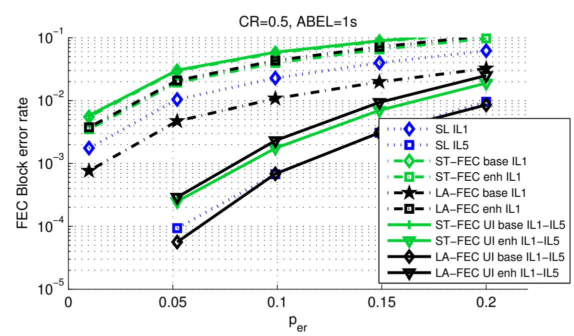

Figure 6. FEC block error rate of selected settings comparing LA-FEC UI, ST-FEC, and SL.

As can be derived from Fig. 6, the LA-FEC UI approach outperforms both the ST-FEC UI and the SL IL1 approaches. As clearly shown in Fig. 6, with IL=1 in both layers, the LA-FEC BL outperforms the SL and ST-FEC approaches like similarly shown in [7]. While the EL performance is weaker with respect to the SL approach. The same trend can be observed when an interleaver length of five seconds (IL5) is applied to the EL. The best performance is provided by the LA-FEC base IL1-IL5, which means no interleaver for BL and an interleaver factor of 5 for EL. With LA-FEC UI, the performance of both layers is significantly improved. This due to the intrinsic nature of the LA-FEC UI approach, which allows the BL to benefit from the longer interleaver applied to the EL. Contrary to that, with ST-FEC only the EL benefits from the longer time interleaving. This is due to the fact that in the ST-FEC approach each layer is protected independently. Despite the increased EL performance in ST-FEC, the overall video quality does not benefit due to the media dependencies between BL and EL, as also shown in the PSNR results in Fig. 7. It is worth noticing that LA-FEC UI IL1-IL5 shows a significantly better performance like the SL IL1 case, where the latter is the only setting for SL that fulfills the tune-in time requirements of one second. Considering the reference case with SL IL5, which has the drawback of a long tune-in time of five seconds, we can observe that at low erasure probability (p_{er}) between 0.01 and 0.05, both the SL with IL5 and the LA-FEC BL IL1-IL5 are always successfully decoded. At higher erasure probabilities, the BL of LA-FEC UI IL1-IL5 shows an equal performance compared to the SL IL5 case. Whereas the EL performs slightly worse compared to BL and SL case. However, this does not have a significant impact on the overall video quality like shown in Fig. 7.

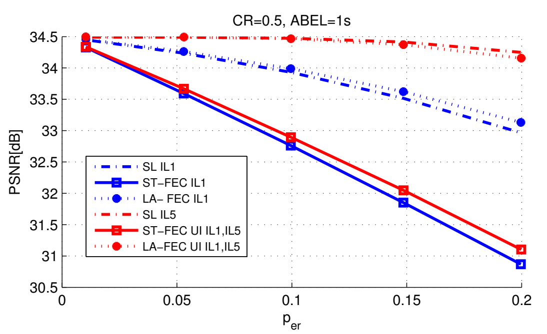

Figure 7. Video quality in terms of PSNR of selected settings comparing LA-FEC UI, ST-FEC, and SL.

As can be observed, in terms of measured quality the performance of LA-FEC UI IL1-IL5 and SL IL5 are very similar and they both largely outperform the ST-FEC scheme. But we should always take into consideration that the SL IL5 case does not fulfill the tune-in requirements of our service and is only used as upper bound for performance comparison. If we want to fulfill the tune-in constraints we need to compare the LA-FEC UI IL1-IL5 results with SL IL1. The gain of the LA-FEC UI scheme is quite significant. At a PSNR quality of 34.1 dB, LA-FEC UI IL1-IL5 allows to overcome a 0.13 higher channel erasure rate p_{er} with respect to the SL IL1 case at the same PSNR quality.

V. Conclusion

This work presents LA-FEC UI as a novel scheme that brings together for the first time long time interleaving with fast zapping. A detailed description of the scheme is shown that combines Scalable Video Coding (SVC) with Layer-Aware FEC (LA-FEC) and unequal time interleaving (UI) with a sophisticated transmission scheme. Simulation results in a Gilbert-Elliot model report the benefit of the LA-FEC UI scheme compared to traditional schemes. The proposed method has a comparable performance with respect to the upper-bound case given by single layer with long time interleaving while providing at the same time a very fast tune-in time to the service, which allows to fulfill real-world system conditions.

References

[1] 3GPP TS 26.346, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs (Release 10)," 2011.

[2] P. Kelley, "Overview of the DVB-SH specifications," *International Journal of Satellite Communications and Networking*, vol. 27, no. 4-5, pp. 198–214, 2009.

[3] 3GPP TS 22.246, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Multimedia Broadcast/Multicast Service (MBMS) user services; Stage 1 (Release 10)," 2011.

[4] M. Rezaei, I. Bouazizi, V.K.M. Vadakital, and M. Gabbouj, "Optimal channel changing delay for mobile tv over dvb-h," in *In Proc. of IEEE International Conference on Portable Information Devices (PORTABLE07)*, 2007, pp. 1–5.

[5] J. Maisonneuve, M. Deschanel, J. Heiles, W. Li, H. Liu, R. Sharpe, and Y. Wu, "An overview of IPTV standards development," *Broadcasting, IEEE Transactions on*, vol. 55, no. 2, pp. 315–328, 2009.

[6] B. Sayadi, Y. Leprovost, S. Kerboeuf, M.L. Alberi-Morel, and L. Roullet, "MPE-IFEC: An enhanced burst error protection for DVB-SH systems," *Bell Labs Technical Journal*, vol. 14, no. 1, pp. 25–40, 2009.

[7] C. Hellge, D. Gómez-Barquero, T. Schierl, and T. Wiegand, "Layer-aware forward error correction for mobile broadcast of layered media," *Multimedia, IEEE Transactions on*, vol. 13, no. 3, pp. 551–562, 2011.

[8] H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the scalable video coding extension of the H.264/AVC standard," *Circuits and Systems for Video Technology, IEEE Transactions on*, vol. 17, no. 9, pp. 1103–1120, 2007.

[9] A. Shokrollahi and Michael Luby, "Raptor codes," *Foundations and Trends in Communications and Information Theory*, vol. 6, no. 3-4, pp. 213–322, 2011.

[10] ISO/IEC JTC1/SC29/WG11 MPEG2012/m23962, "Report of CE on D1-FEC: Delivery 1-Layer AL-FEC," San Jose, USA, February, 2012.