

**Digital Video Broadcasting (DVB);
IP Datacast: Content Delivery Protocols (CDP)
Implementation Guidelines;
Part 2: IP Datacast over DVB-SH**



Reference

DTS/JTC-DVB-274-2

Keywords

broadcast, DVB

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

Individual copies of the present document can be downloaded from:
<http://www.etsi.org>

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status.
Information on the current status of this and other ETSI documents is available at
<http://portal.etsi.org/tb/status/status.asp>

If you find errors in the present document, please send your comment to one of the following services:
http://portal.etsi.org/chaircor/ETSI_support.asp

Copyright Notification

No part may be reproduced except as authorized by written permission.
The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2010.
© European Broadcasting Union 2010.
All rights reserved.

DECT™, PLUGTESTS™, UMTS™, TIPHON™, the TIPHON logo and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members.

3GPP™ is a Trade Mark of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.

LTE™ is a Trade Mark of ETSI currently being registered
for the benefit of its Members and of the 3GPP Organizational Partners.

GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

Contents

Intellectual Property Rights	6
Foreword.....	6
Introduction	6
1 Scope	7
2 References	7
2.1 Normative references	7
2.2 Informative references.....	8
3 Definitions and abbreviations.....	8
3.1 Definitions.....	8
3.2 Abbreviations	8
4 System overview	9
5 Delivery protocol for Real-Time Streaming Services	9
5.1 Specification.....	9
5.1.1 IFEC in early decoding mode technique (normative)	9
5.1.1.1 Proposed Architecture (informative).....	9
5.1.1.2 Parameters (normative)	10
5.1.1.2.1 Table of parameters (informative).....	10
5.1.1.2.2 Slowdown rate (α) (normative)	10
5.1.1.2.3 Other parameters (informative)	10
5.1.1.2.4 Adst_delay_min (informative)	11
5.1.1.2.5 Adst_delay_max (normative)	11
5.1.1.2.6 Examples (informative)	11
5.1.1.2.7 SDP additional parameters (normative).....	12
5.1.1.3 Action at IFEC Decoder level (normative)	12
5.1.1.3.1 Initialisation.....	12
5.1.1.3.2 Delayed decoding	12
5.1.1.4 RTP Proxy Definition (informative)	13
5.1.1.4.1 RTP Retimestamping.....	13
5.1.1.4.2 RTCP	13
5.1.1.5 Example (informative)	13
5.1.1.5.1 D = 0.....	13
5.1.1.5.2 D = B + S.....	14
5.1.1.5.3 Summary	14
5.2 Usage Guidelines (informative)	14
5.2.2 Audio issue	14
5.2.2.1 Description of the problem.....	14
5.2.2.2 Proposals for solving this problem.....	14
5.2.2.2.1 Before player level	15
5.2.2.2.2 At player level	15
6 Delivery protocols for File Delivery Services.....	15
6.1 Specification.....	15
6.2 Usage Guidelines.....	15
6.2.1 General.....	15
6.2.2 Relation to IPDC over DVB-H CDP Implementation Guidelines	16
6.2.3 Use and configuration of application layer FEC.....	16
6.2.3.1 Introduction.....	16
6.2.3.2 Use Cases and Measures of Interest	17
6.2.3.3 Configuration Parameters.....	18
6.2.3.4 Summary of simulation results.....	19
6.2.3.4.1 Overview	19
6.2.3.4.2 Selected Performance Results for LMS Channels	20
6.2.3.4.3 Selected Performance Results for Terrestrial Channels.....	23

6.2.3.4.4	Varying Bitrate Operations.....	24
6.2.3.4.5	Trading Physical Layer FEC with Application Layer FEC.....	25
6.2.3.4.6	Using MPE-IFEC optimized for file delivery.....	26
6.2.3.5	Recommended Parameter Settings.....	27
6.2.3.6	Advanced Use Cases	28
6.2.3.6.1	File interleaving.....	28
6.2.3.6.2	Variable Bitrate Transmission and Statistical Multiplexing.....	29
6.2.3.6.3	Pseudo-Streaming.....	29

Annex A (informative): Simulation results for Streaming Delivery30

A.1	Validation of the early decoding technique.....	30
A.1.1	Simulation setup.....	30
A.1.2	Evolution of correction power over time.....	31
A.1.2.1	LMS-SUB channel.....	31
A.1.2.2	LMS-ITS channel	33
A.1.3	HRD Validation.....	35
A.1.3.1	Principle.....	35
A.1.3.2	Results	35
A.1.3.3	Conclusion	37
A.1.4	Subjective quality evaluation	37
A.1.4.1	Introduction.....	37
A.1.4.2	Comments on video	38
A.1.4.2.1	On the video Errors	38
A.1.4.2.2	Audio impact.....	39
A.1.4.2.3	Channel switching time duration.....	39
A.1.4.2.4	Audio/video quality	39

Annex B (informative): Simulation results for File Delivery Services.....40

B.1	Introduction	40
B.2	System Configuration Options	40
B.2.1	Overview	40
B.2.2	Physical Layer Configuration Options	41
B.2.3	Link Layer Configuration Options	42
B.2.4	Application Layer Configuration Options	43
B.3	Test Cases.....	43
B.4	Evaluation Criteria	44
B.5	Simulation Methodology	44
B.6	Simulation Results.....	45
B.6.1	Reported Data.....	45
B.6.2	LMS.....	45
B.6.2.1	Small File Size: 128 kB	46
B.6.2.2	Medium File Size: 1 024 kB	47
B.6.2.3	Large File Size: 4 096 kB	49
B.6.2.4	File Interleaving: 945 kB, FLUTE symbol size 920 byte	51
B.6.2.5	Results with IFEC optimized for file delivery	53
B.6.3	Terrestrial	54
B.6.3.1	File Size: 118 kB	55
B.6.3.2	File Size: 236 kB	55
B.6.3.3	File Size: 472 kB	56
B.6.3.4	File Size: 945 kB	56
B.6.3.5	File Size: 1 890 kB	58
B.6.3.6	File Size: 3 780 kB	58
B.6.3.7	File Size: 7 562 kB	60

Annex C (informative): Formula demonstration.....61

C.1	IFEC decoder delivery time	61
-----	----------------------------------	----

C.2 Channel switching transition duration.....	62
Annex D (informative): Bibliography.....	63
History	64

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (<http://webapp.etsi.org/IPR/home.asp>).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This Technical Specification (TS) has been produced by Joint Technical Committee (JTC) Broadcast of the European Broadcasting Union (EBU), Comité Européen de Normalisation ELECtrotechnique (CENELEC) and the European Telecommunications Standards Institute (ETSI).

NOTE: The EBU/ETSI JTC Broadcast was established in 1990 to co-ordinate the drafting of standards in the specific field of broadcasting and related fields. Since 1995 the JTC Broadcast became a tripartite body by including in the Memorandum of Understanding also CENELEC, which is responsible for the standardization of radio and television receivers. The EBU is a professional association of broadcasting organizations whose work includes the co-ordination of its members' activities in the technical, legal, programme-making and programme-exchange domains. The EBU has active members in about 60 countries in the European broadcasting area; its headquarters is in Geneva.

European Broadcasting Union
CH-1218 GRAND SACONNEX (Geneva)
Switzerland
Tel: +41 22 717 21 11
Fax: +41 22 717 24 81

The Digital Video Broadcasting Project (DVB) is an industry-led consortium of broadcasters, manufacturers, network operators, software developers, regulatory bodies, content owners and others committed to designing global standards for the delivery of digital television and data services. DVB fosters market driven solutions that meet the needs and economic circumstances of broadcast industry stakeholders and consumers. DVB standards cover all aspects of digital television from transmission through interfacing, conditional access and interactivity for digital video, audio and data. The consortium came together in 1993 to provide global standardisation, interoperability and future proof specifications.

The present document is part 2 of a multi-part deliverable covering the IP Datacast Content Delivery Protocol implementation guidelines over DVB-H and DVB-SH as identified below:

Part 1: "IP Datacast over DVB-H";

Part 2: "IP Datacast over DVB-SH".

Introduction

IP Datacast is an end-to-end broadcast system for delivery of any types of digital content and services using IP-based mechanisms optimized for devices with limitations on computational resources and battery. An inherent part of the IPDC system is that it comprises of a unidirectional DVB broadcast path that may be combined with a bidirectional mobile/cellular interactivity path. The present document gives the implementation guidelines on the use of the Content Delivery Protocols in IP Datacast over DVB-SH system. [5] is a corresponding document guidelining on the use of Content Delivery Protocols in IP Datacast over DVB-H system.

1 Scope

The present document provides necessary adaptations of the Content Delivery Protocols specified [3] for IP Datacast over DVB-H for the usage in DVB-SH. In particular, the document provides specification as well as usage and implementation guidelines on the use of Content Delivery Protocols (CDP) and different reliability control techniques in IP Datacast over DVB-SH system for the whole delivery chain from the network to the terminal in a DVB-SH system.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
 - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
 - for informative references.

Referenced documents which are not found to be publicly available in the expected location might be found at <http://docbox.etsi.org/Reference>.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

- [1] ETSI TS 102 585 (V1.1.2): "Digital Video Broadcasting (DVB); System Specifications for Satellite services to Handheld devices (SH) below 3 GHz".
- [2] 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 3 GHz".
- [3] ETSI TS 102 472 (V1.3.1): "Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Content Delivery Protocols".
- [4] ETSI TS 102 005: "Digital Video Broadcasting (DVB); Specification for the use of Video and Audio Coding in DVB services delivered directly over IP protocols".
- [5] ETSI TS 102 591-1 (V1.3.1): "Digital Video Broadcasting (DVB); IP Datacast: Content Delivery Protocols (CDP) Implementation Guidelines Part 1: IP Datacast over DVB-H".
- [6] ETSI TS 102 471 (V1.3.1): "Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Electronic Service Guide (ESG)".

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

- [i.1] "Implementation of the digital phase vocoder using the fast Fourier transform" by Portnoff, taken from IEEE Transactions on Acoustics, Speech, and Signal Processing, Volume 24, Issue 3, Jun 1976, pages 243 - 248.
- [i.2] An overlap-add technique based on waveform similarity (wsola) for high quality time-scale modification of speech, Werner VERHELST and Marc ROELANDS, Vrije Universiteit Brussel, Faculty of Applied Science, dept. ETRO/DSSP, Pleinlaan 2, B-1050 Brussels, Belgium.
- [i.3] ETSI TS 102 584: "Digital Video Broadcasting (DVB); DVB-SH implementation Guidelines".
- [i.4] ITU-T SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS; Infrastructure of audiovisual services - Coding of moving video; Advanced video coding for generic audiovisual services.
- [i.5] IETF RFC 4566: "SDP: Session Description Protocol by M. Handley, V. Jacobson & C.Perkins (July 2006)".
- [i.6] IETF RFC 4234: "Augmented BNF for Syntax Specifications: ABNF by D. Crocker, Ed. & P. Overell (October 2005)".
- [i.7] ETSI TS 102 772: "Digital Video Broadcasting (DVB); Specification of Multi-Protocol Encapsulation - inter-burst Forward Error Correction".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in [1], [2], [3], [4], [5] and [i.3] apply.

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in [1], [2], [3], [4], [5] and [i.3] and the following apply:

ADST	Application Data Sub Table
FEC	Forward Error Correction
FLUTE	File Delivery over Unidirectional Transport
IFEC	Inter burst FEC
IP	Interne Protocol
LMS	Land Mobile Satellite
LMS-ITS	Land Mobile Satellite Intermediate Tree Shadow
LMS-SUB	Land Mobile Satellite Sub Urban
QEF	Quasi Error Free
RTCP	Real Time Transport Control Protocol
RTP	Real Time Transport protocol
SRTP	Secured RTP

4 System overview

The system overview presented in clause 4 of TS 102 591-1 V1.3.1 [5] fully applies to DVB-SH case.

In this guidelines document, the implementation of different use cases based on the specified delivery methods and in accordance with TS 102 472 [3] and TS 102 471 [6] is discussed.

5 Delivery protocol for Real-Time Streaming Services

5.1 Specification

For real-time streaming services the same procedures as specified in [3], clause 5 apply except for the following modifications that relate to the support of IFEC decoding and related audio issues.

5.1.1 IFEC in early decoding mode technique (normative)

The use of IFEC can greatly improve the quality of the displayed video but it introduces a delay between the emission time and the displayed time. In permanent mode this delay would not be noticed by the users; however at start or when switching channels the delay, which can reach several tens of seconds, can not be tolerated. To avoid this problem the early decoding mode technique is proposed.

5.1.1.1 Proposed Architecture (informative)

The basic idea is to decode the first packets received after switching channels before having received all the IFEC, then progressively using more and more of the IFEC sections so that after a transition period the complete decoding power will be reached. During the transition period the packets will be transmitted slower than normal to the player.

In order to achieve this behaviour the following architecture is proposed.

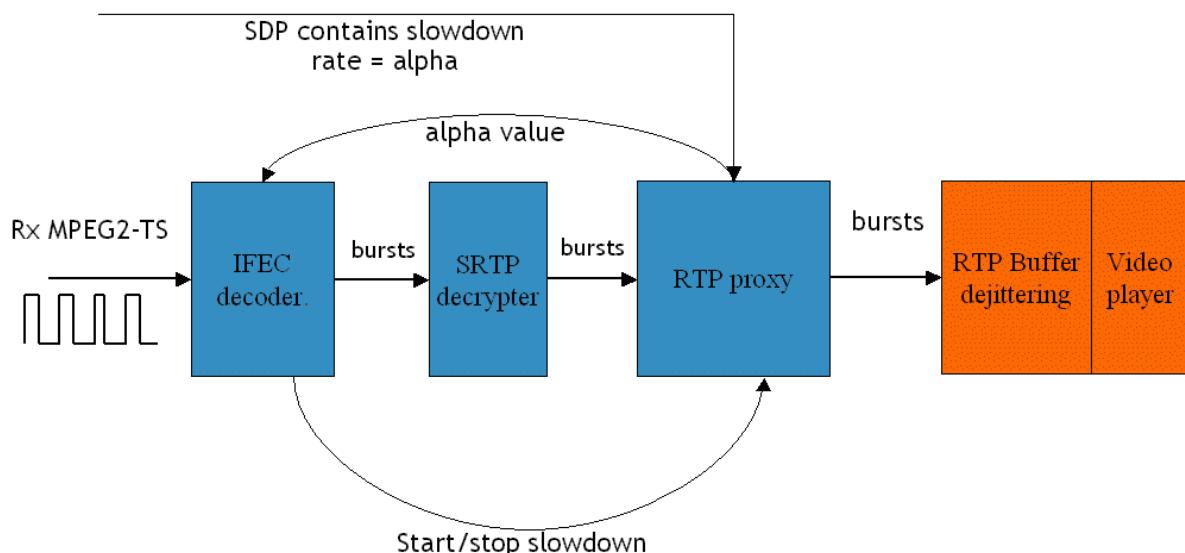


Figure 1: Complete steps from receiving raw bursts to playing the video

The changes compared to normal IFEC decoding are:

- Modification of the IFEC decoder so that it outputs packets without waiting the reception of all the IFEC packets necessary to its reconstruction.
- Introduction of the RTP Proxy module which modifies the RTP timestamps that will be used by the player, so that it handles the slowdown correctly.
- Adding new parameters in the SDP file (see below).

5.1.1.2 Parameters (normative)

5.1.1.2.1 Table of parameters (informative)

The early decoding technique needs the following parameters to be defined.

Table 1: Parameter list

Name	Context	Location	Description	Status
Alpha (α)	Content	SDP	Slowdown rate	New
Adst_delay_min	Receiver	Register	Minimal value of ADST delay (value in seconds)	New
Adst_delay_max	IFEC	Computed from IFEC parameters	Maximal value of ADST delay (value in seconds)	New
RTP_Freq	Content	SDP	RTP timestamp frequency	existing

5.1.1.2.2 Slowdown rate (α) (normative)

The slowdown rate is defined as the ratio of displayed frames per second to original frames per second during the transition mode: $\alpha = \text{fps}_{\text{out}} / \text{fps}_{\text{in}}$

This parameter is specific to the content and is defined in the SDP file (see clause 5.1.1.2.7). The RTP proxy reads the value of the slowdown rate α from the SDP file. Then it is transmitted to the IFEC decoder through the use of an internal register.

Different values are possible:

- $\alpha = 0$ means total slow down is allowed, which is equivalent to re-buffering. This re-buffering is allowed at any time (e.g. at channel switching instant like alpha = 1 but why not later on); it also means that the IFEC decoder can work in early mode (without the full correction capacity).
- For values such as $0 < \alpha < 1$, α is the maximum allowed slowdown rate for this content. This slow down can happen at any time (e.g. at channel switching instant, but also later on); it means that the IFEC decoder can work in early decoding mode (without the full correction capacity).
- $\alpha = 1$ means no slow down allowed; the terminal has to wait for ADSTmax at channel switching instant, before starting displaying; $\alpha = 1$ means than only late decoding is allowed.

5.1.1.2.3 Other parameters (informative)

The IFEC decoder notifies the RTP proxy about when to start and stop the slowdown mechanism.

The IFEC decoder sends to RTP proxy the values of adst_delay_{min} and adst_delay_{max}.

The RTP proxy needs to know the sampling frequency used in the RTP timestamps for each flow, and retrieves this information from the SDP file.

5.1.1.2.4 Adst_delay_min (informative)

adst_delay_{\min} value is defined by configuration of the terminal. The role of adst_delay_{\min} is to ensure a minimal decoding capacity when switching channels, but it should be kept small so that the user will not have to wait too long before having access to his program.

The minimum adst_delay_{\min} value is $1 * \text{burst_period}$ because you need at least one burst to display.

The maximum adst_delay_{\min} value is $\max(B + S - D; B) * \text{burst_period}$.

5.1.1.2.5 Adst_delay_max (normative)

The parameters adst_delay_{\max} is the maximum delay, expressed in seconds, that can be used between the moment of reception of a burst by the IFEC Decoder and the time at which this burst is decoded and emitted towards the player.

adst_delay_{\max} value is derived from IFEC parameters, so that maximum decoding capacity is reached:

$$\text{adst_delay}_{\max} = \max(B + S - D; B) * \text{burst_period}$$

When $D = 0$, $\text{adst_delay}_{\max} = B + S$.

When $D = B + S$, $\text{adst_delay}_{\max} = B$.

Some examples are given below:

- With IFEC parameters $B = 6$ $S = 4$ $D = 0$, $\text{adst_delay}_{\max} = 10 * \text{burst_period}$.
- With IFEC parameters $B = 6$ $S = 4$ $D = 10$, $\text{adst_delay}_{\max} = 6 * \text{burst_period}$.

5.1.1.2.6 Examples (informative)

- $\alpha = 0$:
 - re-buffering is allowed;
 - decoding status depends on adst_delay_{\min} and subsequent re-buffering durations: terminal is in early mode until adst_delay_{\min} and cumulated re-buffering durations equal to adst_delay_{\max} .
- $0 < \alpha < 1$:
 - re-buffering is not allowed, slow-down is possible;
 - slow-down happens after adst_delay_{\min} , total duration of slow-down depends on B , S and D parameters (see clause 5.1.1.3 for explicit formulation);
 - a typical value of α is obtained with $\text{fps}_{\text{in}} = 30$ frames per second, $\text{fps}_{\text{out}} = 24$ frames per second:
 $\alpha = 0,8$.
- $\alpha = 1$:
 - neither slow-down, nor re-buffering are allowed;
 - decoding status depends on adst_delay_{\min} and D :
 - $\text{adst_delay}_{\min} = 1 * \text{burst_period}$: $B-1$ ADT following current burst can never be received:
 - $D = 0$: this is equivalent to working without iFEC because FDT corresponding to current burst can never be received;
 - $D > 0$: some early decoding is possible.
 - $1 * \text{burst_period} < \text{adst_delay}_{\min} < B * \text{burst_period}$: this is equivalent to working in early decoding since complete set of ADT can never be received, whatever the value of D .

- adst_delay_min $\geq B * \text{burst_period}$: the ADT set can be received completely after some bursts, the FDT reception itself is dependent on D value:
 - adst_delay_min $> (B + S + 1 - D) * \text{burst_period}$: some FDT can never be received and the receiver will always be in early decoding;
 - adst_delay_min $\leq (B + S + 1 - D) * \text{burst_period}$: at some point, all FDT will have been received and the receiver will be in late decoding.

5.1.1.2.7 SDP additional parameters (normative)

It is proposed to add a new attribute in the SDP file in order to indicate the value of alpha concerning the current media. The syntax follows the "value attribute" specified page 21 of [i.5]: a=<attribute>:<value>.

The proposed attribute name is the ASCII string "dvbsh-max-slowdown-rate". The valid values are defined below. The formal description of this attribute is defined by the following ABNF [i.6] syntax:

`dvbsh-slowdown-attribute = "a=dvbsh-max-slowdown-rate:" slowdownrate CRLF slowdownrate = "1" ["." *"0"] / 1*"0" / *(("0") "." *(DIGIT))`

This attribute should be defined as a "session-level" attribute, because all the media present in the session should use the same value in order to be synchronized.

EXAMPLE:

- a=dvbsh-max-slowdown-rate: 0,85;
- Sets the maximum slowdown rate for the current session to 85 %.

5.1.1.3 Action at IFEC Decoder level (normative)

5.1.1.3.1 Initialisation

The RTP proxy extracts from SDP file sampling frequency used in the RTP timestamps for each decoded flow.

The IFEC decoder notifies the RTP proxy about when to start and stop the slowdown mechanism.

The IFEC decoder sends to RTP proxy the values of adst_delay_{min} and adst_delay_{max}.

5.1.1.3.2 Delayed decoding

Instead of using a fixed delay between TS packets reception time and IP packets delivery time, the IFEC decoder uses a variable delay varying from adst_delay_{min} to adst_delay_{max}.

Assuming:

- channel change occurs at t = t₀;
- burst number k was received at time t_k.

The IFEC decoder will use all available correction and send burst k at time t'_k such that

t'_k = t_k + adst_delay_{min} + (1 / α - 1) * (t_k - t₀). The demonstration of this formula can be found in annex C.

Therefore all IFEC sections present in bursts received before t'_k will be used to help correct the burst number k.

5.1.1.4 RTP Proxy Definition (informative)

5.1.1.4.1 RTP Retimestamping

The progressively delayed data bursts issued by the IFEC decoder are treated by a module called RTP proxy which role is to modify the RTP packets in order to ensure coherence between the slowed down packet rate and the presentation time (timestamp) present in the RTP header.

The formula used by the RTP proxy to modify the RTP timestamps is:

- $TS' = TS / \alpha + C$.

Where C is a constant defined as:

- $C = TS_0 * (1 - 1 / \alpha) - (adst_delay_{max} - adst_delay_{min}) * rtp_freq$.

The constant C is chosen so that when the slowdown is finished, the RTP proxy does not recalculates the timestamps any more but the continuity exists between the last modified timestamp and the first unchanged timestamp.

The channel switching transition time is given by the formula demonstrated in annex C:

$$t'_{fin} - t_0 = adst_delay_{max} + (adst_delay_{max} - adst_delay_{min}) / (1 / \alpha - 1).$$

NOTE: The RTP proxy does not change the burst rate and resends them as fast as possible. The RTP buffer present in the media player is responsible for the task of sending the packets to the player at the right time.

5.1.1.4.2 RTCP

The sender report present in the RTCP packet contains, among others, the fields:

- RTP Timestamp.
- NTP Timestamp (as MSW and LSW).

Both fields are modified so that they are synchronized with the slowed-down stream.

The modified value for the NTP timestamp is given by the following formula:

$$ntp' = ntp / \alpha + C'$$

where C' and ntp_0 are constant defined as:

- $C' = (1 - 1 / \alpha).ntp_0 + (adst_delay_{max} - adst_delay_{min})$.
- ntp_0 is the equivalent NTP timestamp of the first received packet:

$$ntp_0 = ntp_{\text{first received RTCP packet}} - (t_{\text{first received RTCP packet}} - t_{\text{first received RTP packet}}).$$

The RTP timestamp is modified in the same way as in a normal RTP packet:

$$TS' = TS / \alpha + C$$

5.1.1.5 Example (informative)

5.1.1.5.1 D = 0

At channel switching instant, the IFEC decoder is set up with the minimal ADST-delay value (e.g. ADST delay = 1 s). As there is no IFEC the quality of the first bursts may be low when the signal reception is not so good.

Then ADST-delay used to decode the bursts progressively increases up to the value of $(B + S)$, causing a slowdown of the video playing. IFEC protection is progressively recovered, and the quality of the displayed video increases. For streaming playout, slow-down of 0,9, $B + S = 10$ s, $D = 0$, the delay to get full IFEC protection is $T = 88,8$ s after channel switching instant and $T = 87,7$ s after the first second of display. After this delay, the end user benefits from an improved quality (the ESR5 criterion is fulfilled).

5.1.1.5.2 $D = B + S$

With the same hypothesis as in previous paragraph, the value of $adst_delay_{max}$ is now equal to B .

For streaming playout slow-down of 0,9, $B = 6$, $S = 4$, $D = 10$, the delay to get full IFEC protection is now $T = 49,7$ s after channel switching instant, 48,8 s after first second of display. After this delay, the end user benefits from an improved quality (the ESR5 criterion is fulfilled).

5.1.1.5.3 Summary

**Table 2: Channel switching transition examples
for $B + S = 10$ (given in number of bursts)**

B	6
S	4
adst_min	1

D	0	10
adst_max	10	6

alpha	zapping transition	
1	N/A	N/A
0,9	91	51
0,8	46	26
0,7	31	18
0,6	24	14
0,5	19	11

5.2 Usage Guidelines (informative)

For real-time streaming services the same procedures as specified in [5], clause 5 apply except for the following modifications that relate to the support of IFEC decoding and related audio issues.

5.2.2 Audio issue

5.2.2.1 Description of the problem

At video level, playing frames at a smaller rate than the nominal rate only results to the display of the same frame for a longer delay, which may remain unnoticed to the viewer. But for audio, if the sample is played at its nominal rate, the next sample will not be ready to play immediately after the previous one is finished, and the introduced delay will be silent, resulting in noticeable "clicks" to the listener.

5.2.2.2 Proposals for solving this problem

Trivial time-scaling approach can solve the problem of clicks but change the pitch of the played audio. The pitch difference is spectacular even when changing speed to 90 % of original speed.

The receiver can compensate for the lowered pitch of the audio signal by performing a frequency translation of the decoded audio signal so as to bringing its central frequency to approximately the central frequency of the original signal.

It is well known how to change the pitch of an audio signal without changing its duration ("pitch shifting"), by using such techniques as Time Domain Harmonic Scaling (see for example [i.7]) or Phase Vocoder (see for example [i.1]).

However, in our case, we would like to maintain the pitch while changing the duration, which is the reverse problem. Some algorithms like WSOLA are able to time scale audio samples without affecting the pitch with a very low computational CPU load (see [i.2]).

5.2.2.2.1 Before player level

Processing the audio change before the player level means that the process in charge has to decompress the audio sample, then modify it thanks to the selected algorithm and finally re-encode it into the format specified in the container or SDP.

The advantage is that this approach will be compatible with all players.

5.2.2.2.2 At player level

If the processing is done at player level, the gain as compared to the previous approach is that only one decoding is needed, rather than two decoding phases and one encoding. This means a better performance.

The drawback is that the players will have to support this technique.

6 Delivery protocols for File Delivery Services

6.1 Specification

For the delivery protocol for file delivery services the same procedures as specified in [3], clause 6 shall apply.

For the associated delivery procedures for file delivery services the same procedures as specified in [3], clause 7 shall apply.

For the application layer FEC for file delivery services the same procedures as specified in [3], clause 8 shall apply.

6.2 Usage Guidelines

6.2.1 General

In IPDC file delivery is used to distribute objects, for example ESG data or multimedia files. File delivery is the most common application on today's IP networks. It is used to distribute content such as multimedia clips, high quality music files, digital newspapers, software download, etc. When compared to streaming, file delivery poses even more stringent requirements in terms of reliability and integrity of the data, as even a single bit error can corrupt the whole file and make it useless for the receiver. Hence, error-free reception of the files is typically required. On the other hand, latency and delay constraints are usually significantly relaxed as file delivery applications start processing the information after the entire file has been received.

File delivery services over DVB-SH for the distribution of multimedia data may be quite attractive as delays of "real-time" streaming delivery are anyways expected to be required to compensate the fading phenomena in land-mobile satellite reception. Interleaving depth of 10 seconds and more may have to be considered to compensate satellite obstructions. In this case multimedia data may be delivered in a non real-time to a storage device on the receiver. The user may access the local storage and may consume the multimedia data on demand, for example by browsing a catalogue of the available multimedia data or by user-generated playlists, etc. Also, terminal applications may make use of the file delivery procedures to enable radio and mobile TV services that partly or completely rely on data on the local storage that has been delivered in non-realtime.

For the delivery itself in bidirectional service setups, TCP is generally used for reliable data transmission. However, in IPDC the delivery of files over unidirectional links is based on FLUTE. The delivery may be setup in a scheduled session, for which the start of the delivery of the files is announced to the receiver population beforehand such that all interested receivers can join the download session at the appointed time. In a carousel distribution receivers can join the download session at any point of time independently of other receivers in an asynchronous fashion. The transmission of the files is virtually unbounded and the receivers only leave the carousel until they have received the file. Two types of file carousel services can be distinguished, static and dynamic. Whereas the former delivers a file with the same content, in dynamic carousels individual files may change dynamically.

DVB-SH reuses IPDC over DVB-H Content Delivery Protocols (CDPs) [3]. The usage of these CDPs is strongly aligned to the usage guidelines in DVB-H [5], clause 6. Clause 6.2.2 of the present document provides the relation to IPDC over DVB-H CDP Implementation Guidelines. As the transmission conditions in DVB-SH are generally different, especially in satellite reception, but also for terrestrial reception due to the differences of DVB-H and DVB-SH physical layer, also the configuration, usage and performance of the application layer FEC in FLUTE for IPDC over DVB-SH content delivery protocols is different. This aspect is addressed in clause 6.2.3 of the present document. The guidelines provided in the clause 6.2.3 are supported by selected simulation results in annex B.

6.2.2 Relation to IPDC over DVB-H CDP Implementation Guidelines

For the setup and operation of a file delivery session the procedures as specified in [5], clause 6.1 shall apply.

For the post-delivery procedures of a file delivery session the procedures as specified in [5], clause 6.2 shall apply.

For the delivery of web pages (webcasting) within a file delivery session the procedures as specified in [5], clause 6.4 shall apply.

For the FLUTE sessions and memory management within a file delivery session the procedures as specified in [5], clause 6.5 shall be apply.

For the FDT Instance ID wraparound problem within a file delivery session the procedures as specified in [5], clause 6.6 shall be apply.

6.2.3 Use and configuration of application layer FEC

6.2.3.1 Introduction

Application Layer FEC (AL-FEC) is most valuable in cases where the data set to be delivered is to be spread over more than one DVB-SH Time Slice Burst. In networks where services are provided based on data sets which are delivered over multiple DVB-SH time slice bursts it is strongly recommended that AL-FEC is supported by all terminals.

The data set to be delivered may consist of a single file or multiple files which are required together in order to deliver a service. Data may be spread over more than one Time Slice burst when the data set to be delivered is larger than a single burst, or if multiple files are interleaved such that each file is spread over multiple bursts.

For data sets which are delivered over multiple DVB-SH time slice bursts, in the absence of AL-FEC, efficient file delivery is only possible within the coverage area where the frame error rate (the fraction of MPE frames which could not be recovered using either physical layer FEC, MPE-FEC and/or MPE-IFEC) is close to zero.

The AL-FEC operates by providing additional "repair packets" which can be used to reconstruct the data set as soon as a sufficient quantity of data packets has been received, independent of the mix of source and repair packets that have been received. The key difference from the case of simple repetition is that the terminal in general does not receive a duplicate copy of the same packet (in which case the duplicated reception is useless) and thus the transmission is significantly more efficient. For AL-FEC, a sufficient quantity of received packets is usually only slightly larger than the size of the data to be delivered.

Many of issues that have been discussed in the DVB-H CDP implementation guidelines [5], clause 6.3 are also applicable of DVB-SH based transmission. In particular some configuration guidelines as discussed in [5], clause 6.3.2, the interleaving of multiple files in [5], clause 6.3.5 and the use of sub-blocking feature in [5], clause 6.3.6. equally apply to DVB-SH.

6.2.3.2 Use Cases and Measures of Interest

This clause provides guidelines for configuration of AL-FEC for the recommended case in which all terminals support AL-FEC decoding. In terms of user perception, file download delivery is to a large extent binary, i.e. for each user the file is either fully recovered and the user is satisfied or the file is not fully recovered and the user is unsatisfied. Clearly, not all users can always be satisfied, and file distribution services are usually operated such that a certain percentage of users are satisfied. Unsatisfied users are not necessarily excluded from the download service, and may rely on post-delivery methods on point-to-point systems to complete the file recovery.

Configuration of AL-FEC should be based on the target quality of service for the file delivery session. There are four metrics which measure the quality of service, generally defined for users at the edge of the coverage area:

- 1) the transmission time or file reception time T ;
- 2) the channel bandwidth required B ;
- 3) delivery probability (i.e. the probability that a terminal has received the file);
- 4) the coverage area.

Given any two of these three metrics as fixed parameters, then AL-FEC can be configured to optimize the third. Alternatively, intermediate configurations may be used which provide a balance between the three metrics.

Generally, the delivery time in seconds is a suitable measure, but in case the physical layer configurations are fixed for the transmission of a certain burst size and repetition period, the transmission or file reception time can also suitably be measured in terms of the number of bursts that are required to distribute a file in a reliable manner, i.e. to for example 95 % of the users at the coverage edge.

Another interesting metric results from the combination of the first and second metric from above. For a specific delivery probability and a specific coverate area and file of size F , we are interested in the bandwidth efficiency of the file delivery that is defined as $F / (T * B)$ generally expressed in units bit/s/Hz or in %.

With the CDP download delivery protocols in place, different services can now be realized:

- **Scheduled Distribution without Time Constraints:** In a scheduled broadcast service, files are distributed once within a session and all users should join the session at the very beginning. In this case the main measure of interest is by the necessary radio resources, for example expressed by number of required bursts. Also the bandwidth efficiency can well express the performance of a specific configuration. For simplicity, we consider the case of distributing a single file. Of less relevance in this case is the experienced "download time", i.e. how long it takes to receive the file.
- **Time-Constrained Distribution:** In this service scenario we consider scheduled distribution with the additional constraint that files of a certain aggregate size need to be distributed within a certain amount of time. In this case it is of interest to evaluate the radio resources required to transport a file of a certain size within this amount of time. The hard time-limit may lead in less efficient distribution as time-diversity may not be exploited fully. The bandwidth efficiency may be used to understand the reduction in efficiency due to this time constraint.
- **Carousel Services:** File delivery using carousel is a possibly time-unbounded file delivery session in which a fixed set of files are delivered. When using FLUTE the file delivery carousel is realized as content delivery session whereby file data tables and files are sent continuously during a possibly time-unbounded session. In case no AL-FEC is available, the data must be repeated (referred to as NoCode option in FLUTE environment). In the case that the Raptor code is used, the data transmitted for a given file generally includes repair symbols generated by Raptor encoding in addition to the original source symbols. In particular, file reception time is minimized if symbols are never repeated until all 65,536 possible symbols (source and repair) have been sent. With this, the fountain property of the Raptor codes can be optimally exploited. In terms of configuration, the transmitter has only limited options, basically only the transmit rate can be selected. The measure of interest for the receiver is the amount of time it takes to acquire the file. The objective is to minimize the time that it takes for a receiver to acquire all files in one carousel session to minimize on time and battery consumption.

The simulations as conducted can not directly be assigned to any of the above services, but we can deduct interpretations for the configuration of each of the above services from the simulation results.

6.2.3.3 Configuration Parameters

When implementing a file delivery service over DVB-SH, the service operator does have several design options to enable an efficient and reliable file delivery service, taking into account the requirements of the application that makes use of the file delivery services. The file delivery service itself may be typically have assigned the following parameters:

- The size of the files within the file delivery service.
- Typical and maximum latencies for the file delivery.
- The distribution of one file or multiple files within the service.
- The reliability requirements of the file delivery service.

On the DVB-SH transport level, the system designer can choose or has decided on the following parameters:

The settings in the DVB-SH physical layer:

- Modulation scheme:
 - OFDM (SH-A): {bandwidth; FFT; GI; modulation order};
 - TDM (SH-B): {bandwidth; roll-off factor; modulation order}.
- FEC code rate: there are 8 possibilities from 1/5 to 2/3.
- Interleaver depth and configuration:
 - Class 1 short;
 - Class 2 uniform long, or uniform late.

The settings in the transport and link layer:

- repetition interval;
- number of time-slice bursts;
- size and duration of the time slice bursts.

The settings in the link layer:

- MPE-IFEC parameters: code rate, interleaver (B, S, D), encoding matrices (ADST number of rows).
- Resulting service bitrate at file level as defined by F/T.

The reception environment:

- Vehicular/handheld receiver.
- Satellite/terrestrial coverage with their respective channel models (LMS Sub Urban, LMS Intermediate Tree Shadow; TU6).
- Terminal speed, signal and interference strength, etc.

The CDP for file delivery provides additional configuration options to trade reliability versus bandwidth efficiency, namely.

The payload and packet sizes of the FLUTE packets. These should typically be in the same range or at most the size as the DVB-SH physical layer packets, i.e. in the range of 800 to 1 400 bytes.

Within the IPDC CDP, two different FEC types can be chosen, namely the NoFEC code or the Raptor FEC. For NoFEC code, it typically needs to be decided how often the file is repeated. For Raptor codes it needs to be decided in advance, how much FEC overhead is applied.

Typically also the delivery rate of the CDP FLUTE session is of relevance, but the delivery rate is determined by the resources provided on the physical layer. The delivery rate may even be varying, as no time-critical or real-time aspects need to be considered for file delivery.

Table 3 provides a summary of the configuration parameters for file delivery within DVB-SH.

Table 3: Configuration parameters for the file delivery simulations in DVB-SH

Parameter	Values
DVB-SH Physical Layer Parameters	
Modulation Scheme	SH-A: {bandwidth; FFT; GI; modulation order} SH-B: {bandwidth; roll-off factor; modulation order}
Physical Layer Code Rate	{1/5; 2/9; 1/4; 2/7; 1/3, 2/5; 1/2; 2/3}
Interleaver parameters	(common multiplier; nof late taps; nof slices; slice distance; nof late increments)
DVB-SH transport Layer Parameters	
Repetition interval (also called Time Slice Cycle (s))	OFDM: 975 ms
number of time-slice bursts	Depends on physical layer parameters and targeted bit rate
Burst size (bytes)	For recommendations see DVB-SH guidelines
Burst duration (ms)	For recommendations see DVB-SH guidelines
DVB-SH Link Layer Parameters	
Link Layer FEC Code Rate	For recommendations see DVB-SH guidelines
MPE-IFEC number of rows	For recommendations see DVB-SH guidelines
Application Data Table size	Depends on code rate
MPE-FEC columns	Depends on code rate
MPE-IFEC: Burst Spread <i>B</i>	For recommendations see DVB-SH guidelines
MPE-IFEC: FEC spread <i>S</i>	For recommendations see DVB-SH guidelines
MPE-IFEC: Delay <i>D</i>	The delay is irrelevant for file delivery and can be set to <i>D</i> = 0.
Service bit-rate (kbit/s)	Depends on service, but may be lower than for example used in streaming delivery. Service bitrate may also be variable.
DVB-SH Application Layer Parameters	
FEC-ID	{NoCode, Raptor}
FLUTE payload size	Flexible, typically on the range of 800 - 1 450 byte
FEC symbol size	Typically FLUTE payload size, or FLUTE payload size is integer multiple of the FLUTE payload size. See [3] for details.
FEC overhead	For NoFEC: amount of redundantly sent source symbols. For Raptor: amount of repair symbols divided by the amount of source symbols. Guidelines are provided below and be deducted from the simulation results.

6.2.3.4 Summary of simulation results

6.2.3.4.1 Overview

Let us recall used metrics:

- 1) the transmission time or file reception time *T*;
- 2) the channel bandwidth required *B*;
- 3) delivery probability (i.e. the probability that a terminal has received the file);
- 4) the coverage area.

Let us recall how the different techniques are used:

- No IFEC: you repeat the file as many times as necessary until the delivery probability is reached; read clause B.2.2 for more details.
- IFEC: it is the same as no IFEC but instead of sending the file alone, we send also some IFEC (with the code rate accordingly set). MPE-IFEC sections are interleaved according to the *B* + *S* parameters. Read clause B.2.3 for more details.

- Raptor: we send the fixed amount of systematic symbols and an ever-increasing repair symbols, as many as needed. Please read clause B.2.4 for more details.

Individual techniques performance are evaluated in the following clauses. The following general conclusions can be drawn from these performance:

- It is always better to select the Raptor solution than any other one typically used and optimized for streaming delivery (MPE-IFEC configured for streaming, id est with code rates exceeding 1/2, class 2, etc.). This is due to the fact that, when using physical or link layer parameters optimized for streaming, there are always remaining errors that usually do not impact video streaming but affect seriously the performance of the carousel. On the contrary, the use of the AL-FEC enables to remove all remaining errors in an efficient manner.
- Configuration of the AL-FEC and its performance depends on the size of the file. Virtual bandwidth (and associated AL-FEC overhead) can vary significantly between different sizes. Recommendations are to augment the overhead when the file size lowers. So the AL-FEC configuration is file size dependent.
- When using Raptor, it is always better to increase time diversity by distributing the data over a greater number of time slice bursts, especially by interleaving transmissions (e.g. sending 2 files at a time instead of 2 files sequentially). This is applicable if available bandwidth (when we have more bandwidth than required for 1 transmission at a time) and time constraints (delivery time need not be the one of the max maximum bandwidth usage) permits it.
- MPE-IFEC can offer an alternative to AL-FEC when configured with code rates are lower than $\frac{1}{2}$. In such a case, MPE-IFEC offers following characteristics:
 - the configuration is independent (agnostic) of the file size;
 - the MPE-IFEC exhibits larger end to end delay compared to Raptor;
 - the MPE-IFEC exhibits better efficiencies compared to Raptor without interleaving but similar performances when interleaving is activated.

6.2.3.4.2 Selected Performance Results for LMS Channels

We detail here performance of carousel without FEC, with Raptor and with MPE-IFEC configured for streaming video only, additional MPE-IFEC configured for file delivery can be found in clause 6.2.3.4.6.

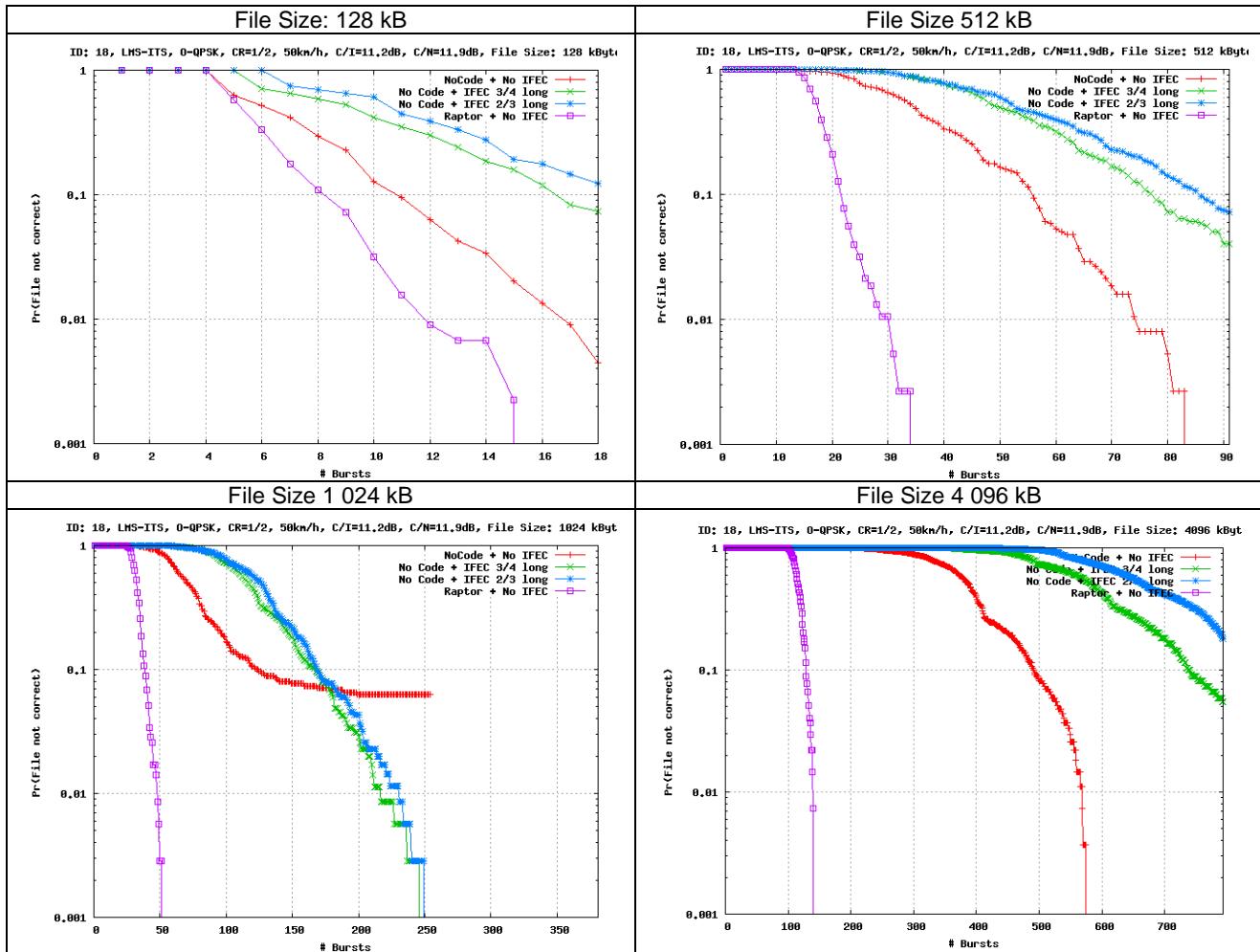


Figure 1a: File delivery simulation results for LMS-ITS, 50 km/h, O-QPSK, R = 1/2 (ID = 18)

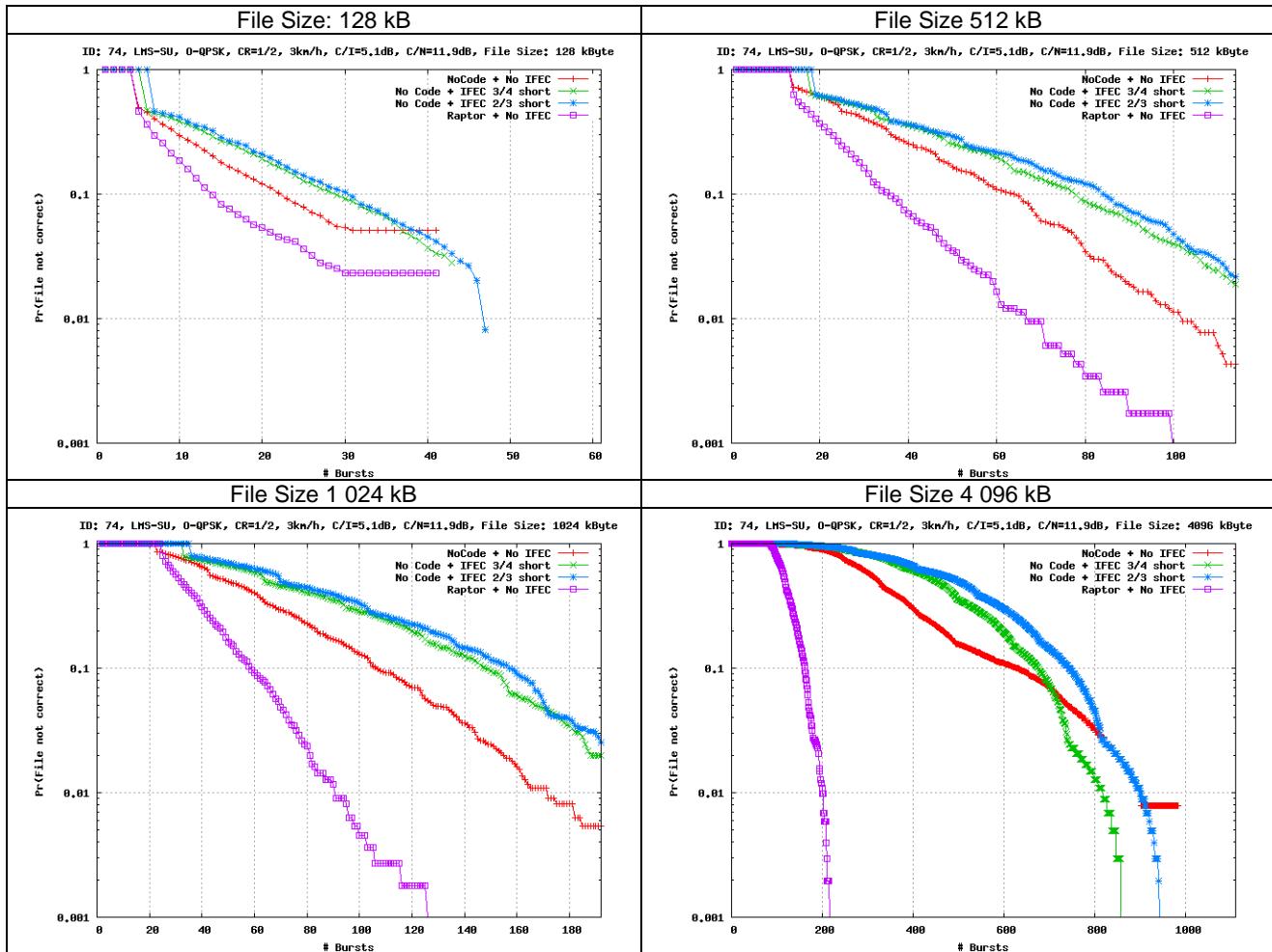


Figure 1b: File delivery simulation Results for LMS-SU, 3 km/h, O-QPSK, R = 1/2 (ID = 74)

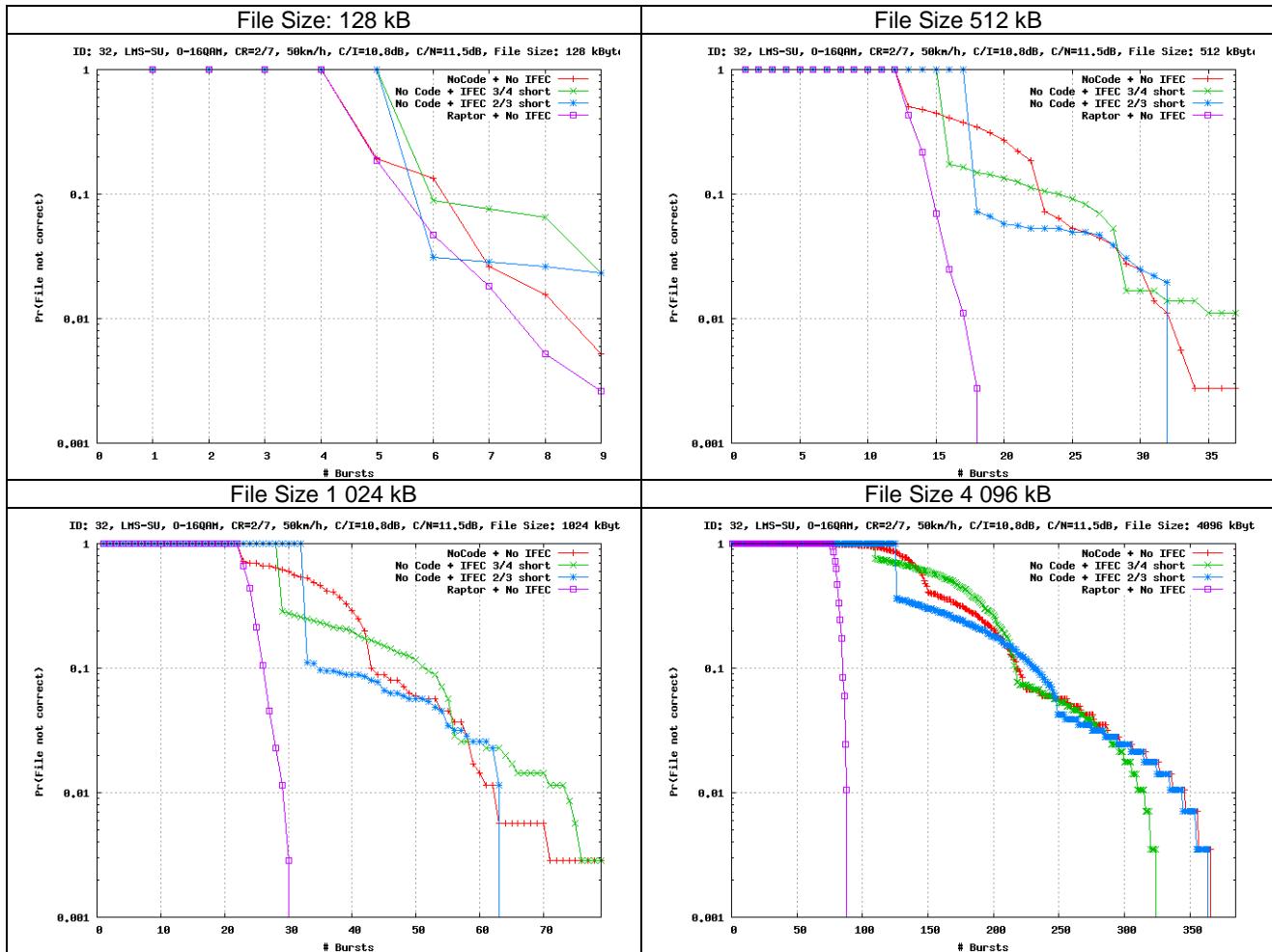


Figure 1c: File delivery simulation results for LMS-SU, 50 km/h, 16QAM, R = 2/7 (ID = 30)

6.2.3.4.3 Selected Performance Results for Terrestrial Channels

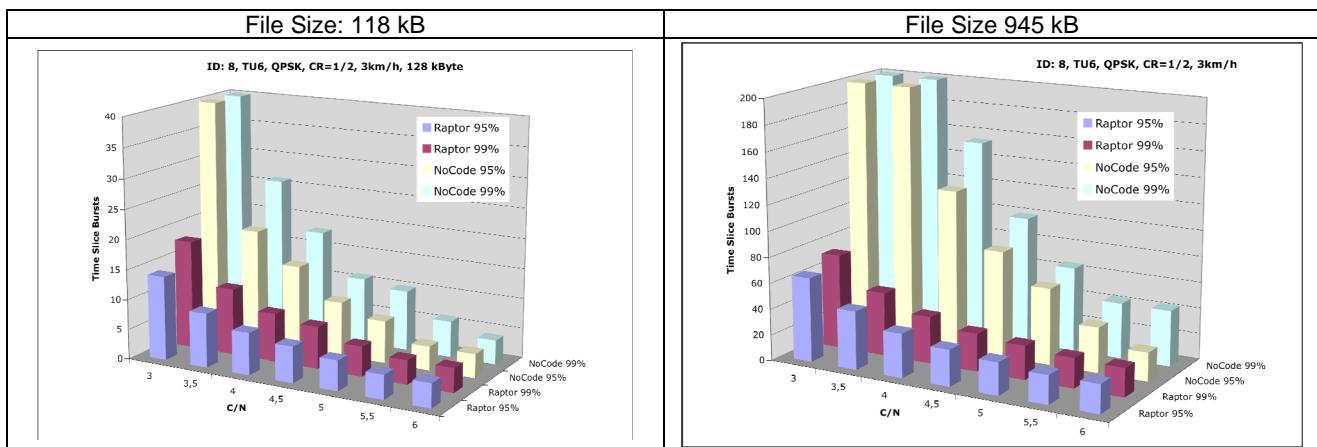


Figure 1d: Simulation Results for TU-6 QPSK 1/2 3km / h (ID = 8)

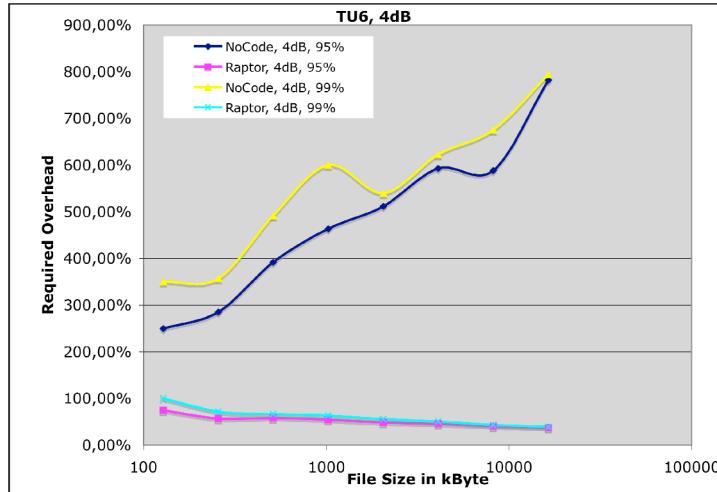


Figure 2: Required Overhead over File Size

6.2.3.4.4 Varying Bitrate Operations

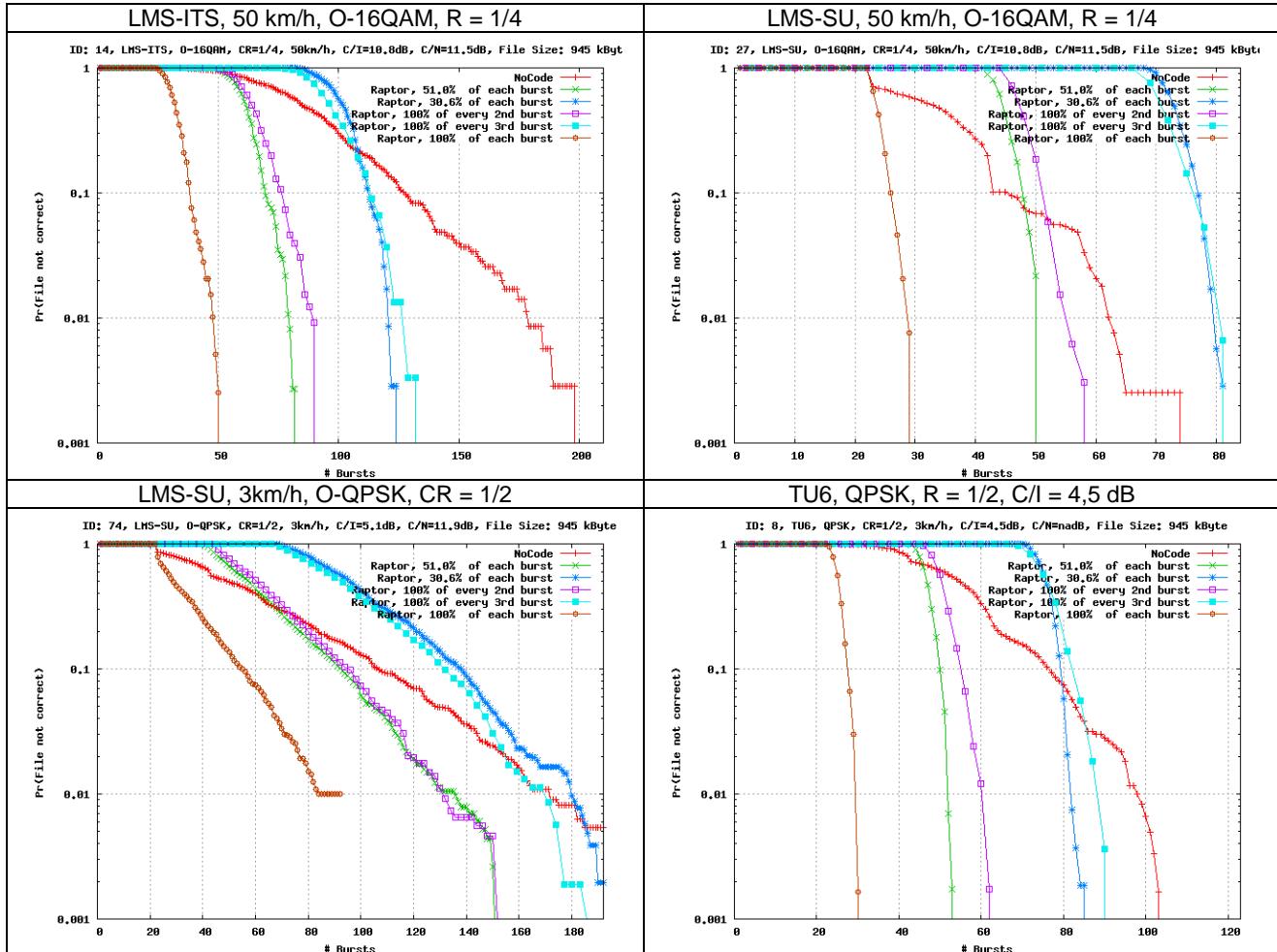


Figure 2a: Simulation Results for different bitrates: file error probability over number of occupied burst for different configurations in terms of burst share and burst time slice cycle

Table 4: Virtual Bitrate and Bandwidth Efficiency

Phy Layer			Link		Application				90		95		99		ViBR kbit/s			BWE bit/s/Hz			
ID	C/N	C/I	ID	CB	FS	FEC	Share	TS	MB	B	Pr	B	Pr	B	Pr	90	95	99	90	95	99
14	10,8	11,5	1	1	945	0	100 %	352	21	126	90,3	139	95,2	177	99,1	62,5	56,7	44,6	10,0 %	9,1 %	7,1 %
14	10,8	11,5	1	1	945	1	100 %	392	21	37	92,3	39	95,2	47	99,5	208,9	198,5	165,4	33,4 %	31,8 %	26,5 %
14	10,8	11,5	1	1	945	1	51 %	371	42	68	90,6	73	96,5	78	99,2	115,1	107,3	100,5	36,1 %	33,7 %	31,5 %
14	10,8	11,5	1	1	945	1	31 %	350	70	111	91,1	116	96,0	119	99,1	70,9	67,9	66,2	37,1 %	35,5 %	34,6 %
14	10,8	11,5	1	2	945	1	100 %	326	21	37	92,6	38	95,4	43	99,1	104,5	101,8	90,2	33,4 %	32,6 %	28,9 %
14	10,8	11,5	1	3	945	1	100 %	300	21	36	91,0	38	96,3	41	99,7	71,5	67,9	63,0	34,3 %	32,6 %	30,3 %

6.2.3.4.5 Trading Physical Layer FEC with Application Layer FEC

Table 5: Trading physical layer FEC with application layer FEC

Phy Layer			Link		Application				90		95		99		ViBR kbit/s			BWE bit/s/Hz			
ID	C/N	C/I	ID	CB	FS	FEC	Share	TS	MB	B	Pr	B	Pr	B	Pr	90	95	99	90	95	99
14	10,8	11,5	1	1	4 096	0	100 %	272	83	507	90,1	544	95,6	568	99,3	67,7	63,1	60,5	10,8 %	10,1 %	9,7 %
14	10,8	11,5	1	1	4 096	1	100 %	272	83	137	90,8	140	96,3	146	99,6	249,4	244,1	234,1	39,9 %	39,1 %	37,5 %
17	10,8	11,5	1	1	4 096	0	100 %	284	74	546	90,1	579	95,1	650	99,3	62,9	59,3	52,9	10,1 %	9,5 %	8,5 %
17	10,8	11,5	1	1	4 096	1	100 %	284	74	133	90,1	137	95,1	142	99,6	256,8	249,4	240,7	41,1 %	39,9 %	38,5 %

Let us recall the different configurations:

- Case 14: O-16QAM1o4_S.
- Case 17: O-16QAM2o7_S.

It is visible that higher phy FEC is more efficient for combination with Raptor, but less efficient with NoCode, even if the variations are limited due to the fact that physical code rate varies little.

6.2.3.4.6 Using MPE-IFEC optimized for file delivery

A link layer configuration offering a quasi error-free transmission condition can be found for each type of medium (LMS-ITS and LMS-SUB):

- For LMS-ITS: FEC code rate of 50 % and B + S of 100 are enough to provide QEF reception.
- For LMS-SUB: FEC code rate of 80 % and B + S of 90 are enough to provide QEF reception.

Using these MPE-IFEC parameters it is possible to distribute the files with a no FEC option in the application layer.

Table 6: Performance comparison for different file delivery options in LMS-ITS

File size		MPE-IFEC for file delivery CR = 1/2, B + S = 100
128 kB	Virtual bandwidth (kbps)	10
	Efficiency	35
1 024 kB	Virtual bandwidth	59
	Efficiency	40
4 096 kB	Virtual bandwidth	126
	Efficiency	40

Table 7: Performance comparison for different file delivery options in LMS-SUB

File size		MPE-IFEC for file delivery CR = 80 %, B + S = 90
128 kB	Virtual bandwidth (kbps)	11
	Latency	56
1 024 kB	Virtual bandwidth	74
	Latency	71
4 096 kB	Virtual bandwidth	185
	Latency	73

6.2.3.5 Recommended Parameter Settings

Performance of different techniques is illustrated in the following figures for different channels and different file sizes.

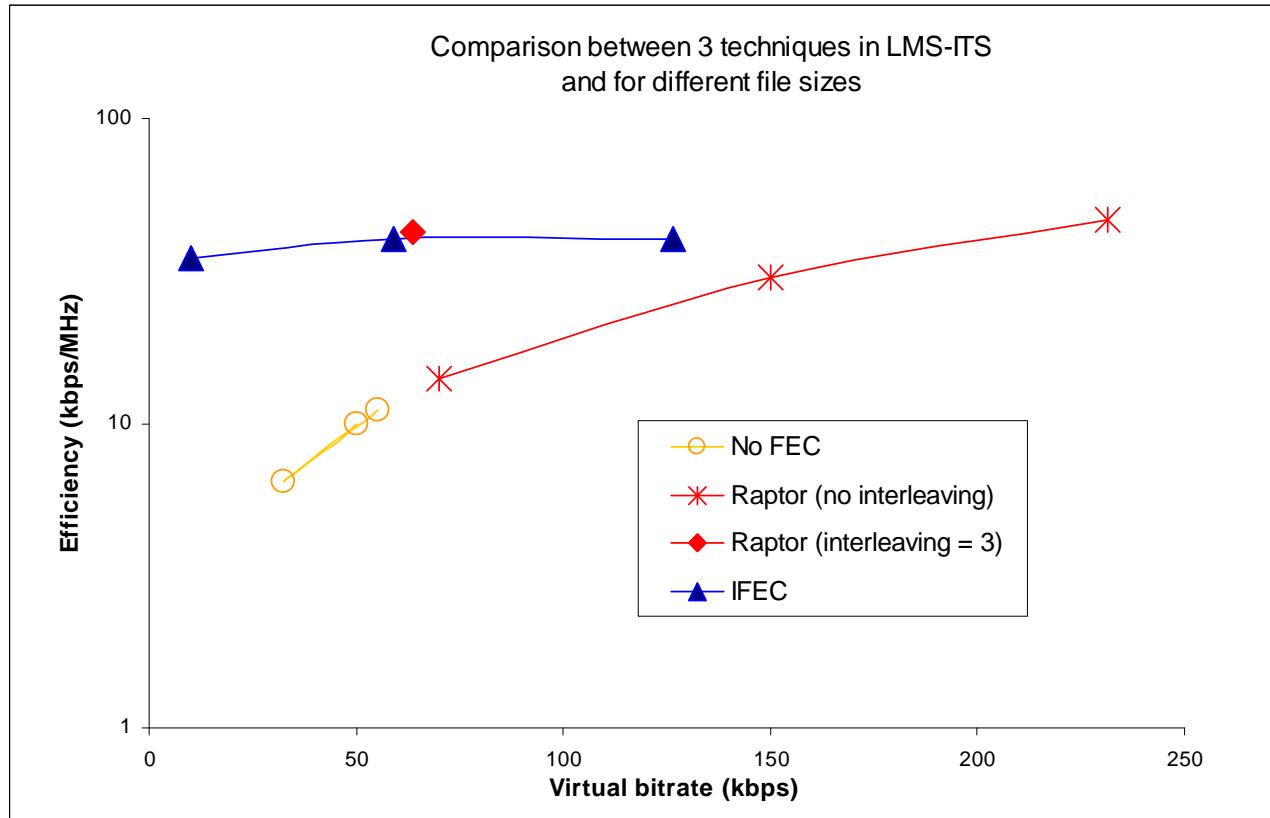


Figure 3: Comparison of 3 file delivery techniques (carousel with no FEC, carousel with MPE-IFEC, carousel with Raptor) in LMS-ITS

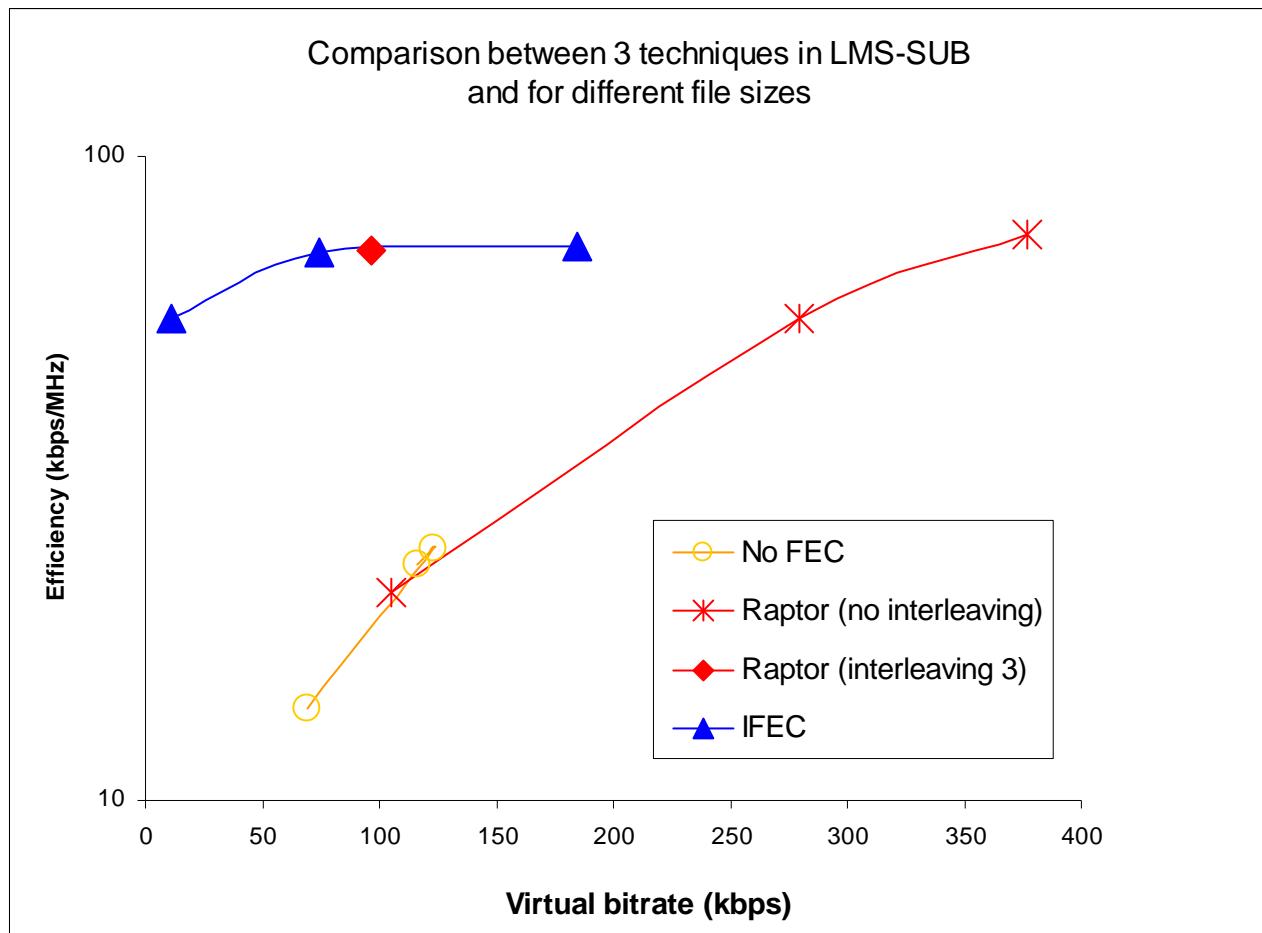


Figure 4: Comparison of 3 file delivery techniques (carousel with no FEC, carousel with MPE-IFEC, carousel with Raptor) in LMS-SUB

NOTE 1: It can be seen that the AL-FEC with Raptor generally outperforms other techniques such as AL-FEC with no FEC or MPE-IFEC.

NOTE 2: MPE-IFEC outperforms Raptor without interleaving at the expense of increased latency and therefore lowered virtual bandwidth.

NOTE 3: Raptor with interleaving can match MPE-IFEC.

NOTE 4: MPE-IFEC efficiency is rather independent of the file size, being a link layer technology, on the contrary of AL-FEC technology. On the other side UL-FEC with Raptor configuration can be tuned to actual file size automatically.

Therefore AL-FEC with Raptor offers best performance in all cases. In the specific case where latency (or virtual bitrate) is not an issue, a solution based on MPE-IFEC can provide an alternative solution agnostic of the file size.

Actual choice of the technique in such a case depends on implementation aspects such as available memory and is out of scope of the present document.

6.2.3.6 Advanced Use Cases

6.2.3.6.1 File interleaving

In case an application requires the distribution of multiple files within the service, and AL-FEC is applied, it is preferable to not sequentially transmit the files in a single FLUTE session but to either multiplex the FLUTE sessions and transmit the multiple files in parallel. This increases the time diversity and therefore generally also improves the performance and the required overhead. Alternatively, the individual files may be transmitted in a single session and the files may be included in a single file package.

6.2.3.6.2 Variable Bitrate Transmission and Statistical Multiplexing

For many applications, file distribution does not require a constant bitrate transmission or constant delivery time. The bitrate and the packet transmission frequency of FLUTE packets is very flexible generally no real-time constraints apply. Therefore, file download applications can for example be efficiently applied in combination with streaming services to compensate any bitrate variations of individual streaming services or the multiplex of streaming services. If AL-FEC is used, the bitrate adaptation may even be done by dropping packets before sending them as only a sufficient amount of symbols need to be delivered, regardless whether the symbols are source or repair symbols.

6.2.3.6.3 Pseudo-Streaming

File delivery may also be used to support services for which the user is not aware the media is not live streamed. In this case the continuous media is encapsulated in one or several 3GPP files and those files are distributed with some timing constraints. This permits that the files or the pieces of the file may be played from the terminal storage. Such a delivery permits the exploitation of long time diversity for more robust and efficient transmission. Delivery and playout are done almost independently and the buffer in the terminal may be quite large in ranges of minutes or hours.

Annex A (informative): Simulation results for Streaming Delivery

A.1 Validation of the early decoding technique

A.1.1 Simulation setup

The chosen simulator consists of several processes run in batch. Each process uses PCAP files in input and output that saves the time information together with the data. Here is a description of the task accomplished by each module:

- **IFEC Encoder:** applies time-slicing, MPE and IFEC protection. Takes as input a stream stored in a PCAP file (compliant with Tcpdump/Ethereal) corresponds to a CBR H.264 AVC video sequence streamed over RTP/UDP/IPv4. Generates at output an MPEG2-TS stream.
- **ApplyError:** applies an error pattern (in DVB SH format) to the input MPEG2-TS stream.
- **IFEC Decoder:** processes the incoming noised TS stream and performs MPE decapsulation and IFEC decoding. Also performs the slowdown: during the slowdown period it outputs packets at a rate different from reception rate. The IFEC Decoder outputs an IP stream stored in a PCAP file.
- **RTP Proxy:** recalculates the values for the RTP timestamps so that coherency is maintained during slowdown phases.
- **Trace Processor:** analyzes both input and output IP streams to compute correction power statistics.

The simulator is fed with the following parameters.

Table A.1: Simulator parameters

Function	Parameter name	Description
IFEC	B	Number of matrices for data interleaving
	S	Number of bursts for IFEC sections interleaving
	D	Transmission delay in number of bursts (used to send IFEC protection before associated data)
Time-Slicing	Burst period	Period of bursts
	Burst duration	Duration of a single burst
	MPEG2TS per burst	Fixed number of MPEG2-TS packets per burst (to be compliant with DVB packet errors trace)
Slowdown	Alpha	Slowdown value
	ADST_delay_min	Minimal time waited by decoder before decoding a burst
	ADST_delay_max	Maximal time waited by decoder before decoding a burst

The IFEC decoder function is detailed in [i.3], clause 6. The time-sliced TS input stream is defragmented to MPE or IFEC sections. ADST recovered from MPE sections are stored in an ADST buffer of fixed time length (controlled by parameter ADST-delay) before being outputted. IFEC sections are fed to the IFEC decoding module which asynchronously repairs or recovers some of the erroneous or lost ADSTs. Hence, the longest ADST-delay, the highest ADST recovery probability. It should be noted that, neither RTP buffering, nor video decoding parts are simulated, and hence, channel switching delay does not include RTP buffering and video decoding time. **The channel switching delay presented in this study stands for the additional delay introduced by the MPE-IFEC long interleaver.**

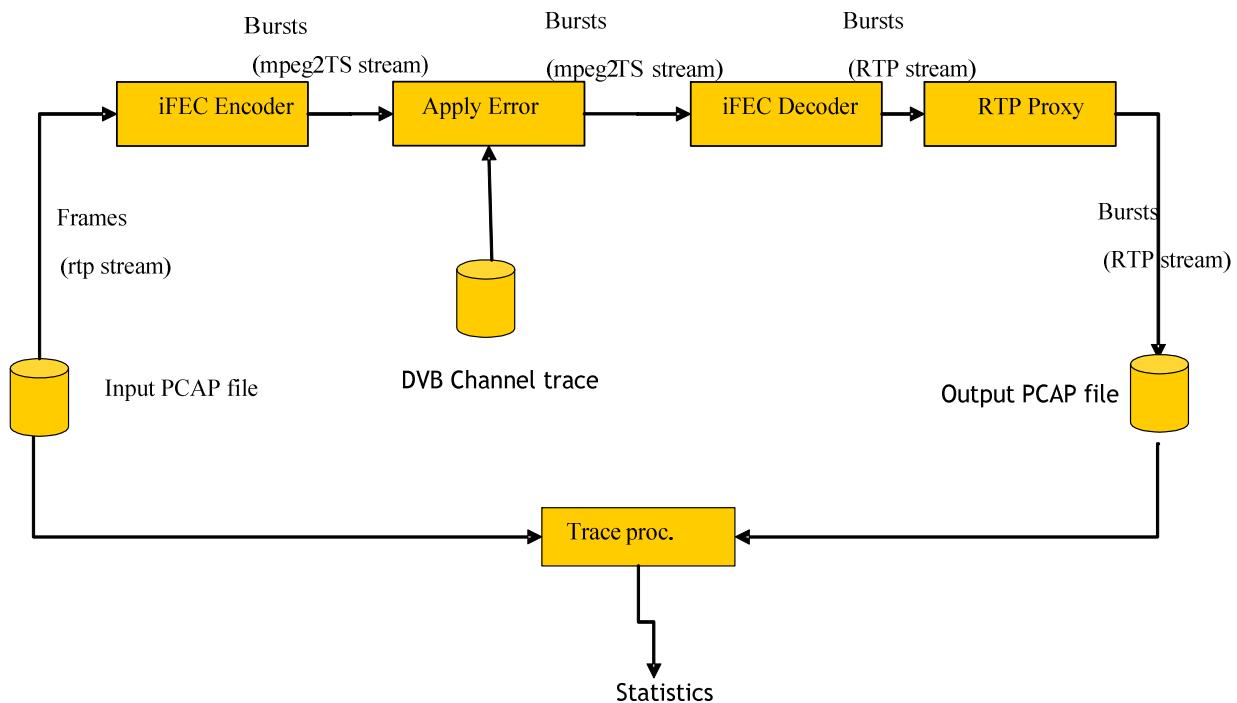


Figure A.1: Simulation set up

During each step some information is stored in text files that are later used for statistic by the TraceProcessor module. The resulting files can finally be played using the PcapReplay process and a media player such as QuickTime using MPEG4 codec is provided.

A.1.2 Evolution of correction power over time

The following graphs are the results of simulations. Decoding is performed with various values of alpha on the same stream containing missing and erroneous packets. The quality is defined here as the percentage of well-received frames computed over a number of simulations performed with different error dumps. Two channel configurations are simulated, one in LMS-SU and one in LMS-ITS. Each channel is simulated with a D equal to 0 (iFEC sections following the data), one with a D equal to B + S (iFEC sections preceding the data).

A.1.2.1 LMS-SUB channel

For this example, a suburban model (dump LMS_SUB_QPSK½_¼_50kmph_2K_8) has been chosen:

- Video at 230 kb/s.
- IFEC parameters are B = 6, S = 4, D = 0 and 10.
- Code rate ~ 0,55 to fit the burst size of 272 TP.

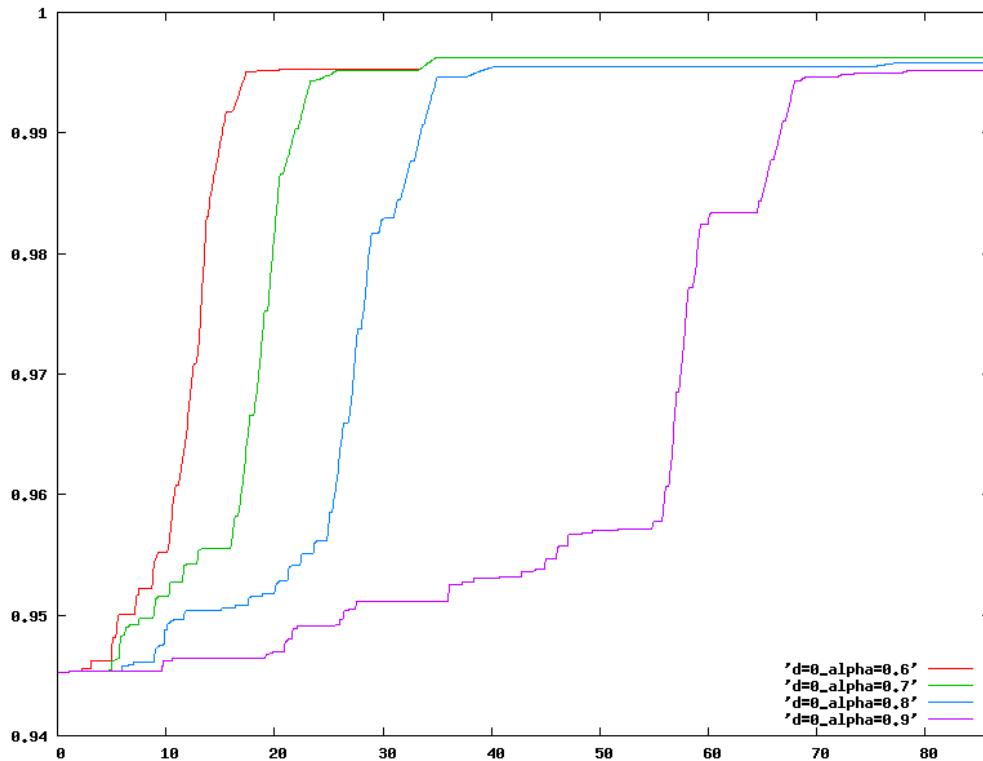


Figure A.2: Evolution of quality over time for different values of alpha, D = 0

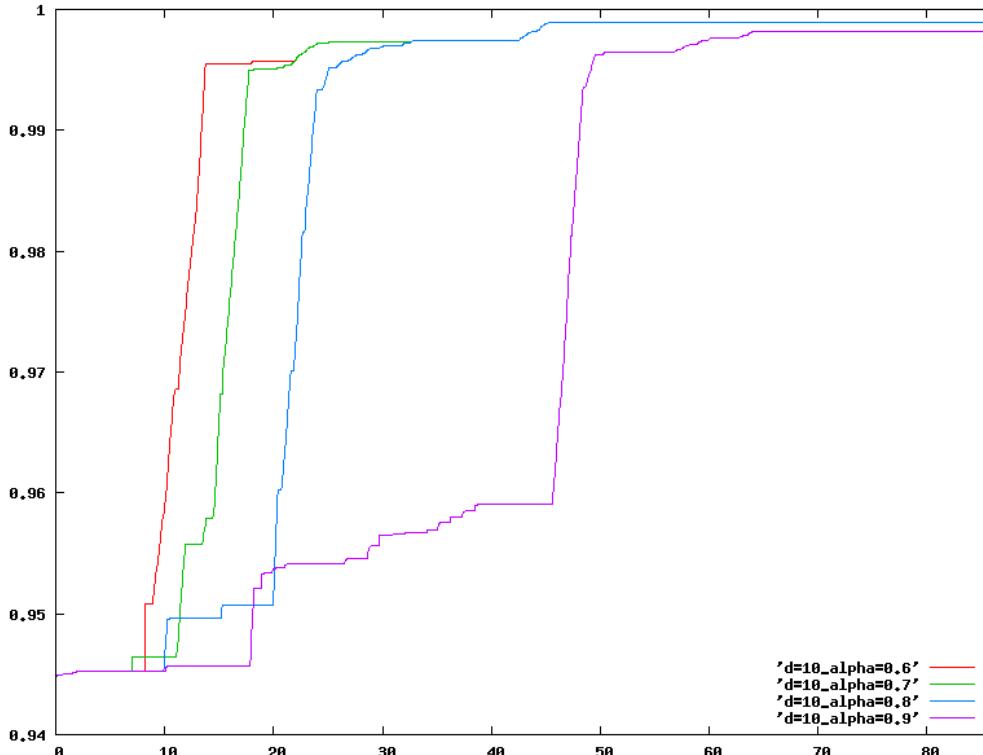


Figure A.3: Evolution of quality over time for different values of alpha, D = B + S = 10

It is visible in this figure that all curves do not reach quality = 1, where all the frames are corrected, once the permanent mode is reached. This is however compatible with the 1-ESR5 criterion equal to 100 % (not more than 1 second erroneous on any window of size 20 seconds).

Figures A.2 and A.3 show that the quality progressively increases with time, the time to reach this steady state being a function of the slow-down factor alpha:

- When alpha = 0,9, the steady state (95 % of free frame) is achieved after 87,8 seconds (after beginning of first image display) when D = 0 and only 48,8 seconds when D = B + S = 10 (reception of slice 50, display of burst 45, decoding using ADT n°41 to 50 and FDT received in slice n° 35 to 44).
- When alpha = 0,8, the steady state (99,5 % of free frame) is achieved after 43,9 seconds when D = 0 and only 24,4 seconds with D = B + S = 10 (reception of burst 25, display of burst 20, decoding using ADT 16 to 25 and FDT received in slice n° 10 to 19).
- When alpha = 0,7, the steady state (99,5 % of free frame) is achieved after 29,3 seconds when D = 0 and only 16,2 seconds with D = B + S = 10 (reception of burst 17, display of burst 12, decoding using ADT 8 to 17 and FDT received in slice n° 2 to 11).
- When alpha = 0,6, the steady state (99,5 % of free frame) is achieved after 21,9 seconds when D = 0 and only 12,2 seconds with D = B + S = 10 (reception of burst 13, display of burst 8, decoding using ADT 4 to 13 and FDT received in slice n° -2 to 7).

Therefore, for values lower than 0,7, the steady state is reached later than expected since IFEC has not yet been received when the slow-down has ended.

These curves confirm the interest of using D = B + S.

The performance variation as a function of time is the following when D = 0:

- Low between channel switching instant and the first inflection point: during this state, no IFEC decoding is possible because not enough ADT are collected at decoding time.
- High when enough ADT are collected: this occurs, when D = 0, when B - 1 ADT after the current burst are collected; this happens at a time that depends on the parameter α ; for instance when $\alpha = 0,6$, this happens at time = 11, corresponding to emission of burst 6. As a matter of fact, when decoding burst 6, we need ADT coming from burst 6 - 5 = 1 to 6 + 5 = 11. So when ADT are completely received, quality increases quickly because IFEC can now be used to improve channel losses.
- Low before end of slow-down phase: a flat curve seems to indicate that last IFEC sections do not significantly improve quality reception.

This general behaviour can also be found when D = B + S with the following main modifications due to the fact that the IFEC have been partially received before the ADT:

- the start of the steep curve happens sooner;
- the curve after the ADT have been received is even steeper.

A.1.2.2 LMS-ITS channel

For this example, an ITS model (LMS_ITS_QPSK½_¼_50kmph_2K_8) has been chosen, for a video at 200 kb/s. IFEC configuration is B + S = 36, B = S = 18, D = 0 and 36, code rate = 0,5.

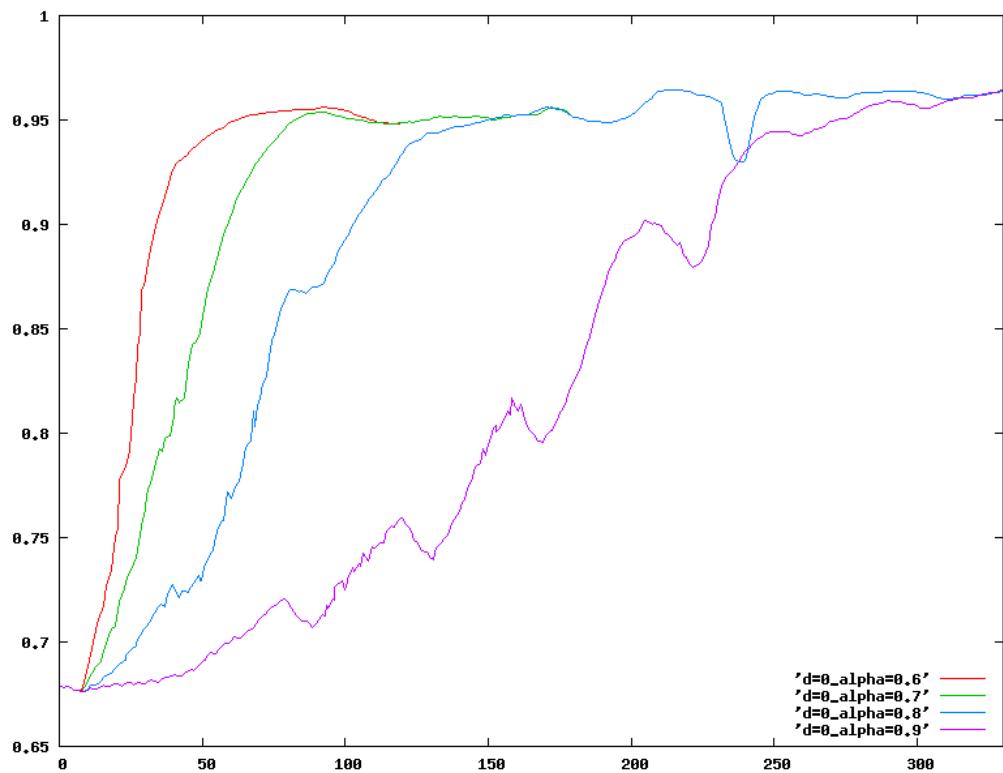


Figure A.4: Evolution of quality over time for ITS channel for different values of alpha D = 0

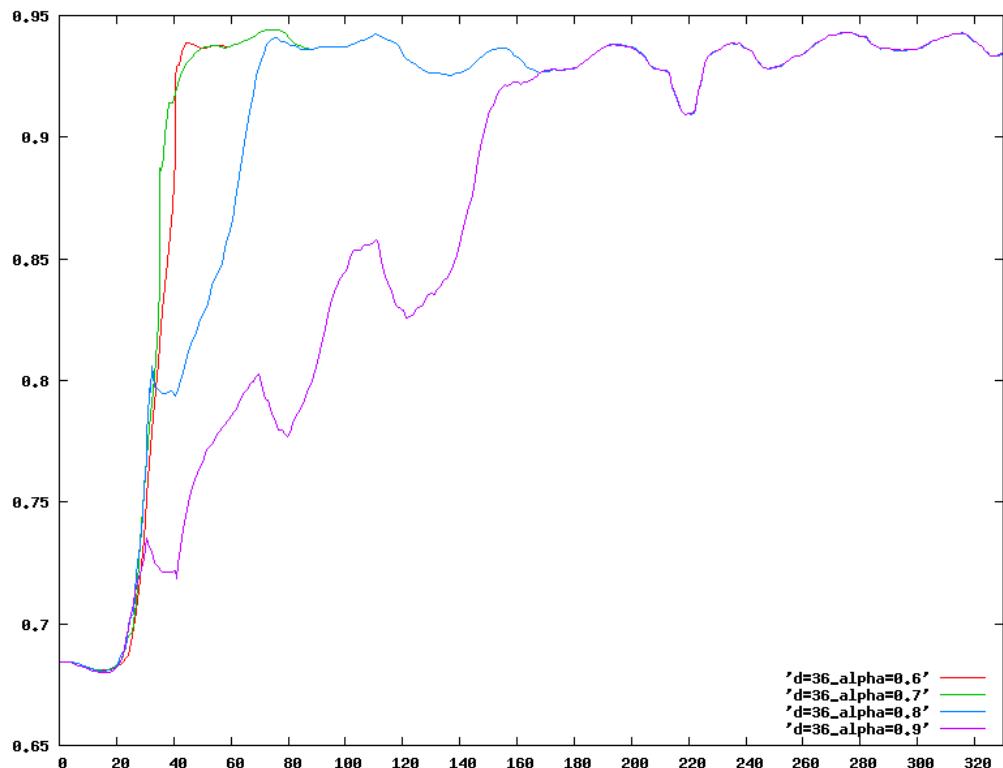


Figure A.5: Evolution of quality over time for ITS channel for different values of alpha D = 36

Figures A.4 and A.5 show that the quality progressively increases with time, the time to reach this steady state being a function of the slow-down factor alpha:

- When alpha = 0,9, the steady state (99,5 % of free frame) is achieved after 341 seconds (after beginning of first image display) when D = 0 and only 166 seconds when D = B + S = 36 (reception of slice 170, display of burst 152, decoding using ADT n°135 to 170 and FDT received in slice n° 116 to 151).
- When alpha = 0,8, the steady state (99,5 % of free frame) is achieved after 171 seconds when D = 0 and only 83 seconds with D = B + S = 36 (reception of burst 85, display of burst 63, decoding using ADT 48 to 83 and FDT received in slice n° 27 to 62).
- When alpha = 0,7, the steady state (99,5 % of free frame) is achieved after 113 seconds when D = 0 and only 55 seconds with D = B + S = 36 (reception of burst 57, display of burst 39, decoding using ADT 22 to 57 and FDT received in slice n° 3 to 38).
- When alpha = 0,6, the steady state (99,5 % of free frame) is achieved after 85 seconds when D = 0 and only 41 seconds with D = B + S = 36 (reception of burst 42, display of burst 24, decoding using ADT 6 to 41 and FDT received in slice n° -12 to 23).

Therefore, for values lower than 0,7, the steady state is reached later than expected since IFEC has not yet been received when the slow-down has ended.

A.1.3 HRD Validation

A.1.3.1 Principle

To validate conformance of the modified bit stream, we need to check HRD buffers. As slowdown method is achieved by the RTP Proxy, we need to check the validity of the RTP de-jittering buffer. There is no need to take into account the Timing SEI messages.

The bit stream conformance is achieved if the RTP buffer used by the player avoids overflow and underflow states:

- overflow state happens when the buffer cannot accommodate new packets which are therefore lost;
- underflow state happens that there is nothing to play and the display freezes.

We therefore need to check that RTP buffer does not exceed maximum value as defined in the H.264 standard in [i.4] clause A.3.1 as MaxCPBSIZE.

A.1.3.2 Results

- **Overflow case:** Different media streams have been simulated that confirm that maximum CPB size required for early decoding mode is always equal to maximum CPB size for late decoding mode, both being below the limit defined H.264 specification.
- **Underflow case:** The RTP de-jittering buffer indeed uses the RTP presentation time to decide when to consume a packet from the buffer. Without the retimestamping method, as the packets are slowed down at the output of the IFEC decoder, it may happen that the packet presentation time happens earlier than its reception time. In this case the received burst is consumed immediately after its reception by the RTP proxy and the player enters a starvation period until next burst is received etc. The RTP buffer is therefore regularly in underflow state, which may result in a bad user experience.

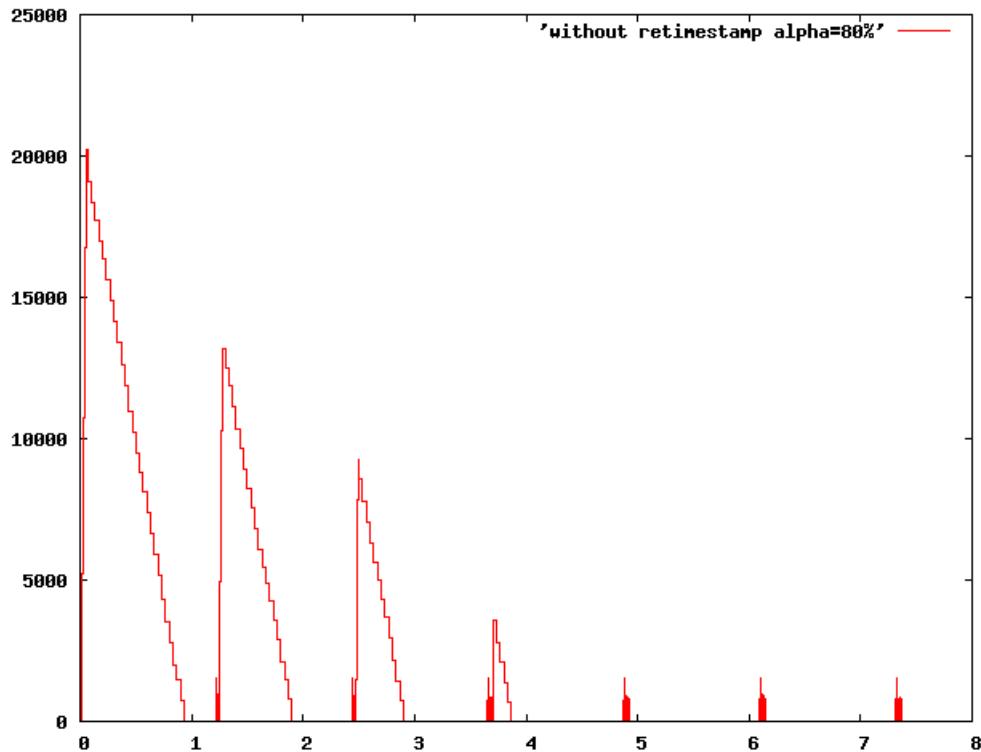


Figure A.6: Evolution of RTP buffer size without RTP proxy

The results shown in figure A.5 correspond to case when initial playback delay is set to 0. However the underflow state also happens when the initial playback delay is not null, although later.

When the RTP retimestamping is activated in RTP proxy, RTP presentation time is synchronized with burst reception time as presented in figure A.7 where it can be seen that there are no more starvation periods:

- used buffer size contains one burst at most at burst reception;
- then RTP buffer progressively empties;
- until it becomes empty for a period less than one frame delay;
- before a new burst is received.

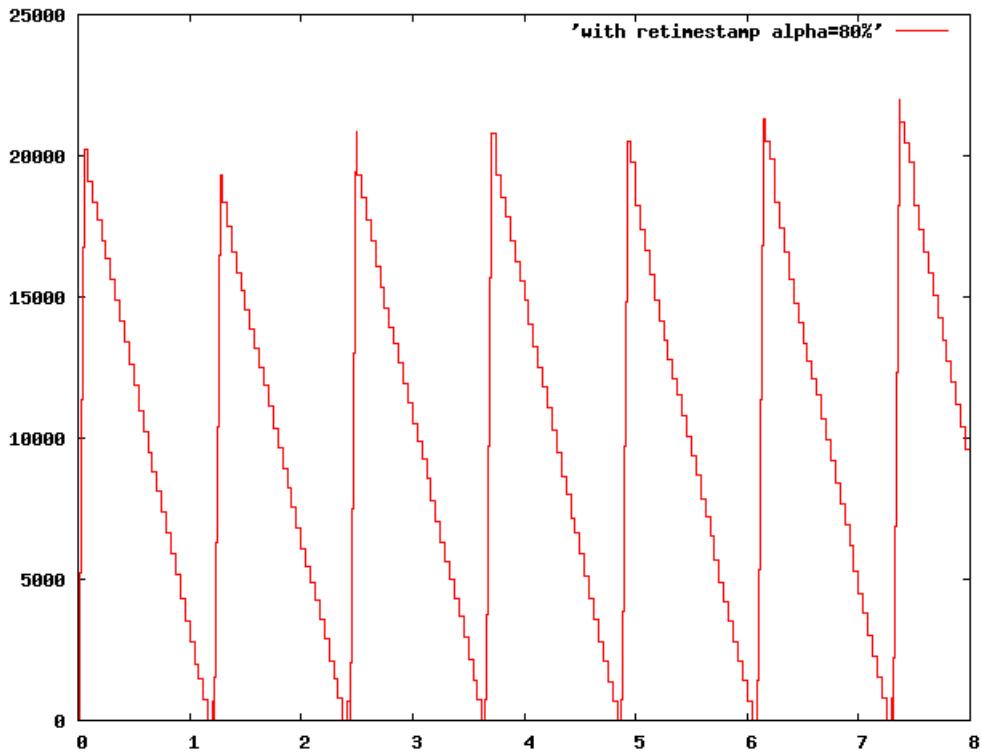


Figure A.7: Evolution of used buffer size with RTP proxy

A.1.3.3 Conclusion

We noticed by simulation that RTP de-jittering buffer, by using the information provided in the RTP presentation timestamps fields never enters underflow or overflow states provided that the RTP proxy re-time-stamping function is activated.

A.1.4 Subjective quality evaluation

A.1.4.1 Introduction

Some videos have been subjectively evaluated. The videos have been encoded in H264 at QVGA (320*240) resolutions with an H264 codec at 256 kbps, sound is AAC stereo 24 kHz.

Three types of content are presented:

- Music: a reportage on a pop singer.
- News: a reportage on London.
- Sport: some ski footage.

For each type of content, the following dumps are provided:

- the content with no error with 3 "slow down factors":
 - 100 %: no frame rate reduction;
 - 90 %: 10 % frame rate reduction;
 - 80 %: 20 % frame rate reduction.

- The content with an LMS-SU error pattern; this LMS dump has been selected to exhibit errors during the channel switching period:
 - errors occur at $t = 2, t = 3, t = 36, t = 37, t = 49, t = 51, t = 58$ et $t = 61$ seconds on the original TS;
 - one error in that case corresponds practically to a complete burst erasure; since the burst is erroneous, the same image in all streams (normal speed and slow-down speed) may be corrupted if not corrected by MPE-IFEC; so visually the same image shall be impacted, even if it is displayed at a different time due to the α parameter;
 - with a 90 % speed, these errors occur at $t = 2,2, t = 3,3, t = 40, t = 41,1, t = 54,4, t = 56,6, t = 64,6$ et $t = 67,7$ seconds;
 - with a 80 % speed, these errors occur at $t = 2,5, t = 3,7, t = 45, t = 46,2, t = 61,2, t = 63,7, t = 72,6$ et $t = 76,2$ seconds;
- the pattern has also been applied with different slow-down factors:
 - 100 %: no frame rate reduction;
 - 90 %: 10 % frame rate reduction;
 - 80 %: 20 % frame rate reduction.

In all cases, IFEC parameters are the same:

- $B = 6, S = 4$.
- $CR_{ota} \sim 2/3$.
- $D = 0$.

All contents have the audio corrected via the WSOLA algorithm. Only one case is provided with no audio correction to illustrate its importance.

For clarity, the naming convention is the following:

- <name_of_content>_<error_pattern>_<speed> <audio_correction>; where
- <name_of_content> is taken from {popsinger;london;tbc};
- <error_pattern> is taken from {error; noerror};
- <speed> is taken from {100;90;80};
- <audio_correction> is taken from {void; no_wsola}.

A.1.4.2 Comments on video

A.1.4.2.1 On the video Errors

One can see on the popsinger_error_90 different errors occurring:

- 1st error: at 3 - 5 seconds corresponding to 2 - 3 seconds on the original TS.
- 2nd error: at 40 - 42 seconds corresponding to 36 - 37 seconds on the original TS.
- 3rd error: at 54 - 58 seconds corresponding to 49 - 51 seconds on the original TS.
- 4th error: at 64 - 68 seconds corresponding to 58 - 61 seconds on the original TS.

On the popsinger_error_80, only the 1st error at 3 - 7 s is noticeable (MPE-IFEC buffer being not filled), all other errors (at 36 - 37, 49 - 51 and 58 - 61 seconds of the original TS) are fully corrected by the IFEC.

A.1.4.2.2 Audio impact

On the dumps without any audio correction, the audio has not been modified. Therefore, the player plays out the audio at nominal speed. Therefore some mute happens on a regular basis, giving a scratchy sound. This can be verified in particular with the music sequence ("popsinger").

When an audio correction algorithm such as WSOLA is used, this scratchy disappears. The WSOLA maintains the audio pitch and only the audio speed change is noticeable.

A.1.4.2.3 Channel switching time duration

The channel switching time duration (delay between the channel switching instant and the time when the normal speed resumes) is fully in-line with theoretical formulas:

- 90 %: the normal speed resuming is at 90 seconds;
- 80 %: the normal speed resuming is at 45 seconds.

A.1.4.2.4 Audio/video quality

From internal review, it is accepted that a 80 % speed can be considered as acceptable.

However, any feedback is welcome.

Annex B (informative): Simulation results for File Delivery Services

B.1 Introduction

This clause describes the simulation environment used to deliver file delivery performance. The simulation environment needs particular attention since it involves different layers:

- physical layer: type of DVB-SH modulation;
- link layer: MPE-IFEC parameters;
- AL layer: FLUTE with its code.

Distribution of FEC among different layers is then an important criteria.

B.2 System Configuration Options

B.2.1 Overview

Figure B.1 shows the IPDC protocol stack over DVB-SH that follows closely the IPDC stack of DVB-H [3]. Specifically highlighted is the File Delivery part of the protocol stack. File Delivery Services in DVB-SH may serve to transport discrete objects such as multimedia clips in file formats such as 3GP file format, still images or other binary data. This may for example allow the distribution of video in a non real-time fashion to support clipcasting services. In addition, the ESG generally uses the File Delivery CDPs. File Delivery in DVB-SH is built on FLUTE. FLUTE supports the delivery of objects over unidirectional transport and maps the file properties to LCT and ALC. It may include Forward Error Correction for reliability and it also allows to realize carousel services. FLUTE delivers data over UDP/IP. According to [1] in DVB-SH the IP datagrams of the respective service are mapped on MPEG2 transport streams using MPE, time-slicing, as well as optionally MPE-IFEC. The resulting MPEG2 transport streams are then transported via DVB-SH physical layer [2] which itself provides several system configuration parameters.

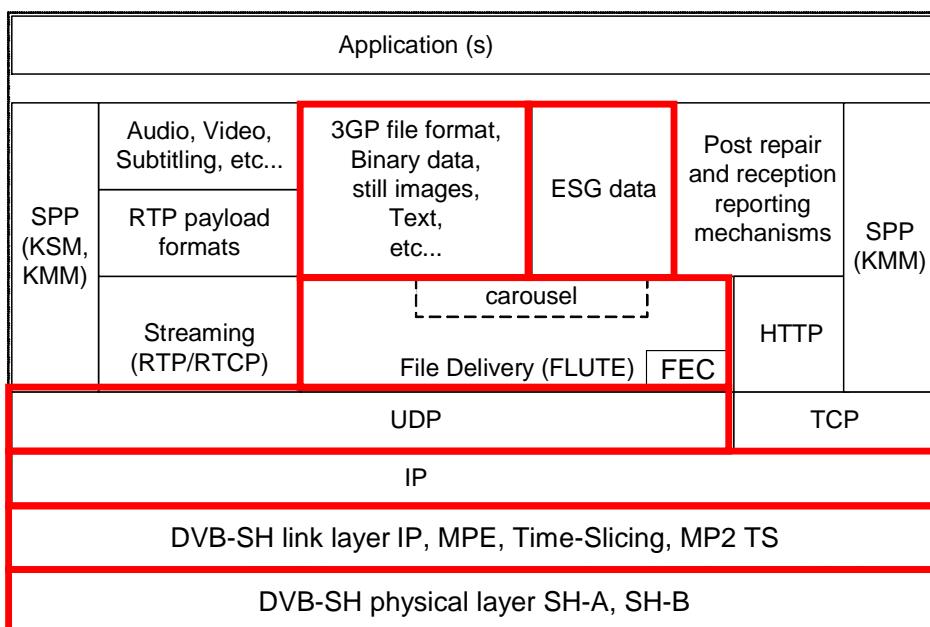


Figure B.1: IPDC over DVB-SH Protocol Stack emphasizing File Delivery Services

The performance of the delivery of these discrete objects depends on many different operating conditions, among others:

- The settings in the physical layer, e.g. system mode, modulation scheme, FEC code rate, interleaver depth, etc.
- The reception environment, e.g. vehicular/handheld, satellite/terrestrial coverage, mobility, speed, environmental conditions, signal strength, etc.
- The settings in the link layer, e.g. the time-slicing mode, service bitrate, MPE-FEC parameters, MPE-IFEC parameters, etc.
- The settings in the application layer for the CDP parameters, e.g. payload and packet sizes, file sizes, FEC type, FEC overhead, delivery rate, etc.
- The evaluation criteria for the file delivery service.

Obviously, not all possible combinations can be evaluated, and only selected representative use cases are simulated. The selected use cases are carefully chosen to reflect most meaningful service categories and to cover a range of deployment scenarios. The configuration options on the different layers as well as some rationales on the selection of the configuration options are discussed in the remainder of the clause B.2.

B.2.2 Physical Layer Configuration Options

The DVB-SH implementation guidelines [i.3] discuss the usage and options of the DVB-SH system in detail. To assess the performance of the DVB-SH system and to compare different modes, the DVB-SH implementation guidelines contain simulation results for representative physical layer configurations. Specifically, clause A.9 in [i.3] contains a comprehensive list of representative simulation cases. For the file delivery simulations this comprehensive collection of configuration and simulation cases are reused as representative settings for physical layer configurations. For completeness, table A.14 is repeated below as table B.1. Details on the configurations of these simulation settings are provided in [i.3].

Table B.1: Selected Physical Layer Configurations (see [i.3], table A.14)

Waveform configuration	Channel	Speed	State-machine	C/I	C/N	PL-ID	Comment
O-16QAM1o4_S	TU6	50 km/h	Off	20 dB	[2:6] dB	1	Basic terrestrial-only link
O-16QAM1o4_S	TU6	3 km/h	Off	20 dB	[2:6] dB	2	Basic terrestrial-only link
O-QPSK1o2_S	TU6	50 km/h	Off	20 dB	[3:5] dB	7	Basic terrestrial-only link
O-QPSK1o2_S	TU6	3 km/h	off	20 dB	[4:6] dB	8	Basic terrestrial-only link
O-16QAM1o4_S	LMS-ITS	50 km/h	on	11,5 dB	10,8 dB	14	Vehicular
O-16QAM2o7_S	LMS-ITS	50 km/h	on	11,5 dB	10,8 dB	17	Vehicular
O-QPSK1o2_S	LMS-ITS	50 km/h	on	11,9 dB	11,2 dB	18	Vehicular
T-8PSK1o3_S	LMS-ITS	50 km/h	On	12,0 dB	11,8 dB	21	Vehicular
T-QPSK1o2_S	LMS-ITS	50 km/h	on	12,5 dB	12,3 dB	24	Vehicular
O-16QAM1o4_S	LMS-SU	50 km/h	on	11,5 dB	10,8 dB	27	Vehicular
O-16QAM1o5_U	LMS-SU	50 km/h	on	11,5 dB	10,8 dB	28	Vehicular
O-16QAM1o5_UL	LMS-SU	50 km/h	On	11,5 dB	10,8 dB	29	Vehicular
O-16QAM2o7_S	LMS-SU	50 km/h	On	11,5 dB	10,8 dB	30	Vehicular
O-QPSK1o2_S	LMS-SU	50 km/h	On	11,9 dB	11,2 dB	31	Vehicular
O-QPSK1o3_U	LMS-SU	50 km/h	On	11,9 dB	11,2 dB	32	Vehicular
T-8PSK1o3_S	LMS-SU	50 km/h	On	12,0 dB	11,8 dB	34	Vehicular
T-QPSK1o2_S	LMS-SU	50 km/h	On	12,5 dB	12,3 dB	37	Vehicular
T-QPSK1o3_U	LMS-SU	50 km/h	On	12,5 dB	12,3 dB	38	Vehicular
T-QPSK1o3_UL	LMS-SU	50 km/h	On	12,5 dB	12,3 dB	39	Vehicular
O-16QAM1o5_U	LMS-SU	3 km/h	On	11,5 dB	4,7 dB	72	Vehicular
O-16QAM1o5_UL	LMS-SU	3 km/h	On	11,5 dB	4,7 dB	73	Vehicular
O-QPSK1o2_S	LMS-SU	3 km/h	On	11,9 dB	5,1 dB	74	Vehicular
T-8PSK1o3_S	LMS-SU	3 km/h	On	12,0 dB	5,7 dB	77	Vehicular
T-QPSK1o2_S	LMS-SU	3 km/h	On	12,5 dB	6,2 dB	80	Vehicular

To avoid extensive simulations of physical layer, error patterns from the physical layer simulations have been provided. These error patterns have been used to simulate the file delivery service including the link layer and the application layer. The generated error patterns contain an index to erroneous MPEG2-TS packet as well as the number of erroneous bits after decoding. This value is a signed integer (text or binary to be defined). The format of the file is as follows:

```
*****
Format of the File

OutputFileFormat,0
FECBlockLength_bits,12282
CodingRate,0
ModulationOrder,2
NbUsefulSubCarriersperOFDMsymbol,1512
OFDMSymbolDuration_seconds,0.000448
WordLength_bits,1504
FrameLength_bits,391168
NbFECBlocks,16283
N<B0>_of_erroneous_MPEG_packet,Number_of_erroneous_bits
...
Simulation ended, real_Nb_of_bits_in_the_simulation
```

One line per erroneous MPEG-2 TS packet is issued providing the packet index in the simulation, the number of binary errors inside the packet. The dumps reflect only MPEG-2 TS packets belonging to the first burst, e.g. corresponding to the reception of a single demodulator listening to a single burst in the time frame.

B.2.3 Link Layer Configuration Options

The DVB-SH provides options on the link layer, in particular usage and settings of the MPE-IFEC [i.7] and the time-slicing parameters. Table B.2 lists sample configuration that may be used for the simulations within this clause. The following cases are represented:

- LLC1: no IFEC protection.
- LLC2 and LLC3: a light IFEC protection typical for streaming video ($CR \geq 1/2$) is used (3/4) with 2 different interleaving durations (10 and 64).
- LLC4 and LLC5: a stronger IFEC protection still typical for streaming video ($CR \geq 1/2$) is used (2/3) with 2 different interleaving durations (10 and 64).
- LLC6: here we use an IFEC protection more typical of a file delivery ($CR \leq 1/2$). The B + S is then selected in order to achieve a quasi error-free burst error rate for different code rates. With a QEF delivery the file is received at first iteration (probability of file loss is null).

Table B.2: Selected DVB-SH link layer configurations (LLCs)

Parameter	LLC1	LLC2	LLC3	LLC4	LLC5	LLC6
Link Layer FEC Code Rate	1	3/4		2/3		Variable
Extended MPE-IFEC rows	n/a		1 024			512 or 1 024
Application Data Table size	n/a	36		32		TBD
MPE-FEC columns	n/a	12		16		Variable
Burst size (bytes)		34*8*188 bytes = 51 136 bytes (ID 1, 2, 7, 8, 14, 18, 24, 27, 31, 37, 74, 80)		35*8*188 bytes = 52 640 bytes (ID 21, 34, 77)		38*8*188 bytes = 57 152 bytes (ID 17, 32)
Burst duration (ms)				122 ms		
Service bit-rate (kbit/s)				To be computed from burst size and time slice cycle		
Time Slice Cycle (s)				Integer multiples of 975 ms (referred as $C_{Burst} = 1, 2, \dots$)		
Burst Spread B	n/a	7	48	6	42	Variable
FEC spread S	n/a	3	16	4	22	Variable
Delay D				0		

B.2.4 Application Layer Configuration Options

This annex deals with the delivery of files. File may be of different size and the file size has influence on the performance and the configuration of the system. To investigate the performance for different use cases, the transmission of different file sizes is tested. Table B.3 provides an overview on selected file sizes with some typical examples. The transmission of other intermediate file sizes is simulated for selected cases to understand the influence of the file size on the performance of the delivery.

Table B.3: Test Cases File Size

File Size	Examples
≈ 100 kB	Web pages, images, etc.
≈ 1 MB	AAC audio files, software updates, ring tones, etc.
≈ 4 MB	video clips, MP3 audio files, etc.
≈ 16 MB	TV show, high-quality video clip, etc.

The file is transported in FLUTE. To accomplish this, the files need to be split in encoding symbols whereby one or several symbol are mapped to an UDP/IP packet. In contrast to streaming applications, the FLUTE symbol size can be chosen flexibly. Therefore, also the IP packet size can be chosen flexibly. For good performance and overhead trade-off a FLUTE payload of either 920 or 1 000 byte is chosen.

In addition, FLUTE and DVB-SH file delivery permits the use of one of two FEC codes, namely the NoCode Option with FecID = 0, i.e. without application layer FEC, the file is simply repeated until it is recovered. For cases with application layer FEC, the parity symbols are sent along with the original data using the rateless feature of the Raptor code with FecID = 1.

Typically, burst sizes and burst periods in DVB-SH are adapted to streaming services to ensure appropriate bitrates, sufficiently low channel switching times and power savings. For file delivery applications, the service does not necessarily have to follow these patterns. For example, files may be delivered at lower bitrate than streaming services. This may be achieved by the following two means:

- The service is setup such that for the delivery of a specific file only a fraction of each burst that is typically used for streaming is occupied for a specific file and multiple files are transmitted in parallel in an interleaved manner. Selected simulation will be carried out for no file interleaving, interleaving of 4 files and interleaving of 16 files.
- The service is setup such that for the delivery of a specific file the burst period is increased, but the file still occupies the entire burst. The increase is typically by a multiple integer of the regular streaming burst period as indicated in table B.2.

Furthermore, file delivery as generally not critical in terms of bitrate guarantees, and may therefore be combined with statistical multiplexing of streaming services. Simulation methodologies for this purpose are for further study.

B.3 Test Cases

Simulation test cases are combinations of setting on the different layers of the protocol stack to understand the performance of file delivery services over different DVB-SH configurations using IPDC CDPs. Specifically the following configurations need to be specified for a test case.

- Physical layer configuration: one of the Physical-Layer IDs according to table B.1 together with a C/N and C/I in dB.
- Link layer configuration: one of the 5 configurations as specified in table B.2 is chosen together with the time slice cycle C_{Burst} .
- Application layer combination: A test case file size, for example according to table B.3 is chosen, the application layer FEC is specified by the FecID and the number of interleaved files is specified.

The simulated test cases are specified in detail along with results in clause B.6.

B.4 Evaluation Criteria

As already highlighted, file delivery asks that the delivered file is received completely and without any error. The reliability is achieved by transmitting as many FLUTE symbols such that as many receivers as possible can reconstruct the file. Obviously, the more FLUTE symbols there are transmitted, the more likely the receivers have sufficient information to reconstruct the file in the FLUTE session. However, it is also acceptable that a small fraction of users do not receive the files. These users may for example use post repair procedures to complete the delivery.

As a evaluation criteria to reflect this performance the number of time slice bursts required to achieve a certain probability of file reception is chosen. This criteria is evaluated by simulations for each test case. As probability we have chosen three different values, namely that 90 %, 95 % and 99 % of the receivers have received the file.

With the size of the time slice burst, the burst period and the test case parameters on the physical layer, the exact resource consumption can be evaluated. The results as provided in this annex can then be used for more detailed evaluation and for the generation of configuration guidelines.

B.5 Simulation Methodology

To simulate the performance of the file delivery the physical layer is replicated by MPEG2 TS traces. The remaining parts of the protocol stack, link layer and application layer have been simulated according to the specifications in [1], [2], [i.3] and [i.7]. The following simulation method has been applied:

- The file delivery starts synchronized with the time slice burst cycle.
- For statistical significance, the file delivery simulation shall be started at least every 10th burst in the MPEG2 TS trace regardless of the delivery time of individual experiments. The number of experiments is therefore at least 1/10th of the number of bursts in each MPEG2 TS trace.
- For files for which the systematic part spans less than 10 bursts, the burst starting period shall be reduced such that at least each burst carries some part of the file.
- For each transmission attempt (starting point in the trace), the number of necessary bursts and used MPEG2 TS packets until full recovery is achieved is computed. At most five times the size of the original data has been transmitted. If recovery is not successful with the is amount of overhead, the attempt is considered as erroneous.
- If the end of the trace is reached before the recovery is successful, the trace is looped, i.e. the experiment continues with the first burst in the trace.

By this, for each of the N experiments the required number of bursts or MPEG2 TS packets is obtained. This forms the basic output of each statistical experiments. From this statistical data, the probability that the file is not recovered after sending it for a certain time (or a certain amount of bursts) can be derived.

For the IFEC parameters optimized for file delivery, a quasi error-free delivery is sought, enabling reception of the file at first.

B.6 Simulation Results

B.6.1 Reported Data

For each of the test cases, the following data is reported in the below tables in the remainder of clause B.6.

Table B.4: Reported data

Acronym	Type	Explanation
PHY-ID	PL Config	Physical Layer ID as specified in table B.1
C/N	PL Config	The carrier-to-noise ratio for this specific test case
C/I	PL Config	The carrier-to-interference ratio for this specific test case
LL-ID	LL Config	Link Layer ID as specified in table B.2
CB	LL Config	The time slice burst cycle integer as a multiple of 975 ms
FS	AL Config	File size of the file under consideration in kB
FEC	AL Config	FEC ID of the FLUTE CDP, with 0 the NoCode and 1 Raptor
Share	AL Config	The amount of data that is used in the time slice for this file - emulates file interleaving
TS	Sim Para	The total amount of simulation runs for this test case
MB	Sim Result	The minimum amount of bursts, i.e. the bursts to transmit the original file
B 90 %	Sim Result	The required bursts to support at least 90 % of the users/experiments
Pr 90 %	Sim Result	The actual probability when transmitting B90 % of the bursts (only for cross-checking)
B 95 %	Sim Result	The required bursts to support at least 95 % of the users/experiments
Pr 95 %	Sim Result	The actual probability when transmitting B95 % of the bursts (only for cross-checking)
B 99 %	Sim Result	The required bursts to support at least 99 % of the users/experiments
Prob 99 %	Sim Result	The actual probability when transmitting B99 % of the bursts (only for cross-checking)
ViBR 90 %	Derivation	The virtual media bitrate determined by the file size and the duration of the transmission to achieve 90 % reliability
ViBR 95 %	Derivation	The virtual media bitrate determined by the file size and the duration of the transmission to achieve 95 % reliability
ViBR 99 %	Derivation	The virtual media bitrate determined by the file size and the duration of the transmission to achieve 99 % reliability
BWE 90 %	Derivation	The bandwidth efficiency determines the bitrate per MHz that can be reliably transmitted under the 90 % reliability constraint
BWE 95 %	Derivation	The bandwidth efficiency determines the bitrate per MHz that can be reliably transmitted under the 95 % reliability constraint
BWE 99 %	Derivation	The bandwidth efficiency determines the bitrate per MHz that can be reliably transmitted under the 99 % reliability constraint

B.6.2 LMS

For LMS simulations, the FLUTE symbol size was chosen as 1 000 bytes except stated otherwise.

Phy Layer			Link		Application					90		95		99		ViBR kbit/s			BWE bit/s/Hz		
ID	C/N	C/I	ID	CB	FS	FEC	Share	TS	MB	B	Pr	B	Pr	B	Pr	90	95	99	90	95	99
37	12,3	12,5	2	1	4 096	0	100 %	263	124	218	90,1	236	95,1	247	100,0	157,1	145,2	138,8	25,1 %	23,2 %	22,2 %
37	12,3	12,5	4	1	4 096	0	100 %	263	136	188	90,1	230	95,1	268	99,2	182,1	149,0	127,9	29,1 %	23,8 %	20,5 %
37	12,3	12,5	1	1	4 096	1	100 %	263	83	88	95,1	88	95,1	90	100,0	386,7	386,7	378,2	61,9 %	61,9 %	60,5 %
74	5,1	11,9	1	1	4 096	0	100 %	1 015	83	628	90,0	740	95,1	900	99,0	54,7	46,4	38,2	8,8 %	7,4 %	6,1 %
74	5,1	11,9	2	1	4 096	0	100 %	1 015	124	676	90,1	720	95,1	816	99,1	50,8	47,7	42,1	8,1 %	7,6 %	6,7 %
74	5,1	11,9	4	1	4 096	0	100 %	1 015	136	742	90,0	792	95,1	899	99,0	46,3	43,4	38,2	7,4 %	6,9 %	6,1 %
74	5,1	11,9	1	1	4 096	1	100 %	1 015	83	160	90,5	168	95,2	199	99,0	213,8	203,6	172,1	34,2 %	32,6 %	27,5 %
77	5,7	12	1	1	4 096	0	100 %	807	81	321	90,5	359	95,2	397	99,0	106,9	95,6	86,5	17,1 %	15,3 %	13,8 %
77	5,7	12	2	1	4 096	0	100 %	807	120	475	90,1	544	95,0	634	99,0	72,3	63,1	54,2	11,6 %	10,1 %	8,7 %
77	5,7	12	4	1	4 096	0	100 %	807	136	411	90,1	479	95,2	531	99,0	83,5	71,7	64,7	13,4 %	11,5 %	10,4 %
77	5,7	12	1	1	4 096	1	100 %	807	81	121	90,8	130	95,0	140	99,0	282,1	262,7	244,1	45,1 %	42,0 %	39,1 %
80	6,2	12,5	1	1	4 096	0	100 %	806	83	334	90,1	396	95,0	473	99,0	102,7	86,7	72,6	16,4 %	13,9 %	11,6 %
80	6,2	12,5	2	1	4 096	0	100 %	806	124	443	90,1	482	95,2	577	99,0	77,5	71,3	59,5	12,4 %	11,4 %	9,5 %
80	6,2	12,5	4	1	4 096	0	100 %	806	136	406	91,4	481	95,0	591	99,0	84,6	71,4	58,1	13,5 %	11,4 %	9,3 %
80	6,2	12,5	1	1	4 096	1	100 %	806	83	120	90,7	128	95,0	138	99,1	284,4	266,8	247,6	45,5 %	42,7 %	39,6 %

B.6.2.5 Results with MPE-IFEC optimized for file delivery

In this clause, we provide results obtained with MPE-IFEC LL6 configuration.

We provide the quasi error-free analysis for the following two selected physical dumps:

- LMS-ITS QPSK 1/2 ID = 18;
- LMS-SUB 16QAM 2/7 ID = 74.

These MPE-IFEC simulations have been performed with a different simulator than the previous FLUTE-based simulations. As we need to provide a comparison point between the MPE-IFEC LL6 simulations and the other simulations, the following procedure has been used:

- LL6 MPE-IFEC did not undergo the full AL-FEC simulation of FLUTE, therefore a direct comparison is not possible.
- To compare LL6 with other cases, we need to extrapolate available FLUTE-based simulations to quasi error free conditions; for this we extent the results available for 10 %, 5 % and 0,1 % to the quasi error free case (0 %).
- In quasi error free situation a fair comparison in terms of virtual bit rate and efficiency is now possible using standard virtual bitrates and efficiency criteria; the virtual bit rate and efficiencies are extended to the LL6 according to following rules:

$$\begin{aligned} \text{- } \text{Virtual_bitrate}_{\text{kbits/s}} &= \frac{\text{file_volume}_{\text{kbytes}} * 8}{\text{nof_bursts} * \text{repetition_interval}_s} \text{ where } \text{nof_bursts} \text{ is given by} \\ &\text{nof_bursts} = \frac{\text{file_volume}_{\text{bursts}}}{1 - \text{fec_ratio}} + (\text{B} + \text{S}). \\ \text{- } \text{Efficiency}_{\text{kbits/s/Hz}} &= \frac{\text{Virtual_bitrate}_{\text{kbits/s}} * \text{nof_bursts}}{5\text{MHz} * \text{tot_nof_burst}} \text{ where } \text{tot_nof_burst} \text{ represents the} \\ &\text{actual volume in bursts to carry the file and its associated FEC:} \\ &\text{tot_nof_bursts} = \frac{\text{file_volume}_{\text{bursts}}}{1 - \text{fec_ratio}}. \end{aligned}$$

- Therefore the efficiency is derived from the virtual_bitrate by a multiplication factor equal to $\frac{\text{nof_bursts}}{\text{tot_nof_burst}} = 1 + (\text{B} + \text{S}) * \frac{1 - \text{fec_ratio}}{\text{file_volume}_{\text{bursts}}}$.
- Other cases than LL6 include: no FEC, Raptor without interleaving, Raptor with interleaving factor of 3.
- For the LL6 case, we distinguish between 2 configurations:
 - the first case corresponds to a code rate equal to the Raptor case which is file size dependent; therefore there are 3 different code rates for the 3 types of file size;
 - the second case corresponds to the best parameters among the 3 file sizes: since MPE-IFEC is not file dependent, we can choose the best configuration among the 3.

Annex C (informative): Formula demonstration

C.1 IFEC decoder delivery time

This formula gives the delivery time of a packet at the output of the IFEC decoder:

$$t'_k = t_k + \text{adst_delay}_{\min} + (1 / \alpha - 1) * (t_k - t_0).$$

To demonstrate this formula, we start from the definition of α . It is defined as the ratio between frame rate per second output and frame rate per second input:

Definition 1:

$$\alpha = \frac{f_{\text{out}}}{f_{\text{in}}}, f_{\text{out}} = \frac{n_{\text{out}} - n_0}{t_{n_{\text{out}}} - t_{0_{\text{out}}}}, f_{\text{in}} = \frac{n_{\text{in}} - n_0}{t_{n_{\text{in}}} - t_{0_{\text{in}}}}.$$

Where:

- n_{out} is a frame number at the output of the IFEC decoder and $t_{n_{\text{out}}}$ is the departure time;
- n_{in} is a frame number at the input of the IFEC decoder and $t_{n_{\text{in}}}$ is the arrival time;
- n_0 is the first frame of the GOP and $t_{0_{\text{in}}}$ and $t_{0_{\text{out}}}$ are the corresponding arrival and departure times of the IFEC decoder.

Hypothesis 1: Let us assume $n_{\text{out}} = n_{\text{in}}$ (frame numbers are the same): $n_{\text{out}} = n_{\text{in}}$, therefore $\alpha = \frac{t_{n_{\text{in}}} - t_{0_{\text{in}}}}{t_{n_{\text{out}}} - t_{0_{\text{out}}}}$; and

$$t_{n_{\text{out}}} - t_{0_{\text{out}}} = \frac{1}{\alpha} * (t_{n_{\text{in}}} - t_{0_{\text{in}}}).$$

Hypothesis 2: Another hypothesis is that the first frame undergoes a fixed delay of ads_min :

$$t_{0_{\text{out}}} = \text{adst_delay}_{\min} + t_{0_{\text{in}}}; \text{ and}$$

$$t_{n_{\text{out}}} - t_{n_{\text{in}}} = \frac{1}{\alpha} * (t_{n_{\text{in}}} - t_{0_{\text{in}}}) - t_{n_{\text{in}}} + t_{0_{\text{out}}} = \left(\frac{1}{\alpha} - 1 \right) * t_{n_{\text{in}}} + \left(1 - \frac{1}{\alpha} \right) * t_{0_{\text{in}}} + \text{adst_delay}_{\min}.$$

Equation 1:

$$t_{n_{\text{out}}} - t_{n_{\text{in}}} = \left(\frac{1}{\alpha} - 1 \right) * (t_{n_{\text{in}}} - t_{0_{\text{in}}}) + \text{adst_delay}_{\min}.$$

C.2 Channel switching transition duration

This formula gives the total duration of the period during which the slow down occurs.

$$t'_{fin} - t_0 = adst_delay_{max} + (adst_delay_{max} - adst_delay_{min}) / (1 / \alpha - 1).$$

To demonstrate this formula, we make the hypothesis that, at the end of the transition, the delivery time of a packet corresponds to the $adst_delay_{max}$ duration:

Hypothesis 3:

$$t'_k = t_k + asdt_delay_{max} .$$

We replace this value of t'_k in the equation 1:

$$t_{n_{out}} = t_{n_{in}} + adst_delay_{max} = t_{n_{in}} + \left(\frac{1}{\alpha} - 1\right) * (t_{n_{in}} - t_{0_{in}}) + adst_delay_{min} .$$

Equation 2:

$$(t_{n_{in}} - t_{0_{in}}) = (adst_delay_{max} - adst_delay_{min}) * \frac{\alpha}{1 - \alpha} .$$

But we are interested by $t_{n_{out}}$ rather than by $t_{n_{in}}$ and we must re-inject $t_{n_{in}}$ inside formula 1:

$$\begin{cases} \text{Equation 1: } t_{n_{out}} - t_{n_{in}} = \left(\frac{1}{\alpha} - 1\right) * (t_{n_{in}} - t_{0_{in}}) + adst_delay_{min} \\ \text{Equation 2: } (t_{n_{in}} - t_{0_{in}}) = (adst_delay_{max} - adst_delay_{min}) * \frac{\alpha}{1 - \alpha} \end{cases}$$

$$\Rightarrow t_{n_{out}} = (adst_delay_{max} - adst_delay_{min}) + adst_delay_{min} + t_{0_{in}} + (adst_delay_{max} - adst_delay_{min}) * \frac{\alpha}{1 - \alpha} .$$

Equation 3:

$$t_{n_{out}} = adst_delay_{max} + t_{0_{in}} + (adst_delay_{max} - adst_delay_{min}) * \frac{\alpha}{1 - \alpha} .$$

Annex D (informative): Bibliography

IEEE Transactions on Acoustics, Speech, and Signal Processing, Volume 27, Issue 2, Apr 1979, pages 121 - 133.

History

Document history		
V1.1.1	February 2010	Publication