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DVB-SH Implementation Guidelines

EBU·UER

Digital Video Broadcasting

Reference

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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).

The present document is complementary to EN 302 583 [1], which provides the waveform-level specifications for the use of TV Broadcast in DVB Satellite to Handhelds applications.

The present document is complementary to TS 102 585 [2], which provides the system-level specifications for the use of TV Broadcast in DVB Satellite to Handhelds applications.

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.

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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.

Introduction

The present document gives the first guidelines for setting up networks and services using the Digital Video Broadcasting - Satellite to Handheld (DVB-SH) specifications. Updates to the present document will be produced when more results become available.

Document summary

Clause 4 gives an outline of the DVB-SH system, the main hybrid satellite/terrestrial architectures, the distinction between SH-A and SH-B as well as between SFN and non-SFN, the possible approaches for frequency planning, the frequency bands and their regulatory aspects, the characteristics of satellite and terrestrial propagation channels, the receivers constraints, and other system considerations.

Clause 5 gives a discussion of possible DVB-SH system configurations and likely scenarios of their deployment sequence.

Clause 6 introduces the elements of the DVB-SH specifications at the link layer and above. These are, at the link layer, time-slicing and LL-FEC and, at above this layer, the supports of VBR/Statmux and of Mobility. Compatibility with DVB-H link layer is discussed.

Clause 7 clarifies the physical layer elements of the DVB-SH specifications that are new to the DVB families of standards, namely the 3GPP2 turbo codes, the long physical time interleaver and its implication on receivers (class 1 and class 2 receivers), the 1K-FFT mode for OFDM and the diversity combining techniques (MFN).

Clause 8 discusses services and usage scenarios. This clause presents concepts such as service categories and attributes and highlights the consequences of the hybrid architectures on a personal mobile TV service offering.

Clause 9 is devoted to DVB-SH network configuration. The following issues are discussed: Network configurations, Service Information, Handover.

Clause 10 provides preliminary information about the DVB-SH reference terminals and reference receivers.

Clause 11 is devoted to Network coverage planning. The main purpose is to present reference data and methodologies for transmitter network (both satellite and terrestrial) sizing as a function of coverage.

Finally three annexes are included:

- Annex A documents the methodology used by TM-SSP and presents the simulation and experimental results that support the recommendations given.
- Annex B details the implications for "convergence terminals" in which broadcast (DVB-SH, DVB-H) and telecommunication (GSM/UMTS) technologies co-exist.
- Annex C documents the methodology and formulas for spectrum efficiency analysis and throughput calculations.

1 Scope

The present document provides guidelines for the use and implementation of ETSI Digital Video Broadcasting-Satellite to Handheld (DVB-SH) standard EN 302 583 [1] in the context of providing an efficient way of carrying multimedia services over digital hybrid satellite/terrestrial broadcasting networks to handheld terminals.

The present document should be read in conjunction with the:

- DVB-SH waveform specifications [1];
- DVB-SH system specifications [2];
- DVB IP Datacast specifications [21] and [22];
- DVB Data broadcasting specifications [9];
- DVB Specifications for Service Information (SI) [24].

Objective

The present document draws attention to the technical questions that need to be answered when setting up DVB-SH services and networks and offers some guidance in finding answers to them. It does not cover in detail, issues linked to the content of the broadcasts such as Coding Formats, Electronic Programme Guides (EPG), Access Control (CA), etc.

Target readers

The present document is aimed at the Technical Departments of organizations that are considering implementing digital hybrid satellite/terrestrial broadcasting to handheld devices. It assumes that readers are familiar with digital satellite and terrestrial networks.

Contributors

The present document was prepared by members of the Ad-hoc group TM-SSP from the DVB Project. Members include broadcasters, network operators and professional as well as domestic equipment manufacturers.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

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 necessary for the application of the present document.

[1] ETSI EN 302 583: "Digital Video Broadcasting (DVB); Framing Structure, channel coding and modulation for Satellite Services to Handheld devices (SH) below 3 GHz".

[2] ETSI TS 102 585: "Digital Video Broadcasting (DVB); System Specifications for Satellite services to Handheld devices (SH) below 3 GHz".

[3] ETSI EN 302 304: "Digital Video Broadcasting (DVB); Transmission System for Handheld Terminals (DVB-H)".

[4] Void.

[5] Void.

[6] ETSI EN 302 307: "Digital Video Broadcasting (DVB); Second generation framing structure, channel coding and modulation systems for Broadcasting, Interactive Services, News Gathering and other broadband satellite applications (DVB-S2)".

[7] Void.

[8] ISO/IEC 13818-1 (2007): "Information technology -- Generic coding of moving pictures and associated audio information: Systems".

[9] ETSI EN 301 192: "Digital Video Broadcasting (DVB); DVB specification for data broadcasting".

[10] Void.

[11] ITU-R Recommendation M.1225 (1997): "Guidelines for evaluation of radio transmission technologies for IMT-2000".

[12] ITU-R Recommendation P.681-7 (2009): "Propagation data required for the design of Earth-space land mobile telecommunication systems".

[13] ITU-R-Recommendation P.1546-4 (2009): "Method for point-to-area predictions for terrestrial services in the frequency range 30 MHz to 3 000 MHz".

[14] EICTA/TAC/MBRAI-02-16 Version 1, 2004: "Mobile and Portable DVB-T Radio Access Interface Specifications".

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[21] ETSI TS 102 470-2: "Digital Video Broadcasting (DVB); IP Datacast: Program Specific Information (PSI)/Service Information (SI); Part 2 : IP Datacast over DVB-SH".

[22] ETSI TS 102 472: "Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Content Delivery Protocols".

[23] ISO/IEC 13818-6: "Information technology -- Generic coding of moving pictures and associated audio information; Part 6: Extensions for DSM-CC".

[24] ETSI EN 300 468: "Digital Video Broadcasting (DVB); Specification for Service Information (SI) in DVB systems".

[25] ITU-R Recommendation P.676-8 (2009): "Attenuation by atmospheric gases".

[26] IEEE 802.11 (2007): "Standard for Information technology-Telecommunications and information exchange between systems-Local and metropolitan area networks-Specific requirements; Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications".

[27] ITU-R Recommendation P.526-11 (2009): "Propagation by diffraction".

[28] ETSI EN 300 744: "Digital Video Broadcasting (DVB); Framing structure, channel coding and modulation for digital terrestrial television".

[29] ETSI TS 145 005: "Digital cellular telecommunications system (Phase 2+); Radio transmission and reception (3GPP TS 45.005)".

[30] ETSI TS 100 910: "Digital cellular telecommunications system (Phase 2+); Radio transmission and reception (3GPP TS 05.05)".

[31] ITU-R Recommendation P.618-10 (2009): "Propagation data and prediction methods required for the design of Earth-space telecommunication systems".

[32] ITU-R Recommendation P.531-10 (2009): "Ionospheric propagation data and prediction methods required for the design of satellite services and systems".

[33] ITU-R Recommendation P.680-3 (1999): "Propagation data required for the design of Earth-space maritime mobile telecommunication systems".

[34] ETSI TS 102 591-2: "Digital Video Broadcasting (DVB); IP Datacast: Content Delivery Protocols (CDP) Implementation Guidelines; Part 2: IP Datacast over DVB-SH".

[35] ETSI TS 102 611-2: "Digital Video Broadcasting (DVB); IP Datacast: Implementation Guidelines for Mobility; Part 2: IP Datacast over DVB-SH".

[36] ETSI TS 102 772: "Digital Video Broadcasting (DVB); Specification of Multi-Protocol Encapsulation - inter-burst Forward Error Correction (MPE-IFEC)".

[37] ETSI TS 102 592-2: "Digital Video Broadcasting (DVB); IP Datacast: Electronic Service Guide (ESG) Implementation Guidelines; Part 2: IP Datacast over DVB-SH".

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] "Antennas and Propagation for Wireless Communication Systems" Chapter 9, Section 9.5.

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[i.5] IEEE Trans. Veh. Technol., Vol. 43, No. 3, August 1994: "A Statistical Model for Land Mobile Satellite Channels and Its Application to Nongeostationary Orbit Systems"; Corazza, G.E. and Vatalaro, F.

[i.6] IEEE 47th, Volume 1, Issue, 4-7 May 1997 Page(s):41; 45 vol.1: "A channel model for nongeostationary orbiting satellite system"; Vehicular Technology Conference, 1997; Seung-Hoon Hwang, Ki-Jun Kim, Jae-Young Ahn and Keum-Chan Whang.

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[i.8] "Simulation of Communication Systems" (Plenum Press, 1992); M. C. Jeruchin, P. Balaban, and K. S. Shanumgan.

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[i.13] "Radio Network Planning and Process and Methods for WCDMA", Jaana Laiho, and Achim Wacker (Nokia Networks).

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[i.20] C. E. Gilchriest: "Signal-to-Noise Monitoring", *JPL Space programs Summary*, No. 37-27, Vol. IV, pp. 169-176.

[i.21] ETSI TR 102 377 (V1.2.1): "Digital Video Broadcasting (DVB); DVB-H Implementation Guidelines".

[i.22] ETSI TR 101 190 (V1.2.1): "Digital Video Broadcasting (DVB); Implementation guidelines for DVB terrestrial services; Transmission aspects".

[i.23] ETSI TR 102 376 (V1.1.1): "Digital Video Broadcasting (DVB) User guidelines for the second generation system for Broadcasting, Interactive Services, News Gathering and other broadband satellite applications (DVB-S2)".

[i.24] ETSI TR 101 112 (V3.2.0): "Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS (UMTS 30.03 version 3.2.0)".

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[i.30] ECC Decision ECC/DEC/(06)10 of 1 December 2006 on transitional arrangements for the Fixed Service and tactical radio relay systems in the bands 1980-2010 MHz and 2170-2200 MHz in order to facilitate the harmonised introduction and development of systems in the Mobile Satellite Service including those supplemented by a Complementary Ground Component.

NOTE: Available at http://www.erodocdb.dk/docs/doc98/Official/word/ECCDec0610.doc.

[i.31] ECC Recommendation (06)05 on the provision of information on the progress of implementation of the mobile satellite systems which are candidates to use the 1980-2010 MHz and 2170-2200 MHz MSS frequency bands.

NOTE: Available at http://www.erodocdb.dk/docs/doc98/official/pdf/Rec0605.pdf.

[i.32] ETSI EN 301 442: "Satellite Earth Stations and Systems (SES); Harmonized EN for Mobile Earth Stations (MESs), including handheld earth stations, for Satellite Personal Communications Networks (S-PCN) in the 2,0 GHz bands under the Mobile Satellite Service (MSS) covering essential requirements under Article 3.2 of the R&TTE directive".

[i.33] Decision 2008/626/EC of the European Parliament and of the Council of 30 June2008 on the selection and authorisation of systems providing mobile satellite services (MSS).

NOTE: Available at http://eur-lex.europa.eu/LexUriServ/LexUriServ.do?uri=OJ:L:2008:172:0015:0024:EN:PDF

[i.34] EICTA, MBRAI documents MOBILE AND PORTABLE DVB-T/H RADIO ACCESS - Part 1: Interface specification.

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[i.38] Void.

[i.39] Agilent Spectrum Analysis Basis. Application note 150, chapter 5.

[i.40] Performance evaluation of channel change for DVB-SH streaming services, ICC 2010, Marie-Line Alberi Morel, Sylvaine Kerboeuf, Bessem Sayadi, Yann Leprovost, and Frederic Faucheux, Alcatel Lucent Bell Labs France, Route de Villejust 91620 NOZAY, France, E-mail: {marie_line.alberi-morel, sylvaine.kerboeuf}@alcatel-lucent.com, {bessem.sayadi, yann.leprovost, frederic.faucheux@alcatel-lucent.com.

[i.41] ETSI EN 302 574-2: "Satellite Earth Stations and Systems (SES); Harmonized Standard for satellite earth stations for MSS operating in the 1 980 MHz to 2 010 MHz (earth-to-space) and 2 170 MHz to 2 200 MHz (space-to-earth) frequency bands; Part 2: User Equipment (UE) for wideband systems: Harmonized EN covering the essential requirements of article 3.2 of the R&TTE Directive".

[i.42] 3GPP2 C.S0002-D: "Physical Layer Standard for cdma2000 Spread Spectrum Systems".

[i.43] Final report on "Study of Satellite Role in 4G Mobile Networks", European Space Agency, Contract No: No. 20194/06/NL/US.

[i.44] K.P. Liolis et al.: "Statistical modeling of dual-polarized MIMO land mobile satellite channels", accepted for publication to IEEE Trans. Commun., 2010.

[i.45] International Journal of Digital Multimedia Broadcasting:"Implementing Statistical Multiplexing in DVB-H", Volume 2009 (2009), Article ID 261231.

NOTE: Available at: http://www.hindawi.com/journals/ijdmb/2009/261231.abs.html.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

common content (or Service): content available via the SC and also obligatorily transmitted in each cell of the CGC

Complementary Ground Component (CGC): terrestrial leg of a hybrid network that uses a portion of the frequency allocated to that network, under a special regulatory regime decided at regional or national levels, to provide a complementary cellular coverage of the regional/national territory

NOTE: It cannot exist independently of the SC it is associated to. Also called ATC in the US.

hybrid frequency: frequency used by a terrestrial transmitter carrying the Common content

NOTE: A Non-Hybrid frequency carries only Local content.

Inter-burst FEC (IFEC): new FEC at link layer standardized in DVB-SH to combat erasures of several complete Bursts

NOTE: It usually operates over protection periods of 10 s or more. It uses the syntax of the conventional MPE-FEC (see EN 301 192 [9]), but extends this technique with spreading and interleaving, to achieve the stated improved protection.

late/early decoding: "Late decoding" refers to the technique of buffering the incoming channel data so that delivery to the source decoders occurs only when all data has been received that ensure maximum FEC protection

NOTE: On the contrary, "Early decoding" refers to the technique of delivering data to the source decoders before all data has been received, for the purpose of improving zapping time. This introduces jitter, in adverse reception conditions, that has to be taken into account by the presentation layer.

Link Layer FEC (LL-FEC): FEC implemented at the Link layer. It usually includes a time interleaver and relies on independent framing and error detection mechanisms to correct long strings of erased bits

NOTE: In DVB-SH, link-layer FEC are constructed using the MPE syntax.

local content: content not available via the SC and the not guaranteed to be available in every cells of the CGC

modulation: process of varying a sine wave to convey a message

NOTE: Four digital modulations are used in DVB-SH: QPSK, 8PSK(TDM only), 16APSK(TDM only) and 16QAM(OFDM only).

OFDM (waveform): waveform composed of a large number of closely packed sine waves each modulated at low speed

NOTE: This waveform is optimized for terrestrial propagation, but can also be used with satellite. This waveform allows SFN operation by several transmitters.

partially available Transport Stream: ransport Stream that carries at least one DVB service that is not transmitted in all cells where the TS is transmitted

NOTE: This is signalled by the service_availability_descriptor in the SDT.

receiver: functional module of a terminal from antenna(s) to IP baseband output

NOTE: TS 102 585 [2] currently specifies two *classes* of receivers, according to their capability and methods to process time diversity elements in the waveform (see clause 10).

Regular IP encapsulator, regular receiver, regular transmitter: IP encapsulator/receiver/modulator that is working according to the present standard, but which is not aware of the low latency extension defined in [1], annexes B and D of this implementation guideline.

Satellite Component (SC): satellite leg of a hybrid network, using a portion of the frequency allocated to satellite services at international level, and which appears as an umbrella cell to the DVB-SH receivers

sub-band: contiguous segment of the allocated spectrum usually dedicated to the transmission of a multiplex

NOTE: In DVB-SH, sub-bands can have 1,7 MHz; 5 MHz; 6 MHz; 7 MHz and 8 MHz of bandwidth.

TDM (waveform): waveform composed of a sine wave modulated at high speed and with a modulation designed to produce a low temporal envelop variation

NOTE: This waveform optimized for use with satellite. It does not allow SFN operations.

terminal: device comprising several functional modules: receiver, video/audio processing, battery/power, display and optionally (non-DVB-SH) bidirectional radio functions

NOTE: For network planning, several terminal *categories* are distinguished in DVB-SH (see clause 10).

time interleaver: physical layer operation introduced for mitigating long fades

NOTE: At the transmitter it breaks one turbo code word into several pieces (also called Interleaver Units) and spread these over time (from several hundred ms to more than 10 s). In the receiver, these interleaver units are collected, reordered and forwarded to the turbo decoder.

Transport Stream (TS): multiplex based on the MPEG2 packet protocol and complete with its EN 300 468 [24] compliant SI signalling

NOTE: A TS has a unique identification through the Transport_Stream_ID and the Original_ network_ID. By extension, two multiplexes having the same unique identification but transmitted in parallel in the SC and the CGC are considered in DVB-SH as forming a unique TS.

waveform: shape and form of a transmitted signal. DVB-SH has defined two waveforms: TDM and OFDM

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

16APSK	16-state APSK
16QAM	please change to 16QAM
3G	Third Generation
3GPP	Third Generation Partnership Program
8PSK	please change to 8-PSK

A3GH	Above 3 GHz (frequency band)
ACIR	Adjacent Channel Interference Ratio
ACLR	Adjacent Channel Leakage Ratio
ACS	Adjacent Channel Selectivity
ADST	Application Data Sub Table
ADT	Application Data Table
AM/AM	Amplitude Modulated to Amplitude Modulated characteristic
AM/PM	Amplitude Modulation/ Phase Modulation
APSK	Amplitude Phase Shift Keying
ASI	Asynchronous Serial Interface
ASIC	Application Specific Integrated Circuit
ATC	Ancillary Terrestrial Component (US)
AVC	Advanced Video Coding
AVHRR	Advanced Very High Resolution Radiometer
AWGN	Additive White Gaussian Noise
B3GH	Below 3 GHz (frequency band)
BER	Bit Error Rate
BFN	Beam Forming Network
BL	Base Layer
BpS	BitsPerSymbol (= modulation order)
BS	Base Station
BTS	Base Transceiver Station
BW	Bandwidth
C/I	Carrier-to-Interference ratio
C/M	Carrier over Multipath ratio
C/N	Carrier-to-Noise ratio
CA	Access Control
CBMS	Convergence of Broadcast and Mobile Systems (DVB specifications)
CBR	Constant Bit Rate
CC	Code-Combining
CCSR	Call Completion Success Rate
CDF	Cumulative Distribution Function
CDP	Content Delivery Protocol
CEPT	Conference Européenne des Postes et Telecommunications
CFT	Cumulated Freeze Time
CGC	Complementary Ground Component
CMOS	Complementary Metal-Oxide Semiconductor
COFDM	Coded Orthogonal Frequency Division Multiplexing
COST	COperation européenne dans le domaine de la recherche Scientifique et Technique
CR	Code Rate
CRC16	16-bit CRC
CSSR	Call Set-up Success Rate
CU	Capacity Unit (DVB-SH)
CW	Code-Word, or Continuous Wave, depending on context
DA-MLE	Data Aided Maximum Likelihood Estimation
DANL	Displayed Average Noise Level
dBic	(Gain) in dB relative to Isotropic with Circular polarization
dBW	Decibel referenced to 1Watt
DFL	DataField Length
DU	Dense Urban
DVB-GBS	DVB Generic data Broadcasting & Service information protocols
DVB-H	Digital Video Broadcast - Handheld
DVB-S	Digital Video Broadcasting Satellite standard
DVB-S2	Digital Video Broadcasting Satellite standard version 2
DVB-T/H	This is a short form for "either DVB-T or DVB-H"
ECC	European Communications Committee
EEP	Equal Error Protection
EGC	Equal Gain Combining
EIRP	Equivalent Isotropic Radiated Power
EPG	Electronic Programme Guide
ERP	Effective Radiated Power
ERS	Empirical Road Side

ESG	Electronic Services Guide
ESR5	Errored-Second-Ratio at % in 20 s (ITU)
ESR5(20)	Performance index computed based on ESR5
FCC	Federal Communications Commission (US)
FDT	FEC Data Table
FEC	Forward Error Correction
FER	Frame Error Rate
FFT	Fast Fourier Transform
G/T	Receiver figure of merit
GBBF	Ground Based Beam Forming
GBS	should be DVB-GBS
GEO	Geosynchronous or Geostationary (orbit)
GI	Guard Interval (for OFDM)
GLCF	Global Land Cover Facility
GMR1 and 2	Geo-Mobile Radio 1 and 2 (ETSI standards)
GOP	Group Of Pictures
GPS	Global Positioning System
GSE	Generic Stream Encapsulation (IETF)
GSM	Global System for Mobile Communications
HEO	Highly Elliptical Orbit
HM	Hierarchical Modulation
HP	High Priority
HPA	High Power Amplifier
HTS	Heavy Tree Shadowing
Hyb	Hybrid
IBO	Input Back Off
IC	Interleaver Cycle
IDR	Instantaneous Decoding Refresh
iFDT	Interburst FEC Data Table
IFEC	Should be always MPE-IFEC
IG	Implementation Guidelines
IMT-2000	International Mobile Telecommunications-2000
IMUX	Input Multiplexer
INT	IP Notification Table (DVB-GBS)
IPDC	IP DataCasting (DVB specifications)
ISI	Inter Symbol Interference
ITS	should be LMS-ITS
ITU	International Telecommunication Union
IU	(Time) Interleaver Unit (DVB-SH)
Ka-band	Fequency band (defined by ITU) in the range of 18 GHz to 30 GHz
KIP	Key Indicator Parameter
Ku-band	Frequency band (defined by ITU) in the range of 10 GHz to 14 GHz
LEO	Low Earth Orbit
LHCP	Left Hand Circular Polarisation
LL	Low Latency
LL-FEC	Link Layer Forward Error Correction
LLR	Log-Likelihood Ratio
LMS	Land Mobile Satellite (channel)
LMS-ITS	LMS- Intermediate Tree Shadowing
LMS-SU	LMS- SubUrban
LNA	Low Noise Amplifier
LNB	Low Noise Block converter
LOS	Line Of Sight
LP	Low Priority
LSB	Low Significant Bits
MAC	Medium Access Control (layer)
MAPL	Maximum Allowable Path Loss
MBRAI	Mobile Radio Access Interface (specifications)
MEO	Medium Earth Orbit
MER	Modulation Error Ratio
MFER	see DVB-H implementation guidelines
MFN	Multi Frequency Network

MIP	Megaframe Initialisation Packet
MPA	Multiport Amplifier
MPE	Multi Protocol Encapsulation
MPE-FEC	see DVB-H implementation guidelines
MPEG TS	please change to MPEG-2 TS
MPEG2 TS	please change to MPEG-2 TS
MPEG2	please change to MPEG-2
MPEG-TS	please change to MPEG-2 TS
MPE-IFEC	MPE Inter-burst Forward Error Correction (DVB-SH)
MRC	Maximum Ratio Combining
MRP	Milestone Review Process
MSB	Most Significant Bit
msec	millisecond
MSS	Mobile Satellite System
mux_assoc	multiplex-association-vector
N_{CW}	Number of Coded Words (per SH-Frame)
NF	Noise Figure
NGSO	Non Geostationnary Satellite Orbit
NIT	Network Identification Table (DVB)
nm	Nano meter
NOAA	National Oceanic and Atmospheric Administration
NPR	Noise Power Ratio
OBO	Output Back-Off
OFDM	Orthogonal Frequency Division Multiple access
OFDM/TDM	OFDM or TDM
OMUX	Output Multiplexer
PA	Power Amplifier
PAT/PMT	Program Association Table/ Program Map Table
PDA	Personal Digital Assistant
PDF	Probability Density Function
PDP	Power Delay Profile
PFD	Power Flux Density
PHY	Physical Layer
PI	Portable Indoor
PID	Program Identifier (MPEG)
PIPO	Pedestrian Outdoor and Pedestrian Indoor
PL SLOTS	Physical Layer Slots
PL	Physical Layer
Pps	(one) pulse-per-second
PRBS	Pseudo Random Binary Sequence
PSD	Power Spectrum Density
PSI/SI	Program Specific Information / Service Information
PVR	Personal Video Recording
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
RHCP	Right Hand Circular Polarisation
RL	Regular Latency
RNPT	Radio Network Planning Tool
RS	Reed-Solomon (FEC)
RSC	Recursive Systematic Convolutional
RU	Rural
Rx	Receive
Sat	Satellite
SC	Satellite Component (DVB-SH)
SDARS	Satellite Digital Audio Radio System (US)
SDT	Service Description Table (DVB)
SF	Signalling Field
SFN	Single Frequency Network
SG	System Gain
SH	Satellite to Handheld (specification)

SH-A	SH- waveform configuration A (OFDM/OFDM)
SHA	Should be SH-A
SH-B	SH- waveform configuration B (TDM/OFDM)
SHB	Should be SH-B
SH-F	Same as SH-FRAME
SH-FRAME	A baseband frame defined in the present document
SHG	Soft Handover Gain
SHIP	SH frame Information Packet (DVB-SH)
SI	Service Information (DVB)
SLA	Service Level Agreement
SMS	Short Messages Service
SNIR	Signal to Noise + Interference Ratio
SNORE	Signal to Noise Ratio estimator
SNR	Signal-to-Noise Ratio
SRRC	Square-Root Raised-Cosine
STS	Synchronisation Time Stamp
SU	Sub-Urban
S-UMTS	Satellite UMTS
SVC	Scalable Video Coding
TBC	To Be Confirmed
TDL	Tapped Delay Line
TDM	Time-Division Multiplex
TDMA	Time Division Multiple Access
TER	Terrestrial
TMA	Tower Mounted Amplifier
TPS	Transmission Parameter Signalling
TS PER	Transport Stream Packet Error Rate
TS	Transport Stream
TU6	Typical Urban channel with 6 taps
TWTA	Travelling-Wave Tube Amplifier
TX	Transmitter
U	Urban
UEP	Unequal Error Protection
UHF	Ultra High Frequency
UL	Uniform Late
UL	Up Link
UMTS	Universal Mobile Telecommunications System
UP	User Packets
VBR	Variable Bit Rate (video)
VoD	Video on Demand
VSWR	Voltage Standing Wave Ratio
w.r.t.	With respect to
WER	Word Error Rate
WLAN	Wireless Local Area Network
xDSL	x type of Digital Subscriber Line
XML	eXtensible Markup Language

4 System outline

4.1 DVB-SH system capability outline

DVB-SH system provides an efficient and flexible mean of carrying broadcast services over an hybrid satellite and terrestrial infrastructure operating at frequencies below 3 GHz to a variety of portable, mobile and fixed terminals having compact antennas with very limited or no directivity. Target terminals include handheld defined as light-weight and battery-powered apparatus (e.g. PDAs, mobile phones), vehicle-mounted, nomadic (e.g. laptops, palmtops, etc.) and stationary terminals.

The broadcast services encompass streaming services such as television, radio programs as well as download services enabling for example Personal Video Recorder services.

The DVB-SH system coverage is obtained by combining a Satellite Component (SC) and, where necessary, a Complementary Ground Component (CGC) to ensure service continuity in areas where the satellite alone cannot provide the required QoS. The SC ensures wide area coverage while the CGC provides cellular-type coverage. All types of environment (outdoor, indoor, urban, suburban and rural) can then be served. It should be noted that the area served by a beam of currently planned multibeam satellites is in the order of 600 000 Km^2.

DVB-SH can provide the following theoretical total capacity per 5 MHz (typical) satellite bandwidth and per beam.

Table 4.1: DVB-SH typical and maximum net bit rates (Mbps) in Satellite-only coverage (Sat) and in Terrestrial coverage (TER)

Waveform		SH-A				SH-B	
Hybrid network frequency configuration		SFN		MFN		MFN	
		Typ	Max	Typ	Max	Typ	Max
Multibeam satellite system with 3x5 MHz spectrum assigned, 5 MHz allocated to each satellite beam in a 3-color reuse pattern	SAT/beam TER/beam	2,5 10,0	10,0 30,0	2,5 7,5	10,0 20,0	2,7 7,4	10,6 20,5
Multibeam satellite system with 4x5 MHz spectrum assigned, 5 MHz allocated to each satellite beam in a 4-color reuse pattern	SAT/beam TER/beam	2,5 13,7	10,0 40,0	2,5 11,2	10,0 30,0	2,7 11,1	10,6 30,43
NOTE 1: Values are indicative as different optimizations allowed by the standard lead to slightly different results. NOTE 2: The TER capacity includes the repetition of the SAT capacity.							

The total system capacity for a multibeam system is obtained by multiplying each number above by the number of beams. For example a hypothetical 9-beam satellite with 20 MHz bandwidth can offer, in terrestrial coverage typically 100 Mbps total (MFN/SH-B/Typ) and up to 360 Mbps total (SFN/SH-A/Max) when there is no power constraint. The same satellite can offer, in satellite-only coverage typically 24 Mbps total (MFN/SH-B/Typ) and up to 96 Mbps total (MFN/SH-B/Max) when there is no power constraint.

Currently planned multibeam satellite systems target a minimum of 6,6 Mbps total per 5 MHz of satellite bandwidth per beam (of which 2,2 Mbps capacity when in satellite-only reception conditions). Higher total capacity can be achieved through more satellite frequency-reuse, more powerful satellites, more CGC transmitters density or more advanced terminal technologies (e.g. antenna diversity).

4.1.1 System overview

A typical DVB-SH system (see figure 4.1) is based on a hybrid architecture combining a SC and, where necessary, a CGC consisting of terrestrial repeaters fed by a broadcast distribution network of various kinds (satellite [DVB-S/S2], terrestrial [fibre, xDSL, etc.]). Three kinds of terrestrial repeaters are envisaged:

- TR(a) are broadcast infrastructure transmitters which complement reception in areas where satellite reception is difficult, especially in urban areas; they may be collocated with mobile cell sites or standalone. Local content insertion at that level is possible, relying on adequate radio frequency planning and/or waveform optimizations.

- TR(b) are personal gap-fillers of limited coverage providing local re-transmission, on-frequency and/or with frequency conversion; typical application is indoor coverage provision, locally repeating the satellite signal available outdoor. No local content insertion is foreseen.

- TR(c) are mobile broadcast infrastructure transmitters creating a "moving complementary infrastructure" on board moving platforms (cars, trains, bus). Depending on waveform configuration and radio frequency planning, local content insertion may be possible.

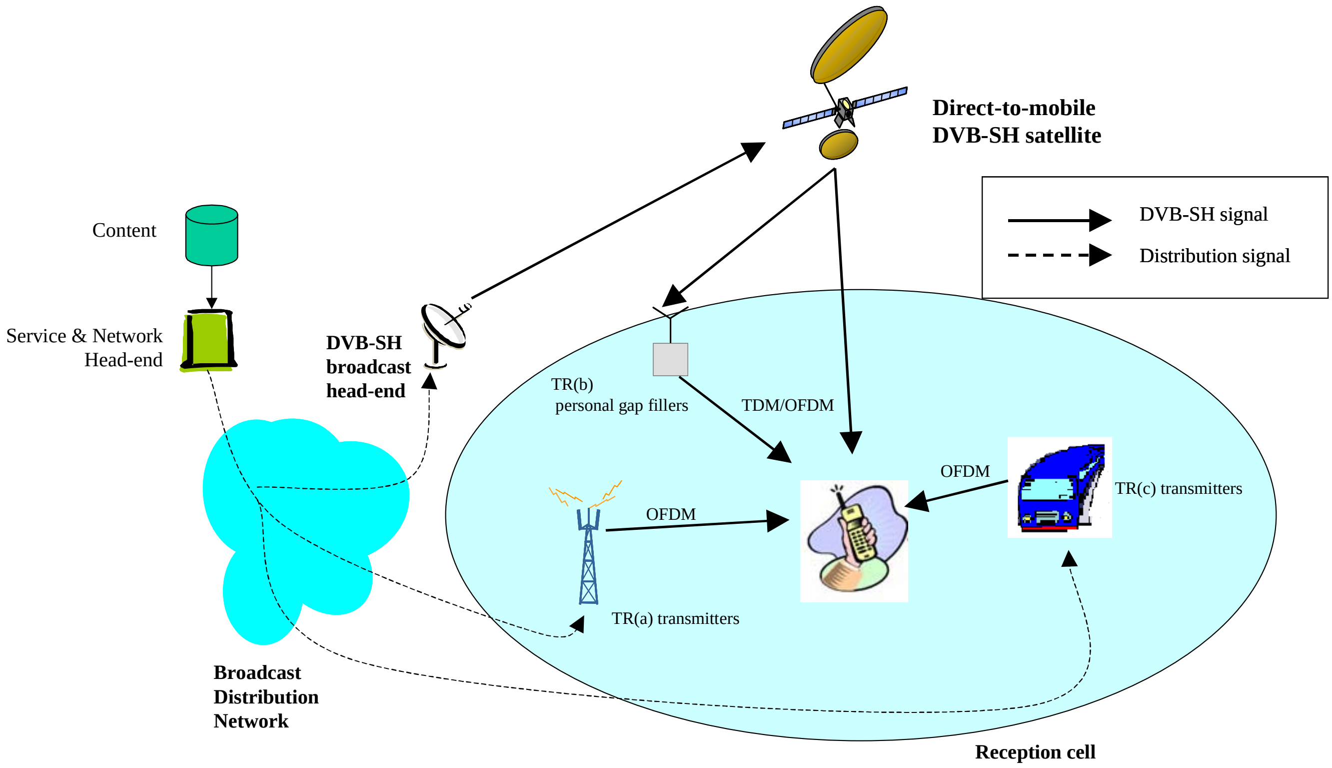


Figure 4.1: Overall DVB-SH transmission system reference architecture

The DVB-SH transmission system key features are the following:

- seamless service continuity between SC and CGC coverage;
- support of all reception conditions associated to portable and mobile terminals: indoor/outdoor, urban/suburban/rural, static/mobile conditions. Typical mobility conditions covers pedestrian as well as land vehicular scenarios;
- possible implementation of power saving schemes to minimize the power consumption of battery activated terminals in order to maximize autonomy;
- local insertion of broadcast services on CGC;
- use different kinds of distribution network to feed the CGC repeaters, such as Satellite (DVB-S/S2) and/or terrestrial (Optical fibre, Wireless Local Loop, xDSL, etc.) resources;
- possible offering of interactive services by inter-working with mobile or wireless systems at service, network head-ends and user terminal levels;
- possible use of low-latency services in parallel to regular latency services (see Annex D).

Two main different physical layer configurations are supported by the DVB-SH waveform standard [1]:

- SH-A exploiting OFDM transmission mode for both SC and CGC. SH-A allows (but does not impose) a Single Frequency Network (SFN) between the SC signal and the CGC signal carrying the same content. If SFN is implemented then FEC, Channel Interleaver, modulation and Guard Interval cannot be optimized separately for the SC and CGC (by definition of SFN). If it is desirable to optimize these parameters separately for the SC and the CGC, a distinct frequency channel for the SC and the CGC can also be used in SH-A, leading to reduced spectrum efficiency and some handover complications for receivers having only one RF front-end. If the receiver has two RF-front-ends, soft combining (see clause 4.2.6) and easier handover (see clause 9), can also be implemented with SH-A.

- SH-B exploiting TDM transmission mode for the SC and OFDM transmission mode for the CGC. SH-B requires a distinct frequency band for the SC and the CGC since they transmit signals based on two different physical layers. The impacts on the system frequency plan for this physical layer configuration are discussed in the next clause. Each component of the transmission system can be optimized separately to its respective transmission path.

The OFDM waveform is known to exhibit a larger peak-to-average signal envelope fluctuation compared to the TDM waveform. Therefore, SH-A is in general recommended for spectrum limited systems while SH-B is of interest in power limited satellite systems. However, the final choice depends on the detailed architecture of the payload (e.g. single or multiple amplifiers per beam, number of carriers/beam, etc.). Differently from SH-A in SFN configuration, SH-A in *non-SFN configuration* and SH-B also allow an independent optimization of the waveform and LL-FEC parameters for the SC and CGC.

4.1.2 Frequency planning aspects

Being a hybrid satellite/terrestrial system, the frequency planning for DVB-SH is typically more complex than for a pure terrestrial system or a pure satellite system. In fact due to the large geographical coverage provided by the satellite and the possible linguistic segmentation of the market, multibeam satellite may be used in certain areas (e.g. Europe). The use of a multibeam satellite with linguistic areas and/or nation-wide broadcasting requires the segmentation of the DVB-SH downlink spectrum. Let first introduce some key definition:

Sub-band: elementary unit of frequency resource in which the overall satellite bandwidth is partitioned.

Colour scheme: it represents the number of colours N_c adopted for allowing frequency reuse in the system. The colours can be provided by different frequency bands.

Frequency reuse factor f_R*:* it represents the number of times the same frequency sub-band is reused in the system thanks to antenna beams spatial or cross-polar isolation.

If the number of beams, N_b is large enough (e.g. greater than 3), then frequency reuse among satellite beams can be exploited. The multibeam solution brings the following advantages:

a) higher satellite antenna gain providing higher user terminal C/N for the same in-orbit RF power. The higher receiver C/N translates in higher spectral efficiency and/or link margin;

b) frequency reuse between non-adjacent beams in the SC (independently of the existence of the CGC);

c) frequency reuse between the SC and CGC.

To facilitate understanding we will assume in the subsequent that the system is using a single polarization and that the bandwidth allocated for the satellite transmission is the same for each beam and denote this unit of resource as a frequency "sub-band". Then the overall satellite downlink spectrum is partitioned into N_c sub-bands, with N_c=1 for a single-beam and typically, N_c=3 for a multibeam as shown in figure 4.2.

The same sub-band can be reused only by non adjacent satellite beams with a certain reuse pattern that depends on the antenna design. Typically a three-color frequency reuse scheme is sufficient to overcome interference between adjacent beams. The entire available spectrum (F_G in the global beam case) is then split into 3 frequency sub-bands denoted F_A, F_B and F_C. It should be kept in mind that, in the most general case colour scheme may require up to (but no greater than) four colours.

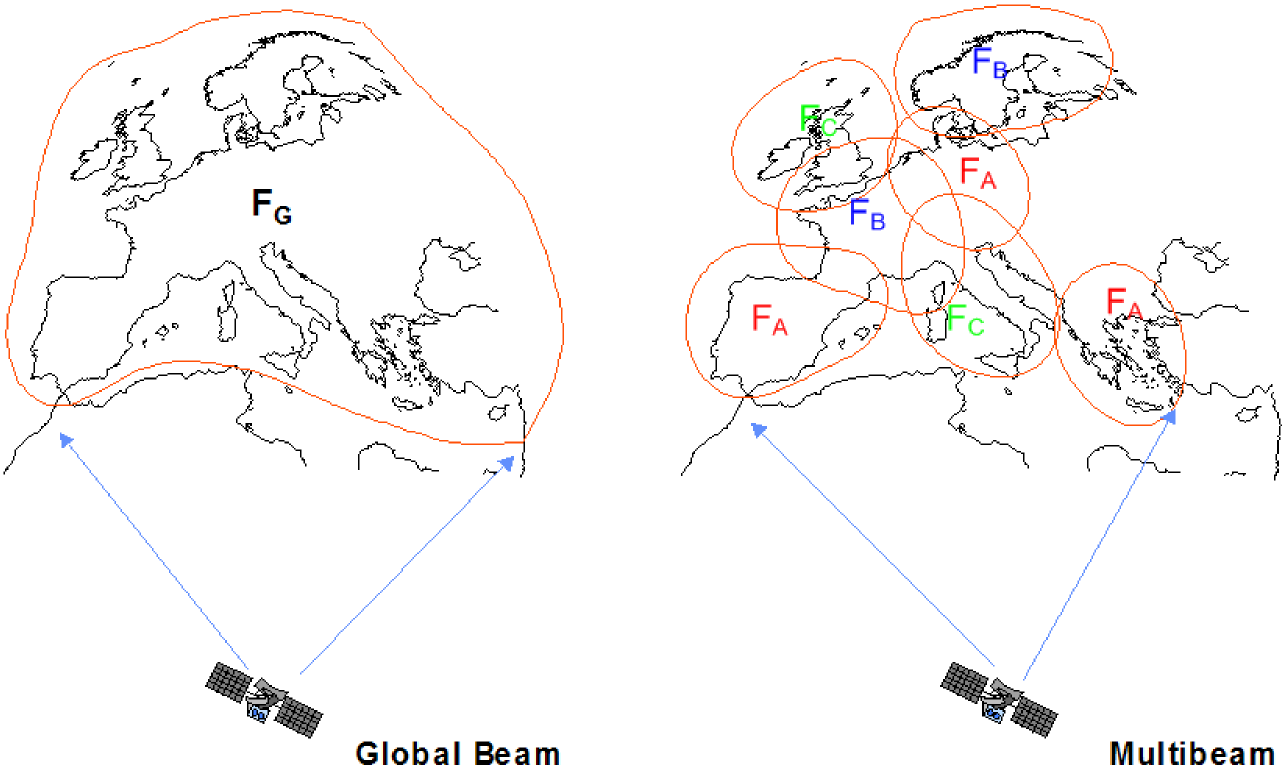


Figure 4.2: Example of satellite single and multibeam

The apparent drawback due to satellite spectrum segmentation is compensated by the fact that the satellite power can be focused over the wanted (and smaller) regions in a multibeam scenario. Furthermore, the global beam scenario would require broadcasting the whole multilingual multiplex to the global footprint, wasting of bandwidth and power in areas where reception of the signal content is not required (or even forbidden).

In the more general case also polarization (Left Hand Circular (LHC) and Right Hand Circular (RHC)) may be used to create isolation or frequency reuse in the system. Looking at the following figure one can distinguish four different cases for N_b=6 beams European coverage:

a) There are N_c=3 colours obtained dividing the system spectrum in 3 different sub-bands. The antenna polarization is alternated for the three sub-bands with no specific advantage obtained by using two different polarizations. There is an overall frequency reuse f_R=2 as each sub-band is reused twice over the coverage region.

b) There are N_c=3 colours obtained dividing the system spectrum in 3 different sub-bands each with different polarisation. The antenna polarization is alternated for the three sub-bands with different polarization patters for the first and the second triplet of beams. This polarization reuse scheme allows to increase the co-polar beam isolation compared to case a) by an amount equal to the cross-polar isolation factor. There is an overall frequency reuse f_R=2 as each sub-band is reused twice over the coverage region.

c) There are N_c=2 colours obtained dividing the system spectrum in 2 different sub-bands each with different polarisations. The antenna polarization is alternated for the two sub-bands with different polarization patters for the first and the second pair of beams. Same inversion of polarisation pattern is adopted for the second and the third couple of beams. This colour scheme allows to increase the frequency reuse scheme compared to cases a) and b) at the expenses of a lower antenna co-polar beam isolation. There is an overall frequency reuse f_R=3 as each sub-band is reused three times over the coverage region.

d) There are N_c=3 colours obtained dividing the system spectrum in 3 different sub-bands each using the two possible polarisations in each beam. This polarization reuse scheme allows to double the bandiwidth associated to each beam compared to cases a) and b) while obtaining a beam co-polar isolation slightly lower to case a) due to the extra cross-polar co-frequency other beams interference. There is an overall frequency reuse f_R=4 as each sub-band is reused four times over the coverage region.

Figure 4.3: Examples of satellite colour schemes and frequency reuse for a 6 beams (N_b=6) European coverage

When the frequency is reused among different beams or polarisations it is recommended to use different physical layer scrambling sequences for each beam and/or polarisation. The rationale for that and more detailed guidelines are provided in Annex E.

The resulting spectral efficiency improvement factor is given by:

$$\gamma(N_b, N_c) = \frac{N_b}{N_c} f_R$$

It is observed that for N_b=6 beams there is:

- A four-fold spectral efficiency improvement compared to the global beam case for cases a) and b).
- A nine-fold spectral efficiency improvement compared to the global beam case for case c).
- An eight-fold spectral efficiency improvement compared to the global beam case for case d).

In conclusion although more complex from the satellite payload perspective the colour and frequency reuse scheme d) appears a good solution in terms of spectral efficiency and co-polar beam isolation.

The system frequency plan and spectrum efficiency are also impacted by the decision to use or not the SFN technique to complement the satellite coverage with the CGC coverage.

If SFN is used by the CGC to complement the satellite coverage, the common content is transmitted by the CGC using the same sub-band as the satellite signal. By definition of SFN, no additional local content can be added in this sub-band. However, within each beam, the remaining N_c-1 sub-bands, which cannot be used by satellite due to adjacent beams, may be used by the CGC to carry local content.

Alternatively, if SFN is not used, the spectrum must be divided into at least 2 sub-bands: one for the direct-to-user transmission from satellite and the other for the complementary and indirect transmission from the CGC. In this case, this second sub-band may be used to carry additional local content by the CGC since the SFN constraint does not apply. If N_c>2, the remaining N_c-2 sub-bands may be used by the CGC to carry additional local.

When planning additional local content transmission it should be kept in mind that strong interference from satellite into terrestrial may occur in quite large *exclusion zones* (see for example, in figure 4.2, the overlap of the beam centred on France to the south of UK territory). This means that, for sub-bands not reserved for satellite in each beam, the terrestrial transmission parameters and the corresponding terrestrial coverage/capacity cannot be identical within a territory and must be carefully engineered, in coordination with the satellite operator.

If Multi Frequency Network (MFN) is used, different physical-layer configurations will be possible in different areas of a territory (e.g. higher spectral efficiencies modulation and FEC and/or guard-times and/or hierarchical modulation). Therefore MFN allows optimization for different transmitter densities, for different terrestrial coverage and different capacity requirements. In SFN instead, the terrestrial signal complementing the satellite signal has to use exactly and uniformly the common configuration with the satellite signal.

For the various reasons above DVB-SH does not impose the use of SFN for SH-A.

4.1.3 Other considerations for spectrum efficiency

A satellite customer traditionally leases a satellite beam with an associated satellite frequency sub-band. To convert the leased resource into a program offer, he needs a measure of system performance called spectrum efficiency which is given in bits/s/Hz. For example an efficiency of 0,5 bits/s/Hz means that, using 5 MHz satellite bandwidth the customer can build a bouquet of programs with data rates summing up to 2,5 Mbits/s (8 to 10 Mobile TV programs).

In DVB-SH, this customer also gets a frequency "bonus" which is the frequency resource available for potential use terrestrially. This by-product is a virtual asset which becomes a real resource when:

i) the customer obtains the authorization from his government authority to use this spectrum terrestrially; and

ii) he deploys/leases the appropriate CGC infrastructure.

Therefore in DVB-SH, three spectrum efficiencies can be distinguished, all normalized to the *leased satellite bandwidth*:

- The $\eta[COM]$ is the spectrum efficiency for the "common" programs, per Hz of satellite bandwidth in a beam.
- The $\eta[LOC]$ is the spectrum efficiency for the "local" programs, per Hz of satellite bandwidth in a beam.
- The $\eta[TOT]$, or system spectrum efficiency, is the sum of the above efficiency parameters.

The $\eta[TOT]$ gives an indication of the total programs offer that can be provided with satellite and terrestrial transmissions combined, normalized to the leased satellite bandwidth.

Note that in the "exclusion zones", coexistence of terrestrial transmissions with satellite transmissions in adjacent beams can impose lower $\eta[LOC]$. Also, in MFN between satellite and CGC, the OFDM guard-time can be engineered differently from one local zone to another, resulting in different $\eta[LOC]$. Therefore $\eta[LOC]$ and consequently $\eta[TOT]$ should be used as local performance indexes and not a nation-wide index.

To evaluate and/or optimize DVB-SH system spectral efficiency as defined above, first the following must be determined/computed.

Table 4.2

Parameter/ Questions	Explanation	Main Dependencies	General target values
N_{beams}	The number of satellite beams	Number of countries/linguistic regions, satellite antenna sub-system cost, inter-beam interferences	≥ 3 (for regional coverage) (for large countries, Nbeams = 1)
f_R	Satellite frequency and/or polarization reuse over the beams (also called number of "colours")	N_{beams} , national/ linguistic boundaries, satellite power, frequency, polarization allocation and channelization.	3
Are there regulatory limitations for local content?	Local content may be forbidden or its amount restricted	Regulatory context	Country dependent
Is SFN used by CGC to complement satellite?	For SH-B, the answer is negative by definition. For SH-A it is possible to choose between SFN and MFN	SH-A/SH-B, handover complexity, chipset availability	Equally likely
Does the satellite allow nonlinear operation of TWTA?	In general yes with different levels of operating HPA OBO depending on selected waveform (SH-A or SH-B) or payload architecture (single HPA/beam or shared HPA/beam)	System and payload design. Market definition	Equally likely
$\eta_{OFDM}[SAT]$	Spectral efficiency of the OFDM waveform when used by satellite, given minimum terminal characteristics and Quality of Service	Satellite effective EIRP, Terminal characteristics, required link margin (QoS), allowed end-to-end max delay	0,5 b/s/Hz
$\eta_{TDM}[SAT]$	Spectral efficiency of the TDM waveform when used by satellite, given minimum terminal characteristics and Quality of Service	Satellite effective EIRP, Terminal characteristics, required link margin (QoS), allowed end-to-end max delay	0,5 b/s/Hz
$\eta_{OFDM}[TER]$	Spectral efficiency of the OFDM waveform when used by CGC, given minimum terminal characteristics, coverage type (indoor/outdoor) and Quality of Service	CGC network cost, indoor-outdoor operations, terminal types, required link margin (QoS)	0,7 b/s/Hz
Total payload equivalent signal power degradation D_{TOT}	Total payload losses due to payload nonlinear effects (output back off, signal distortion, intermodulation noise)	Payload characteristics, physical layer settings, required margin (QoS), interference scenario	1 dB to 1,5 dB for single carrier TWTA operation, 2,5 dB to 3,5 dB for multi-carrier TWTA operation

Several frequency plans are possible depending on the SH-A or SH-B architecture and SFN or MFN configurations. Clause C.1 provides examples and a methodology for spectral efficiency computations and comparisons.

4.1.4 Frequency bands and their regulatory aspects

4.1.4.1 Frequency Allocations and coordination procedures

Frequency spectrum is regulated at the international level by a binding treaty called the Radio Regulations. The Radio Regulations deal with two aspects: frequency allocation and regulatory procedures for accessing the spectrum/orbit resources. The binding character of the Radio Regulations implies that national or regional (e.g. European) regulations shall be defined within the framework of this treaty.

Frequencies between 1 GHz and 3 GHz are the most suitable, considering the satellite, terminal and mobility constraints. Within this frequency range, the following bands are candidates for the provision of multimedia services based on DVB-SH (the uplink bands are given for completeness and their use are out of the scope of the DVB-SH specifications).

Table 4.3

Frequency band designation	Frequency range	Others common names
L band	1 626,5 MHz to 1 660,5 MHz, 1 668 MHz to 1 675 MHz (uplink) 1 518 MHz to 1 559 MHz (downlink)	MSS GEO L band
	1 610 MHz to 1 626,5 MHz (up and downlink)	MSS Big LEO L band
	1 452 MHz to 1 492 MHz (downlink)	S-DAB band
2 GHz S band	1 980 MHz to 2 010 MHz (uplink) 2 170 MHz to 2 200 MHz (downlink)	MSS S-Band
S-DARS S band	2 320 MHz to 2 345 MHz (downlink)	
2,5 GHz S band	2 670 MHz to 2 690 MHz (uplink) 2 500 MHz to 2 520 MHz (downlink)	
	2 520 MHz to 2 670 MHz (downlink)	

4.1.4.1.1 L band

Frequency allocations

In the field of satellite communications, the term "L band" encompasses three frequency bands:

- the bands 1 626,5 MHz to 1 660,5 MHz and 1 668 MHz to 1 675 MHz (uplink)/1 518 MHz to 1 559 MHz (downlink): these bands are allocated almost worldwide to the mobile-satellite service and are mainly used by incumbent MSS operators using geostationary satellites to provide two-ways mobile satellite communications;
- the band 1 610 MHz to 1 626,5 MHz (up and downlink): this band is allocated worldwide to the mobile-satellite service and is used by incumbent MSS operators using non-geostationary satellite systems in a low Earth orbit (LEO) to provide the same type of services as in the previous bands. It should be noted that one non-geostationary system uses this band only for its uplinks (its downlink using the band 2 483,5 MHz to 2 500 MHz);
- the band 1 452 MHz to 1 492 MHz (downlink): this band is allocated worldwide (with the only exception of the USA) to the broadcasting-satellite service but only the upper 25 MHz (i.e. 1 467 MHz to 1 492 MHz) can actually be used since the lower 15 MHz are not usable until a decision by a future ITU World Radiocommunications Conference. This band is also allocated on a co-primary basis with the broadcasting-satellite service to the terrestrial fixed, mobile and broadcasting services. The use of this band by the broadcasting-satellite and terrestrial broadcasting services is limited to digital audio broadcasting.

Access to the spectrum/orbit resources

In the bands 1 518 MHz to 1 559 MHz, 1 610 MHz to 1 626,5 MHz, 1 626,5 MHz to 1 660,5 MHz and 1 668 MHz to 1 675 MHz, geostationary satellites and non-geostationary systems are on an equal footing with regards to spectrum access (see provision No. 9.11A of the Radio Regulations [20]). The basic principle governing the coordination under No. 9.11A [20] is "first come, first served".

In the band 1 467 MHz to 1 492 MHz (since the band 1 452 MHz to 1467 MHz is de facto unusable), geostationary satellites have regulatory priority with regards to non-geostationary systems (see provision No. 22.2 of the Radio Regulations [20]). Among themselves, geostationary satellites are required to coordinate according to Radio Regulations provision No. 9.7 [20] ("first come, first served" but only between geostationary satellites). Moreover, all systems are required to coordinate with terrestrial services under provision No. 9.11 [20]. It should be noted that the protection of terrestrial services in the United States of America renders operations of a geostationary satellite in visibility of the US territory very challenging if not unfeasible.

4.1.4.1.2 2 GHz S band (1 980 MHz to 2 010 MHz/2 170 MHz to 2 200 MHz)

Frequency allocations

At the international level, the band 2 170 MHz to 2 200 MHz is allocated worldwide to the fixed, mobile and mobile-satellite (space-to-Earth) services on a co-primary basis. It means that, from a regulatory point of view, these three services enjoy the same status. It does not mean that applications provided through these services are necessarily compatible from a technical point of view (i.e. that they can use the same band without interfering each other). On the contrary, numerous studies have shown that terrestrial mobile applications and satellite mobile applications are not compatible if they are operated independently in the same geographical area. Complementary ground components (CGC) are compatible with the provision of satellite applications to mobile terminals since they are operated in a coordinated manner with the satellite system.

It should be noted that in the Americas, the adjacent band 2 160 MHz to 2 170 MHz is also allocated, inter alia, to the mobile-satellite service (space-to-Earth).

Some countries have chosen to favour terrestrial systems in these bands: in Algeria, Benin, Cape Verde, Egypt, Iran, Mali, Syria and Tunisia, the use of these bands by the mobile-satellite service shall neither cause harmful interference to the fixed and mobile services, nor hamper the development of those services prior to 1 January 2005, nor shall the former service request protection from the latter services. While this has not a direct impact on the overall design and operations of MSS systems intended to cover Europe, it may have an impact on the design of satellite antennas compatible with the protection of terrestrial services in these bands.

The fact that a country is not listed above does not imply that it intends to give priority to satellite services but simply that it has chosen to keep satellite and terrestrial services on an equal footing from the point of view of international regulations. National regulators may chose to restrict nationally the use of these bands to a specific service. For more information on the regulatory status of these bands, the national table of frequency allocations shall be consulted.

NOTE: National tables of frequency allocations of most of European countries can be accessed in: www.efis.dk.

Access to the spectrum/orbit resources at the international level

In these bands, geostationary satellites and non-geostationary systems are on an equal footing with regards to spectrum access (see provision No. 9.11A of the Radio Regulations [20]). The basic principle governing the coordination under No. 9.11A [20] is "first come, first served".

Moreover, in the band 2 170 MHz to 2 200 MHz, a coordination procedure between transmitting space stations and terrestrial services is established through the provision No. 9.14 of the Radio Regulations [20]. The PFD thresholds used to determine whether there is a need to coordinate with terrestrial services in any country are:

Table 4.4

	For geostationary satellites	For non-geostationary satellites
For 0° ≤ δ ≤ 5°	-128 dBW/m²/MHz	-123 dBW/m²/MHz
For 5° ≤ δ ≤ 25°	-128 + 0,5(δ - 5) dBW/m²/MHz	-123 + 0,5(δ - 5) dBW/m²/MHz
For 25° ≤ δ ≤ 90°	-118 dBW/m²/MHz	-113 dBW/m²/MHz
NOTE: Where δ is the angle of arrival (degrees).		

It should be noted that these pfd levels are not compatible with MSS operations towards handheld terminals. Therefore this coordination threshold will always be exceeded over the territories of countries within the service area of the MSS system. It will not be a constraint as such, since it is widely recognized that such systems cannot share the same geographical area with terrestrial services. Therefore countries willing to encourage the provision of MSS services in these bands over their territory have taken steps to free the bands from any terrestrial systems. However, the same does not apply in countries not belonging to the MSS service area. On their territories, these administrations may wish to continue to deploy terrestrial services as planned and may require protection from MSS transmitting satellites. This request for protection has to be explicit: following the publication of the MSS system special section by the ITU, any administration considering itself as potentially affected has four months to explicitly request to be included in the coordination process. Otherwise, it is deemed to be unaffected. This mechanism substantially reduces the number of administrations actually included in the process of coordination with respect to their terrestrial services.

4.1.4.1.3 Sound Broadcasting Satellite (SDARS) S band (2 310 MHz to 2 360 MHz)

The band 2 310 MHz to 2 360 MHz is allocated to the sound broadcasting-satellite service in only three countries: India, Mexico and the USA. Only in the latter is the band currently used for such service. More particularly, only 25 MHz are available in the USA for such a service (2 320 MHz to 2 345 MHz). This 25 MHz band was divided into two blocks of 12,5 MHz and auctioned by the FCC in April 1997. At the time of writing these guidelines, the two blocks were fully used by their respective owners for the provision of digital radio via satellite supplemented by ancillary ground repeaters in areas where the satellite signal is too weak to be received with the required quality. These ground repeaters are using part of the frequencies available in the 12,5 MHz blocks.

4.1.4.1.4 2,5 GHz S band (2 500 MHz to 2 690 MHz)

The band 2 500 MHz to 2 690 MHz includes various allocations to satellite services. Of particular interest for the introduction of DVB-SH based services are the following:

- 2 500 MHz to 2 520 MHz (downlink)/2 670 MHz to 2 690 MHz (uplink): these paired bands are allocated worldwide to the mobile-satellite service (see note 1) and are shared, inter alia, with terrestrial (fixed and mobile) services. As for the bands 1 980 MHz to 2 010 MHz/2 170 MHz to 2 200 MHz (see above), coordination under No. 9.11A [20] (i.e. "first come, first served") applies in these bands and therefore geostationary satellites and non-geostationary systems are on an equal footing with regards to spectrum access. Similarly to the band 2 170 MHz to 2 200 MHz, in the band 2 500 MHz to 2 520 MHz, a coordination procedure between transmitting space stations and terrestrial services is established through the provision No. 9.14 of the Radio Regulations [20]. The pfd thresholds used to determine whether there is a need to coordinate with terrestrial services in any country are -128 MHz/-118 dBW/m²/MHz depending on the elevation angle.

NOTE 1: With the exception of Azerbaijan, Bulgaria, Kyrgyzstan and Turkmenistan.

- 2 520 MHz to 2 670 MHz: this band is allocated to the broadcasting-satellite service and is also shared, inter-alia, with terrestrial (fixed and mobile) services.

It should be noted that the European Conference of Postal and Telecommunications administrations (CEPT) has decided in 2005 that the frequency band 2 500 MHz to 2 690 MHz is designated for terrestrial IMT-2 000/UMTS systems only (see note 2). This band is therefore not suitable for the provisions of satellite services within Europe.

NOTE 2: See ECC Decision of 18 March 2005 on the harmonized utilisation of spectrum for IMT-2000/UMTS operating within the band 2 500 MHz to 2 690 MHz (ECC/DEC/(05)05 [i.25]).

4.1.4.2 Regulatory aspects

4.1.4.2.1 Europe

4.1.4.2.1.1 European Union

L band and 2,5 GHz S-band:

No specific regulatory instruments on the designation of the frequency bands for satellite use have been adopted by the European Union in these bands.

2 GHz S-band:

On 14 February 2007, the European Commission adopted a Decision "on the harmonized use of radio spectrum in the 2 GHz frequency bands for the implementation of systems providing mobile satellite services" (see note 1) which gives priority to the development of MSS in the 2 GHz bands. Pursuant to the decision, Member States will be obliged to designate and make available as of 1 July 2007 the 2 GHz S-bands for systems providing mobile satellite services, which can include the use of complementary ground components. A European Decision is mandatory for all Member States of the European Union. It gives therefore more regulatory certainty than CEPT Decisions for the use of these bands.

NOTE 1: This Decision 2007/98/EC [i.26] was published in the Official Journal of 14 February 2007.

On 30 June 2008, the European Parliament and the Council adopted jointly a Decision on the selection and authorisation of systems providing mobile satellite services (MSS) in the 2 GHz band. Pursuant to this decision, Member States will be obliged to ensure that the selected applicants have the right to use the specific radio frequency identified in the Commission decision and the right to operate a mobile satellite system. In particular, Member States shall ensure that their competent authorities grant to the selected applicants the authorisations necessary for the provision of the complementary ground components of mobile satellite systems on their territories. In Recital 8 of the Decision, it was recalled that "Complementary ground components are an integral part of a mobile satellite system and are used, typically, to enhance the services offered via the satellite in areas where it may not be possible to retain a continuous line of sight with the satellite due to obstructions in the skyline caused by buildings and terrain".

NOTE 2: This Decision 2008/626/EC [i.33] was published in the Official Journal of 2 July 2008.

4.1.4.2.1.2 European Conference of Postal and Telecommunications administrations (CEPT)

L band:

CEPT has decided in 2004 that the frequency bands 1 518 MHz to 1 525 MHz and 1 670 MHz to 1 675 MHz are designated to the mobile-satellite service from 1 April 2007 (see note 1 above). While no specific CEPT instruments designate the frequency bands 1 525 MHz to 1 559 MHz and 1 610 MHz to 1 660,5 MHz to the mobile-satellite service, other CEPT decisions on free circulation and use or on licence exemption of terminals imply that these bands are, de facto, designated to such service.

NOTE 1: See ECC Decision of 12 November 2004 on the designation of the bands 1 518 MHz to 1 525 MHz and 1 670 MHz to 1 675 MHz for the Mobile-Satellite Service (ECC/DEC/(04)09 [i.27]).

CEPT has decided in 2003 that the frequency band 1 479,5 MHz to 1 492 MHz is designated for use by satellite digital audio broadcasting systems (see note 2). It should be noted that the band 1 467 MHz to 1 479,5 MHz is part of the Maastricht Plan for the introduction of terrestrial T-DAB systems in L band. The use by satellite systems of this lower band in Europe is generally not compatible with T-DAB systems.

NOTE 2: See ECC Decision of 17 October 2003 on the designation of the frequency band 1 479,5 MHz to 1 492 MHz for use by Satellite Digital Audio Broadcasting (S-DAB) systems (ECC/DEC/(03)02 [i.28]).

No specific regulatory instruments have been adopted by the CEPT in these bands concerning complementary ground components.

2 GHz S band:

In December 2006, the European Communications Committee (ECC) adopted two Decisions and one Recommendation designed to foster the regulatory certainty of the 2 GHz bands.

- Band designation for MSS systems, including those supplemented by a Complementary Ground Component (CGC) (see note 1): the bands 1 980 MHz to 2 010 MHz and 2 170 MHz to 2 200 MHz are designated for systems of the mobile-satellite service. Since no other ECC Decision exists to designate these bands for other services, it means that these bands are exclusive for MSS systems. The same Decision explicitly mentions the possibility for MSS systems in these bands to incorporate a complementary ground component (CGC). This explicit mention gives more regulatory certainty for deploying such ground-based infrastructure across the whole Europe.

NOTE 3: See Decision ECC/DEC/(06)09 [i.29].

- Harmonized regulatory conditions for complementary ground components: within the same Decision, CGC is defined as follows: "CGC is an integral part of a mobile satellite system and consists of ground based stations used at fixed locations to improve the availability of the mobile satellite system in zones where the communications with one or several space stations cannot be ensured with the required quality. CGC uses the same portions of the mobile-satellite service frequency bands (1 980 MHz to 2 010 MHz/2 170 MHz to 2 200 MHz) as authorized for the associated space station(s)". Moreover, some conditions are attached to the use of CGC:
    - the frequency band to be used by the CGC of a particular satellite system shall be accommodated within the same frequency band used by the satellite component of that satellite system;
    - the use of CGC shall not increase the spectrum requirement of the satellite component of that particular mobile satellite system;

- the CGC shall only be deployed in the geographical areas where the mobile earth stations of the associated MSS system are authorized to operate;
- the same direction of transmission by CGC and the satellite component shall be used so as to decrease the number and complexity of compatibility issues;
- the satellite segment shall be re-established as soon as possible in case of failure of the satellite segment, and no later than 18 months after such a failure, unless justified otherwise on considerations based on reasonableness and/or proportionality. Otherwise, CGC shall cease operation;
- compatibility with terrestrial IMT-2 000/UMTS operational systems in adjacent bands should be ensured;
- the CGC shall not operate independently from the satellite resource/network management system.

• Spectrum reframing of the 2 GHz bands (see note 2): in addition to the Decision designating the 2 GHz bands for MSS, the ECC also agreed on a Decision on "transitional arrangements for the fixed service and tactical radio relay systems in the bands 1 980 MHz to 2 010 MHz and 2 170 MHz to 2 200 MHz in order to facilitate the harmonized introduction and development of systems in the mobile satellite service including those supplemented by a complementary ground component". This Decision is threefold:
  - finalizing the transitions of existing fixed service systems and tactical radio relay systems operating in or overlapping the 1 980 MHz to 2 010 MHz and 2 170 MHz to 2 200 MHz bands;
  - not implementing any new FS networks and tactical radio relay systems which are incompatible with MSS operations in the bands 1 980 MHz to 2 010 MHz and 2 170 MHz to 2 200 MHz;
  - not requiring any longer coordination of MSS systems with respect to fixed service and tactical radio-relay systems (see section 4.1.3.1.2, coordination under No. 9.14 [20]).

NOTE 4: See Decision ECC/DEC/(06)10 [i.30].

• Recommendation on Milestone Review Process (see note 3): at its December 2006 meeting, the ECC adopted a Recommendation establishing a Milestone Review Process (MRP) in order to gather precise information about the development of the various satellite projects intending to use the bands. This ECC Recommendation lists the milestones for which MSS systems proponents are invited to submit information in order to monitor the development of projects in these bands. It was initially envisaged to use this MRP as a binding process to choose MSS systems eligible for a frequency authorization in these bands; however the lack of legal power of CEPT to enforce such a process was deterrent to its implementation. The MRP is now only of an informative nature. All declarations are based on good faith and will not be checked against. Any binding decision should be transferred to the European Commission. The milestones consists in evidence of submission of ITU request for co-ordination, satellite manufacturing contract, completion of the Critical Design Review of the MSS system, satellite launch agreement, gateway earth stations procurement, successful satellite mating, successful satellite launches, completion of frequency co-ordination as well as actual provision of satellite service within the territories of CEPT countries.

NOTE 5: See ECC/Recommendation (06)05 [i.31].

2,5 GHz S band:

As mentioned above, CEPT has decided in 2005 that the frequency band 2 500 MHz to 2 690 MHz is designated for terrestrial IMT-2 000/UMTS systems only (see note). This band is therefore not suitable for the provisions of satellite services within Europe.

NOTE 6: See ECC Decision of 18 March 2005 on the harmonized utilisation of spectrum for IMT-2 000/UMTS operating within the band 2 500 MHz to 2 690 MHz (ECC/DEC/(05)05 [i.25]).

4.1.4.2.2 North America (all bands)

In the United States of America, the available bandwidth is limited by the Table of Frequency Allocations to 20 MHz (i.e. 2 180 MHz to 2 200 MHz coupled with the uplink band 2 000 MHz to 2 020 MHz). In these bands, ancillary terrestrial components may be operated in conjunction with MSS networks, subject to the Commission's rules for ancillary terrestrial components (ATC) and subject to all applicable conditions and provisions of the MSS authorization. The FCC generally imposes to an MSS licensee that wishes to include ATC to meet five requirements (see note 1 in clause 4.1.4.2.1.2):

- geographic coverage: an MSS licensee must provide space-segment service across the entire geographic area where the ATC can be deployed;
- coverage continuity: MSS operators must maintain space station coverage over the relevant geographic area, which implies timely replacement of satellites in the event coverage should degrade as a result of satellite failure. In order to implement this condition, a non-geostationary MSS system licensee is required to maintain an in-orbit spare, while a geostationary MSS system licensee is required to maintain a spare satellite on the ground within one year of commencing operations and launch it into orbit during the next commercially reasonable launch window following a satellite failure;
- commercial availability: the MSS service via satellite must be commercially available as a perquisite to any offering of the ATC service;
- an integrated offering: MSS licensees must offer an integrated service. MSS licensees must make an affirmative showing to the FCC that demonstrates that their ATC service offering is truly integrated with their MSS offering. As an example, MSS licensees that wish to provide ATC services could demonstrate that they use a dual-mode handset to provide the proposed ATC service; and
- in-band operation: the ATC operations must remain limited to the precise frequency assignments authorized for the MSS system.

NOTE 1: For more information, see Federal Communications Commission - Report and Order and Notice of Proposed Rulemaking 03-15 - In the matter of Flexibility for Delivery of Communications by Mobile Satellite Service Providers in the 2 GHz Band, the L-Band, and the 1,6 GHz to 2,4 GHz Bands (released 10 February 2003).

In Canada, the available bands for mobile-satellite services are 2 165 MHz to 2 200 MHz (space-to-Earth) and 1 990 MHz to 2 025 MHz (Earth-to-space). Industry Canada has issued generic regulations for Ancillary Terrestrial Components (ATC) (see note 2) that can be used in conjunction with MSS systems in the frequency range 1 GHz to 3 GHz. These regulations are based on the following principles:

- the ATC mobile service will be an integral part of MSS service offerings. A substantial level of mobile-satellite services will be provided with the ATC service;
- the frequencies used for the ATC system will be within the assigned spectrum for a particular MSS network and the ATC service will be limited to the satellite serving areas. The use of the MSS spectrum for ATC operation will be subordinate to the spectrum being available for mobile-satellite service;
- the ATC mobile service will be required to cease operation, within a reasonable period, should the mobile-satellite service or network be discontinued;
- the ATC operation will be authorized such that it will neither cause harmful interference to, nor claim protection from, MSS services and other primary radio services operating in adjacent bands. ATC operations will be subject to technical and operational requirements considered appropriate to mitigate potential interference;
- complete applications as radiocommunication carriers will need to be submitted to seek authorization to operate an ATC mobile system as an integral and indefeasible part of the MSS service offerings. Specific information will be required as part of the applications to demonstrate adherence to policy, operational and regulatory principles;
- spectrum area licences will be issued for ATC systems and will be subject to spectrum fees.

NOTE 2: See Industry Canada - Notice No. DGTP-006-04 - Spectrum and Licensing Policy to Permit Ancillary Terrestrial Mobile Services as Part of Mobile-Satellite Service Offerings (May 2004).

4.2 Main issues addressed by DVB-SH

4.2.1 Differences in propagation characteristics between satellite and terrestrial channels

For frequency below 3 GHz, the only significant effects are those caused by the environment close to the user terminal as atmospheric effects are negligible (see ITU-R Recommendation P.676-8 [25], ITU-R Recommendation P.618-10 [31], ITU-R Recommendation P.531-10 [32] and ITU-R Recommendation P.680-3 [33]).

For DVB-SH, three main mobile environments may be considered:

- *rural environment*: the propagation is mainly affected by the vegetation. The coverage is mainly provided by the SC;
- *urban environment*: the propagation is mostly affected by dense buildings or other constructions with height of 4 storeys or more. The coverage is mainly provided by the CGC; and
- *suburban environment*: representing an intermediate case with medium density of buildings, lower structures (2 to 3 storeys) and roads which are wider than in an urban environment. The SC and the CGC contribute to the desired coverage. Small villages may be treated as suburban areas where, if the population density is low, satellite may be the only source for service provision.

For both LMS and terrestrial mobile channels, the effects are conventionally divided into three types according to the scale of distances to be considered:

- *path loss at large scale* (very slow fluctuations): the signal suffers variations due to modifications in the geometry of the propagation path. This loss is usually assumed to be proportional to d^n, d being the distance from the transmitter to the user terminal and n being an empirical exponent, based on theory and measurements. For satellite channels n=2 while for terrestrial channels typically n=4;
- *shadowing at mid-scale* (slow fluctuations): it corresponds to amplitude variations due to nearby obstacles on the ground. The signal suffers variations due to obstructions caused by buildings, trees etc., the scale here being similar to the dimensions of these obstacles; and
- *multipath fading at small scale* (fast fluctuations): the scale of the variations of the signal is about one wavelength, as a result of the constructive or destructive addition of multiple paths. For wideband systems, it is necessary to consider the multipath fading as frequency selective. The satellite propagation channel in DVB-SH is generally non-frequency selective. This assumption is very accurate for 1,5 MHz, pretty correct for 5 MHz and fairly correct for 8 MHz channelization. It means that the satellite propagation channel can be considered as a single complex multiplicative process. Instead the terrestrial repeaters propagation channel is considered to be frequency selective because of their respective coherence bandwidths. A frequency selective fading is classically characterized through a Power Delay Profile (PDP) which gives the relative time of arrival, the relative power and the type (Ricean or Rayleigh distribution, spectrum) of each group of unresolved echoes (also called tap). These PDP are then used to parameterize Tapped Delay Line (TDL) models. For SFN operation between satellite and terrestrial OFDM (SH-A, SFN), all contributions have to be taken into account.

In summary, the LMS channel has very different characteristics compared to the terrestrial one. In particular, the channel can be considered as non frequency selective and link margins are not as large as for terrestrial networks. More specifically, the satellite distance from the user is basically constant within the beam and the satellite power limitations make the link margin bounded to a value typically ranging from 5 dB to 15 dB.

4.2.2 LMS channels (in the L and S bands)

The LMS channel models can be grouped into three classes:

1) *empirical models* are obtained from experimental data. They are very close to reality for the environment type in which the measurements have been done but are difficult to generalize to other environment types;

2) *statistical models* are based on the use of canonical statistical distributions. Like empirical models, statistical models are applied to environment classes (rural, suburban, urban, etc.). The subsequent classification problem is not straightforward, since an environment classified as urban in some countries may look a little more like small town elsewhere. Therefore statistical models are also difficult to generalize; and

3) *physical models* rely on a deterministic modelling of the propagation phenomena (reflection, diffraction, refraction), but also of the considered environment. These models have been efficiently used for planning purposes in terrestrial radio-communication or broadcast networks.

In the following the main models belonging to the two first categories are presented, the third being mainly used for terrestrial networks.

4.2.2.1 Empirical Models

The ERS (Empirical Road Side) model was elaborated based on numerous experiments performed since 1983 by J.Goldhirsh and W.J.Vogel [i.2]. This model mathematically describes attenuation from tree-lined roads, multipath in mountainous and tree lined roads. It also provides a frequency scaling relationship for static tree attenuation from UHF to K Band. This model is recommended by ITU for rural environment (see ITU-R Recommendation P.681-7 [12]). Examples of ERS model link margin calculations for S-band (2,2 GHz) are reported in table 4.5.

Table 4.5: Additional link margin versus elevation angle for rural landscapes according to the ERS model, at f_c = 2,2 GHz

Elevation angle	Margin for 80 % availability [dB]	Margin for 95 % availability [dB]	Margin for 99 % availability [dB]
41,5°	5	12	20
39°	6	13,5	21,5
30°	9,5	17,5	26,5
25°	12,5	20	29

It can be seen that by using an ERS model, calculation based on instantaneous signal availability results in a link margin that, for availability greater or equal than 95 %, is incompatible with practical satellite capabilities. As shown in Annex A, by using the DVB-SH waveform features (powerful FEC, long time interleaving at physical layer or inter-burst upper layer FEC) one can achieve high availability with smaller margins. This is possible as short term link unavailability is recovered through physical FEC and LL-FEC. Thus the DVB-SH link margins shall be based on a more elaborated satellite channel model able to represent the first and the second order statistics of the fading/shadowing process. Therefore the ERS model is not recommended for DVB-SH network planning.

4.2.2.2 Statistical models (for narrow-band signals)

The statistical channel models illustrated in the following clause are capable of generating time series of the LMS signal amplitude fading/shadowing process through a propagation model that can be readily implemented in a computer simulation or in a laboratory test environment. The statistical LMS propagation model is characterized by parameters that have been derived through synthetic time series matching with experimental data obtained in different LMS propagation environments. The experimental data matching corresponds to first and second order statistics of the signal attenuation, thus is in principle suitable to our DVB-SH performance assessment scope.

4.2.2.2.1 Single-state models

Statistical models for satellite channels make the assumption that the received signal is composed of two parts: a coherent part associated with the LOS path and a diffuse part arising from multipath components. Table 4.6 summarizes the best known statistical satellite mobile channels in increasing order of complexity.

Table 4.6: Statistical satellite mobile channel models

Model	Coherent part	Correlation	Diffuse part
Rice	Constant	Zero	Rayleigh
Loo [i.4]	Log-normal	Variable	Rayleigh
Corazza [i.5]	Log-normal	Unity	Log-normal, Rayleigh
Hwang [i.6]	Log-normal	Zero	Log-normal, Rayleigh

The Rice model assumes that the LOS component is only affected by Rayleigh distributed multipath and is characterized by the carrier-to-multipath ratio (C/M or Rice factor K). The Loo model assumes that the LOS is affected by lognormal shadowing (typical for tree shadowing) which makes the instantaneous C/M time variant. The Corazza model assumes that both the LOS and the multipath components are affected by the same lognormal shadowing. Thus in the Corazza model the C/M is constant although both the LOS and the average multipath power components are lognormal distributed. The Hwang model also considers lognormal process affecting both the LOS and the multipath component but with total decorrelation between them. The Hwang model has been shown to include the Rice, Loo and Corazza models as special cases.

As DVB-SH is supposed to operate in different mobile environments, where channel conditions are time variant, a single-state channel model is considered inadequate.

4.2.2.2.2 Two-state (Lutz) model

The Lutz model [i.3] is a statistical model which represents the channel by a two states Markov chain. The "good" state occurs when a dominant line-of-sight component is received. In that state, the channel can be considered as a Ricean channel. The "bad" state occurs when no LOS component is received. In this state, the channel can be modelled as a Log-normal Rayleigh channel without coherent component. The parameters associated with each state and the transition probabilities from one state to another are empirically derived. This approach allows the generation of time series representing different environments. However, two-state channels were considered not enough to represent the variety of propagation conditions experienced in the different environments.

4.2.2.2.3 Three-state (Fontan) Model

A further refinement of the Lutz model is represented by the Fontan model [16] which includes a three-state Markov chain:

- state 1: line-of-sight (LOS);
- state 2: moderate shadowing; and
- state 3: deep shadowing.

The statistical model adopted for all three states is a Rice/lognormal distribution with parameters that have been experimentally derived from propagation campaigns performed in different European locations. The model parameters are reported in [16] for different environments, elevation angles and locations. Being recognized as the most accurate statistical LMS channel model available today, encompassing the widest set of environments and elevation angles, this model has been adopted for the DVB-SH performance evaluation detailed in clause A.7.

Figure 4.4 shows a number of time series generated by using the Fontan's model for various environments and satellite elevations angles at S-band. For open area the received signal power fluctuations are rather limited except for the 20 degrees elevation.

The situation is radically different for S-band intermediate tree-shadowed (ITS) environment. In this case the multi-state channel nature is evident in particular up to 40 degrees elevation. When no LOS signal is present large shadowing fluctuations on top of multipath fading are visible. Suburban areas shows similar two-state nature with longer LOS condition compared to the ITS case. Urban environment keeps a remarked on-off channel nature even at high satellite elevation angles due to the presence of close buildings.

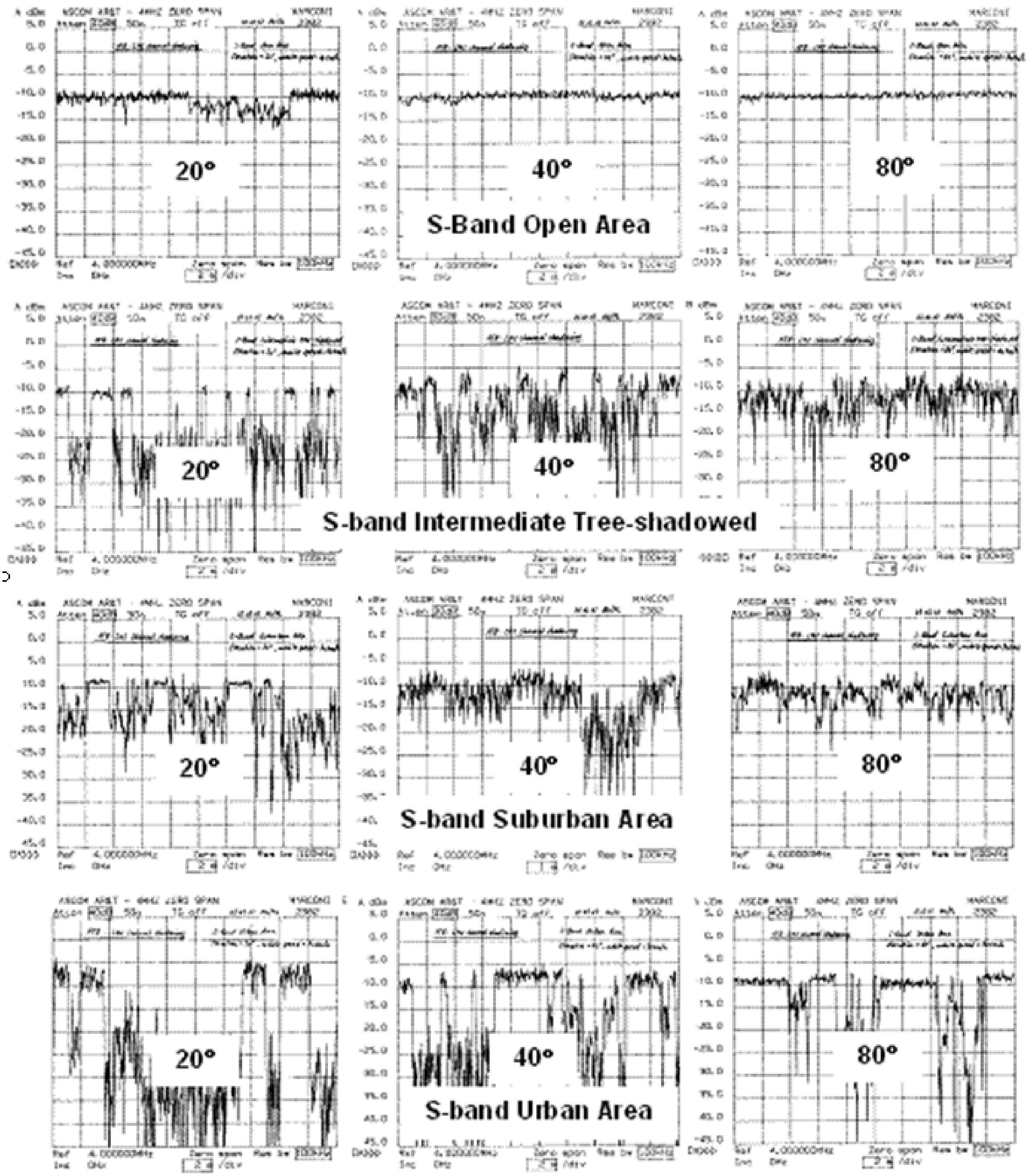


Figure 4.4: Examples of Fontan's channel model time series

4.2.2.2.4 Quasi-static Channel Model

The use of DVB-SH for direct satellite operation reception by hand-held terminals operating in quasi static conditions with user cooperation calls for an extra channel model. This channel model corresponds to the line of sight state of the Fontan model described before. Being the terminal quasi static, the resulting channel can be described by a very slow Ricean fading process (see clause A.7). Being the fading correlation time larger than the physical layer or upper layer decorrelation capabilities, link availability can be computed according to first order fading statistics as described in clause 11.

4.2.3 Terrestrial Channels

Path loss:

Several models have been developed to describe path loss for a single terrestrial transmitter in both urban and suburban environments in the past [i.1]. Two of them have been used in contexts close to the DVB-SH one:

a) the COST 231-Hata classically used for terrestrial broadcast network planning;

b) the Xia Bertoni model used for the assessment of UMTS performances (see TR 101 112 [i.24] and ITU-R Recommendation M.1225 [11].

Shadowing effects:

For a single terrestrial transmitter, shadowing is generally considered to be lognormal distributed. This lognormal law is characterized by its median value (corresponding to the path loss) and by its standard deviation or location variability, σ_L. Location variability is generally higher in suburban areas than in urban areas. For outdoor users in urban and suburban environments, the value of σ_L recommended by ITU-R Recommendation M.1225 [11] is 10 dB at 2,2 GHz. However, in urban environment, this value can be slightly pessimistic: models described in [i.1] give a value of about 8,1 dB in urban environment and of 9,5 dB in suburban environment. Nevertheless, other references that are used by Mobile Telephony Operator also propose 8 dB for suburban areas (see clause 11). On the other hand, it is acknowledged that the value of 5,5 dB for σ_L is commonly used by broadcasters [13], [19] and [i.21]. So the proposed values for network planning in clause 11 are σ_L = 5,5 dB and σ_L =8 dB, for the "broadcasters' approach" and the "cellular approach" respectively. TM-SSP plans to publish refreshed recommendations on this parameter as soon as experimental results are made available.

Physical layer performance assessment:

Single cell environment: for physical layer performance assessment in terrestrial environment DVB-SH in common to DVB-H has selected the TU6 model [18]. The Typical Urban 6-paths model (TU6) is proven to be representative of the typical mobile reception with Doppler frequency above 10 Hz. In TU6 the channel model consists of tapped delay line (TDL) made of 5 delayed plus one non delayed path each affected by Rayleigh distributed fading with different average power. The Rayleigh fading bandwidth is dependent on the mobile speed.

In addition to TU6 we also considered the so called Pedestrian Outdoor and Pedestrian Indoor (PIPO) channels described in clause A.13 which are better suited to represent pedestrian type of terrestrial channels.

Although very popular the TU6 channel shows limitations when used to represent SFN type of networks, indoor operations and low speed mobile conditions. Low fading effects (also called large scale fading) modelled for example with lognormal fading has not been included in the simulation setups for the evaluation of the terrestrial reception. These effects are covered by the margin added by the network planning. For the satellite reception the slow and very slow fading effects are covered by the LMS models. The evaluation of the fading characteristics for terrestrial networks typical for DVB-SH is subject of several on-going projects and field trials. In the future these models may be used to provide complementary performance data.

It should be remarked that Rayleigh and Rice channel models definitions are also used in the context of DVB-T/H but they refer to two different contexts as explained detailed in clause A.7.

Multi-cell environments: the DVB-SH terrestrial re-transmitter network is characterized by Single Frequency Network (SFN) architecture. This characteristic must be taken into account when assessing the system performances in realistic propagation conditions when coverage is fully or partly ensured by gap-fillers as it is the case in urban and suburban environments. Contrary to the satellite-only case, the terrestrial propagation channel must be considered as frequency selective. Therefore, the assessment of waveform performances must be performed using a TDL model parameterized with a Power Delay Profile giving the stationary impulse response of the propagation channel. This PDP for such SFN configurations can be obtained through two types of methods:

- for a given site for which a 3D representation is available, a site-specific deterministic tool can be used (ray launching or ray tracing tools). For a generic urban/suburban context, a macro-cellular geometric configuration is set for a given cell radius. Then, given a position of the user terminal, the channel SFN Power Delay Profile (PDP) is obtained by combining elementary PDP coming from each gap-filler taking into account the path losses from the gap-fillers to the user terminal. This second approach allows the cell radius to be tuned and requires an elementary PDP repeated for each gap-filler. Two standardized PDP corresponding to a macro-cellular single emitter case have been used in contexts close to the DVB-SH one:
    - ITU Vehicular A recommended by ITU-Recommendation M.1225 [11]. This PDP has been used for UMTS performance assessment;
    - GSM-TU6 as defined in [18]. This PDP has also been used for DVB-H performance assessment;
- for path loss, the Xia-Bertoni model gives average results between dense urban and suburban cases of the COST 231-Hata model. Therefore this model is preferably used for such an analytical approach that requires a unique cell radius for the whole SFN configuration.

A SFN channel model for DVB-SH physical layer performance assessment is illustrated in clause A.13.1.

4.2.4 Satellite specific issues

Another important element in the DVB-SH transmission chain to the user is the satellite payload that together with the land mobile satellite channel greatly contributes to the end-to-end system performance.

4.2.4.1 Satellite Payload architectures

Satellite payloads for mobile broadcasting systems comprise two major elements: a feeder-link receive section and a high power S-Band transmit section. The outstanding EIRP requirements obviously make the last element the most demanding and worth of a dedicated discussion.

The payload architecture of a high power transmit section largely depends on the required service performance (coverage, EIRP, number of beams, etc.) and on the satellite constellation typology (GEO/HEO). Systems aiming at wide coverage regions are based on medium/large on-board antenna apertures which are composed of light-weight deployable rigid reflectors (these technologies are typically available up to about 5 meters of reflector diameter at S-band). On the contrary, regional or multi-regional coverage areas as seen from a GEO/HEO are of very limited angular extension and called for a large deployable reflector of typically 12 meters projected aperture diameter, at S-band. EIRP requirements and the difficulty of embarking large deployable antennas lead in most cases to a demanding RF power requirement that generally exceeds the generation capabilities of single amplifiers available for space applications and "power combining" must be adopted. The approach consists in adding up a plurality of amplified replicas of the desired signal individually amplified by highly-efficient/high-power devices such as Travelling Wave Tubes Amplifiers (TWTAs).

The problem dimensionality is further increased if power re-configurability is required to match varying and un-predictable market demand of transmitted capacity (i.e. EIRP) among the different beams, or to adjust in-orbit the beam coverage with the use of Beam Forming Network (BFN) or Ground Based Beam Forming (GBBF) antennas.

To meet these objectives, the distributed amplification concept plays a fundamental role. The general concept foresees several power amplifiers working together not only to achieve the total amount of power otherwise non-achievable with a single source, but also to constitute a single power pool to which each signal can get its own amount in a variable and reconfigurable manner. This leads to the question of whether the satellite can or cannot be used in non-linear mode as further discussed in the next clause. More details about typical satellite payload architectures are provided in clause A.1.

4.2.4.2 Nonlinear Payload Distortion Effects

For assessing payload distortion impacts on DVB-SH waveforms, the optimal operating point (OBO) of the TWTA must be evaluated. Details about the satellite nonlinearity impact and the TWTA operating point optimization methodology is provided in clause A.1.2.

For this, two main satellite payload configurations should be distinguished (see clause A.1.1):

1) single multiplex amplification per TWTA (network and polarization combining);

2) multiple multiplexes amplification per TWTA (antenna spatial combing and multibeam satellite).

As shown in clause A.1.3 in the first case, the satellite TWTA can be driven close to saturation if the TDM waveform is selected i.e. SH-B. In this case, SH-B may have a power advantage of 1 dB to 1,5 dB with respect to SH-A.

In the second case, since each satellite TWTA is used to amplify several multiplexes, these amplifiers have to be backed-off away from saturation in order to avoid the creation of intermodulation products. Therefore the use of SH-B is not so advantageous in terms of link-budget for this class of payloads.

4.2.4.3 Phase Noise Aspects

In the DVB-SH direct path (i.e. satellite-to-user) system phase noise is dominated by the contributions from the satellite local oscillators used for on-board frequency conversion and the user terminal RF front-end Local Oscillators. This is because of the usually high frequency band used for the feeder link, the need of frequency conversion agility and the space environment. Table 4.7 reports a typical/worst-case Ka- to S-band satellite transponder DVB-SH phase noise mask which has been used to derive the impact of phase noise on DVB-SH waveform.

Table 4.7: S-band DVB-SH phase noise mask (Satellite)

	10 Hz	100 Hz	1 000 Hz	10 kHz	100 kHz	1 MHz	10 MHz
Phase noise (dBc/Hz)	-29	-59	-69	-74	-83	-95	-101

There exist simplified approaches to compute the impact of the phase noise residual phase error on the digital demodulator i.e. [i.7] for OFDM or [i.8] for TDM. In practice the phase noise that cannot be tracked is dominated by the high frequency components of the phase noise i.e. the components from the equivalent phase estimator phase noise bandwidth B_N to the baud rate.

In SH-B two pilots groups having length L=80 symbols and separated by L_{TOT}/2=1 088 symbols being L_{TOT} the PL slot length. Consequently the equivalent feedforward phase estimator can be computed as $B_N = \frac{R_s}{L_{TOT}}$ kHz which corresponds to $B_N =$ 1,8 KHz for the 4 Mbaud case. The DVB-SH phase noise mask of table 4.7 produces a typical amount of 4,5° rms untrackable phase noise due to the almost flat phase noise PSD between 1 KHz and 100 KHz. This untrackable error should be not causing any appreciable impact on coded QPSK performance.

The impact of phase noise error depends also on its dynamic compared to the physical layer FEC block size. Typically for the same standard deviation "fast" phase noise has much less impact compared to "slow" phase noise being the speed related to the FEC block duration. For this reason when higher than QPSK modulations are used, it is recommended to simulate the phase noise impact on the end-to-end system rather than just basing impact estimation on the simplified approach described above.

4.2.4.4 Satellite induced Doppler Shift

Satellite induced Doppler shift manifests itself as a change in signal frequency which cycles through a range of frequencies and which reaches a given ± maximum value. This cycling results in a smooth and readily predictable frequency shift as a function of time. This effect should not be confused with the Doppler spread induced by multipath and the receiver speed, *except in SFN mode*. In SFN mode, the satellite-induced Doppler must be corrected (locally) by the CGC, otherwise it manifests itself as a Doppler spread in the hybrid reception coverage. Clause 7.5 gives recommendations for this correction.

In considering Doppler shift here it is tacitly implied that any Doppler shift in the uplink can be pre-compensated for at the transmitting station. Unfortunately such pre-compensation approaches are not feasible for the downlink where the Doppler shift varies as a function of geographical location within the coverage footprint.

For GEO satellites the Doppler shift is very small and presents no problems for the receiver. For the case of a GEO system with an inclined orbit (up to around 5 degrees) the Doppler Shift may be of the order of ±250 Hz.

For HEO and MEO satellite constellations the downlink Doppler shift at S-band can be of the order of tens of kHz and requires that the receive can handle any initial frequency shift at acquisition and can track the frequency shift during the orbit pass. Figure 4.5 shows typical ground tracks of HEO systems targeted for European service based upon the parameters given in table 4.8.

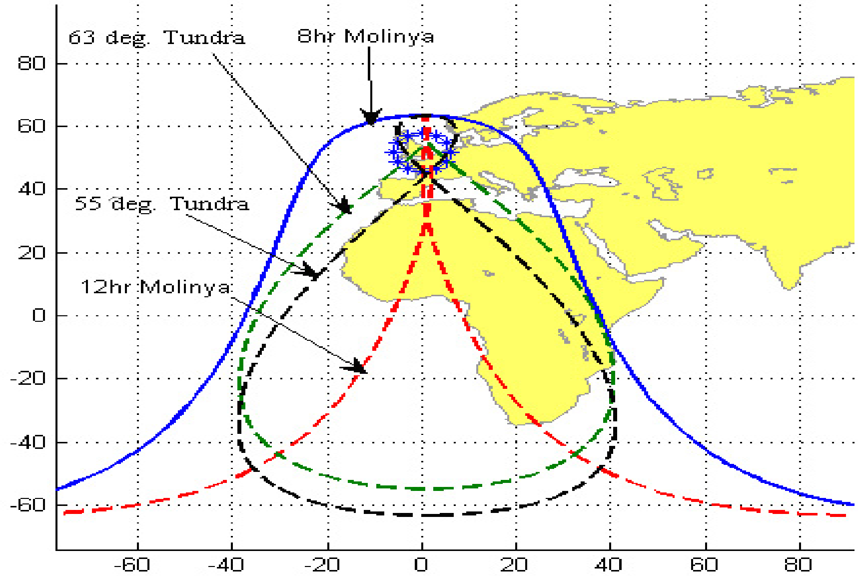


Figure 4.5: Ground track of typical HEO orbits

Table 4.8: Representative parameters of typical HEO Orbits

Orbit	Orbit Period (hrs)	Semi-major Axis (km)	Apogee Radius (km)	Apogee height (km)	Perigee Radius (km)	Perigee height (km)	eccentricity	Inclination (°)	Argument of Perigee (°)	Range of Elevation Angles (°)	Minimum number of satellites
8 hr Molnya	8	2 027	33 163	26 785	7 378	1 000	0,636	63,435	270	55 to 90	8
12 hr Molnya	12	26 561	45 804	39 426	7 307	929	0,7249	63,4	270	60 to 90	4
63° Tundra	12	42 164	53 480	47 102	30 847	24 469	0,2684	63,4	270	at least 60	3
55° Tundra	24	42 164	55 656	49 278	28 672	22 294	0,32	55	270	65 to 90	3

The maximum value of S-band downlink Doppler shift and the maximum rate of change of this Doppler shift with time are given in table 4.9.

Table 4.9: Typical maximum Doppler shift parameters for downlink at S-Band

Orbit	Min elevation angle (°)	Maximum Doppler shift (KHz)	Maximum rate of change of Doppler shift (Hz/s)
8 hr Molnya	50	17	2,0
	60	15	2,0
12 hr Molnya	50	22	1,8
	60	21	1,8
63° Tundra (24 hr)	50	3,9	0,35
	60	3,9	0,3
55° Tundra (24 hr)	50	4,8	0,3
	60	4,3	0,3
MEO	10	11,0	4,5
GEO, stable orbit	Any	Very small	Very small
GEO, inclined orbit (5°)	Any	0,25	0,02

4.2.5 DVB-SH Impairments Countermeasures

4.2.5.1 LMS Mobile Channel Impact Mitigation

To mitigate the impact of LMS propagation impairments, two specific techniques are recommended: exploit diversity and optimize demodulator robustness.

4.2.5.1.1 Exploitation of Time and Space Diversity

The DVB-SH standard provides inherent methods for both time and space diversity:

Method 1: Time diversity

It is supported in DVB-SH by the highly scalable Physical time interleaver. In addition, a solution based on LL-FEC, with possible splitting of the interleaving function between the physical and the link layers is further detailed in clauses 6 and 7.

Method 2: Space diversity (simultaneous reception of SC and CGC)

For the SH-A (SFN), the space diversity is intrinsic in the demodulation process as the isofrequency satellite and terrestrial OFDM signals will be combined at the user terminal antenna.

For the SH-B and SH-A (MFN) cases, the user terminal needs to separately demodulate the satellite and terrestrial signals and may implement different combining options depending on its capability (clause 7.6).

In addition, the DVB-SH standard supports space and time diversity from "extrinsic" methods:

Method 3: Multiple satellite space diversity

The same combining strategy for MFN of Method 2 applies to this case too. For SFN combining there are restrictions.

Method 4: Multiple receive antenna space diversity

It has particular benefits in terrestrial conditions, mainly indoor (quasi-static). For the satellite direct reception antenna diversity may bring valuable performance improvement for (very) slow mobility conditions if the separation is sufficiently large (e.g. 1 meter or more) which is often difficult to support in practice, except for vehicular applications. Polarization diversity (e.g. horizontal/vertical linear or left-handed/right-handed circular) may be beneficial as soon as the reflected components still carry sufficient power.

4.2.5.1.2 Robust Demodulator Operations

In a mobile fading channel, it is important to have very robust signal re-acquisition after signal loss as well as a fast and reliable signal state tracking. Although this aspect is also common to terrestrial standards such as DVB-H some specific aspects must be addressed by DVB-SH:

- satellite operations with limited link margins compared to terrestrial systems;

- demodulator operation at lower SNR than terrestrial systems.

Guidelines for the demodulator implementations and performance for this aspect are given in clause 10.

4.2.5.2 Nonlinear Channel Impairments Countermeasures

In case of multiple-carriers amplification, the use of linearized TWTA can bring appreciable advantages. Similar performance improvement can be obtained by on-ground pre-distortion to compensate for the satellite TWTA nonlinear characteristic.

In case of single-carrier amplification, the use of linearized TWTAs provides limited performance improvement in particular for QPSK with low code rate as the TWTA will be operated in the vicinity of its saturation (where the linearizer has little effect). However, linearized TWTA may provide appreciable benefits for 16APSK in particular if the operating point corresponding to a given modulated signal drive is not negatively impacted by the linearizer presence.

For single carrier TWTA operation, 16-APSK being multi-ring can benefit from simple static pre-distortion techniques which are able to mitigate the constellation warping effect [i.16]. Dynamic pre-distortion techniques are particularly suitable to reduce the clustering effects of higher order modulations such as 16APSK. Their benefit for QPSK and 8PSK is instead limited.

4.2.6 Receiver characteristics

This clause discusses, in general terms, the architectures of DVB-SH receivers. The detailed Reference Designs for the terminal classes and their recommended performance are given in clause 10.

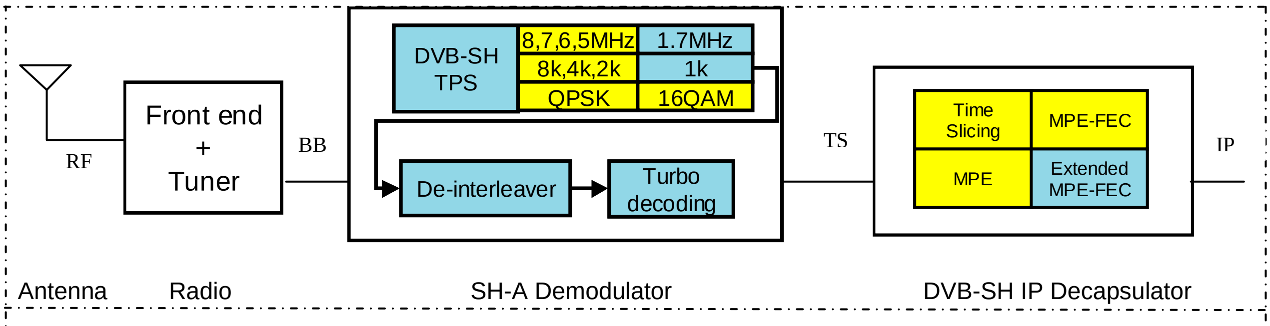
DVB-SH receivers must be able to:

- receive seamlessly a signal when moving between satellite(s) and CGC(s) coverage;
- implement battery saving schemes when necessary;
- exploit diversity mode with several receiver branches in parallel when appropriate;
- co-exist with other active radio functions (2G, 3G, IEEE 802.11 [26], Bluetooth®, etc.) embedded in a terminal; and
- be relatively immune to high-level signals from various cellular network base stations (2G, 3G) or/and ISM systems.

Two different receiver architectures can be distinguished according to the DVB-SH waveform options, respectively the OFDM/OFDM (SH-A see figure 4.6) and the TDM/OFDM (SH-B see figure 4.7) system architectures.

The SH-B architecture encompasses the SH-A architecture in the sense that SH-B receivers can be used in an SH-A configuration. The converse is obviously not true.

Receivers designed for the SH-A architecture are in general intended to be used with SFN between satellite and CGC. However, it is recommended that they should be also compatible with an MFN configuration (transmissions of the OFDM satellite signal and its OFDM CGC counterpart on two different sub-bands).



NOTE: The dual tuner may have a common front-end and downconversion.

Figure 4.6: SH-A Receiver architecture

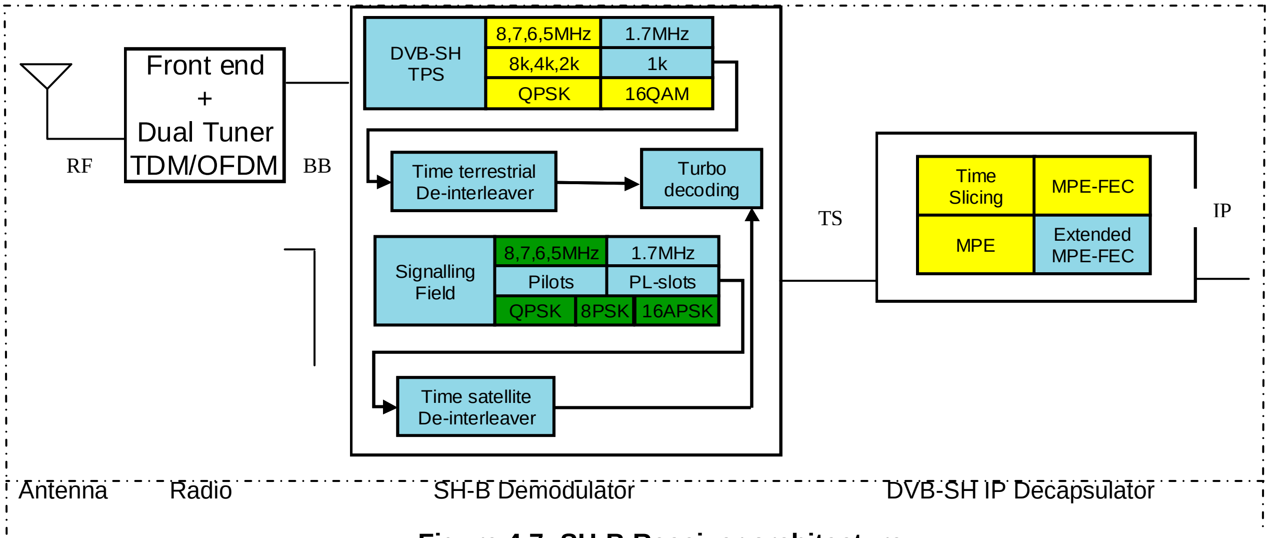


Figure 4.7: SH-B Receiver architecture

4.2.6.1 Vehicular reception constraints

Vehicular receivers must work at high moving speeds and stay longer in a satellite-only reception mode. However, vehicular receivers can exploit the following features not generally possible for receivers used in a handset:

- the terminal has adequate power supply able to support more complex receiver processing;
- the terminal allows better antenna diversity (order 2 or more) implementation (form factor and antenna spacing);
- one antenna can be optimized for satellite reception (directivity and matching polarization);
- Low Noise Amplifiers (LNAs) can be integrated with the antenna(s) to reduce sensitivity loss (the underlying assumption is that the vehicular receiver does not coexist with embedded Telecom modems in the same electronic equipment and therefore the RF filtering protection for the LNA is removed or relaxed, therefore improving noise factor);
- larger memory can be embedded in the receiver so that longer Physical Layer interleaving can be supported;
- the higher speed of the terminal allows a better exploitation of time diversity (either at Physical Layer or at Upper Layers), at equal memory resource.

4.2.6.2 Handheld reception constraints (pedestrian)

In most common cases the channel conditions are that produced by a pedestrian user (< 3 kmph). Due to the relatively low speed, continuity of service is in general achieved by increasing the link margin, rather than by increasing the time interleaving depth. When in satellite-only reception mode, some cooperation may be required from the user, i.e. to maintain good LOS with the satellite.

Some challenges are associated to the specificities of handheld terminals. These include:

- antenna diversity (order more than 2 is very challenging);
- small battery requires an efficient power saving management;
- antenna gain is in general low (can be less than -3 dBi);
- antenna polarization is most often linear and not optimized to satellite reception;
- embedding telecom modems like GSM or 3G inside terminal without reducing the satellite receiver sensitivity;
- RF filtering, antenna design rules and compactness constraints have an impact on the achievable receiver sensitivity and immunity to high level blockers coming from the terminal;

- memory limitation may, in some architectures, not allow the support of a large Physical Layer interleaver.

4.2.7 Terminal High-level architecture

In theory, all types of terminal can embed large memory and all options defined above. Furthermore, no limitation is imposed on the number of embedded multi-band or/and multimode 2G, 3G communication modem(s) or other wireless features like Bluetooth® or WiFi. Also GPS could be integrated in a terminal. In practice, compactness, supply source and price considerations split terminal in categories. Categories chosen in DVB-H Implementation Guidelines [i.21] and in EICTA/TAC/MBRAI-02-16 [14] are kept (see clause 10). Figure 4.8 shows the terminal high-level architecture for all classes of terminals. Terminal sub-systems will be designed according to their terminal categories and manufacturer marketing requirements.

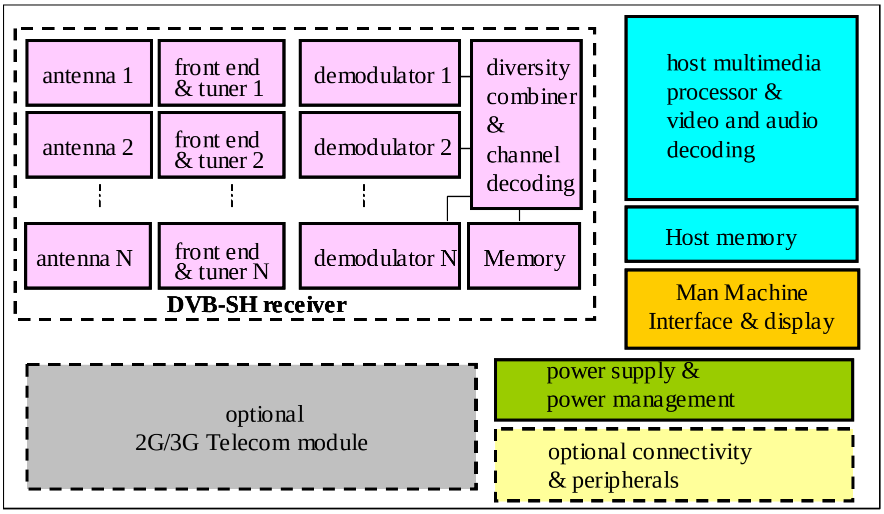


Figure 4.8: Terminal high-level architecture

4.2.8 Compatibility with Existing DVB-H System Features

4.2.8.1 Battery-powered receivers

This clause deals with receivers embedded in handheld "convergence" terminals and pocketable digital TV receiver devices (see clause 10 for definitions of receiver categories). For these devices, battery saving is essential.

4.2.8.1.1 Time slicing

The DVB-H group has defined time slicing as a key mechanism for battery saving purposes. DVB-SH receivers also use this mechanism for the same purposes. Burst size and off time, maximum burst duration, power saving calculations are based on same assumptions than DVB-H.

4.2.8.1.2 Low-power Microelectronics

State-of-the-art technologies like the newest RF BiCMOS or SiGe for tuner and CMOS 65 nm process for channel decoder and sometimes full CMOS process for one chip solutions have significantly reduced the contribution of the front-end in the overall terminal power consumption, compared to what it used to be some years ago.

4.2.8.1.3 Antenna Diversity (for S-band)

Operation at S-band enables the use of antenna diversity, even in small handheld terminals.

Antenna diversity does increase power requirements since more than one receiving chains operate in parallel. However, this power consumption increase can be kept acceptable compared to the gain in QoS improvement with the use of new low-power technology and time-slicing mechanism as stated above. In addition, a signal quality indicator could be used to turn off or reduced diversity processing when this is not required.

4.2.8.2 Service Switching time (Zapping time)

Zapping time is an important subjective quality criterion for the user in a multi-program offer. Due to the combined effect of time-slicing for battery-life conservation and time-diversity techniques to overcome long fadings, possible degradation of zapping time may be incurred if appropriate counter-measure techniques are not applied. In DVB-SH it should be noted first that zapping time may be different depending on the coverage (CGC or Satellite) and the particular length and structure of the time interleaver chosen for the propagation channel in each coverage.

In the CGC coverage, DVB-SH zapping time, normally, does not exceed 2 s on average.

In the satellite coverage, one can distinguish between good reception conditions (e.g. about > 5 dB above threshold) and critical reception conditions. A "smart" terminal can optimize zapping time according to its measured reception margin. In good reception conditions, such terminals can immediately play out any source data without waiting for maximum FEC protection to be fulfilled. This strategy is made possible by special sending arrangements of FEC parity and source data allowed by DVB-SH. Techniques for transitioning between "immediate" play out strategy (in good reception) and maximum error protection (in mobility) may be needed to be implemented in the terminal. Clauses 6 and 7 describe the use of these techniques, for the link and the physical parameters respectively, and also discuss their possible side implications.

4.2.8.3 Support for Variable Bit Rate (VBR) and Statistical Multiplexing (Statmux)

The use of VBR and statistical multiplexing (Statmux) of the different services transmitted in a single TS allow either bandwidth saving or better video quality. The actual gain provided by VBR depends on the system configuration, on the types of contents and on the number of services multiplexed statistically on the TS.

The main element impacting VBR implementation with DVB-SH is the combination of the Time interleaver together with Time Slicing:

- with short interleaving (i.e. slightly exceeding the length of the Burst), VBR is supported with time slicing thanks to the "delta-T" mechanism defined for the same purpose in the DVB-H standard;
- when long interleaving is used, Burst content (and hence "delta-T" information) is spread over several Bursts. This adds complexity in handling of Time Slicing, but does not preclude the use of VBR nor reduce its bandwidth saving gain.

VBR implementation is described in more detail in clause 6.3.4.

4.2.8.4 Multiple QoS support

4.2.8.4.1 Introduction

This clause deals with the support of different quality of service in a DVB-SH environment. The overall quality of service results from the combination of different layers. Streaming protection is mainly achieved by physical and link layers, whereas physical and application layers achieve file delivery protection.

Depending on the chosen systems configurations, different options are possible at different layers as presented in table 4.10.

Table 4.10: Configuration options for the different classes

Terminal option		Parameters	Relevance
class 1	Physical layer	Rate	++
	Link layer	Rate; depth	++++
	Application layer	Rate, repetition	++
class 2	Physical layer	Code rate; interleaver	++++
	Link layer	Code rate; interleaver	+
	Application layer	Rate, repetition	++

4.2.8.4.2 At physical layer

DVB-SH does not provide variable quality of service at physical layer. The following parameters must be configured according to the expected performance of the most demanding service and as a function of the worst channel model scenario:

- code rate;
- interleaver configuration, in particular interleaving depth.

4.2.8.4.3 At link layer

DVB-SH does provide at link layer a per-service protection based on MPE sections. It is possible to configure a streaming service with a long interleaving while a "push" service may be configured with no protection at the link layer since the protection is provided at the application level.

4.2.8.4.4 At application layer

DVB-SH does not provide any specificity for the application layer since the interface with the "application world" is IP. However, the DVB-SH application layer is fully compliant with all CBMS specifications and, in particular, with the file delivery specified in document [Content Delivery Protocols]. It is also expected that other techniques may be applied to the streaming services such as scalable video encoding. Scalable video encoding must be coordinated with the link layer protection since the former is based on a differential protection applied to two parallel streams making the video stream.

4.2.8.4.5 Recommendations

The procedure displayed in figure 4.9 should be followed when configuring the parameters of the FEC protection.

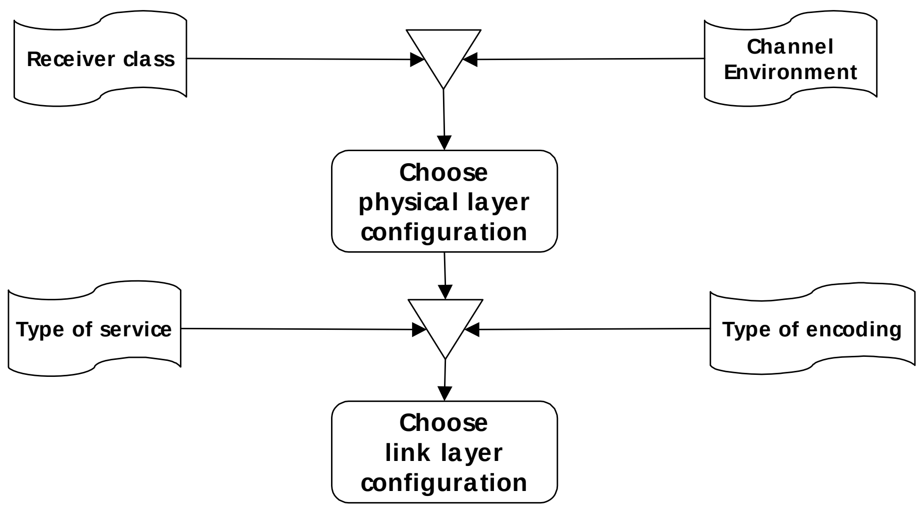


Figure 4.9: Procedure for choosing FEC protection parameters

Several configurations are possible while offering the same useful throughput but not always the same performance:

- protection at physical layer only compared to joint physical and link layer protections;
- different splits of the joint protection.

The network operator, depending on the deployed network and receivers, will choose the physical layer configuration. The service operators will then be able to choose if they want to activate the link layer and application layer protections, trading performance versus available bandwidth.

Differentiation between services at application and link layer may usually happen in the following circumstances:

- when different kinds of services are broadcasted in parallel: this is for instance the case between the streaming application (which will highly benefit from the link layer protection) and the file download application (which may benefit better from the content delivery protection and then will less use the link protection);

- when, for a specific service, different levels of protection are required: this is the case of scalable video encoding which requires two levels (or more) of protection to protect better the main stream than the others.

4.2.8.5 Service announcements in hybrid networks

This clause deals with announcement of services within the DVB-SH context.

4.2.8.5.1 Introduction

Signalling within the DVB-SH system follows the layered approach described in figure 4.10.

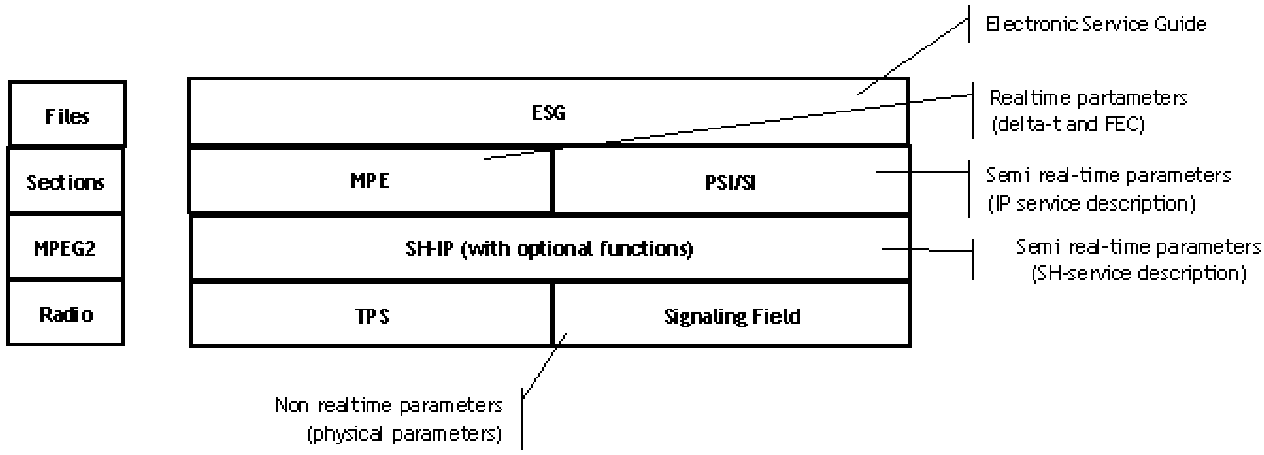


Figure 4.10: Layered signalling approach

According to this layered approach, a terminal learns all information about the network and services at first switch-on time in the following steps:

- radio configuration: the terminal can discover the physical layer parameters by scanning relevant frequencies and recovering either TPS (in OFDM waveform) or Signalling Field (in TDM waveform). These fields will deliver all information about the baseband configuration like FEC ratio, interleaver depth, etc.;

- TS configuration: the terminal is then able to decode the MPEG2 flow. If required it will discover the SH services structure by reading the SHIP packets;

- section configuration: the terminal will read the PSI/SI tables that will give all system related information to be able to derive from a target IP address the relevant PID: Given a target IP address, the IP Notification Table (INT) will deliver the relevant component tag of the target elementary stream;

- then using the PMT and PAT, the relevant PID will be found;

- programme configuration: the terminal is then able to read to the bootstrap ESG IP address to discover the list of ESG providers, choose one and then listen to the relevant IP address to discover its electronic service guide. The ESG has the format of an XML file describing what content can be watched when, typically <name_of_the_programme><IP_address>. Then, using PSI/SI the relevant PID is fetched by the terminal.

Compared to DVB-H, the only difference is the introduction of the SH service layer. The service announcement is performed, as in DVB-H, according to the Electronic Service Guide specification of the DVB-CBMS protocol suite. In general, there is nothing specific compared to the ESG used for the DVB-H, in particular for the SFN case that is very similar to the DVB-H case (no need for SH services). In the MFN case, the announcement of the local content and of its availability at particular locations will require the use of the "SH service" concept (see clause 6).

4.2.8.6 Handover issues

This clause describes in general terms the handover issues in DVB-SH. DVB-SH has been designed to take advantage of the mechanisms for handover that have been defined in the DVB-H framework (see TS 102 470-2 [21]). In particular, the following principles are maintained in DVB-SH:

- **use of physical layer signalling:** for instance, the TPS provides information such as cell_id to quicken the detection of the target network. A similar signalling mechanism has been defined for the DVB-SH TDM waveform;
- **use of spare time in time-slicing for scanning available frequencies:** this is also possible in DVB-SH, even with long physical layer interleaving;
- **PSI/SI signalling:** the same philosophy of signalling has been maintained, with some adaptations to cater for the hybrid architecture. The NIT describes the full network in a beam, including the target frequencies and waveforms. The INT describes the availability of the session on the target transport stream.

Nevertheless, DVB-SH introduces also some specificities that are worth keeping in mind:

- presence of the satellite "umbrella" cell that covers all the terrestrial cells;
- terrestrial cells can be much smaller than in DVB-H, especially in S-band (less than a Km in radius);
- terrestrial cells and satellite umbrella cell may use different time interleaver lengths for the same service (for SH-B and MFN-SH-A;
- terminals with diversity reception have more than one front-end and demodulator;
- coherency of PSI/SI signalling must be addressed carefully when code diversity combining (with SH-B only) is used.

A detailed analysis of handover in DVB-SH is given in reference [i.9].

4.2.9 Support of Other Frequency Bands

The DVB-SH applicability should not be limited to above mentioned bands. The standard should be scalable to other frequency bands and channelization schemes that could be allocated to satellite broadcasting services.

Possible other frequency bands to be supported by DVB-SH are the following:

1) Ku band: non planned FSS, planned FSS, planned BSS: 10,7 GHz to 12,75 GHz;

2) Ka band: non planned: 17,3 GHz to 20,2 GHz. BSS: 21,4 GHz to 22 GHz;

3) other bands: e.g. C-band.

A high-level survey about the standard applicability to the above listed frequency bands above 3 GHz led to the following preliminary conclusions:

- the SH-A configuration is definitely not suitable for A3GH applications as the OFDM waveform will not be able to cope with the higher Doppler spread of A3GH systems (in particular Ku and Ka-band) unless a drastic reduction in mobile user speed can be accepted;
- the satellite TDM component of the SH-B configuration can also be exploited for A3GH bands with no expected impact on performance. In fact the mobile channel countermeasures adopted for SH-B are also valid for A3GH operations. The possible Ku/Ka-band receiver different phase noise mask compared to B3GH applications must be compatible with SH-B TDM pilots structure;
- for the CGC other solutions must be used (different waveform compared to the SH-B) or different frequency bands (e.g. B3GH whereby SH-B OFDM waveform can be exploited).

5 System Configurations and possible Deployment

5.1 Definitions and concepts

System: combination of physical elements (satellite, CGC, terminals, etc.) implementing the DVB-SH specifications.

Radio configuration: defines the selected parameters of the DVB-SH waveform. Among available options we find: SH-A (OFDM/OFDM) and SH-B (OFDM/TDM), interleaving (physical, link layer), FEC ratios, modulation mode, hierarchical modulation.

Frequency configuration: spectrum allocation to the satellite and terrestrial component. For the terrestrial transmitters, it is useful to distinguish between hybrid and non-hybrid frequency sub-bands. A non-hybrid frequency sub-band is defined as one that carries only local content. One main option in the frequency configuration is the choice between SFN and non SFN. In a SFN configuration, the hybrid sub-bands carry only the Common content. In a non SFN configuration, hybrid sub-bands include also those that carry a mix of Common and Local content.

Topological configuration: used in conjunction with frequency configuration, it defines where the spectrum is available, on a satellite ("beam") or terrestrial ("cell") basis. For satellite, it gives the location and shape of the beams. For a terrestrial cell, it gives the centre (in latitude, longitude) and extension of the cell.

Local Content Insertion configuration: defines the way the local content is inserted by the CGC transmitters, in the context of a non-SFN network.

CGC Transmitter configuration: hardware options placed on the transmission site; it concerns the location (co-sited with a 3G base station, in a new specific site), the antenna (reused new) and the power.

Receiver configuration: hardware options; it concerns essentially the available memory for physical and link layer interleaving (using class 1 and class 2 categories) and the antenna options (diversity, gain, etc.).

System configuration: set of options retained for a DVB-SH system; this includes *radio, frequency, topological, content, receiver and transmitter configurations.*

System transition: evolution of a system when one of its configurations changes. Not all transitions are relevant and/or foreseen in the system deployment.

Compatibility: a receiver is *compatible* with "radio configuration X" when it can successfully decode the useful information using only a sub-set of the waveform elements allowed in this configuration. A lower reception margin or a more restricted coverage area is implied compared to that of a compliant receiver.

Compliancy: a receiver is *compliant* with "radio configuration X" when it can exploit ALL the waveform elements allowed in configuration X to maximize reception margin and/or coverage area.

5.2 Configurations options definitions

This clause discusses in more details on the options that are offered.

5.2.1 Radio configuration

The air interface has many available parameters as reflected by the flexibility of the SH delivery system descriptor. Only three of them are discussed in this clause to provide visibility about the main classes. The complete set of possible scenarios is more complex than what illustrated hereafter.

Choice of TDM (SH-B) or OFDM (SH-A) for the satellite signal: this choice is dictated mainly by spectral efficiency considerations (satellite, terrestrial, overall) and the satellite nonlinearity characteristics as discussed in clauses 4.1.3, 4.2.3.2 and C.1.

Choice of between physical layer or link layer techniques to combat long duration fading: comparison of these techniques is given in clauses 6 and 7. The choice is dictated by the performance required in a long duration fading environment as well as the performance for achieving fast zapping; however, the cost and required foot-print of the memory to implement a long interleaver at the physical layer must be considered. The combination of a short physical interleaver with a long link layer interleaver is typically preferable for handheld terminals (see clause 10) whereas a long physical interleaver can suit transportable receivers like those installed in cars. When different modulation parameters and frequencies are used between the satellite and the terrestrial signal, it is possible to resort to different interleavers (physical and link layer) for the satellite and terrestrial signals hence extending available options for the system.

Choice of Modulation and FEC: guidelines for this choice are given in clause 7.

5.2.2 Frequency configuration

It is important to plan how the frequency can be used and, more importantly, reused.

- firstly, the bandwidth of a frequency sub-band must be decided between the values allowed in the standard (1,7 MHz, 5 MHz, 6 MHz, 7 MHz and 8 MHz). This bandwidth is most of the time implied by the frequency band selection;
- secondly, the choice between SFN and non-SFN must be made, on a beam-by-beam basis. In SFN case, a minimum of 1 sub-band is needed per beam, whereas in the non-SFN case this minimum is 2 sub-bands. Since the sub-band that repeats the Common content in the non-SFN CGC may need to be at different frequencies in different places, this minimum may be higher than 2;
- thirdly, the remaining sub-bands can be allocated to local content transmission, on a per-cell basis.

5.2.3 Topological configuration

This consists in allocating the sub-bands to the topological elements, i.e. the satellite beam(s) and the terrestrial cell(s). Adjacent beams must not have common frequencies for interfering avoidance reasons. This is the same for adjacent cells. However it is possible to reuse these frequencies:

- a frequency used in a beam/cell can be reused in another beam/cell if this beam/cell is sufficiently separated from the first;
- terrestrial cells can reuse an adjacent beam frequency provided that these cells are sufficiently far away from this beam.

Actual frequency allocation to sub-bands may be quite complex due to possible interferences from the satellite adjacent spot beams. In particular, for reusing adjacent spot beam frequency, terrestrial reuse is sought in the centre of the spot (far from the borders) but this is not always possible.

5.2.4 Local content insertion configuration

Two different techiques can be used for inserting Local content:

Technique 1: Local content insertion by usage of the special hierarchical modulation:

- As shown in figure 5.1 hierarchical modulation can be used by splitting content into two TS at the input of the terrestrial transmitter: the first TS is input to the primary interface of the terrestrial modulator; this TS is exactly the same as the one going to the satellite transmitter. The Second TS is input to the secondary interface of the terrestrial modulator to carry local contents. There is no specific need for synchronizing the two TS at the service level since they are essentially independent. However, there is a need to synchronize the two HP and LP streams at the SH frame level so that SFN can also be ensured. This latter synchronization is ensured is described in clause 7.5.4.
- Such technique enables in principle to double the content at the expense of some interferences generated by the local content signal on the satellite primary signal. Such interferences can be assimilated to an extra additive noise and it is usually negligible compared to the terrestrial transmitter power.

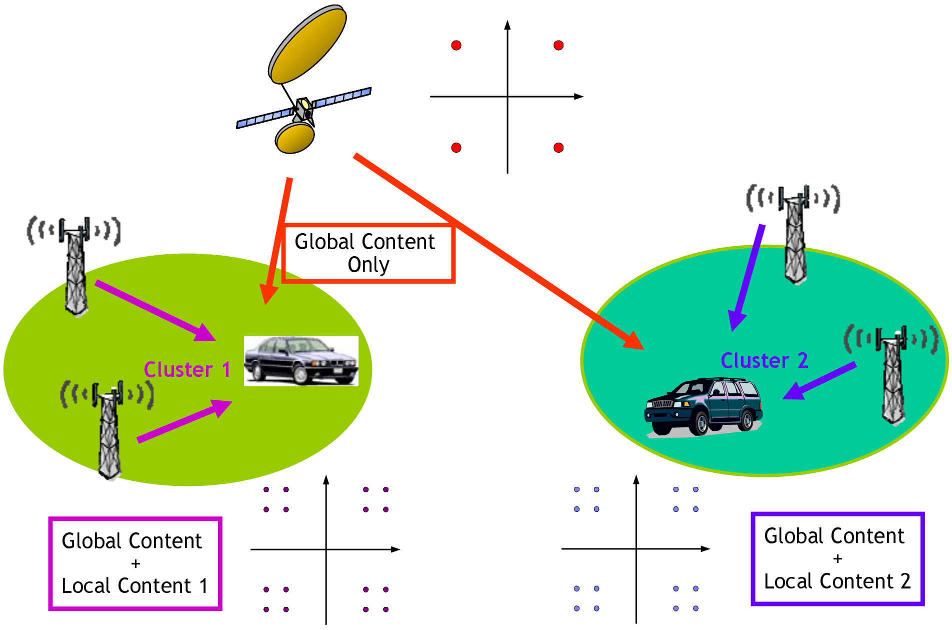


Figure 5.1: Hierarchical modulation for local content insertion

Figure 5.2 provides a block diagram of a network using terrestrial transmitter to broadcast local contents, on top of the global content delivered by the satellite, by the use of a hierarchically modulated signal. Code rates in the High Priority (HP) stream carrying Global content and the Low Priority (LP) stream carrying Local content may be different. But in any cases, the satellite signal carries only the HP stream. Accordingly, as it can be seen in figure 5.2, HP-hierarchical content is available in any part of the service area while the LP-hierarchical content should be available only in areas covered by terrestrial transmitters where some additional content is transmitted. Whatever the location of reception, the information carried by the TPS (Transmission Parameter Signalling : physical layer signalling) signals a hierarchical transmission even if the LP stream is missing in the received signal, under the satellite coverage and possibly terrestrial coverage where there is no additional content.

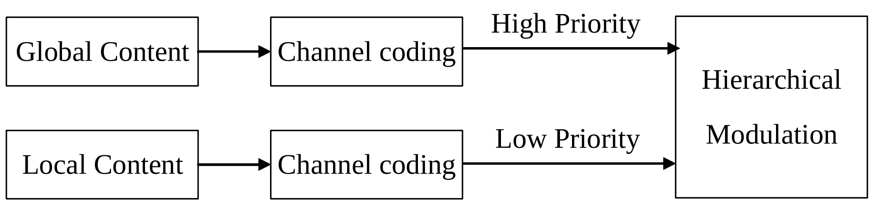


Figure 5.2: Hierarchical modulator for local content insertion

Figure 5.3 provides a block diagram of a receiver capable of demodulating hierarchically modulated signal and extracting both global and local content.

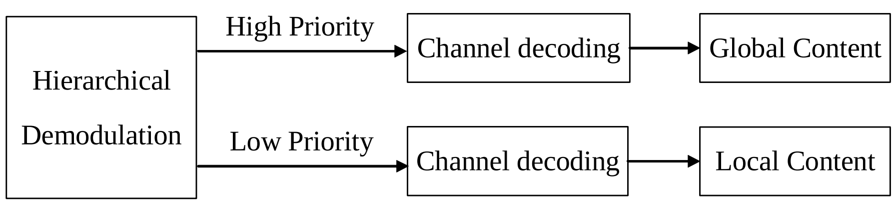


Figure 5.3: Hierarchical demodulator supporting local content insertion

NOTE: Hierarchical modulation can be used also for other purposes. These cases are described in clause 5.4.

Technique 2: Local content insertion by usage of the partially available transport stream.

- Another possibility is to use the notion of partially available Transport Stream for service regionalization as specified in TS 102 470-2 [21]. Using the SHIP signalling, the transmitters will forward only the relevant part of the TS. The satellite transmitter removes all the local content. The terrestrial transmitters remove the part of the local content they need not forward. The concept is exemplified in figure 5.4.

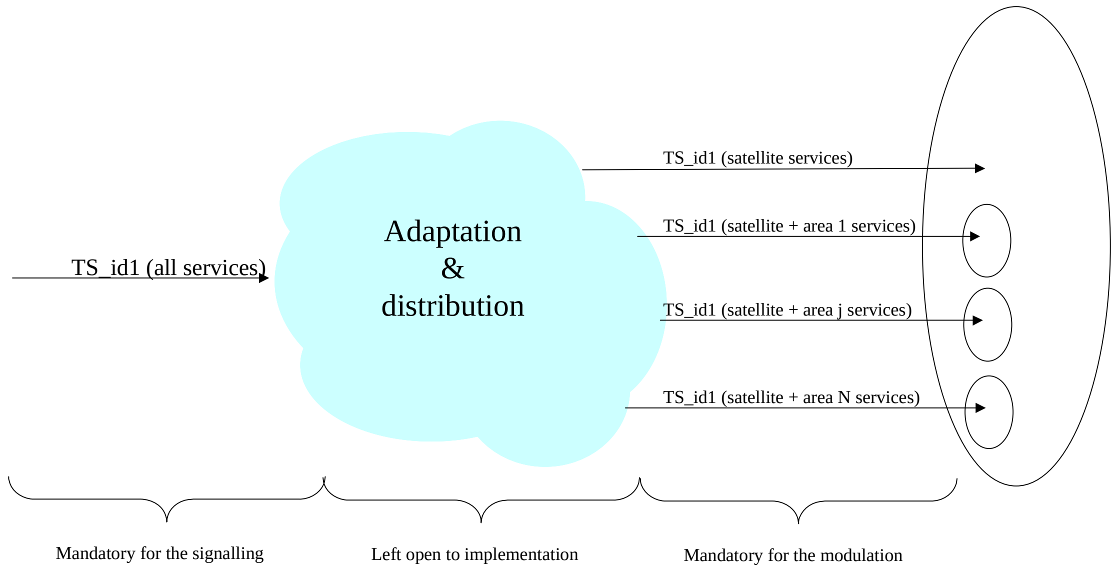


Figure 5.4: Service regionalization concept

- Service regionalization is fully specified in TS 102 470-2 [21].

5.2.5 CGC transmitters configuration

There are two main approaches for configuring the CGC network of transmitters:

- The "high-density", "low-power" approach attempts to reuse, all or partially, existing 3G/2G transmitter sites, or to build an equivalent low to medium height type of transmitters network. These networks are characterized by transmitter towers of typically 30 meter high delivering from 200 W to 1 kW ERP for dense urban coverage in the range from 0,5 km (deep indoor) to 2 km (outdoor).

- The "low-density", "high-power" approach attempts to reuse existing digital terrestrial TV transmitter sites, or to build an equivalent high-altitude transmitters networks. These networks are characterized by transmitter towers from 100-meter to 300-meter high, delivering from 1 kW to 4 kW ERP for typical coverage in the range from 5 Km to 7 Km.

5.2.6 Receiver configuration

The standard identifies two receiver classes based on their capability to process **time diversity** (refer to clause 10 for the memory sizing guidelines for each Class). Class 2 receivers offer maximum flexibility in terms of time diversity. class 1 receivers are compatible with the time diversity transmitted for class 2 (refer to clauses 6 and 7 for the different trade-offs between memory complexity and time-diversity performance).

5.2.7 Configuration naming

Among the various possible configurations described above, 4 are considered as critical in system deployment: Radio configuration, Receiver Class, SFN/MFN configuration, Local content handling.

We propose the following nomenclature for a given configuration.

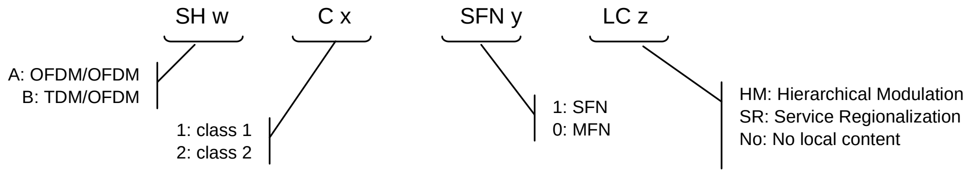


Figure 5.5: DVB-SH system configuration classification

The rules for referencing the configurations are the following:

- when the option is not set, the character is set to this value w, x, y, z; for instance if we want to refer to SH-A configurations in SFN, whatever the other parameters may be (Receiver class and Local content), we refer to "SH-A-C-x-SFN-1-LC-z";
- wildcards are always 1 character long and lower cap ("w", "x", "y", "z");
- exact values are either 1 character long in higher cap ("A", "B", "1", "2", etc.) or several characters long in either high or low cap ("hm", "sr", "HR", "SR", etc.).

5.2.8 Low Latency Extension

Low Latency: DVB-SH system using the optional low-latency extension as specified in Annex B of [1]. A system or equipment supporting the low latency extension shall be named DVB-SH-LL.

Low Latency DVB/SH service: special, partially available DVB/SH service (similar to local content).

Details on the low-latency extension are handled in Annex D.

5.3 Examples of system deployment

Below is a non-exhaustive list of relevant transitions:

- from **SH-w-C-1-SFN-1-LC-z** to **SH-w-C-2-SFN-1-LC-z:** this transition corresponds to a terminal upgrading in terms of memory, in particular by replacing the long upper layer with physical layer long interleaving;
- from **SH-w-C-x-SFN-0-LC-z** to **SH-A-C-x-SFN-1-LC-no:** this transition corresponds to a bandwidth upgrading by moving from non SFN conditions to SFN conditions;
- from **SH-A-C-x-SFN-1-LC-no** to **SH-w-C-x-SFN-0-LC-z:** it would make non sense downgrading overall bandwidth by changing from SFN to non-SFN unless terrestrial network is dense enough to enable important local content insertion that mitigates bandwidth loss. So this transition corresponds to an increase of local content by a strengthening of the CGC network;
- from **SH-A-C-x-SFN-1-LC-no** to **SH-A-C-x-SFN-1-LC-hm:** when the terrestrial network is dense enough, it makes sense to introduce some local content on the hierarchical modulation.

These transitions will be used in the following descriptions of system scenarios.

5.3.1 SH-A deployment scenario

Below is one typical SH-A deployment.

Step 1: early deployment of CGC

- SH-A-C-1-SFN-0-LC-no configuration enables to deploy receivers with the constraint of limited memory, both on physical and link layers. Antenna diversity and physical short time diversity offer quite good performance under CGC coverage. Frequency reuse is possible between all CGC cells.

Step 2: launch of a satellite with limited power

- no transition: LL-FEC is used for time diversity, depending on the link layer memory deployed in Step 1, early terminals may remain compliant;
- change to a stronger physical layer protection through SH-A-C-2-SFN-1-LC-no: early terminals are only "compatible" (inside CGC coverage, in good satellite only reception conditions);
- provide differentiated contents through SH-A-C-x-SFN-0-LC-sr: early terminals continue to receive as before under the CGC and correctly receive satellite only under good reception conditions;
- a transition to SH-B leads to the replacement of early terminals. The viability of this transition is therefore dependent on the latter installed base;
- in the case of NGSO satellite, SFN-yes may no longer be possible.

Step 3: launch a second generation terrestrial network we have 2 basic options

- no transition;
- increase local content insertion by increasing ratio between satellite and terrestrial bandwidth on the hybrid frequency through SH-A-C-x-SFN-1-LC-hm: early terminals continue to receive as before under the CGC and correctly receive satellite only under good reception conditions; in addition some local content can be received on the hierarchical modulation.

Step 4: launch of a satellite with extended power

- different Modulation and FEC parameters may be fine tuned to take advantage of the increase satellite margin. Note that SFN operation may limit the freedom to tune these parameters.

5.3.2 SH-B deployment scenario

In a typical SH-B deployment, we have the following steps:

Step 1: early deployment of CGC

- SH-B-C-1-SFN-0-LC-z configuration enables to deploy receivers having limited memory constraints, both on physical and link layers. Physical short time diversity offer quite good performance under CGC coverage.

Step 2: launch of a satellite with limited power capacity; 3 possible configurations are possible

- no transition: LL-FEC is used for time diversity, depending on the link layer memory deployed in Step 1, early terminals may remain compliant;
- transitions to a new configuration with better time diversity through SH-B-C-2-SFN-0-LC-z: early terminals are only "compatible" (inside CGC coverage, in good satellite only reception conditions);
- transitions to a new configuration with better local content insertion: transition through SH-B-C-x-SFN-0-LC-sr: early terminals are still compliant; more local content is available.

Step 3: launch a second generation terrestrial network we have 2 basic options

- no transition;

- increase local content insertion by increasing ratio between satellite and terrestrial bandwidth on the hybrid frequency and go to SH-B-C-x-SFN-0-LC-sr.

Step 4: launch of a satellite with extended power capacity

- different Modulation and FEC parameters can be easily fine tuned to take advantage of the increase satellite margin;
- depending on the relative satellite power, it could also be envisaged to transition to SH-A if local content insertion is not sufficient and satellite power is large enough.

5.3.3 Less likely transitions

These are listed hereafter:

- from **SH-w-C-2-SFN-1-LC-z** to **SH-w-C-1-SFN-1-LC-z:** this transition is relevant only when the network/market/use cases/receiver technology removes the need of long physical interleaver introduced in the earlier stage;
- from **SH-B-C-x-SFN-0-LC-z** to **SH-A-C-x-SFN-0-LC-z:** the underlying logic is that a transition from SH-B to SH-A is accompanied by a transition from non-SFN to SFN. Note however that a transition from non-SFN to SFN is not always easy if the early CGC network takes advantage of the non-SFN characteristic and is not configured uniformly;
- from **SH-w-C-x-SFN-0-LC-hm** to **SH-w-C-x-SFN-0-LC-sr:** it makes non sense decreasing terrestrial/satellite bandwidth ratio since the terrestrial network density can generally be improved over time and its bandwidth increased.

5.3.4 Forbidden configurations

Configurations that are not technically possible are listed below:

- **SH-B-C-x-SFN-1:** it is not possible with the SH-B radio configuration to have SFN since the modulation is not the same; however with tight synchronization between satellite and terrestrial signals, it is possible to have a code-combinable signal. This kind of detail is not listed here.
- **SH-A-C-x-SFN-yes-LC-rm:** it is not possible in the SFN case to have different radio configurations between the satellite and terrestrial transmitters and so enable content removal.

5.4 Other "advanced" hierarchical modulation use cases

5.4.1 Introduction

The following advanced use cases have been identified and listed in table 5.1.

Table 5.1: hierarchical modulation "advanced" use case list

Case	Name	Area	Comment
1	Quality increase	Satellite	This technique can can also be used also for providing support of heterogenous terminals.
2	Capacity increase	Satellite	This technique is already used by commercial systems for adding additional video programs on top of audio programs.
3	Coverage extension	Terrestrial; Satellite	This technique is already used by commercial systems.
4	Zapping time	both	

In the next paragraphs and for each use case an application scenario example is derived and their gain evaluated.

5.4.2 Use case 1: quality increase

The principle is to use a hierarchical 16QAM modulation configuration similar to a non-hierarchial QPSK one where the HP bit rate is maintained equal to the non-hierarchical QPSK stream while increasing the quality of the reception via an additional stream received on the LP.

- In a first operational mode, the video quality improvement is obtained by transporting over the LP channel additional MPE-IFECprotection for HP stream (typically bringing MPE-IFEC code rate from 2/3 to 1/.2).
- In a second operational mode, the video quality improvement is obtained by transporting over the LP channel an enhancement of video layers in using SVC technique. The flexibility of SVC enables to bring different usages:
  - PSNR scalability: Assume that a base layer transmitted with a bit rate r_{BL} yields an average PSNR of Q_{BL} for a certain video sequence. When an enhancement layer with an additional data rate r_{EL} is received, the PSNR can be increased to Q_{EL}. The equivalent single layer AVC stream of same bit rate $r_{BL} + r_{EL}$ would yield approximately same quality Q_{EL}.
  - Spatial scalability: Assume that the base layer provides video at QVGA resolution with an average PSNR of Q_{BL} at a bit rate r_{BL}. The enhancement layer brings an additional data rate r_{EL} which enables to provide roughly same quality at a larger resolution, e.g. VGA.
- In a third operational mode, LP bit rate can also be used for providing support to heterogeneous terminals without resorting to simulcasting. The additional SVC layers enable spatial resolution extension.

This use case is proposed to offer graceful degradation: on a low-protected TS, additional data can be exploited to improve content quality, either by sending enhancement layer data (as in scalable video coding), or by sending additional MPE-IFEC protection using a second source (see clause 6). The benefit on the signal quality will depend mainly on the position of the terminal under the satellite-only coverage:

- under good reception quality, the terminal will benefit from both TS signals, the high priority that conveys usual video and IFEC protection, and the low priority signal that conveys enhancement layer data and/or additional MPE-IFEC source;
- under bad reception quality, the terminal will benefit only from the high priority giving the basic video layer and primary source of MPE-IFEC protection.

5.4.3 Use case 2: capacity increase

The principle is to use the HM and H.264 SVC video coding so that HP TS carries the AVC-compliant whereas LP carries additional SVC enhancement layers. Since the base layer requires less bit rate than the AVC layer at the quality of the highest SVC enhancement layer, this will turn into more available channels. A different code rate can be used on the HP to mitigate the HP penalty with regard to QPSK and therefore roughly keep same coverage and availability for dual HP and LP reception as for QPSK.

On typical scenarios, using HM, we can provide:

- 33 % capacity increase (e.g. from 6 to 8 programs) with same quality as baseline during 1/3 of the total time (this is the due to LP reception penalty compared to HP stream).
- 30 % service disruption reduction due to better C/N requested level in HP compared to QPSK.
- This comes at the cost of a quality degradation during 60 % of service time since during 50 % of total time, we receive the only HP stream.

5.4.4 Use case 3: coverage extension

The principle is to benefit from intrinsic UEP from HM added to a complementary UEP with FEC (lower code rate) to:

- extend HP coverage to size larger than 16QAM coverage: under this HP coverage the bit rate will be lower than the 16QAM bit rate;

- limit the LP coverage degradation with regards to 16QAM coverage: under this LP coverage, the added bit rates of HP and LP will be greater or equal to the 16QAM bit rate.

Using HM with the application scenario suggested enables the operator to provide:

- same number of programs at base layer quality over an 10 % increase of the baseline coverage;
- same number of programs at same quality (base and enhanced layers) over 85 % of the baseline coverage.

So the cost of the extended coverage with lower quality is the diminution of the coverage with normal quality.

5.4.5 Use case 4: zapping time reduction

In cases of systems that use long interleavers, the principle is the use of the LP channel with a shorter interleaver to transmit auxiliary information, resulting in faster resynchronization of the video and audio decoders. For example, the LP interleaver might be reduced to one second, resulting in an average interleaver delay of 1,5 s. It is proposed that the LP channel contain a copy of the audio channel. This would allow the audio portion of the multimedia stream to begin to play much sooner after a channel change request. Additionally, it is proposed that the LP channel contain periodic intra coded frames, known in the H.264 coding standard as SI frames. The average delay before the first video frame is half the time between SI frames.

The consumer's experience of a channel change could therefore be the following:

- after a fixed delay of approximately two seconds, the user gets full audio and low frame rate (1 Hz) video from LP;
- this reception condition lasts for a period of 1,5 times the delay of HP interleaver;
- after this delay, the HP de-interleaver can deliver full rate video to the user an the channel change time is over.

This solution also gives the service provider the option to optimize the rate of IDR frames in both the HP and LP channels. Reducing the frequency of IDR frames in the HP channel (increasing GOP size) will result in either reduced HP bandwidth or improved video quality. Increasing the frequency of SI frames in the LP channel will provide still more improvements in user experience:

- at zapping or after a very long signal interruption (when all data, HP and LP, are lost): when signal resumes, LP will be faster to display than HP;
- some errors could happen in HP but not in LP that would then provide some diversity information (sse note).

NOTE: Statistics of such events is FFS.

Note that the above solution improves the zapping time only for terminals that are able to receive the LP channel. Assuming that the protection of the latter is weaker than on the HP channel, this might limit the effectiveness of the mechanism to few terminals in the coverage area.

Another possibility is the following: The base layer operates with the desired frequency of IDR frames, while the enhancement layer contains less IDR frames which saves a significant portion of the total data rate. In this case, all terminals benefit from the reduced zapping time, albeit at a slightly reduced quality immediately after channel switching until reception of the first IDR in the enhancement layer on the LP channel.

5.5 Examples of unequal bandwith configurations

We give hereunder typical examples of Unequal Bandwidth usage with DVB-SH technology. The overall assumptions are:

- Unequal Bandwidth is allowed only applicable for SH-B (see note 2 in clause C.2.1.2).
- We assume a global bandwidth of 15 MHz to be allocated to the SC.
- For the CGC the resources can be taken from the same spectrum block as SC or from another separated block.
- Any satellite path MUST be complemented by a terrestrial path, therefore the capacity of the terrestrial path MUST be superior or equal to the capacity of the satellite path.

The Unequal Bandwidth cases are of three kinds:

- Case 1: "increased satellite capacity with a reduced frequency reuse".
- Case 2: "terrestrial spectrum saving".
- Case 3: "digital dividend CGC".

5.5.1 Case 1: "increased satellite capacity with a reduced frequency reuse"

Instead of using a 3 × 5 MHz channelization on the satellite and a reuse factor of 3, we can use a reuse factor of 2 such as 8 MHz and 7 MHz, for a satellite with a reduced number of spot beams:

- Satellite beam 1: BW=8 MHz; roll-off=0,15; mod=8PSK; CR=1/2 => MPEG2 bitrate 9 282 kbps.
- Satellite beam 2: BW=7 MHz; roll-off=0,15; mod=8PSK; CR=1/2 => MPEG2 bitrate 8 195 kbps.
- On the CGC we keep the same 5 MHz channelization: BW=5 MHz; GI=1/4; FFT=2k; mod=16QAM; CR=1/2 => MPEG2 bitrate 9 400 kbps.
- This configuration enables to increase the satellite capacity on each spot beam.

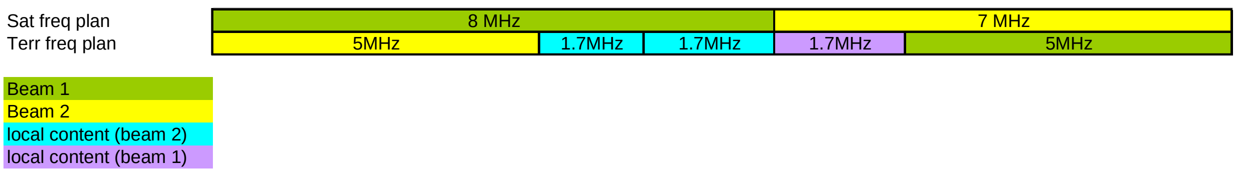


Figure 5.6: Case 1 frequency plan

5.5.2 Case 2: "terrestrial spectrum saving"

This configurations is the most bandwidth efficiency for SH-B since the terrestrial bandwidth required for repeating the satellite signal is reduced to its minimum value, 1,7 MHz and the totally occupied bandwidth is 5 MHz + 1,6 MHz:

- Terrestrial: BW=1,7 MHz, G/I=1/4, FFT=2k, modulation=16QAM, CR=2/3 => MPEG2 bitrate=2,84 Mbps.
- Satellite: BW=5MHz; roll-off=0,15; mod=QPSK; CR=1/3 => MPEG2 bitrate=2,591 Mbps.

This configuration has therefore a bandwidth occupancy almost low as the SH-A SFN one.

5.5.3 Case 3: "digital dividend CGC"

In this configuration, we assume the satellite bandwidth is in the S band whereas the CGC is in the UHF band, e.g. using the digital dividend resources. Depending on the local regulations, the terrestrial channelization is imposed by the multiplex allotting (e.g. 8 MHz in France):

- Ter: BW=8 MHz; GI=1/4; FFT=2k; mod=16QAM; CR=1/3 => MPEG2 bitrate 7 109 kbps.
- Sat: BW=5 MHz; roll-off=0,15; mod=16APSK; CR=2/7 => MPEG2 bitrate 4 424 kbps.

In this configuration, code combining is possible between satellite and terrestrial path, and some resources is available for local content on the terrestrial path.

6 Elements at Link and Service layers

6.1 Introduction

DVB-H presents a layered system structure: equipments operating on a specific layer can easily interconnect to equipments operating on an adjacent layer. Acknowledging this approach, the DVB-SH reuses to the most extent the DVB-H link and service layer in order to achieve seamless interoperability with DVB-H and to benefit from all available DVB-H link layer features as well as the already developed DVB-H ecosystem. This layered approach is presented in figure 6.1:

- a set of IPDC servers deliver IP streams, including the video streams;
- these IP streams are encapsulated by a DVB-SH IP encapsulator; the latter performs IP to MPE encapsulation according to EN 301 192 [9], PSI/SI insertion and MPE-IFEC protection and delivers an MPEG2 TS for the DVB-SH modulator;
- the DVB-SH modulators delivers a radio signal ultimately received by the DVB-SH receiver which performs baseband demodulation and decoding and processes the MPEG2 TS in the link layer client;
- the latter processes sections, MPE, MPE-FEC, MPE-IFEC, PSI/SI, and delivers an IP stream to the IPDC client;
- the IPDC client processes the IP streams, for example to deliver the ESG, the security decryption and the video and audio play out.

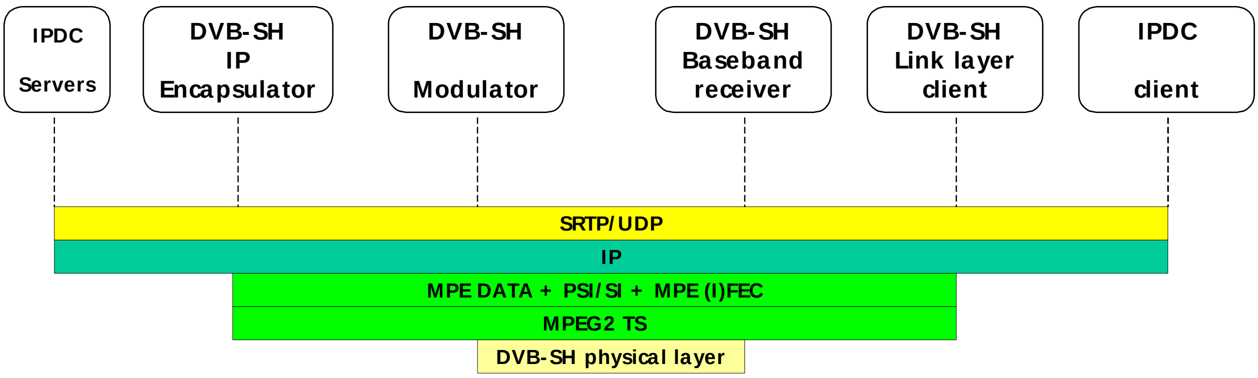


Figure 6.1: DVB-SH layered approach

Key features of DVB-SH link and service layers are:

- Support of Multi-Protocol Encapsulation:
  - DVB-H provides an IP multicast transport on top of MPEG2 Transport Streams (TS). To encapsulate the IP datagrams over MPEG2 TS, Multi-Protocol Encapsulation (MPE), (see EN 301 192 [9], clause 7) is applied. As the DVB-SH physical layer is also MPEG2 TS based, DVB-SH reuses MPE for the transport of IP datagrams over DVB-SH physical layers;
  - MPEG2 TS-based transport and MPE enable to reuse most signalling concept of DVB-H also for DVB-SH;
  - note that DVB-SH physical layer also supports the GSE but this is for further study.
- Support of time slicing:
  - DVB-H uses the real-time parameters, specifically the Delta-t information, conveyed within MPE and MPE-FEC headers in order to inform the start of the next burst. DVB-SH re-uses this concept: each MPE, MPE-IFEC and MPE-IFEC section carried by the MPEG2 TS over DVB-SH physical layer includes the same Delta-t information;

  - this mechanism enables to power off the terminal during periods where no relevant bursts for this service are transmitted. This also enables hand-over even for receivers with a single demodulator in case the infrastructure appropriately synchronizes the transmitted TS;
  - in addition, time slicing enables the efficient support of variable bit rate services since Delta-t can be adapted for each burst size. This is one way to efficiently support statistical multiplexing.

- Support of link layer protection:
  - DVB-H permits the use of link layer protection by applying MPE-FEC (see EN 301 192 [9], clause 9) to counteract terrestrial fading. DVB-SH also supports the use of MPE-FEC;
  - in addition, MPE-IFEC may be required in satellite coverage, especially with class 1 receivers;
  - by doing so, individual protection for each service is enabled. Depending on the service requirements and the physical layer performance, the transmitter can select from a variety of link layer parameters, e.g. using single burst MPE-FEC or multi-burst MPE-IFEC. Each FEC protection scheme can be fully configured to the service requirements thanks to a number of parameters;
  - simultaneous use of MPE-FEC and MPE-IFEC on the same elementary stream is for further study and therefore not allowed.

- Support of IPDC features:
  - DVB-SH is fully compatible with the DVB IPDC specifications, enabling a fast deployment of services on top of DVB-SH physical and link layers through the reuse of the IPDC protocol stack. Recommendations on the use of IPDC specifications in a DVB-SH context are given in [CDP over SH IG];
  - DVB-SH uses updated PSI/SI to convey system and program parameters. This enables smooth transition scenarios between DVB-SH and DVB-H networks, in particular for handovers: dual-mode receivers may receive content on one or the other technique seamlessly, provided PSI/SI are coherently signalled. The PSI/SI updates to support DVB-SH signalling are under study.

As DVB-SH relies to the most extent on DVB-H link layer technologies, this clause will only highlight the additions and differences of DVB-SH link layer compared to DVB-H. In case information on specific aspects on the DVB-SH link layer cannot be found in this clause, the DVB-H link layer specification has to be used.

To address the modification and additions of DVB-SH link layer compared to DVB-H link layer, this clause is organized in three parts:

- the first part introduces of a new link layer forward error correction referred to as MPE-IFEC. This new technology is motivated by the fact that DVB-SH networks without terrestrial repeaters can results in challenging propagation channels ("Land Mobile Satellite"). Signal outages in the order of several seconds may be quite frequent, in particular due to blockage. These outage durations generally exceed conventional physical or link layer error correction capabilities. The interleaving depth of such systems is generally in the order of several milliseconds for DVB-T or at most several hundreds of milliseconds with MPE-FEC. In order to counteract signal outages of several seconds, it is necessary to extend the protection capabilities of the forward error correction schemes. On may either use physical layer protection with very long interleavers or may use a link layer forward error correction scheme. DVB-SH provides both solutions: when transmitting to class 1 receivers, one should rely on a combination of short physical layer FEC and extended link layer FEC, referred to as IFEC and introduced in clause 6.2 MPE-IFEC; when transmitting to class 2 receivers, one may rely exclusively on a physical layer FEC with extended interleaving as described in clause 7;

- the second one addresses the backward compatibility of the DVB-SH physical layer with existing DVB-H time slicing and related legacy features. These include Time-slicing signalling, power saving, variable bit rate support and statistical multiplexing support. Clause 6.3 Time-Slicing describes what level of compatibility (such legacy DVB-H features) is introduced by the DVB-SH physical layer for both terminal classes;

- the third part deals with mobility aspects and IPDC features (clause 6.4 Mobility).

6.2 MPE-IFEC

6.2.1 Framework description

The MPE-IFEC has been designed taking account:

- the necessity to have a protection scheme to counteract the disturbances in DVB-SH transmission and reception environments;
- to achieve this, the MPE-IFEC, contrarily to MPE-FEC, encodes over several time-slice bursts, which enables to increase the interleaver duration;
- to address legacy requirements to existing DVB-H equipment;
- to enable a flexible solution which permits service specific adjustments;
- to provide the option to do further adjustments and optimizations during deployments.

The features supported by the framework are presented below.

- **Compatibility with DVB-H link layer (MPE sections):** MPE-IFEC is introduced in a way that it does not modify MPE section format, but only introduces one new sections type, the MPE-IFEC section. As MPE sections, MPE-IFEC sections convey almost identical real-time parameters. MPE-IFEC conveys exactly the same Time-slicing information as MPE. In addition, MPE-IFEC sections introduce additional information on the respective positions of MPE sections within the burst via the use of `MPE_boundary` bit (`frame_boundary` bit signals the end of the MPE-IFEC section part inside the burst).
- **Support of MPE-FEC:** permits the concurrent use of MPE-FEC and MPE-IFEC sections in one burst. However the support of simultaneous MPE-FEC and MPE-IFEC decoding in DVB-SH is for further study and therefore the parallel sending of MPE-FEC and MPE-IFEC on the same elementary stream is not allowed. This choice can be done on a per elementary stream bases so that MPE-FEC can be applied, on the same transport stream, on one elementary stream while another has MPE-IFEC. MPE-IFEC also relies on the same Application Data Tables (ADT) as the MPE-FEC and the transmitter may use the ADT of the MPE-FEC as the ADST of the MPE-IFEC. As a consequence, the number of rows of MPE-IFEC ADST is the same as the number of rows of MPE-FEC ADT. In case of losses of MPE sections, the receiver may use MPE-FEC sections, (exclusive) or MPE-IFEC sections, depending on the time_slice_fec_identifier signalling, but not both simultaneously, to recover the missing data.
- **Support long interleaving:** MPE-IFEC allows significantly enhanced performance (as measured by ESR5(20)) in LMS channels when compared to MPE-FEC as demonstrated by simulations. This performance benefits can be achieved as the MPE-IFEC encoding process spans several time-slice bursts. MPE-IFEC requires a burst numbering which is signalled in MPE-IFEC headers.
- **Support of different service requirements:** the MPE-IFEC can be configured to enable a variety of configurations providing flexibility for the network operator. Guidelines are provided later in this clause on how to select these parameters.
- **Support of fast zapping:** inter burst FEC protection is adversely affecting latency, and therefore also has influence on channel zapping times. This is due to the fact that for making use of all MPE-IFEC sections protection, the receiver must perform a late decoding and wait for reception of all MPE sections and MPE-IFEC sections relating to the encoding matrix to which a certain MPE section is assigned to. However, by sending MPE sections in each burst, immediate access and processing of these MPE sections is possible in case no errors have occurred. As reception continues, additional parity is received and MPE-IFEC protection can progressively protect the data by doing an early decoding. After some time, late decoding is possible.
- **Support a variety of FEC codes:** the framework can support a variety of codes, currently Reed Solomon is the only code supported by DVB-SH, other FEC codes are for further study.

The MPE-IFEC provides an inter burst protection. Compared to MPE-FEC, this is achieved by either:

- the increase of the encoding matrix to sizes larger than one burst (one encoding matrix is filled during several successive bursts instead of a unique burst as in the DVB-H case);

- the parallelization of the encoding mechanism (instead of using only one matrix, data are distributed over a number of B parallel matrices, among M possible);
- the combination of the above both concepts.

The framework encoding process is shown in figure 6.2.

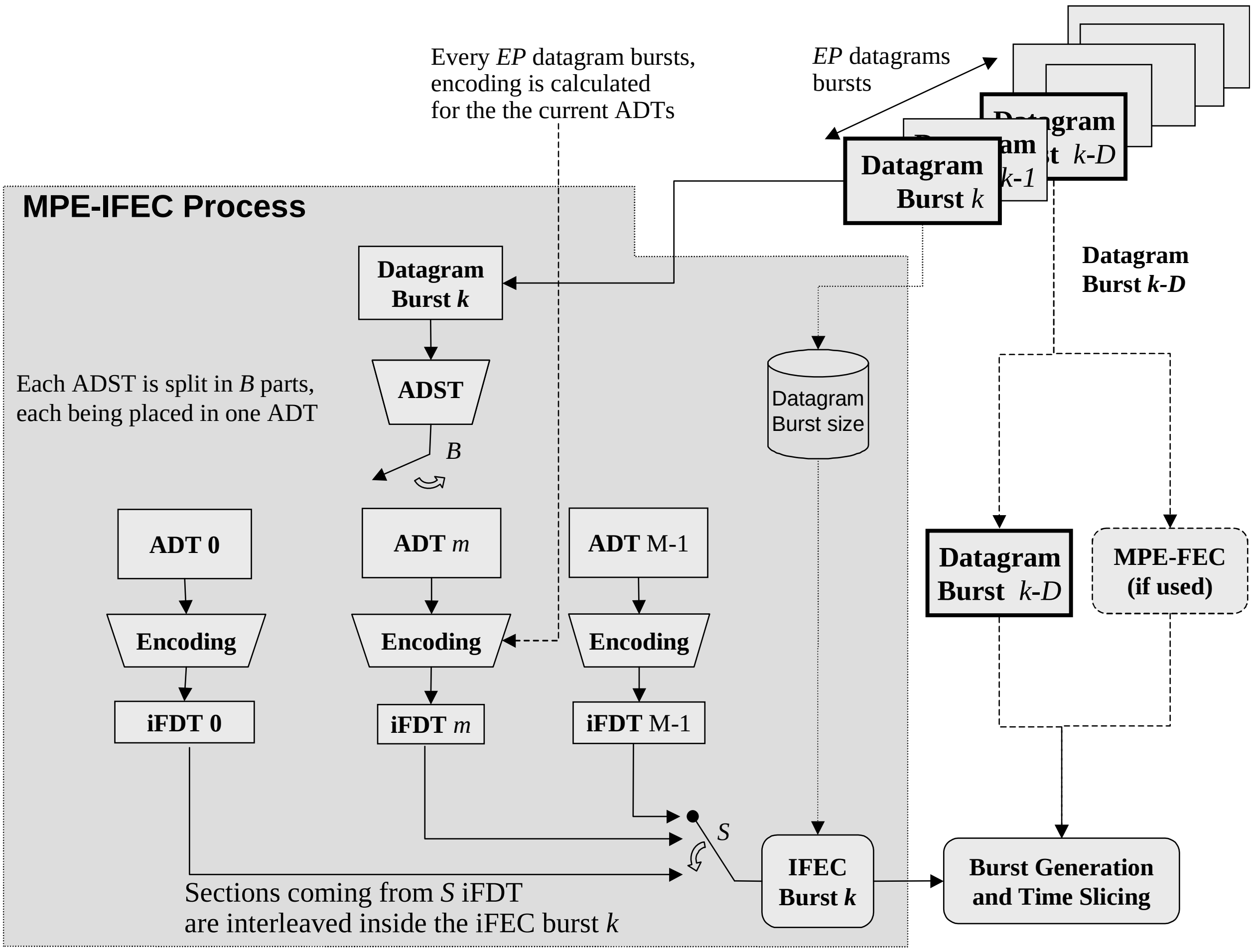


Figure 6.2: MPE-IFEC encoding process

The MPE-IFEC encoding process operates on a sequence of datagram bursts. Datagram bursts consist of a collection of datagrams, i.e. generally IP datagrams. Each datagram burst may be of different size and may comprise a different number of IP datagrams. For each datagram burst, a corresponding IFEC burst is created which encompasses MPE-IFEC sections from iFDTs of S encoding matrices. An MPE-IFEC section is comprised of a header, the parity symbols from one or multiple columns from the same iFDT, and a checksum. This IFEC burst is then multiplexed with all MPE sections of an original datagram burst, including the corresponding MPE-FEC sections if present. This multiplexing is done with the datagram just received in case the delay parameter D is set to D=0, or with a previously received datagram when D>0. This collection of MPE, MPE-FEC, and MPE-IFEC sections forms an MPE-IFEC time slice burst that is further processed. The sections within the IFEC time-slice burst are mapped to MPEG2 TS packets and then encapsulated in the time-slice bursts. Each datagram burst is mapped by an ADST (Application Data Sub Table) function on to the B ADTs (Application Data Tables) selected out of M parallel encoding matrices. Every EP bursts, an encoding process is carried out on one of the encoding matrices, generating from the ADT the corresponding iFDT (IFEC Data Tables) by applying an FEC encoding function. The resulting protection is achieved at the cost of some latency for generating and receiving parity data since the parity data is computed and spread over different bursts instead of one single burst in the MPE-FEC case.

The link layer encoding key parameters are given below:

- **EP:** this is the *encoding period* of the FEC process expressed in burst units. It refers to the frequency with which FEC is computed: an EP of 1 means that the encoding process occurs at every burst whereas an EP greater than 1 means that the encoding process occurs every EP bursts and the encoding matrix capacity is EP times greater so it takes EP times longer to fill the matrix with data. This EP normalizes all other parameters except D.

- **B:** this is the *encoding parallelization* expressed in encoding matrix units. Every burst is split into B parts distributed over B parallel encoding matrices. So actual interleaving depth is B*EP bursts, or a computed FEC has correction capabilities spanning EP*B successive bursts. The B matrices used for the encoding are updated every EP bursts and the list is completely refreshed after EP*B bursts.

- **S:** this is the depth of the FEC spreading factor; it means that produced FEC is interleaved over S iFDT. This enables to better protect the produced FEC.

- **D:** this is the delay applied to the data. Since FEC is computed with the received data, the normal sending order (send data immediately) would imply sending the FEC after the data. This parameters influences the zapping quality but has impact on end-to-end latency (see clause 6.2.4).

These four parameters enable to configure the encoding process in a flexible way. A typical configuration is obtained by setting EP=1 and B>1. This results in a sliding encoding process which enables to reuse several components of an MPE-FEC implementation: every received data burst is interleaved over B encoding matrices and one iFDT is computed applying the MPE-FEC Reed Solomon code.

To familiarize the reader with the MPE IFEC concept, figure 6.3 provides an example encoding and sending process with EP=2, B=9, S=2, and D=0. The horizontal axis represents the M=11 encoding matrices, numbered from 0 to 10, and the vertical axis represents the continuous datagram and MPE-IFEC time slice burst numbers (right column from 0 to 33).

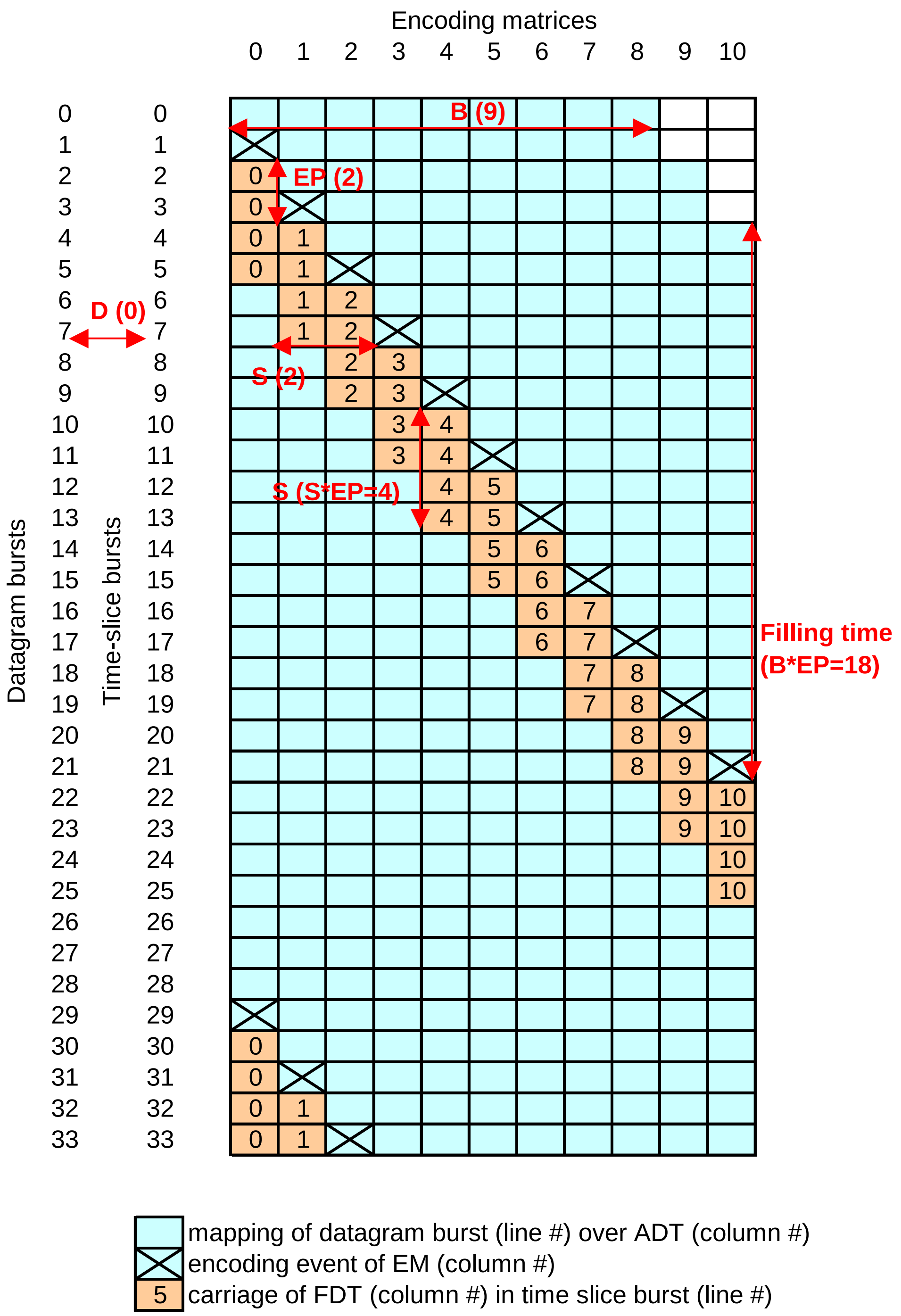


Figure 6.3: One example (EP=2, S=2, B=9)

Figure 6.3 can be read either horizontally or vertically:

- Horizontally:
  - the horizontal reading gives a representation of the sending logic:
    - the first two columns give the impact of the parameter *D* on the datagram burst sending: if *D* is set to 0 (here this is the case), the datagram burst is sent at the same time as the MPE-IFEC time slice burst. For instance, datagram 7 is sent with burst 7. Should the *D* parameter be greater than 0, then the datagram burst *k* would be sent in burst *k*+*D*;

  - for a given MPE-IFEC time slice burst number, if there is an orange cell on the corresponding line, it means that some FEC excerpted from the corresponding FDT is sent in the burst. For instance, at burst 7, MPE-IFEC sections from FDT 1 and 2 are sent. The number of "source" FDT interleaved in the current burst is dictated by the *S* parameter (here set to 2). The choice of the FDT matrix number is dictated by `ifdt_function` in MPE-IFEC specifications [attachment];

- the horizontal view indicates also how the datagram burst is mapped on the ADST:

  - those ADT that receive columns from the current datagram burst are represented by the cyan colour: for example ADT 0 to 8 receive data from datagram burst 0 and 1, ADT 1 to 9 receive data from datagram burst 1 and 2, etc.;

  - one can see the influence of the *EP* parameter: the list of ADT receiving data is "updated" every EP datagram burst received. The number of the ADT used for mapping is given by `adt_function` in MPE-IFEC specifications [attachment].

• Vertically:

- the vertical reading gives a representation of the encoding logic:

  - every *EP* bursts, there is an "encoding event" indicated by a cross: one particular matrix encodes the data (it generates the iFDT from the ADT). After the completion of the encoding, the ADT can be freed from its data so that this ADT can be used to host new datagram bursts again. For instance, encoding matrix 4 has an encoding event at MPE-IFEC time slice burst 9. Its data can be reset after then; indeed this ADT it is not used during the 6 following bursts, then it is again filled starting at burst 20;

  - the encoding event creates FEC information that is transmitted during a successive number of bursts that is equal to *S*EP*. For instance, FDT 4, computed before burst 9 is sent, is sent during bursts 10 to 13 in 4 pieces;

  - since all other parameters are normalized by *EP*, the change of list of encoding matrices happens every *EP* bursts and the list has completely be renewed after *EP*B* bursts, which is also the time it takes to fill a matrix. For instance ADT 10 starts to be filled at burst 4 and stops at burst 21, 17 bursts later, which makes 18 bursts overall.

6.2.2 Usage in the context of the DVB-SH

In the DVB-SH context of the current implementation guideline release, only RS(255;191) (see EN 301 192 [9], clause 9.5.1) is used as encoding scheme. The MPE-IFEC framework is instantiated on this code with the following mandatory parameters: EP = 1, B ≥ 1, S ≥ 1, D ≥ 0.

Again, to provide some insight, we discuss an even simpler example with *B* = 3 and *S* = 1 where a completely lost burst can be recovered with a link layer FEC code rate of only 3/4. Table 6.1 shows, for each burst, what happens at transmitter and receiver sides and represent the event of burst "n" complete loss:

• on transmitter side:

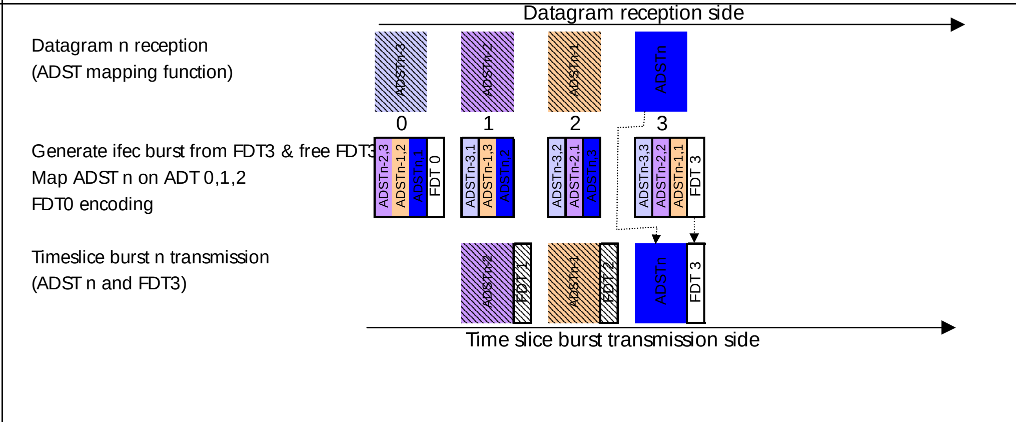
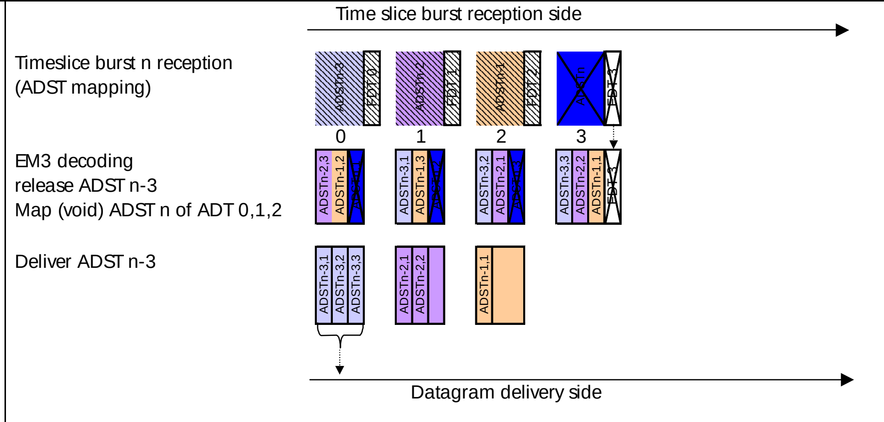
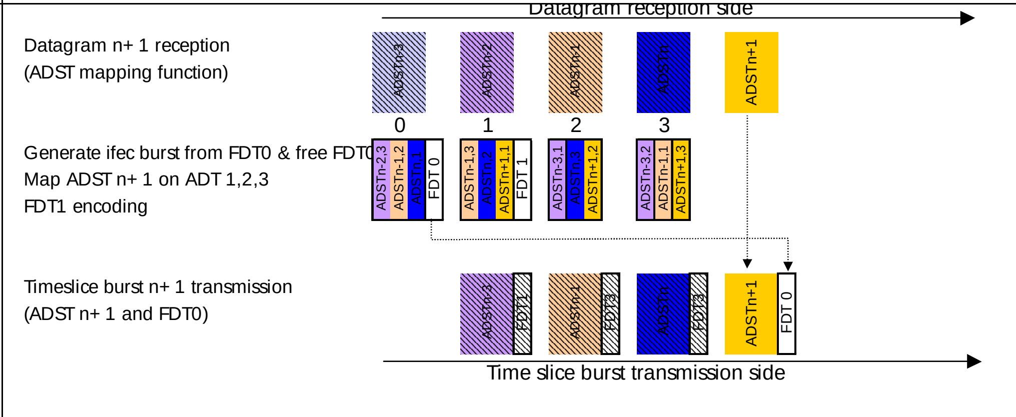
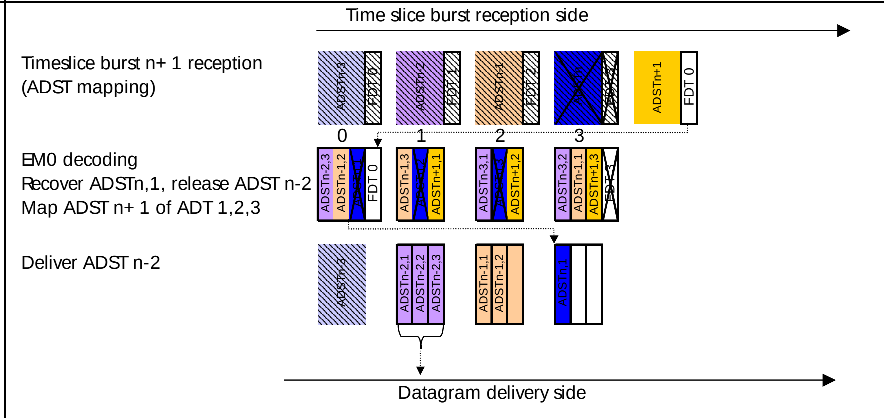
- the first row represents datagram bursts, the data to be transmitted;

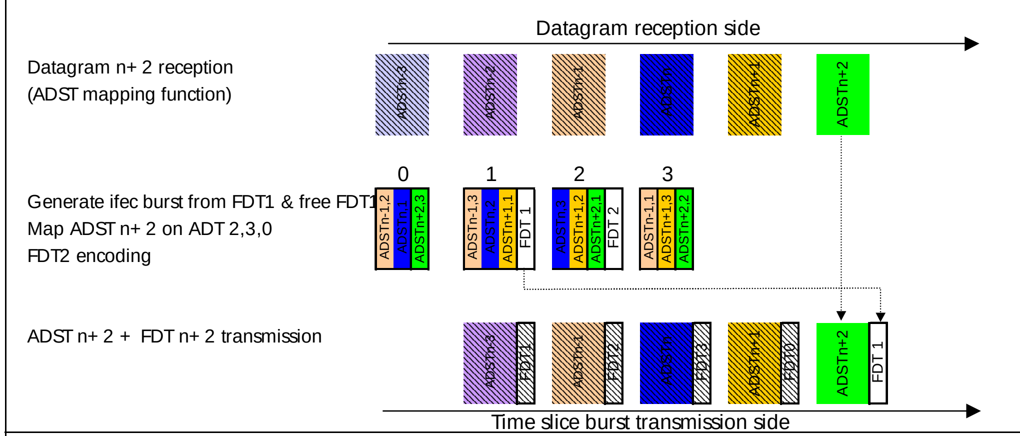
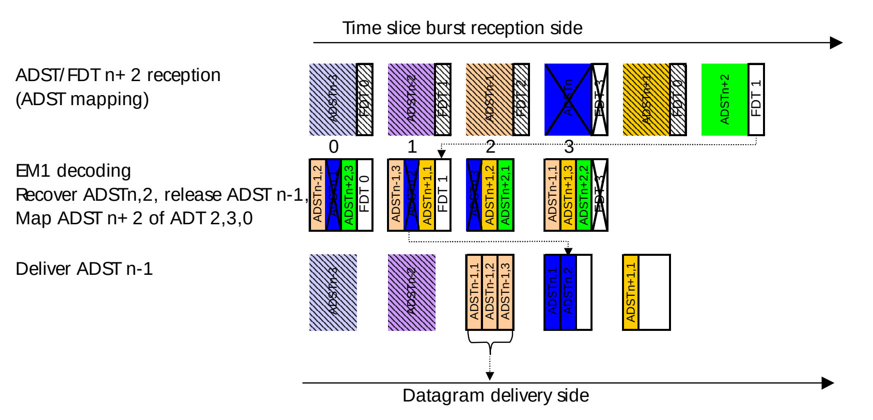
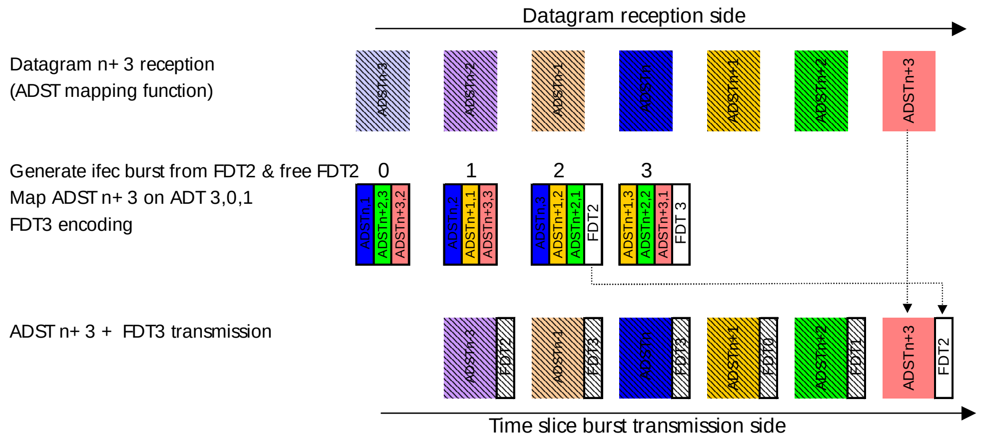
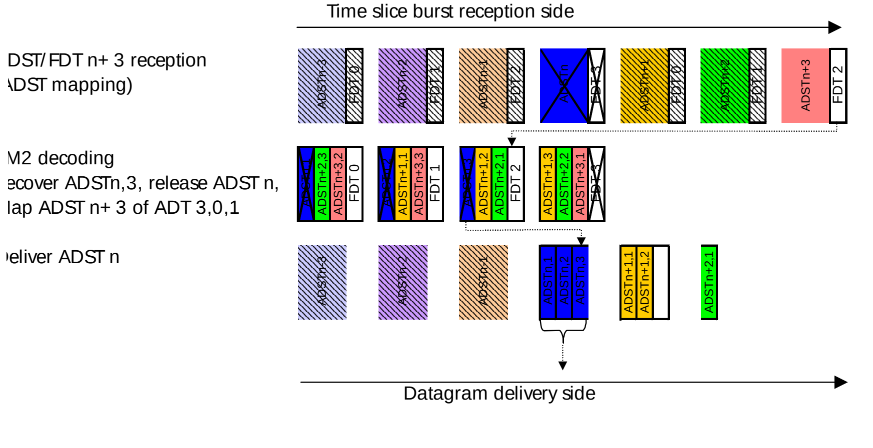
- second row represents the 4 parallel encoding matrices used by the link layer: ADT are filled by received datagram bursts and FDT are computed from the ADT;

- third row represents the actually sent MPE-IFEC time slice bursts made of datagram bursts and IFEC bursts derived from one unique FDT. As *S* = 1, the IFEC burst is constructed from a single FDT;

- on receiver side:
  - first row represents received time slice MPE-IFEC burst, possibly with their losses;
  - second row represents the 4 parallel decoding matrices: ADT are filled by received datagram burst and FDT are filled by received MPE-IFEC burst;
  - third row represents the data to be sent to the higher layer, whether it may be received "lossless" or recovered by FEC decoding.

Table 6.1: Example of complete burst recovery

Step	Transmission side	Reception side
n		
	Transmission of datagram burst n; previously computed FDT 3 is sent in MPE-IFEC time slice burst n; datagram burst n is being mapped over ADT 0, 1, 2; FDT 0 is computed.	We assume previous bursts k<n have been well received but burst n is completely lost (data and FEC). We need to map a void ADST to avoid glitches inside ADT.
n+1		
	Transmission of datagram burst n+1; previously computed FDT 0 is sent in MPE-IFEC time slice burst n+1; datagram burst n+1 is mapped over ADT 1,2,3; compute FDT 1.	Reception of MPE-IFEC time slice burst n+1; decode FDT 0 and recover ADT(n,1); map ADST n+1 into ADT 1,2 and 3.

<table>
<tr><th>Step</th><th>Transmission side</th><th>Reception side</th></tr>
<tr><td rowspan="2">n+2</td><td>
</td><td>
</td></tr>
<tr><td>Transmission of datagram burst n+2; previously computed FDT 1 is sent in MPE-IFEC time slice burst n+2; datagram ADST n+2 is mapped over ADT 2,3,0; compute FDT 2.</td><td>Reception of MPE-IFEC time slice burst n+2; decode FDT 1 to recover ADT(n,2); map data burst n+2 into ADT 2,3 and 4.</td></tr>
<tr><td rowspan="2">n+3</td><td>
</td><td>
</td></tr>
<tr><td>Transmission of datagram burst n+3; previously computed FDT 2 is send in MPE-IFEC time slice burst n+3; datagram burst n+3 is mapped over ADT 3,0,1; compute FDT 3.</td><td>Reception of burst n+3; decode FDT 2 to recover ADT(n,3) ⇒ ADT n has been completely recovered; map ADST n+3 in ADT 3, 0 and 1.</td></tr>
</table>

6.2.3 A practical example

6.2.3.1 Introduction

In the rest of the document, we will continuously refer to the following typical example: EP=1, B=6, S=4, D=0, MPE-IFEC code rate = 2/3. Assume for further simplicity that the service to be protected has a constant bit rate of 300 kbps and IP packets of 1 000 bytes are sent.

The ADST is determined by its T=1 024 rows and has on average 37 data columns (a data column is a column with at least 1 non padding byte). To accommodate with local variations, we fix C to 40 so that 4 more additional data columns could be absorbed in case of small traffic variations.

The number of ADT over which the ADST are padded is equal to B+S (9).

Each time an encoding event happens, an FDT of N columns is created. Based on the code rate 2/3 and the 37 non padded columns, 19 columns on average are excerpted from this FDT for transmission.

6.2.3.2 Time_slice_fec_identifier

The time_slice_fec_identifier provides the following information about the stream inside the INT.

Table 6.2: Time_slice_fec_identifier

Syntax	Number of bits	Value	Comment
time_slice_fec_identifier_descriptor () {			
descriptor_tag	8	01110111	0x77
descriptor_length	8	00001011	11
time_slicing	1	1	Time slicing is used
mpe_fec	2	00	Mpe fec is not used
reserved_for_future_use	2	11	N/A
frame_size	3	11	1 024 rows
max_burst_duration	8	00001001	200 ms
max_average_rate	4	0101	512 kbps
time_slice_fec_id	4	0001	MPE-IFEC is used
T_code	2	00	RS(255,191) is used
G_code	3	000	G=1
Reserved for future use	3	111	N/A
R	8	01000000	R=64
C	13	00101000	C=40
Reserved for future use	3	111	N/A
B	8	00000110	B=6
S	8	00000100	S=4
D	8	000000	D=0
EP	8	00000001	EP=1
`Max_rate_averaged_over_B`	8	100101100	300 kbps
}			

This information is firstly retrieved by the receiver to determine:

- what type of link layer protection is active on the elementary stream of interest (none, MPE-FEC, MPE-IFEC, exclusive choices);
- if one link layer is active, which are its main parameters.

Note that the time_slice_fec_identifier parameters of MPE-FEC have not been modified at all: only additional parameters have been added inside the id_selector_bytes (not used by DVB-H) and 1 reserved for further study bit has been used in `mpe_fec` and `time_slice_fec_id` fields. Therefore, when new values are not needed, parameters used by MPE-FEC are reused by the MPE-IFEC with exactly the same definition (for instance `frame_size`, `max_burst_duration`, `max_average_rate`).

6.2.3.3 MPE-IFEC parameter derivation

The receiver can now derive from the time_slice_fec_identifier the required parameters for performing the MPE-IFEC decoding. Applying the parameters selection as specified in MPE-IFEC specifications [attachment], clause 5.3 we have the following list of MPE-IFEC parameters useful for decoding.

Table 6.3: Example of DVB-SH MPE-IFEC parameters selection

Parameter	Unit	Value	Description	Signalling	Scoping
EP	Datagram burst	1	IFEC Encoding Period	Direct via Time_slice_fec_identifier	Time_slice_fec_identifier
D	Datagram burst	0	Datagram burst sending delay	Direct via Time_slice_fec_identifier	Time_slice_fec_identifier
T	rows	1 024	Number of ADST, ADT, iFDT rows: T= MPE-FEC Frame rows /G	Indirect via Time_slice_fec_identifier	Time_slice_fec_identifier
C	columns	40	Number of ADST columns	Direct via Time_slice_fec_identifier	Time_slice_fec_identifier
R	sections	64	Maximum number of MPE IFEC sections per MPE-IFEC Time-Slice Burst	Direct via Time_slice_fec_identifier	Time_slice_fec_identifier
K	columns	40	Number of ADT columns = EP*C	Indirect via Time_slice_fec_identifier	Time_slice_fec_identifier
N	columns	64	Number of iFDT columns = EP*R*G	Indirect via Time_slice_fec_identifier	Time_slice_fec_identifier
G	columns	1	Maximum number of iFDT columns per IFEC section	Direct	Time_slice_fec_identifier
M	ADT	M = B + max(0,S - D) + max(0,D - B) = 10	Number of concurrent encoding matrices M	Indirect (formula dependent on T_code and given in the parameter definition of clause 5)	Time_slice_fec_identifier
k_{max}	N/A	256 - 256 [i.24] = 250	Modulo operator for MPE-IFEC time slice burst counter	Indirect (formula dependent on T_code and given in the parameter definition of clause 5)	Time_slice_fec_identifier
j_{max}	N/A	10	Maximum backward pointing for datagram burst size used in PREV_BURST_SIZE parameter in clause3.5	Indirect (formula dependent on T_code and given in the parameter definition of clause 5)	Time_slice_fec_identifier

This parameter derivation, in particular, enables the receiver to know if it has the capacity to process the MPE-IFEC decoding extensively. The memory is derived from encoding matrix size given by *T*, *C*, *K*, *N* and the number of encoding matrices (for more information on memory sizing, please refer to clause 6.2.6 Memory requirements).

6.2.3.4 ADST mapping function

ADST mapping and padding: the ADST function maps the received IP packets on a matrix of size 1 024 rows by 40 columns. This results in figure 6.4.

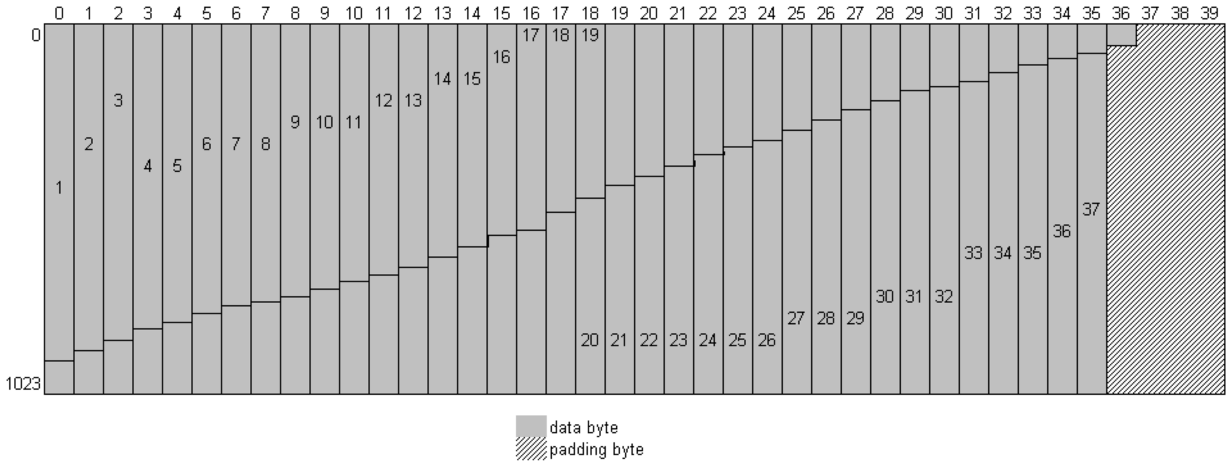


Figure 6.4: ADST and padding

In this figure, one can see that 37 columns (from number 0 to 36) have at least 1 data byte and are so called data columns (even if one, the N°36 has very few data bytes) and 3 are padding columns (from 37 to 38) because all their bytes are padding. Another remark is that, generally, the loss of 1 IP packet during transmission will occur losses on two columns, with the exception of the first IP packet.

6.2.3.5 ADT mapping

To illustrate the ADST to ADT mapping, we take two views, the ADST and then the ADT.

6.2.3.6 ADST view

From the ADST point, it is observed that the ADST columns are inserted ("shifted" as said in MPE-IFEC specifications [attachment]) into different ADTs. An example of such ADST columns distribution is given in figure 6.5, for the case of datagram burst 0.

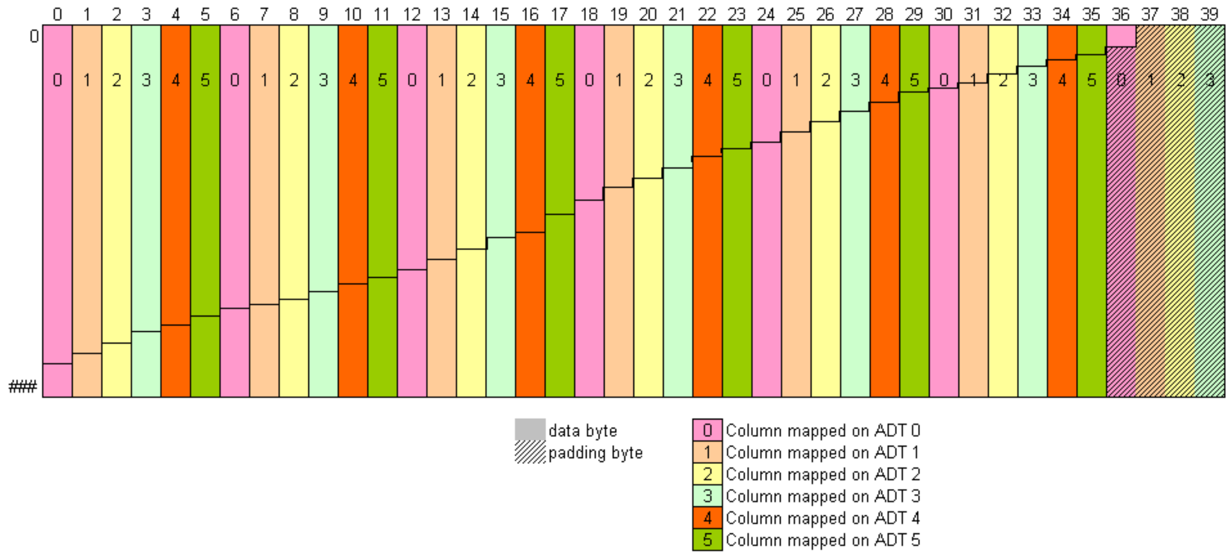


Figure 6.5: ADST to ADT mapping (ADST view)

It can be seen in figure 6.5 that the columns are interleaved between the different ADTs (they are not taken "in block"). Additionally, they are distributed over B ADTs (here B=6 so ADT 0 to ADT 5). The last columns, that are padding columns, are (and must be) also distributed over the B ADTs.

6.2.3.7 ADT view

Structure of an ADT:

It is observed that an ADT within a single encoding matrix hosts columns from *B* different datagram bursts. The mapping is defined by the ADST profile (its number of data and padding columns) and the `adt_index` function which determines in which ADT the ADST columns are shifted (inserted).

Due to the shifting done at each burst (at each received burst, new ADST columns are introduced in *B* ADTs), the "population" of one ADT is progressively done between two successive encoding events on this ADT (once the encoding event has occurred, the ADT is reset and ready to accept new columns). Due to the fact that *M* can be larger than *B*, all ADTs do not include columns from all ADSTs: only when *B*=*M* do we have the situation when all ADTs receive columns from any ADST. In the general case when *B*>*M,* during a number of burst equal to *M*-*B*, the recently encoded and reset ADT does not receive any data. When data are again inserted from burst k and ADT m, the following relationship is valid:

adt_index(k+B-1,0)=m, or (k+B-1)[M]=m, or there exist an "i" such that k = M*i + m-B+1

The ADT will then be filled by next *B* bursts, so bursts k, k+1,…k+*B*-1

We have presented as an example in figure 6.6 the ADT 0 after first 20 datagram bursts have been received. In this figure, we can see that the last columns to have been "shifted" in the ADT 0 (the ones that are on the rightmost part), are coming from burst 20, and the first ones to have been shifted in are on the leftmost part (due to the "push" from right to left of newer columns) and are coming from burst 15, which respects the above relationship.

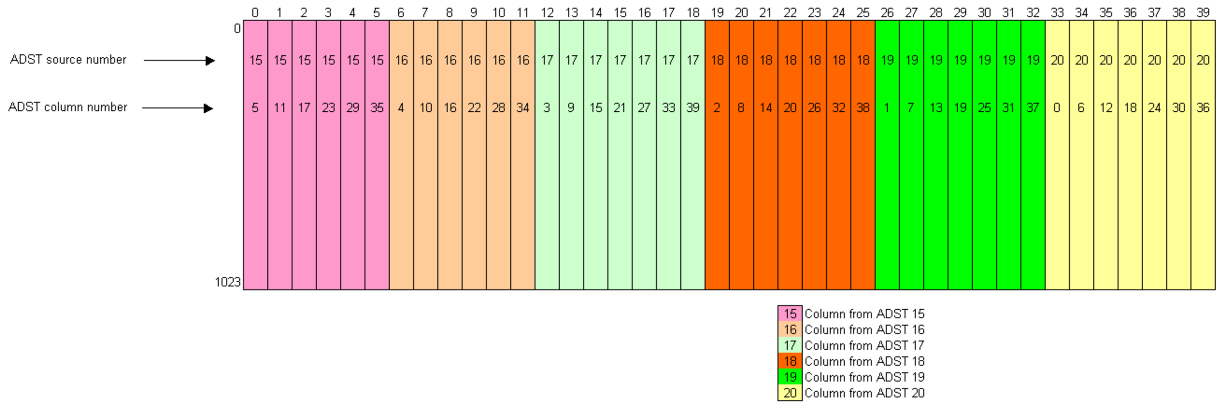


Figure 6.6: ADST to ADT mapping (ADT view)

It can be observed in this figure that the source ADST columns are grouped by "sub-block". This is due to the "shifting" function that inserts a certain number of columns from the same burst. Note also that the source ADST columns indexes are not continuous but separated by *B* (for example see column indexes coming from ADST 15: 5, 11, 17, 23, 29).

The `adt_column` function

More generally, a complete ADT ready for encoding has the following structure presented in figure 6.7.

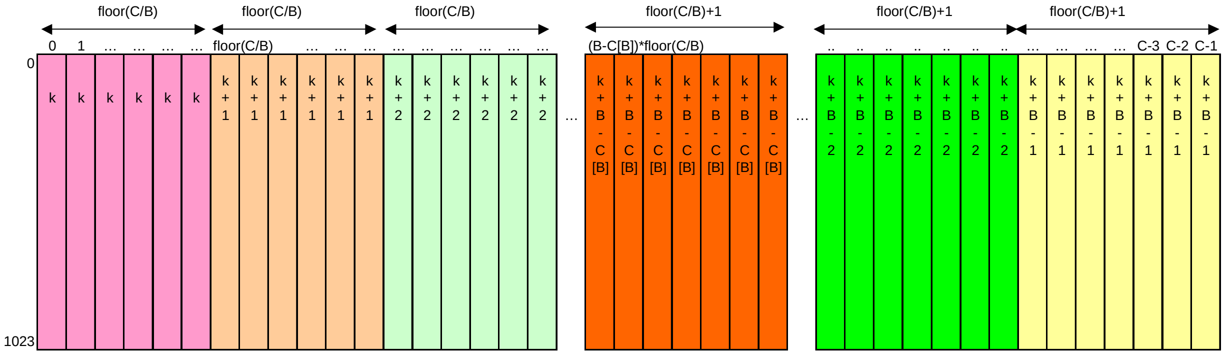


Figure 6.7: Generic ADT structure

The ADT is comprised of B sub-blocks, each sub-block being consisting of a number of columns coming from the same datagram burst and corresponding ADST:

- Each of the first (*B*-*C*[*B*]) sub-blocks hosts floor(*C/B*) columns from same ADST; the floor(*C/B*) ADST columns of the k^{th} block, k∈[1;*B*-*C*[*B*]], are the columns j, j∈[0;*C*-1], such that k = *B*-j[*B*], so the formulation of the column numbers is the following:

  For k = 1:1:B-C[B], for l = 0:1:floor(C/B)-1, j(k,l)=l*B+(k-1)

- Each of the last C[B] sub-blocks host floor(C/B)+1 columns of the same ADST; the floor(C/B)+1 ADST columns of the k^{th} block, k∈[B-C[B]+1;B], are the columns j, j∈[0;C-1], such that k = B-j[B], so the formulation of the column numbers is the following:

For k= B-C[B]:1:B, for l = 0:1:floor(C/B)-1, j(k,l)= l*B+(k-1) and j(k, floor(C/B))= floor(C/B)*B + (B-k)

Hence, the shifting operation in the MPE IFEC specification for each ADST to ADT mapping in case of *EP*=1 may be realized as ***deterministic positions*** inside the ADT when the time to encode the matrix has arrived. These positions are given by the following formula:

adt_column(j) = (B-(j[B]+1))*floor(C/B) + max(0;(C[B])-(j[B]+1)) + floor(j/B)

The use of such `adt_column` formula is easier than implementing the "shifting" method since it establishes a bijection between ADST and ADT columns. An ADST can be then considered as a collection of *C* "pointers" to the ADT columns. This approach is presented in figure 6.7 where the column number of the original ADST are presented below the ADST number (this is the pointer origin) and the column number of the destination ADT are presented at the top of the column (this is the pointer destination). Mapping ADST results in setting the pointers destination to the correct ADT column using the `adt_column` function and this update occurs only once instead of *B* times with the original shifting mechanism. The advantages of this approach are twofold:

- This helps storage optimization since storage is done inside the ADTs rather than on the ADSTs, each ADST being a list of *C* pointers to *C* columns in *B* ADTs. More memory implementation discussion that uses this pointer approach can be found in clause 6.2.6 Memory requirements.

- This helps the decoding since when an ADT has been decoded and some of its columns been corrected, the ADST is automatically "refreshed" because the pointer points now to a correct column. More decoder implementation discussion using this concept can be found in clause A.3.4.2.

Padding:

One ADT columns may not necessarily host data, but also padding bytes and padding columns. Due to the construction of the ADST function that positions padding columns at the end of the ADST, these padding columns always occur at the end of the "sub-blocks" and therefore padding columns are inserted inside the ADT at different positions, but certainly not at the end of the ADT as it is usually the case in MPE-FEC encoding. Assuming in our example that the 4 last columns of each ADST are concerned by padding (the first one is a data column with some data at the top of the column, whereas the 3 last ADST columns are complete padding columns), the ADT will have the following aspect whereby the shaded area represents padding.

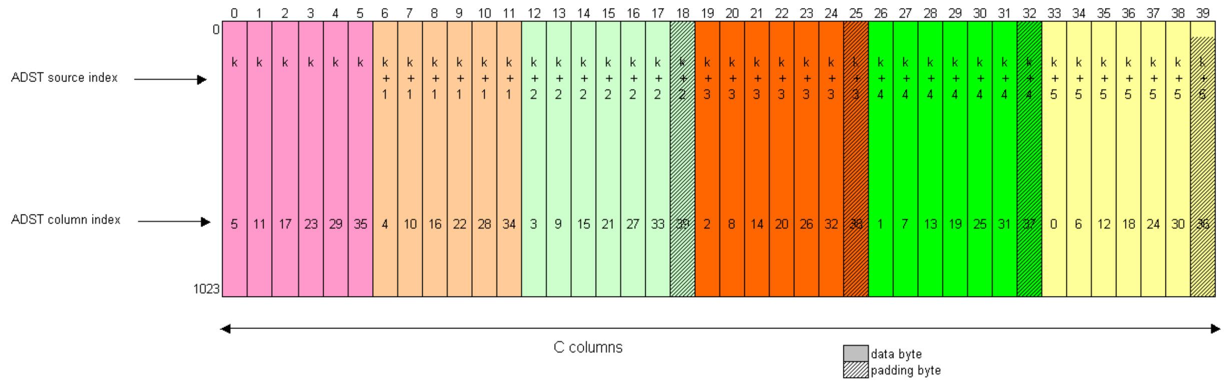


Figure 6.8: ADT aspect with interleaved padding columns

The generation of the Reed Solomon operates on a matrix of 191 data columns, so there is an additional padding of 191-C columns so that resulting ADT is as shown in figure 6.9.

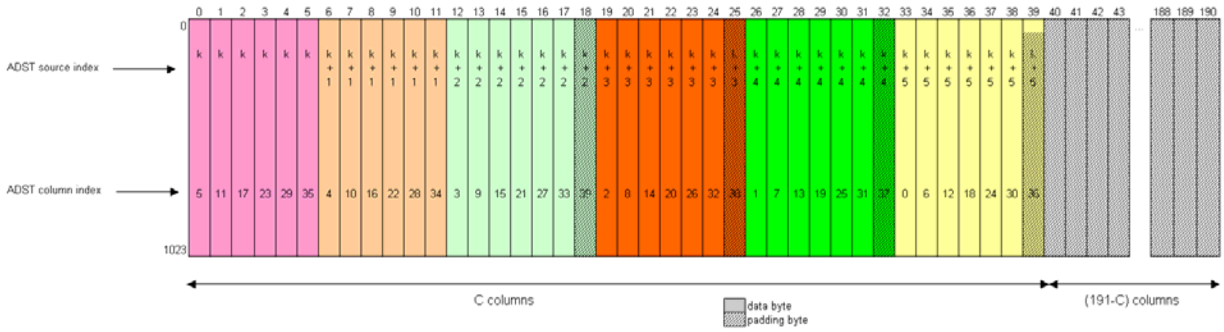


Figure 6.9: Final ADT aspect

The padding is equivalent to a code shortening as defined in EN 301 192 [9], clause 9.3.3.1. Code shortening is required to obtained the signalled ADT size *K*.

6.2.3.8 FDT generation and code rate computation

As soon as the ADT has been constituted, the FEC encoding function is applied to generate the iFDT using the Reed-Solomon code (255, 191). The FEC columns can be as high as 64 but usually only a fraction of these FEC columns is actually transmitted over the air (the first ones), which is equivalent to code puncturing as defined in EN 301 192 [9], clause 9.3.3.2. Puncturing is required to obtain the signalled iFDT size *N*.

With *EP*=1, for each processed datagram burst, exactly one encoding matrix is processed, i.e. the iFDT is generated.

Definition: the actual code rate is computed on a per encoding matrix basis based on the structure of the ADT and the iFDT. So for each encoding matrix m, we have:

$$code\_rate_{actual}(m) = \frac{nof\_data\_columns(m)}{(nof\_data\_columns(m) + nof\_fec\_columns(m))}$$

The actual code rate is usually taken as near as possible to a code rate target, and expected to be the same for all encoding matrices: $\forall m \in [0; M-1], code\_rate_{actual}(m) \approx code\_rate_{target}$. This target code rate is used to compute the number of FEC columns within each iFDT:

$nof\_fec\_columns = ceil\left(nof\_data\_columns * \frac{1 - code\_rate_{target}}{code\_rate_{target}} \right)$.There are at least two ways for computing this number of columns:

Method 1: a fixed number of FEC columns is transmitted in each IFEC slice burst by assuming, for all m, nof_data_column(m)=C' where C'≤C and C' is fixed. Whatever the real completion of the ADT, the transmitted number of RS columns will remain the same. In CBR situation (nof_data_columns~C), this will generate a $code\_rate_{actual}$ almost equal to $code\_rate_{target}$. But in case of VBR, it will make the code rate variable, possibly by large means, depending on variations of nof_data_columns datagram burst by datagram burst and encoding matrix by encoding matrix: when nof_data_columns is equal to C, $code\_rate_{actual}$ is almost equal to the target value $code\_rate_{target}$, but when nof_data_colums is small compared to C, then $code\_rate_{actual}$ may become quite low compared to $code\_rate_{target}$ (nearing 0).

Method 2: another way is to compute the nof_fec_columns based on the actual nof_data_column(m) and target code rate. This ensures a quasi constant $code\_rate_{actual}$, whatever the actual nof_data_columns. The precision of the difference between actual and target is a function of the nof_data_column and is better when this latter increases.

These two options for the selection of the RS columns and their impact on $code\_rate_{actual}$ are presented in figure 6.10.

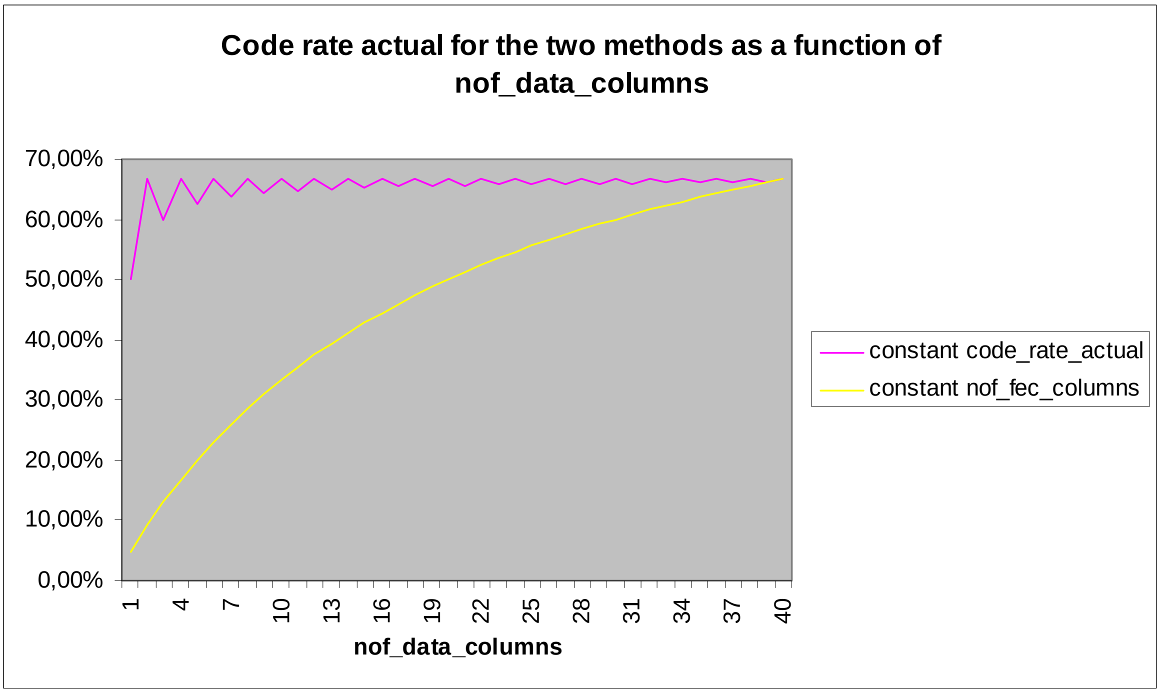


Figure 6.10: Different strategies for RS columns generation

In general for correct support of VBR and statistical multiplexing, the second option is recommended. The first options is recommended when a pure CBR traffic is protected. In our typical case, we assume a $code\_rate_{target}$ of 2/3. Based on the first option, we will assume that the number of FEC columns is equal to $ceil\left(37 * \frac{1-2/3}{2/3}\right) = 19$.

6.2.3.9 IFEC burst generation

Once the RS columns are generated inside one FDT, these columns are stored and interleaved before being inserted inside a time slice MPE-IFEC burst. This is where the S parameter is used: the MPE-IFEC sections inside one IFEC burst are coming from S successive FDT and the index position of one section inside the IFEC burst enables to derive the original FDT. This spreading effect is shown in figure 6.11. It can be seen that, in every IFEC burst i+4, there are:

- 5 IFEC sections coming from FDT i+3.
- 5 IFEC sections coming from FDT i+2.
- 5 IFEC sections coming from FDT i+1.
- 4 IFEC sections coming from FDT i.

More generally, given the MPE-IFEC time slice burst number k', $k' \in [0;k_{max}-1]$, the IFEC section index i, $i \in [0;18]$, the functions `ifd_index` and `ifdt_column` enable to derive the original iFDT index and iFDT column:

ifdt_index(k',j)=(k'-j[S]+M)[M]

ifdt_column(k',j)=j

If nof_fec_columns(i) gives the number of FEC columns present in FDT i, the sending will be as follows:

- in the nof_fec_columns(i)[S] first IFEC bursts, we send floor(nof_fec_columns(i)/S)+1 IFEC sections;
- in the remaining S-nof_fec_columns(i)[S] IFEC bursts, we send floor(nof_fec_columns(i)/S) IFEC sections.

One representation of IFEC burst i+4 is presented in figure 6.11.

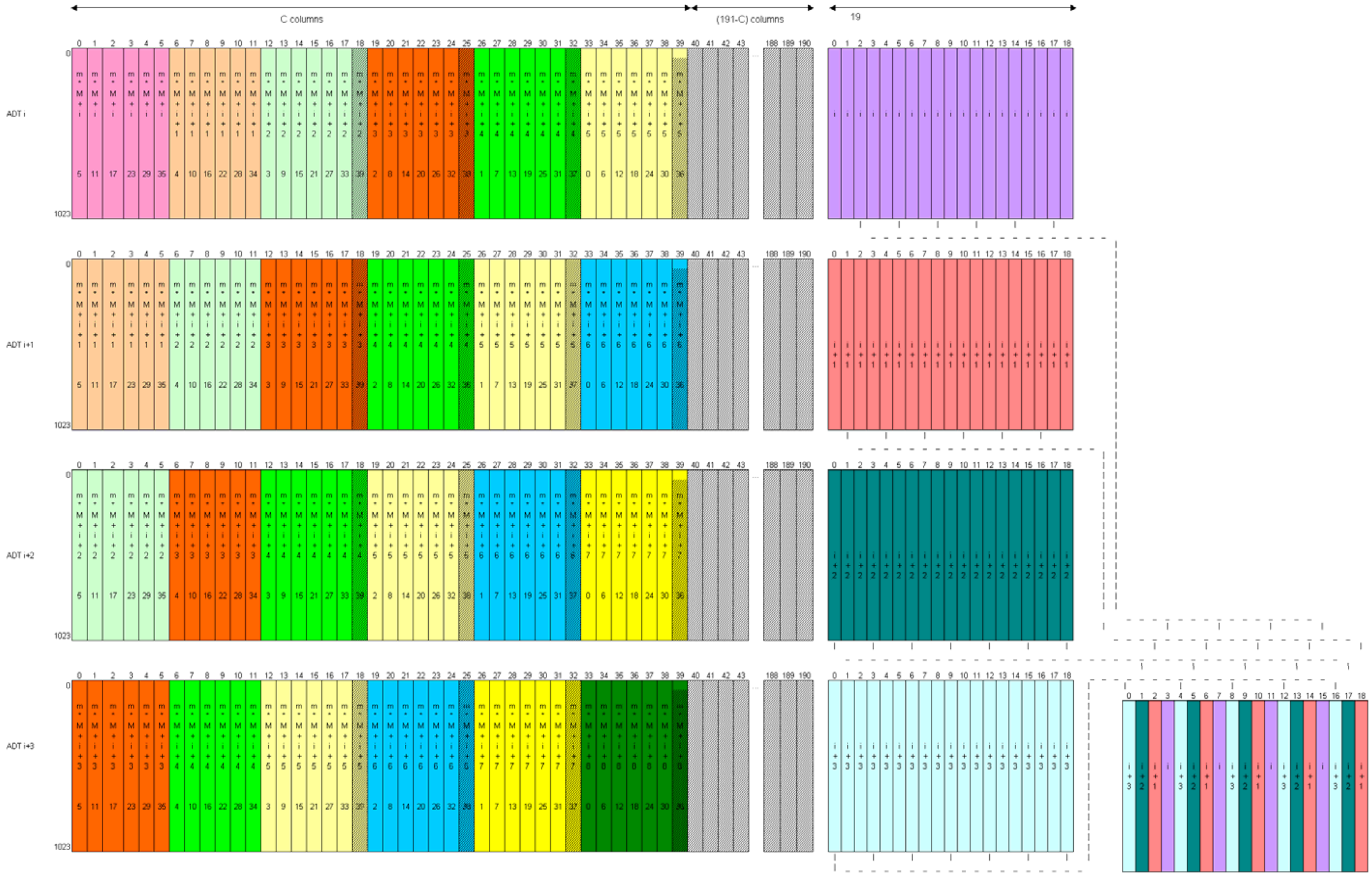


Figure 6.11: Sending arrangement of IFEC burst i+4

Please note that the section index may not be consecutive and there could be index discontinuities when an iFDT has "run out of available columns". Such situations may occur in VBR situations with a fixed code rate: one ADT i has fewer data columns than the following ADT i+1 so that their corresponding iFDT have also different FEC columns available for transmission. Take the current example where the number of column is 19 on all iFDT except iFDT i where only 18 columns are available. The IFEC bursts composition are represented hereafter.

Table 6.4: Source of MPE-IFEC sections in different IFEC bursts

Source iFDT IFEC burst	i-3 (19)	i-2 (19)	i-1 (19)	i (18)	i+1 (19)	i+2 (19)	i+3 (19)	i+4 (19)
i+1	4	5	5	5	-	-	-	-
i+2	-	4	5	5	5	-	-	-
i+3	-	-	4	5	5	5	-	-
i+4	-	-	-	3	5	5	5	-
i+5	-	-	-	-	4	5	5	5
…								

The MPE-IFEC time slice burst i+4 will have only 18 IFEC sections with the following indices.

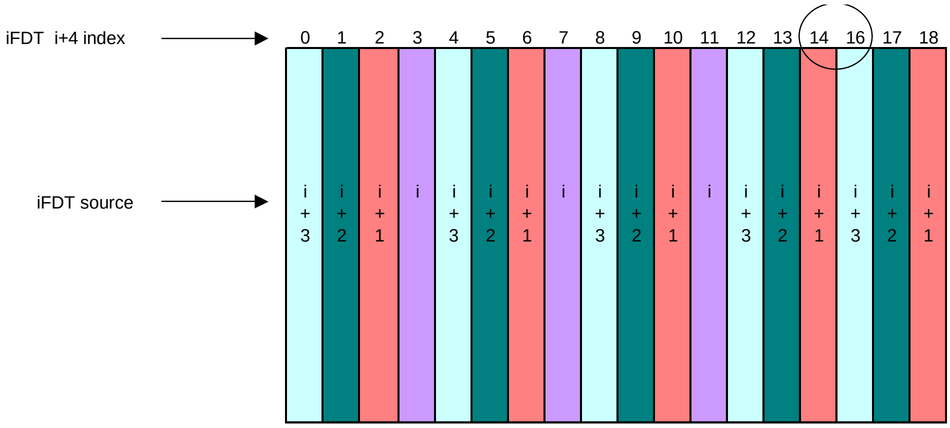


Figure 6.12: IFEC burst I+4 in case of section discontinuity

It is also to be noted that the interleaving over the S IFEC bursts will smooth IFEC burst variations. More scenarios on usage of the section indices are discussed in clause 6.2.7MPE-IFEC usage scenarios.

6.2.3.10 MPE-IFEC section header

In addition to the static configuration of the time_slice_fec_identifier, real-time signalling is conveyed in the MPE-IFEC headers. Figure 6.13 provides an overview on the syntax of the MPE-IFEC section and its header.

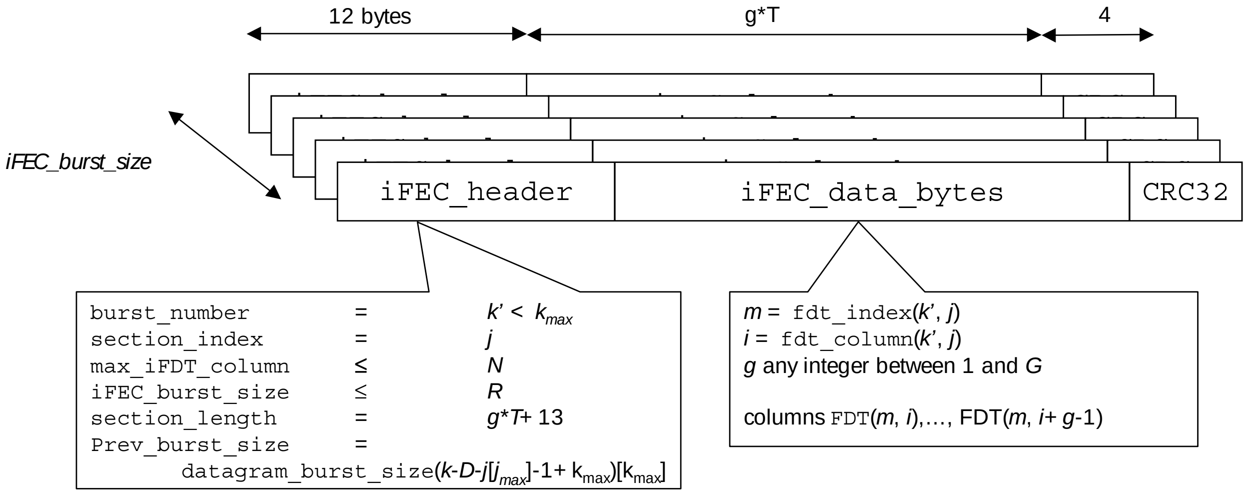


Figure 6.13: MPE-IFEC header and payload structure

For our particular example, the indices have the following information:

- `burst_number` = k', where $k' \in [0;249]$.
- `section_index` = j, where $j' \in [0;18]$.
- The iFDT index is obtained as m = `ifdt_index`(k', j).
- IFEC_data_bytes are obtained from `iFDT`(m) and correspond to iFDT columns `ifdt_column`(k', j).
- `max_iFDT_column` = floor(255**max_iFDT_column*(m)/N)=floor(255*19/64)=75.
- `IFEC_burst_size` = 19 (total number of IFEC sections included in this IFEC burst).
- `section_length` = 1 037.
- `prev_burst_size` = *datagram_burst_size*((k-D-j[j_{max}]-1+k_{max})[k_{max}]).

Usage of these fields is multiple and several examples are provided in clause 6.2.7 MPE-IFEC usage scenarios.

Real-time parameters according to table 6.5 are inserted. These real-time parameters usage are detailed in clause 6.2.3.11 Burst sending arrangement as they relate to the sending arrangement within an MPE-IFEC time slice burst.

Table 6.5: MPE-IFEC real-time parameters

Syntax	Number of bits	Identifier
real_time_parameters () {		
delta_t	12	uimsbf
MPE_boundary	1	bslbf
frame_boundary	1	bslbf
prev_burst_size	18	uimsbf
}		

6.2.3.11 Burst sending arrangement

Considerations on the burst, Time-slicing and Delta-t parameter

In the MPE-IFEC specifications [MPE-IFEC], there is a reference to an (MPE-IFEC) time slice burst ([MPE-IFEC], clause 2.3.7). This MPE-IFEC time slice burst has the same definition as the burst in DVB-H, as specified in EN 301 192 [9], clause 9.1: *"a burst is a set of sections delivered on an elementary stream. Between two consecutive bursts there is a period of time when no sections are transmitted on the particular elementary stream. Each burst indicates the start time of the next burst within the elementary stream".* This definition of burst includes potentially sections of different kinds, like MPE, MPE-FEC and MPE-IFEC, so that the burst is by nature an open concept. In [MPE-IFEC], the IFEC time-slice burst is therefore used to designate all the sections, including MPE, MPE-FEC and MPE-IFEC, that age timely grouped. For simplification and clarity reasons, we will in the following refer to the IFEC time-slice burst by the simple term *burst*, not to confuse it with the DVB-H time-slice burst that is referred by the term time-slice burst. Another kind of burst has been defined in MPE-IFEC specifications [MPE-IFEC] called the IFEC burst. This includes the only MPE-IFEC sections. The terminology is presented in table 6.6.

Table 6.6: Terminology

Name in the present document	Concerned section kinds	Name in DVB-H	Name in IFEC
Burst	All	burst	Burst or MPE-IFEC time slice burst
Time slice burst	MPE and MPE-FEC	Time slice burst	N/A
IFEC burst or MPE-IFEC burst	MPE-IFEC	N/A	IFEC burst or MPE-IFEC burst

Contrarily to the burst, the definition of Time-slicing given in the same document (*"Time Slicing: Method to deliver MPE sections and MPE-FEC sections in bursts"*) refers to the only MPE and MPE-FEC sections, excluding any additional types of sections (like MPE-IFEC). Additionally, the definition of Delta-t given in clause 9.10 of EN 301 192 [9] makes a clear reference to a *time-slice burst* (*"The field indicates the time (Delta-t) to the next Time Slice burst within the elementary stream"*).

Note that Delta-t information provided by MPE-IFEC sections **must** be exactly the same as Delta-t information provided by MPE and MPE-FEC sections since, otherwise, the Delta-t provided by different sections within same burst-MPE/MPE-FEC on one side, MPE-IFEC on the other side could point to different positions within the next burst and hence create inconsistent signalling. This is why the definition of the Delta-t field in MPE-IFEC sections is exactly the same as the definition of the Delta-t field in MPE and MPE-FEC sections: *"Delta-t is a time offset indicating the time from the start of the transport packet carrying the first byte of the current IFEC section to the start of the transport packet carrying the first byte of next* ***time-slice burst****"*.

Therefore, the burst (union of the time-slice burst and the IFEC burst) and the time-slice burst **may** not coincide in time due to the exclusion of IFEC sections from time-slice burst definition while, at the same time, their Delta-t signalling points to the same temporal structure, the time-slice burst.

Some examples are given below to clarify, in the case **when no MPE-FEC is used in the elementary stream** (please note that simultaneous used of MPE-FEC and MPE-IFEC is not allowed and is for further study).

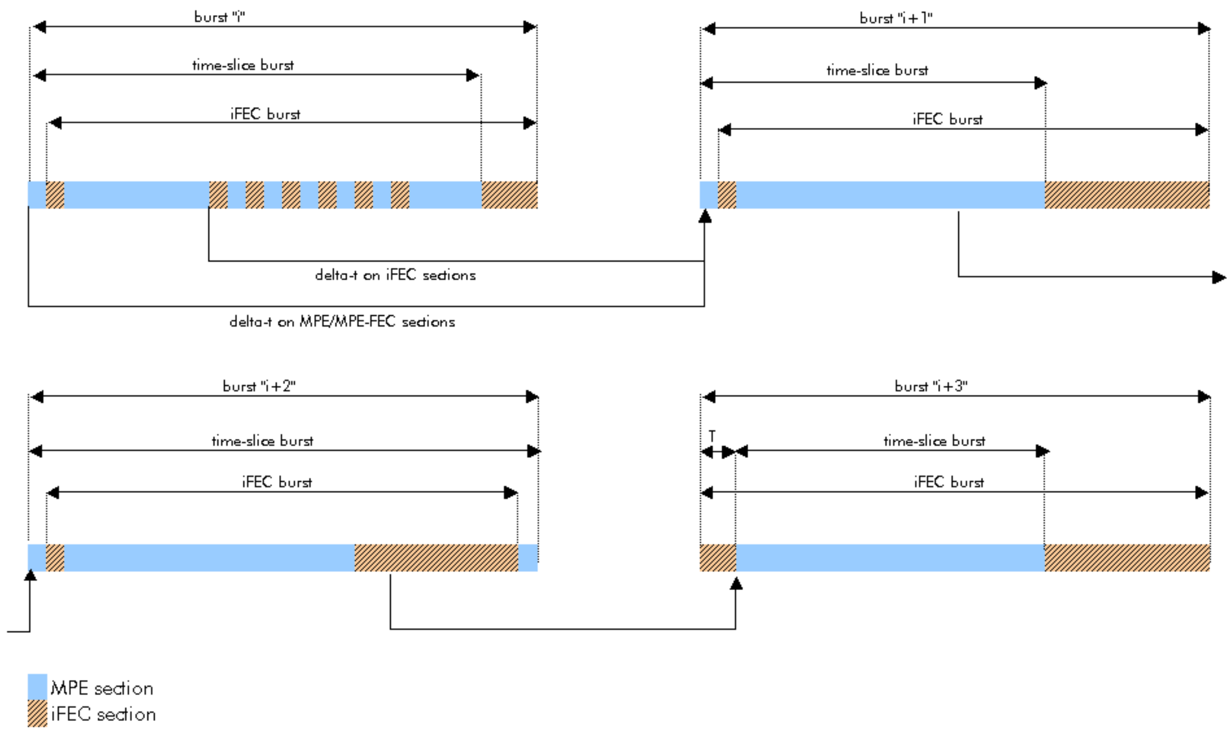


Figure 6.14: Burst sending arrangement (w/o MPE-FEC)

In figure 6.14, we can see three successive bursts of the same elementary stream (bursts i to i+2):

- Delta-t signalling throughout the same burst always points to the first MPE section of the following burst, and not the first MPE-IFEC section, because Delta-t has the same definition, whatever the section, and always refers to the time-slice burst;
- as a consequence of this signalling, if the burst begins with MPE-IFEC sections, they may be missed, unless the delay T between their sending and the actual sending of the time-slice burst is small compared to the Delta-t unit (10 ms).
- the time-slice burst and the burst do not necessarily coincide in time. Many configurations are possible and, in this example, the only burst i+3 sees the matching between both.

Recommendations:

Since the burst can differ from the time-slice burst but the Delta-t signalling is in any case related to the time-slice burst, the following recommendations are given **for cases when no MPE-FEC decoding applies**:

- At least one MPE section **must** be positioned in the vicinity of each burst beginning so that no MPE-IFEC sections are missed. The vicinity is considered such that the delay between the burst and the time-slice burst beginnings is below 1/10th of the Delta-t unit so 1 ms. Depending on the transmission bit rate, this time may allow to send an MPE-IFEC section or not. For instance with a bit rate of 2,4 Mbits, the time it takes to send 1 TS is 0,623 ms which allows sending 1 full MPE-IFEC section in the only case of 256 rows. For simplification, the first section **should** therefore be an MPE section.
- Since the only MPE-IFEC section conveys burst numbering information, at least one MPE-IFEC section **should** be positioned close to the beginning of the burst, such that the receiver may detect the burst number as early as possible while receiving the burst, and position immediately the received MPE sections in their correct ADT, without waiting for reception of late MPE-IFEC sections. Note that this recommendation is not a hard one since it is always possible to receive MPE sections without previously knowing their burst number: these sections are stored until their burst number can be determined by the receiver, by receiving a later MPE-IFEC section conveying this burst number or other means, for instance based on timing.

- After the beginning of the burst, the MPE and MPE-IFEC **may** be freely interleaved during a time interval not exceeding max_burst_duration signalled in time_slice_fec_identifier. After max_burst_duration, no more sections (MPE, MPE-IFEC) **must** be present. The receiver **must** ignore these sections received lately and can switch off the receiver, hence enabling power saving.

- To detect the end of the burst, the receiver **must** check that the end of the time-slice burst and IFEC burst are reached. For that purpose, the receiver checks the received sections flags and looks whether one of the conditions presented in table 6.7 are respected for all lines. These conditions are the results of time-slice burst sending arrangement rules found in EN 301 192 [9], clause 9.10 (*"All sections carrying Application data datagrams of a given MPE-FEC Frame shall be transmitted prior to the first section carrying RS-data of the MPE-FEC Frame (i.e. sections carrying Application data datagrams shall not be interleaved with sections carrying RS-data within a single MPE-FEC frame).[…] Within an elementary stream, sections delivering data of different MPE-FEC Frames shall not be interleaved. […] Note that for each MPE-FEC Frame, MPE sections are delivered before MPE-FEC sections"*) and MPE-IFEC time-slice burst generation and sending arrangement rules found in MPE-IFEC specifications [MPE-IFEC], clause 3.6.

Table 6.7: Conditions for end of IFEC burst detection

<table>
<tr><th></th><th></th><th>Condition 1</th><th>Condition 2</th><th>Condition 3</th><th>Condition 4</th><th>Condition 5</th></tr>
<tr><td rowspan="2">Time-slice burst</td><td>MPE</td><td>An MPE section with a table_boundary set to 1 has been received.</td><td>An MPE-FEC section has been received.</td><td>An MPE-IFEC section has been received with an MPE_boundary set to 1.</td><td rowspan="3">Current time has exceeded previous Delta-t plus max_burst_duration.</td><td rowspan="3">Another service PID has been found, signalling change in the service reception.</td></tr>
<tr><td>MPE-FEC (this is for information only since MPE-FEC cannot be present at the same time as MPE-IFEC in the ES)</td><td>An MPE or MPE-FEC section has been received with the frame_boundary set to 1.</td><td>-</td><td>An MPE-IFEC section has been received with an MPE_boundary set to 1.</td></tr>
<tr><td>IFEC burst</td><td>MPE-IFEC</td><td>An MPE-IFEC section with frame_boundary set to 1 has been received.</td><td>-</td><td>-</td></tr>
</table>

This table can be used as follows on a particular case where the IFEC burst is made of a first MPE section, then one IFEC section, then all remaining MPE sections, then all remaining IFEC sections.

- In case no section at all is lost:
  - Time-slice burst end will be known with the last MPE section (condition 1).
  - IFEC burst end will be known with the last IFEC section (condition 1)
- In case some MPE sections, including the last, and some IFEC section, but not all, are lost:
  - Time-slice burst end will be known with the first MPE-IFEC section received after the last MPE section (condition 3).
  - IFEC burst end will be known with the last IFEC section (condition 1).
- In case some MPE-IFEC sections, including the last, are lost:
  - Time-slice burst end will be known with the last MPE section (condition 1).
  - IFEC burst end will be known with the time out (condition 4).

- Note that one consequence of these rules is the need for the receiver to potentially stay on after last MPE section has been received (to receive MPE-IFEC sections if any) or after last MPE-IFEC section has been received (to receive MPE and MPE-FEC sections if any). This specific behaviour must be applied on a per stream basis: for those streams where no IFEC protection is applied, the same rules as DVB-H will apply (when the last MPE or MPE-FEC frame has been received, the receiver will not have to wait for other MPE-IFEC sections and so will be able to switch off immediately).

6.2.4 Parameters selection

6.2.4.1 Introduction

This clause provides explanations and recommendations for MPE-IFEC parameter selection. To configure the MPE-IFEC, the operator needs to fix the 3 main parameters of the MPE-IFEC framework, namely B, S, D and the code rate: D is selected based on heuristics and the impact on zapping time, B and S are selected based on latency (B+S) and the code rate is selected on performance criteria. All performance results are measures at interface R1 shown on with a D equal either to 0 or B+S.

The reader will see how these parameters can be easily established depending on the sought performance and the channel behaviour. For that purpose:

- we first discuss the D parameter and its dependence with other B and S in clause 6.2.4.2 Recommendations on D; we show that D can be selected between only two typical values;
- we then discuss how B and S can be derived in clause 6.2.4.3 Recommendation on B and S from link layer latency and code rate;
- we finally show how link layer latency and code rate can be selected based on simulation abacuses in clause 6.2.4.4 Selection of M and code rate;
- we also give a possible global procedure in clause 6.2.4.5 Parameter overall selection logic.

We introduce the following latency definitions:

- "encoder latency/delay": time between reception, at the encoder side, of an IP packet and actual emission of the last (data or IFEC) section related to this IP packet; the data section is the one carrying the IP packet whereas the IFEC section has correction capability on the ADT column(s) where the IP packet is located; the encoder latency can be computed over a set of IP packets belonging to one burst; we then talk of "burst encoder latency":

$$\text{encoder\_latency}(\text{IP\_packet}) = \text{last}\begin{pmatrix} \text{MPE section sending time} \\ \text{MPE - IFEC section sending time} \end{pmatrix} - \text{IP\_packet arrival time}$$

$$\text{encoder\_latency}(\text{burst}) = \max(\text{encoder\_latency}(\text{IP\_packet}) \text{ for all IP\_packets belonging to the burst })$$

- "receiver latency/delay": this is the time between delivery of an IP packet at the decoder side and reception of the first information on this IP packet, whether it is the MPE data section carrying this IP packet or an IFEC section providing parity for an ADT column that includes data bytes for this IP packet. The receiver latency can be computed on a complete burst, we then talk on the burst receiver latency.

$$\text{receiver\_latency}(\text{IP\_packet}) = \text{IP\_packet delivery time} - \text{first}\begin{pmatrix} \text{MPE section arrival time} \\ \text{MPE - IFEC section related to an ADT column} \\ \text{that includes data bytes for this IP packet arrival time} \end{pmatrix}$$

$$\text{receiver\_latency}(\text{burst}) = \max(\text{receiver\_latency}(\text{IP\_packet}) \text{ for all IP\_packets included in the burst})$$

- "end-to-end latency/delay": time between delivery of an IP packet at the decoder side and reception of the IP packet at the encoder site. The end-to-end delay is equal to the encoder latency, the transmission delay and the receiver latency. So end-to-end delay varies only due to the receiver latency since encoder latency and transmission delays are considered as fixed. In steady state, the end-to-end delay is equal to its maximal value, but during non steady states (like zapping) this end-to-end delay may be shortened due to the possible reduction of receiver latency.

6.2.4.2 Recommendations on D

B,S and D are related through the function giving $M = B + \max(0, S - D) + \max(0, D - B)$. This M, in addition to giving the number of encoding matrices, gives the receiver latency and the memory requirements:

- Assuming the simplification that burst are regularly repeated with a temporal period of burst_repetition_interval, the receiver latency at the jitter free interface is equal to $\text{burst\_repetition\_interval} * M$. The jitter-free interface is the R1 interface when late decoding is used.
- Assuming that ADT and FDT sizes are given by ADT_size and FDT_size, the memory required for complete IFEC decoding is equal to $(\text{ADT\_size} + \text{FDT\_size}) * M$.

These formula show the importance of the parameter D and its impact on user perception (via the latency) and terminal sizing (via memory), all other things being equal. For instance for a B equal to 6 and a S equal to 4, we obtain the curve presented in figure 6.15.

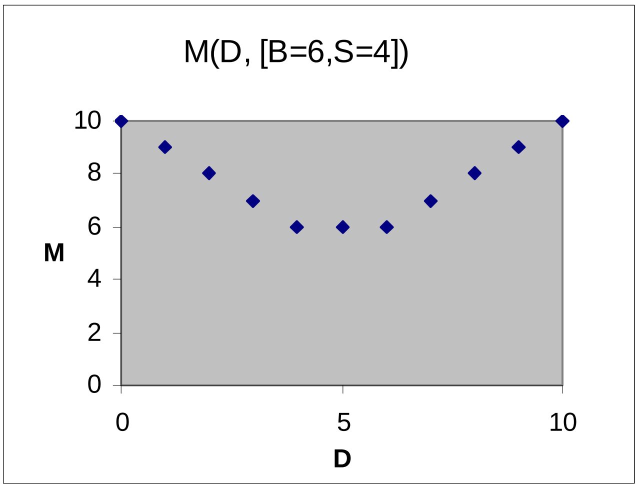


Figure 6.15: M(D, [B=6, S=4])

It can be seen that:

- when D is equal to 0, M is rather large and equal to B+S=10; this configuration corresponds to the situation when the MPE-IFEC sections are always sent after the data they protect, which is good for error correction performance. But the high M value implies long jitter-free receiver interface delay and large memory requirements;
- a minimum value of M is obtained for D taken in the range [4; 6]. Obviously this enables to reduce memory requirements in receiver and latency since the receiver needs to wait less for MPE-IFEC section reception at the jitter-free interface compared to when the sections are not sent with the data they protect. However the MPE-IFEC sections will be mixed with the datagram bursts, which reduces performance because an error affecting a burst likely affects at the same time, the data it conveys, and the FEC that corrects the errors. Increasing D will also increase end-to-end latency;
- after the value 6, M increases again. Data and MPE-IFEC are still mixed, up to the situation when D=B+S (10 in our case). In this case, the data and IFEC sections are again not mixed, which provides good performance results. The receiver jitter-free latency is the same as D=0 but since the IFEC always precedes the data, this enables "early decoding" during zapping (see clause 6.2.5.2 Zapping time performance).

The influence of the D parameter on different criteria is presented in table 6.8.

Table 6.8: Influence of the selection of D parameter on different criteria

D value	Performance	Receiver latency	Encoder latency	Receiver memory
D=0	+	-	+	0
D=D_{min_sizing}	-	0	0	+
D=B+S	+	+	-	0
NOTE 1: D_{min_sizing} is the lowest D that minimizes the M function. For instance, on our previous example, D_{min_sizing} = 4. NOTE 2: "+" means the criteria is matched with a good performance, 0 means a neutral performance and "-" means a worse performance. NOTE 3: Performance is the correction capability at the jitter free interface; receiver latency includes both jitter-free and non jitter-free (during zapping); all other criteria are classical. NOTE 4: D=B+S provides a good compromise in performance, including receiver latency during zapping, at the detriment of encoder latency. This configuration should be used for contents not delay constrained whereas D=0 should be kept for delay constrained contents, at the detriment of receiver latency and zapping time.				

6.2.4.3 Recommendation on B and S

The code rate is defined on a per encoding matrix basis by the formula combining, for a given encoding matrix m, the number of ADT data columns (adt_data_columns(m)) and the number of FDT FEC columns (nof_fdt_fec_columns(m)). The target code rate is defined in clause 6.2.3.8 FDT generation and code rate computation. We also assume, as in this clause, that the same target code rate applies to all encoding matrices.

Code_rate can be selected with complete freedom within the range authorized by C (using padding) and R (using puncturing). If we take our current example, C is fixed to 40 but the number of data columns is on average 37. Depending on the number of FEC columns (from 0 to 64), 64 code rates from 1 to 0,36 are possible as presented in clause 6.2.3.8 FDT generation and code rate computation.

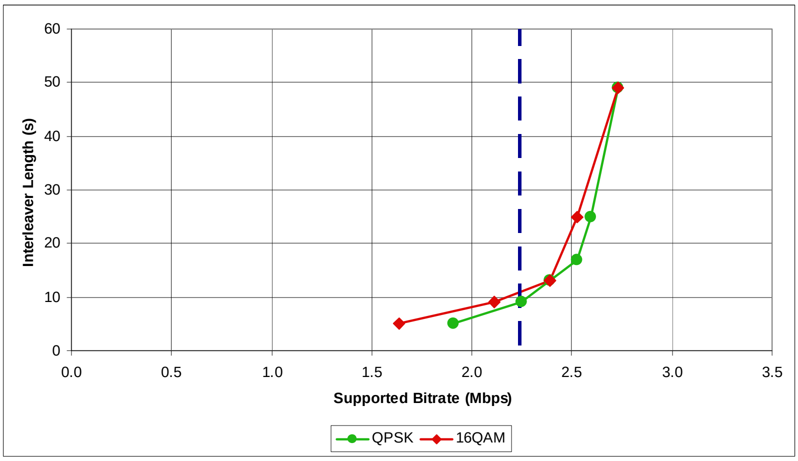
However, an exhaustive simulation campaign determined that the configuration offering the best performance $(B,S,code\_rate)_{actual}$ follows the rule $\left[\frac{B}{B+S}\right]_{actual} \approx code\_rate_{actual}$. Based on this assumption, near optimal parameters B_{opt} and S_{opt} can be derived from target FEC rate and B+S for cases where D=0 by using the following formulas:

$$S_{opt} = ceil((1 - code\_rate) * (B_{actual} + S_{actual}))$$
$$B_{opt} = (B+S) - ceil((1 - code\_rate) * (B_{actual} + S_{actual}))$$

Since B and S can be easily derived from the couple (B+S), which is equal to M when D=0 and noted M(0,B,S), and the code rate, we will use these two parameters in the following.

6.2.4.4 Selection of M and code rate

Simulation campaigns have enabled to derive the rule between M(0,B,S) and the code rate for a given ESR5(20) fulfilment, on a per channel basis. These complete results are presented in the clause A.12. We take the example of a suburban channel with figure 6.16.



NOTE: M(0,B,D)=B+S for D=0 and D=B+S; the curve are valid for these two values of D.

Figure 6.16: SH-A, class 1 - QPSK 1/2 and 16QAM 1/4 - LMS-SU - 50 kmph - 63 dBW EIRP Satellite

Figure 6.16 can be interpreted as follows:

- all points on the curve match an ESR5(20) fulfilment of 90 %;
- all points above have a better quality;
- all points below have a lower quality.

The curve is then an iso-ESR5(20) one.

The iso-ESR5(20) curve has always the same shape with a slope that is increasing as code rate increases, an inflexion point and then an asymptotic point. In the example given, the following points can be found:

- first value is around 1,7 Mbps/5 s (interleaver length directly related to M);
- the inflexion point is at 2,55 Mbps/25 s;
- the asymptotic value is at 2,75 Mbps/50 s.

Two different approaches can be followed for determining which parameters must be selected for a particular system:

- the terminal is memory constrained (compared to the asymptotic value) to a certain M(0,B,S), for instance 25 s: the selected code rate is the code rate found at the intersection of the iso-ESR5(20) curve and the horizontal line at the target M(0,B,S); in our case the resulting useful bit rate is 2,5 Mbps, which turns into a code rate of 2/3 because the total bit rate in 5 MHz QPSK1/2 GI=1/4 is 3,357 Mbps. These bit rates are measured at the R2 interface at MPEG2 TS layer;

- the terminal is not memory constrained (compared to the asymptotic value), the objective is to maximize the capacity and so position at the inflexion point of the curve. In our case, the target capacity could be 2,7 Mbps (code rate 4/5); the M(0,B,S) is found at the intersection of the curve and the vertical line at target code rate. In our example the value is 50 s.

A mobile system is also designed to work in a set of different propagation channels. The selection of the link layer parameters must be done for the worst case, typically LMS-ITS. Once selected, the configuration should perform well under less challenging channels. We give the complete example below.

We start with an LMS-ITS dump excerpted from clause A.12.

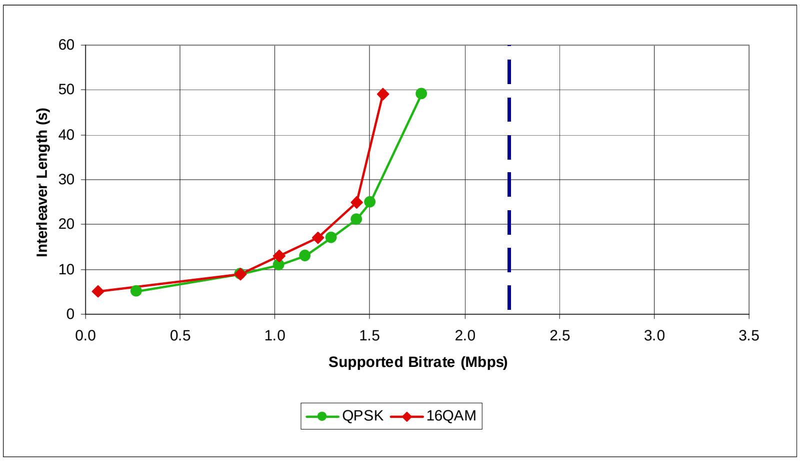


Figure 6.17: SH-A, class 1: - 16QAM 1/4 and QPSK 1/2 - LMS-ITS - 50 kmph - 63 dBW EIRP Satellite

If we are with a M(0,B,S) equal to 25, this gives a code rate of 0,44. We can see immediately on this figure that the performance will be much higher than the target in LMS-SUB (we are above the curve). Finally, since we are in the hybrid frequency in that case, we can also check the performance under TU6 channel: in clause 6.2.5.1.4 TU6 for hybrid frequency we show that a configuration for LMS-ITS or LMS-SU performs perfectly under TU6 channels.

6.2.4.5 Parameter overall selection logic

So the general parameter selection logic is presented in figure 6.18.

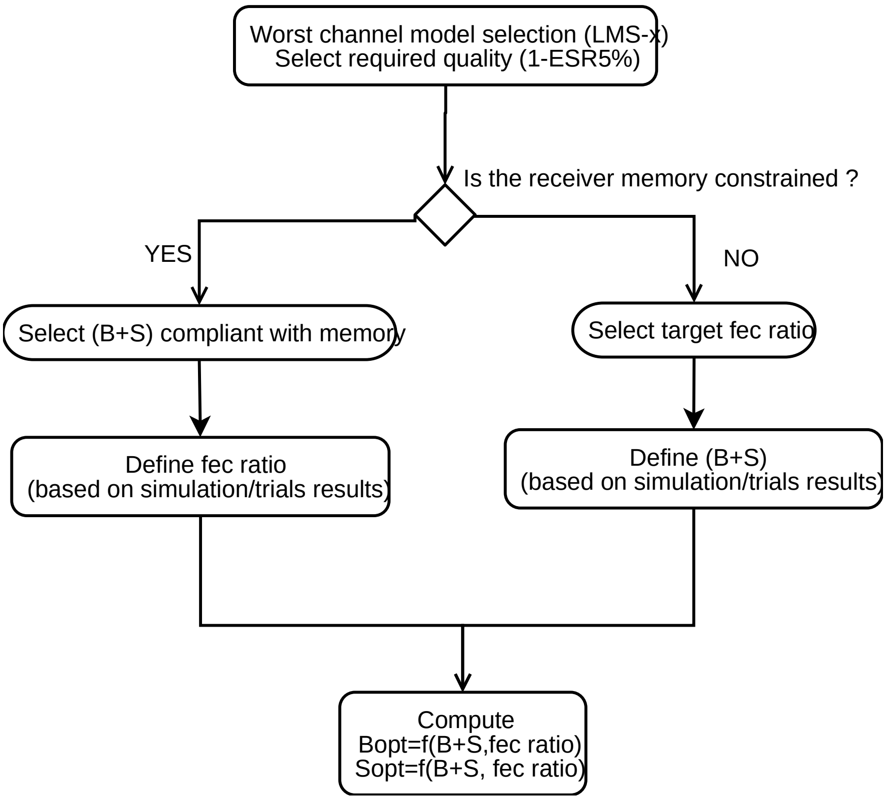


Figure 6.18: Link layer parameter selection logic

This selection logic is valid for D values such that MPE-IFEC sections are not transported in the MPE-IFEC same time-slice burst as the MPE they are protecting, so for D=0 and D≥B+S. For intermediate values, the performance will be slightly less requiring higher B+S and/or lower code rate and is for further study.

This procedure is used to recommend parameters in clause 6.2.5.1.3 Recommended class 1 configuration.

6.2.4.6 Recommendations on signalling parameters

The following recommendations are made on the way to configure the signalling fields:

- max_burst_duration shall not exceed the repetition_interval to avoid confusing the receiver reception.

6.2.5 Simulated performance

Link layer simulation configuration elements can be found in clause A.6. All detailed simulations results can be found in clause A.12. We give in this clause a summary of the clause A.12 results.

6.2.5.1 ESR5(20) performance

This clause excerpts some results from clause A.12. The logic for these results is the following:

- targeted quality:
  - for LMS channels, ESR5(20) fulfilment of 90 % is sought;
  - for terrestrial (TU6) channels, ESR5(20) fulfilment of 99 %, equivalent to a FER of 1 % is sought;
- memory constraints:
  - depending on terminal category (handheld or vehicular), the memory constraints are different; vehicular terminal can be considered to have "unlimited memory" whereas handheld receiver can be considered to have memory limitations;
- code rate:
  - the target code rate is the one that enables to support similar spectral efficiency as the class 2 receivers (full protection at physical layer); for instance, a QPSK1/2 and 16QAM1/4 will have a target link layer code rate of 2/3;
  - if the quality criteria for a maximum B+S value cannot be fulfilled, then the code rate is reduced until the quality is achieved.

The objective of the simulation procedure is to find the ((B+S);code rate) combination that enables to meet expected quality of service under the double constraint of (B+S) stays below maximum memory and code rate does not drop below 30 % of capacity loss. This exercise is performed for different channel environments and receiver categories.

6.2.5.1.1 LMS-ITS 50 kmph

Please refer to clause A.12 for details. There is always an asymptotic value for the interleaving duration, around 25 s to 30 s: increasing the duration beyond this value does not bring any help. The recommended configuration for the link layer is therefore given in table 6.9.

Table 6.9: Recommended link layer configuration in ITS (50 kmph)

Hybrid Frequency modulation	C/N used for simulation	Recommended B+S	Recommended Code rate
SH-A (QPSK1/2 and 16QAM1/4 16QAM2/7 GI=1/4)	11,2 dB	25	1/2
SH-B (TDM 8PSK1/3, QPSK1/2)	12,3 dB	25	2/3

Additional configurations are for further study, in particular sensitivity of this results as a function of C/N.

6.2.5.1.2 LMS-SU

Please refer to clause A.12 for details. An interleaver of 25 s gives the asymptotic performance. Minimum recommended configuration is given in table 6.10.

Table 6.10: Minimum configuration in SU (50 kmph)

Hybrid Frequency modulation	C/N	B+S	Code rate
All	> 8	25	¾
SH-A (QPSK1/2, 16QAM ¼)	< 8	25	2/3

6.2.5.1.3 Recommended class 1 configuration

Based on previous results and procedure explained in clause 6.2.4 Parameters selection, the following class 1 configurations are proposed.

Table 6.11: Recommended class 1 configuration

Hybrid Frequency modulation	Threshold C/N
SH-A	12
SH-B	11

For C/N below the threshold, the following configuration shall be chosen B+S=25 and code rate = ½ whereas for C/N above the threshold, B+S=25 and code rate = 2/3 can be chosen.

For services which require lower end-to-end latency, lower values of B+S may be used. The appropriate configuration can be derived according to the above procedure (in clause 6.2.4 Parameters selection) and using simulation results in clause A.12.

The parameters can be freely chosen provided the required memory, as computed in clause 6.2.6 Memory requirements, does not exceed the class 1 memory capability as given in clause 10.

6.2.5.1.4 TU6 for hybrid frequency

This clause refers only to the SFN case: having selected a class 1 configuration, we know its performance under LMS-ITS and LMS-SU channels. We need now to check also the performance over the terrestrial TU6 channel since the *same* MPE-IFEC configuration is also used over the terrestrial repeaters because of the SFN nature of the network: will the user experience different quality while performing a hand over between the hybrid and non-hybrid channels under the same CGC network (the hybrid frequencies do have an IFEC protection, which is not supposed to be the case of a non-hybrid frequency)?

To evaluate the performance difference, we assume that CGC is working at C/N that enables ESR5(20) fulfilment of 99 % without MPE-IFEC and we need to check at this C/N if the hybrid signal, having additional MPE-IFEC protection but lower physical layer code rate, will have lower, same or better quality than non MPE-IFEC protected ones.

As an example, excerpted from clause A.12, we use a terrestrial configuration in QPSK1/3 that has an error-free C/N_{NHF} at 3 kmph of 3,5 dB and 1,5 dB at 50 kmph. We now check the performance of the hybrid frequency (still at 3 kmph/50 kmph respectively) but with lower physical code rate (1/2) and the selected link layer configuration (e.g. 2/3), for a 25 s interleaver.

It can be checked in clause A.12 that the MPE-IFEC hybrid frequency will need 1 dB more than equivalent physical layer configuration at 3 kmph (the required C/N is 4,5 dB) and 2 dB at 50 kmph. So the channel in hybrid frequency will require a slightly higher C/N than the equivalent one of the non hybrid frequency to ensure same quality as the non hybrid one. The MPE-IFEC compensate partly the higher physical code rate.

Note also that, due to the SFN between satellite and CGC, this added C/N can be compensated in areas where the CGC coverage starts to be less efficiently received. Such SFN gain is introduced in clause 11.

6.2.5.1.5 TU6 for non-hybrid frequency

We check the interest of the MPE-IFEC for non-hybrid channels (content that is not broadcasted over the satellite, only sent over the terrestrial network). For this purpose, we display the useful bit rate as a function of the C/N in clause A.12 and see with MPE-IFEC how dBs can be traded for bandwidth.

This figure shows that a network operator, we can trade capacity with dB:

- in QPSK1/3:
    - 20 % capacity with 1 dB of C/N.
- in QPSK1/2:
    - 15 % capacity with 1 dB of C/N (for comparison, with MPE-FEC we would need 33 % for 1 dB gain);
    - 30 % of capacity with 2 dB.
- in 16QAM1/5:
    - 13 % capacity with 1 dB of C/N;
    - 25 % of capacity with 2 dB of C/N.
- in 16QAM1/4:
    - 25 % capacity with 1 dB of C/N.

Generally speaking, the short physical interleaver performs well in TU6 environment and MPE-IFEC enables to provide intermediate C/N operational points in between the values, in a similar way as, but better than, MPE-FEC.

6.2.5.2 Zapping time performance

6.2.5.2.1 Introduction

This clause deals with zapping performance for only class 1 terminals. For a more general zapping performance discussion, including class 2 please refer to clause 6.3.2 Zapping time impact. However, some definitions are given or recalled that are also applicable for both classes. After these definitions, we explain the concept of zapping in an MPE-IFEC context and then give a complete example. These are conceptual considerations, zapping simulation results is for further study.

6.2.5.2.2 Definitions

The definitions are either new ones or precision on already given definitions in the light of the newly defined.

Précised definitions:

Late decoding: refers to the techniques used for decoding with maximum protection. The receiver latency is always maximal, the IP packet being delivered to the video decoder only when the maximum protection is achieved (received). Late decoding time implies the end-to-end delay is always maximized whatever the situation because the receiver latency is maximized. Late decoding is explained for class 2 in clause 7.3.3.5.4 and for class 1 in clause A.6. Only class 2 can afford late decoding and fast zapping.

Early decoding: refers to the set of techniques used for decoding with less protection than the maximum one. The receiver latency may be reduced compared to the late decoding case, the IP packet being delivered only when the minimum parity has been received and not all the parity. This enables to lower end-to-end delay and accelerate IP packet delivery in case of good reception condition but this induces variable end-to-end delay when more parity is requested during fading events. Early decoding for class 2 is explained in clause 7.3.3.5.5 and for class 1 in clause A.6. Early decoding is requested for class 1 for supporting fast zapping.

New definitions:

Zapping instant: time when the user selects a new program.

Zapping time/delay: delay between the zapping instant and the time this program is actually received by the video decoder (we take into account only the link layer latency); by definition zapping time is mostly equal to the receiver latency. Zapping time can depend on many parameters like air interface configuration (physical and link layer interleaver: do we use a uniform/late profile, is D>0, etc.) but also on local reception condition (is reception quality good?) and receiver strategy (does the receiver supports fast zapping?). In lossless reception situation, zapping time can be as short as a burst reception for terminals applying fast zapping (class 1 with any D value but preferably > 0 and class 2 with a uniform late profile), or as long as encoder latency for terminals without any fast zapping support (class 1 with late decoding strategy or without the capability to recover the parity later-on and class 2 with uniform long profile). In bad reception condition, zapping time can exceed these values and last as long as the signal blockage.

Fast zapping: techniques used for reducing zapping time to acceptable delay even in the presence of long interleavers, be they at physical or link layer. They generally involve specific waveform configuration (uniform late for class 2, D preferably > 0 for class 1), terminal specific behaviour (late decoding for class 2 and early decoding plus parity recovery for class 1). Not taking into account other delays in the system (video decoding, etc.), the fast zapping time can be as low as the time to receive a single burst.

- In class 1 context, early decoding is usually used for fast zapping: under good reception condition, it is always possible to display content immediately since datagram bursts are transmitted as usual. The receiver can watch the video without waiting for late decoding latency, but may not have all the parity information to sustain a potential loss. Air interface configuration (use of D parameter) and parity recovery techniques can help the receiver to recover the parity progressively so that after a certain period, the receiver has recovered same full protection as in late decoding.

- In class 2 context, fast zapping refers to the interleaver uniform/late configuration in conjunction with late decoding that enables displaying first image immediately under good reception condition; parity is progressively recovered and after the encoder latency, full protection is ensured. This technique is described in clause 7.3.3.5.

Parity recovery: technique used by a receiver applying early decoding to progressively recover the same level of protection as late decoding one. For the link layer, different techniques are presented in clause 6.2.5.2 Zapping time performance. Note that parity recovery is not requested for the uniform/late physical interleaver since fast zapping is compatible with the late decoding as explained in clause 7.3.3.5.4.

6.2.5.2.3 Principles for zapping time analysis in class 1

The typical following scenario happens:

- the end user zaps by selecting one PID on a list, the PID being chosen via the ESG;
- the receiver decodes the MPEG2 TS and looks for corresponding PID;
- once it has found the corresponding PID, it starts receiving MPE and MPE-IFEC sections;
- with the included Delta-t, it can delineates the time-slice bursts and achieve power saving;
- it stores the received sections in their respective bursts, maps them on their respective ADTs and iFDTs so that the Encoding Matrices are progressively "populated";
- after a certain delay, the receiver can decode and deliver IP packets to the video player.

The zapping delay depends on this delay and different receiver behaviours are possible:

- late decoding:
  - the receiver will wait for all the parity before decoding;
  - the waiting delay will depend on the sending arrangement:
    - it will be equal to M when D=0;
    - it will lower to a minimum value when D equals D_{min_sizing};
    - for larger D, the delay will be kept to this value;
  - for instance in the example case (B=6, S=4), the initial delay is 10 when D=0 and 6 when D=10. Thanks to this delay, all the parity coming from the data part can be processed and, due to the fact that the FEC parity precedes the data, the FEC is also available;
  - however, with late decoding, fast zapping is then not possible (we need to wait for D_{min_sizing}): we need to resort to early decoding;
- early decoding:
  - the receiver does not wait for all parity before decoding; the receiver can start decoding before parity has been received, for instance as soon as it has received a burst. Obviously, this has a cost since the protection may not be enough to protect against normal impairments;
  - as bursts are received, if D is large enough, since FEC is sent before data, FEC will be accumulated to provide some protection in conjunction with previously received burst (for FEC computation, current burst is interleaved with B-1 previous bursts). However, full protection is not achievable since the protection of the currently received burst depends also on following B-1 bursts (the interleaving is a convolutional one);
  - for achieving normal protection, the receiver must perform a late decoding and, for this, has to wait for same delay as in late decoding, for instance 5 burst in addition of the current one;

- this additional waiting time can be "spread" over time thanks to recovery techniques and/or split with an initial small additional buffering. During this "spreading time", the end-to-end delay will not be constant and will increase from the initial value to the maximum one. So the interface is not jitter-free at the beginning. One technique for achieving this spreading is presented in [i.46]: the data from the link layer is delivered with a reduced rate to the IP layer. A media decoder may be informed to slow down the media playout. More precisely, after zapping, the data signal partially corrected by the MPE-IFEC with a slightly slowed rate than the transmitted rate is played out; the receiver corrects the signal by the MPE-IFEC, and, after a certain period, inversely proportional to the slowing rate, switches to the fully protected signal, at the transmitted rate. [i.47] suggests that 20 % of speed reduction is hardly perceivable by the user, allowing full protection and display at the transmission rate after 25 s, for a B equal to 6 (we need to recover B-1 bursts in addition of the initial one, so 5 bursts and, with a repetition interval of 1 burst per second, this leads to 5 s recovered in 5/0,2 = 25 s). Such techniques may require modification to existing media decoders, and especially audio ones;

- it is also important to mention what happens if the receiver detects that during the slow down, the FEC is not sufficient. Then an immediate rebuffering may be reasonable.

To summarize, correctly decoding with a long interleaving imposes a buffering delay that is a function of D and can be lowered to values typical of B-1 in addition of the current burst reception. However in many cases, the burst received at zapping instant is correct and can be forwarded to the media player for immediate decoding and display. This action is referred to a "fast zapping". If the terminals plays the signal at normal speed, the IFEC decoding will never benefit from the B-1 datagram bursts that are also used to create the parity. So techniques for spreading the delay are considered.

EXAMPLE: We give hereunder a first analysis of these techniques, assuming only MPE-IFEC is used and excluding therefore MPE-FEC. If we consider an encoding depth of B, a FEC spreading of S and a given code rate, we have the following theoretical characteristics during the zapping time on MPE-IFEC time slice burst k.

- at zapping instant, if D=0 or D>B+S, the MPE-IFEC time slice burst k is received without any protection; typical probability of having an erroneous burst before MPE-IFEC decoding is in the order of 20 % to 30 % in worst LMS-intermediate-tree-shadowed cases but only 8 % in LMS-suburban;

- for subsequent MPE-IFEC time slice bursts, more protection is received that enables better recovery as, in average, more data and FEC for the corresponding encoding matrices is received through early decoding (decoding without having received full parity, either FEC or DATA):

  - if D≥B+S, between zapping instant (MPE-IFEC time slice burst k) and MPE-IFEC time slice burst k+B+S-1, useful MPE-IFEC sections are progressively received up to full reception when MPE-IFEC time slice burst k+(B+S-1) is received);

  - between zapping instant (burst k) and burst k+B-1, MPE sections are received that progressively fill the ADTs up to a state of partial completion at burst k+B-1 that then stagnates at this level: even if we wait indefinitely, the ADTs will never be completely filled, which could be considered as a loss, because the missing columns are coming from datagram burst to be received after currently received datagram burst. The level of achievable completion in ADT depends on the initial buffering time D_{Buf}: the more the initial time, the better;

- after a number of burst, the received FEC columns can compensate for the missing ADT columns and can even provide some additional protection for "real" losses. In a typical case (B=6, S=4), during 4 bursts, no error can be corrected but starting with the 6^{th} burst, the correction capacity starts to be positive and a complete burst loss is can be sustained at the 7^{th} position;

- the number of artificially missing columns due to non completion of the data part remains but decreases quickly when the initial buffering increases. So it can be interesting to increase slightly the initial buffering time to ensure that a good level of protection is ensured. In any case, a complementary buffering time is needed to reach full protection (40 columns in each ADT in average).

So during B+S bursts, FEC protection increases regularly but does not reach maximal capacity unless some additional buffering of B-1 is activated.

- the initial buffering before starting to deliver IP packets, as counted in number of bursts, Buf_{init}, can be increased (Buf_{init}>1), this has a good impact on protection of the stream during zapping time;

- complementary buffering can be done by slowing the playout rate by a slowdown_rate factor: the slowdown_rate is the percentage of speed reduction at the output of the IFEC decoder (TR 102 377 [i.21] suggests that 20 % can be supported);
- total duration to recover the data parity (missing datagram burst to fill completely the ADTs) is then $(B\text{-}Buf_{init})$/slow_down_rate.

All these delays have been represented in the following figure as a function of (B+S) and taking for initial buffering delay Dbuf a value of 4 (systematic delay of 4 burst). For instance for (B+S)=10, we recover the initial data after 6 bursts, then the FEC and the final data are completely recovered after 10 bursts. So we can display first image 4 bursts after zapping and be completely protected after 10 burst. Another example for (B+S)=20: after 12 bursts, we have, thanks to the initial delay of 4 bursts, already a good protection over the data. We must receive 20 bursts to benefit from full FEC parity and the final protection is achieved after 45 bursts. Please note the these delays are not additive: they are time thresholds.

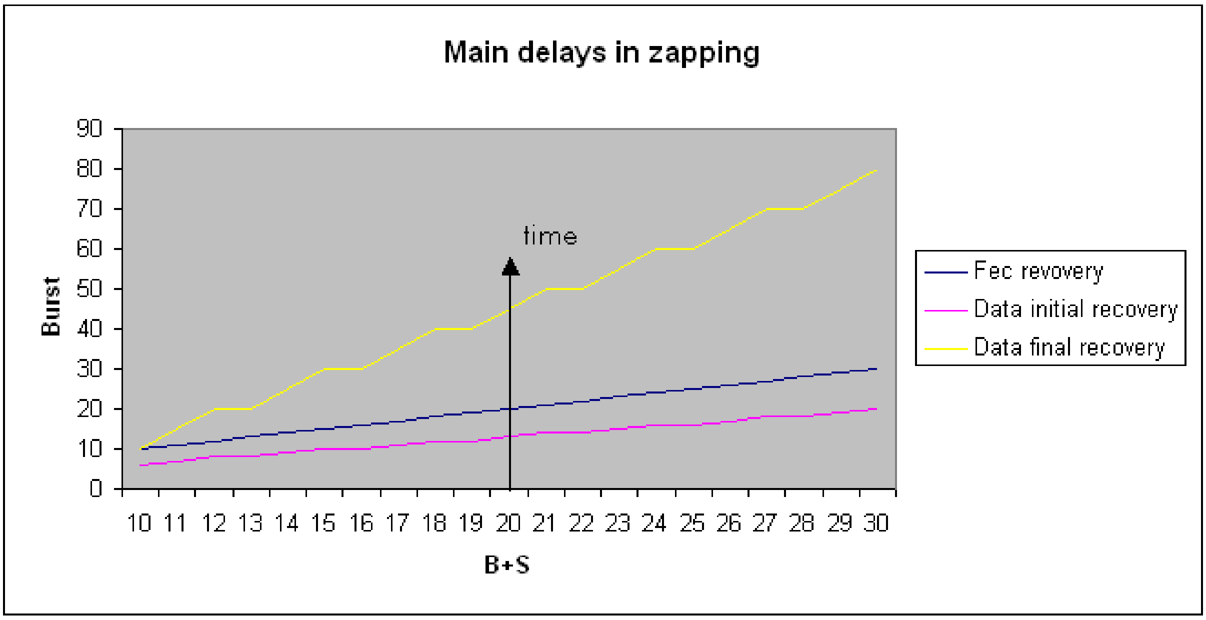


Figure 6.19: Main zapping delays

6.2.5.2.4 Zapping performance with measured quality during the zapping period

6.2.5.2.4.1 Zapping performance in class 1

Introduction:

The zapping performance for class 1 has been carefully studied in [15] to which the user shall refer for more details.

Over DVB-SH, transmit data and parities are delayed. Delay results from the aggregation of multiple effects including, among others, network latency and jitter, receiver buffering, and random access point picture acquisition. This leads to possibly important channel change delay !!!

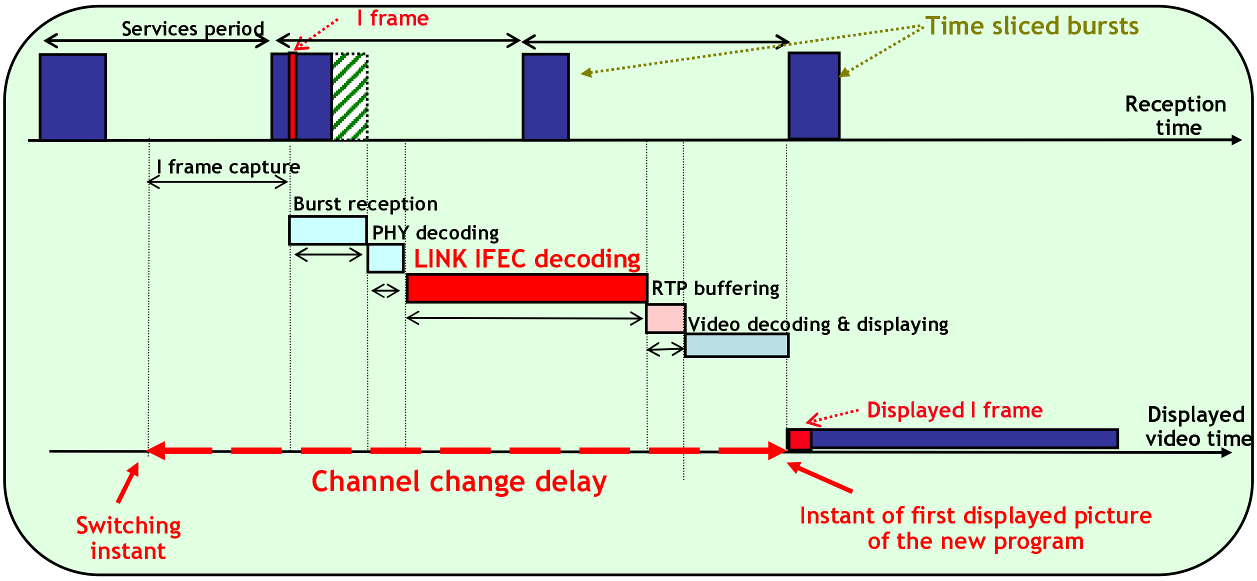


Figure 6.20: Zapping delay with introduction of MPE-IFEC

Zapping performance metrics:

To evaluate numerically this delay, we define performance metrics:

- Channel change delay:

This is the duration between the instant of the channel change request and the instant when MPE-IFEC delivers the first Intra-coded (I) frame to the player (T_{CH}). Tch measures only the delay introduced by the decoder - MPEG2-TS packet acquisition, buffering time required for RTP and video decoding not taken into account.

$$T_{ch} = A_D + (m-1).T_s + \varepsilon$$

where:

- AD = ADST Delay = delay between data arrival and data decoding;
- m = index of the burst containing the first decodable I frame ($m \geq 1$);
- Ts = service period, ε = the tune time depending on the location of I frame in the time-slice.

- Cumulated Freeze Time: an indicator of the subjective video quality. Cumulated time during which the decoder was unable to deliver a displayable frame over 20 s observation windows (CFT).

$$CFT = \sum_{n=0}^{N-1}\sum_{j=0}^{J-1}\left(\overline{F_{j,n}}.\Delta t\right)$$

where:

- N = number of Group of Pictures (GOPs) over 20 seconds, J = number of frames per GOP.
- $F_{j,n}$ = 1 if frame j is non-displayable, otherwise 0.
- Δt = duration of the frame.

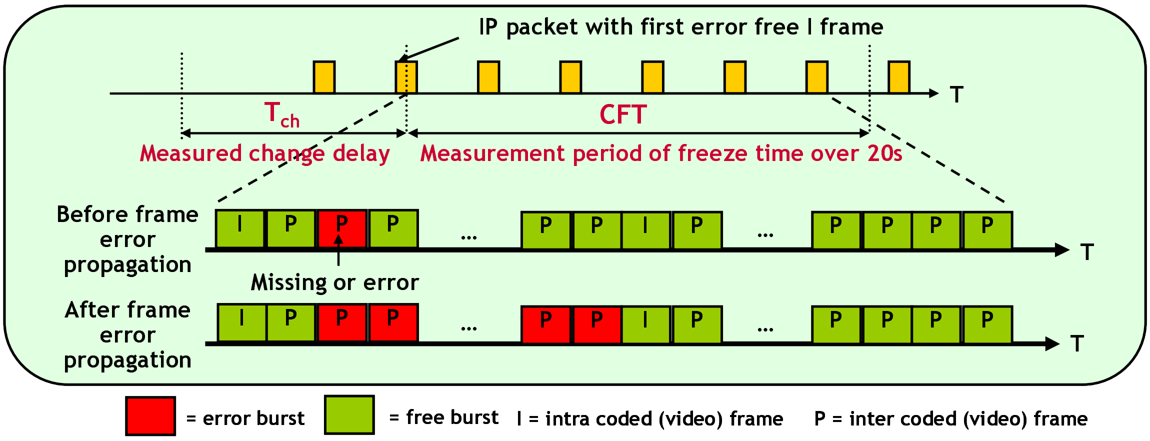
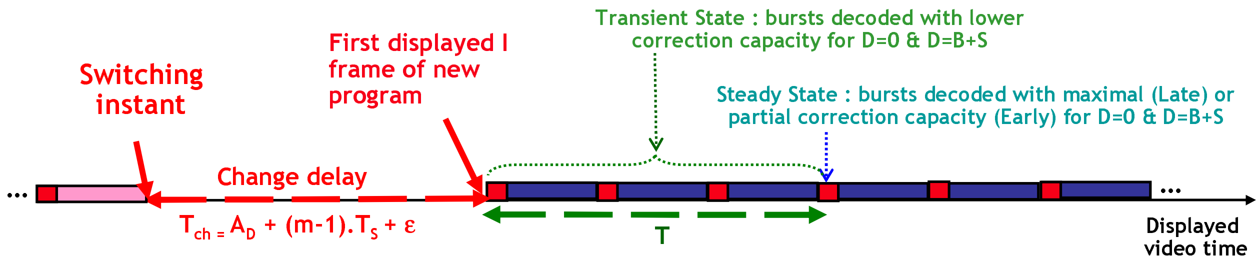


Figure 6.21: Zapping performance metrics concept

Reducing zapping time techniques:

In order to limit the delay introduced intrinsically by the MPE-IFEC technique, the fast zapping technique depicted in clause 6.2.5.2.3 is introduced whereby received data bursts are delivered to the upper layers without waiting for reception of all data and parity (this is called the "early decoding mode") while the data transmitted to the upper layer is slightly delayed in order to fill the MPE-IFEC buffer until the steady stae is reached (this is the late decoding mode) when all data and parities related to a datagram burst are normally retrieved by the MPE-IFEC decoder before decoding a current data burst.

Early decoding mode implementation is based on a new parameter, referred as the ADST delay and denoted by A_D. It fixes the buffering time of a received data burst before being delivered to the upper layers. According to A_D MPE-IFEC operates in an early or late decoding mode.



Decoding mode	LATE		EARLY	
Transmission mode	D = 0	D = B+S	D = 0	D = B+S
ADST delay : A_D	$= (B+S) . T_s + t_{max}$	$= B . T_s + t_{max}$	$< (B+S) . T_s + t_{max}$	$< B . T_s + t_{max}$
Recovery quality at Steady State	Maximal correction capacity		Partial correction capacity Lower !!!	
From the first displayed picture, Steady State achieved at T	$T = (B + S) . T_s$	$T = B . T_s$	$T = A_D . T_s$ Lower !!!	$T = A_D . T_s$ Lower !!!

Figure 6.22: Trade-off between the reduction of channel change delay and the subjective video quality with Early decoding mode

Zapping time simulation set up:

The support simulations is made of:

- a video streaming server delivering Constant Bit Rate, GOP duration = 1 s, H.264/AVC format, for 8 TV programs multiplexed in Ts = 1 second;
- a set of radio features of OFDM transmission with 2K FFT mode, Guard interval of 1/4, QPSK 1/2, 5 MHz channel, LMS-ITS or LMS-SU at 50 km/h at SNR = 11,2 dB and TU at 3 km/h and SNR = 7 dB.

- MPE-IFEC sizing for maximal performance at steady state in Late mode: the Goal is to guarantee 90 % fulfillment of ESR5 leading to the interleaving depth B+S, where optimal values for B and S are derived from B+S and code rate CR = B / (B+S) as follows:
  - LMS-ITS: B = 9, S = 11 (B+S = 20) - for CR = 0,45 (LMS-ITS = 91,6 % - LMS-SU = 100 % - TU = 100 %).
  - LMS-SU: B = 4 & S = 3 (B+S = 7) - for CR = 0,5 (LMS-ITS = 92,8 % - TU = 100 %).

NOTE: Simulations shows same results for both modes D = 0 and D = B+S.

Simulated zapping time results:

Simulations shown the contribution of the three parts in the change delay: A_D and a variable part (m-1).TS + ε:

- In LMS-SU and TU environments, in 90 % of channel change cases, the first error free I frame is displayed after the first received burst: (m-1).TS + ε is measured to a value lower than the GOP duration (1 s).
- In LMS-ITS environment, the first received bursts are more frequently non-displayable since severe burst losses occur more often in the first seconds after the channel change.

Table 6.12: Simulated channel delay

Delay (s) in 90 % of cases	LMS-ITS dimensioning: (B,S) = (9,11)		LMS-SU dimensioning: (B,S) = (3,4)
	LMS-ITS	LMS-SU & TU	LMS-SU & TU
Tch	A_D + 1,131	A_D + 0,555	A_D + 0,319

Simulated video quality results:

Video quality performance split into different 3 behaviour regions with regards to A_D:

- For D = 0:
  - In A: change delay is ok, video quality is nok;
  - In B: compromise between change delay and video quality;
  - In C: video quality is ok,, change delay is nok.
- For D = B+S:
  - In A: compromise between change delay and video quality;
  - In B&C: video quality ok, change delay nok but quality degraded in the first 20 s because errors in the first bursts are very prejudicial compared to D = 0.

In any case, D = B+S more sensible to errors than D = 0 just after the switching request (first seconds of video).

For larger B+S values, IFEC enters in the steady state later (late decoding): 20 s in LMS-ITS instead of 7 s in LMS-SU for D = 0, 9 s in LMS-ITS instead of 3 s in LMS-SU for D = B+S.

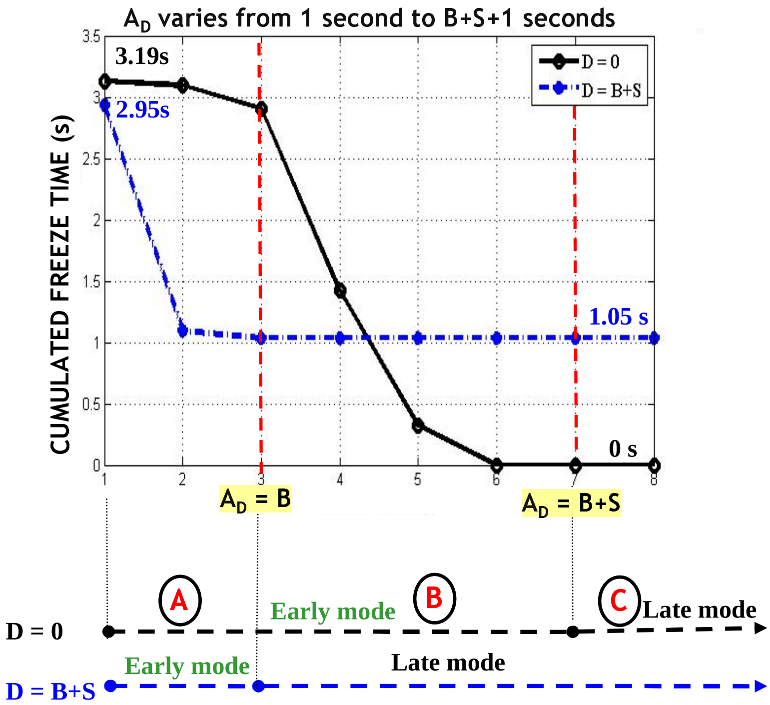


Figure 6.23: Video quality as a function of ADST delay in LMS-SU (B,S,D) = (3,4,0) and (3,4,7) and 90 % of cases

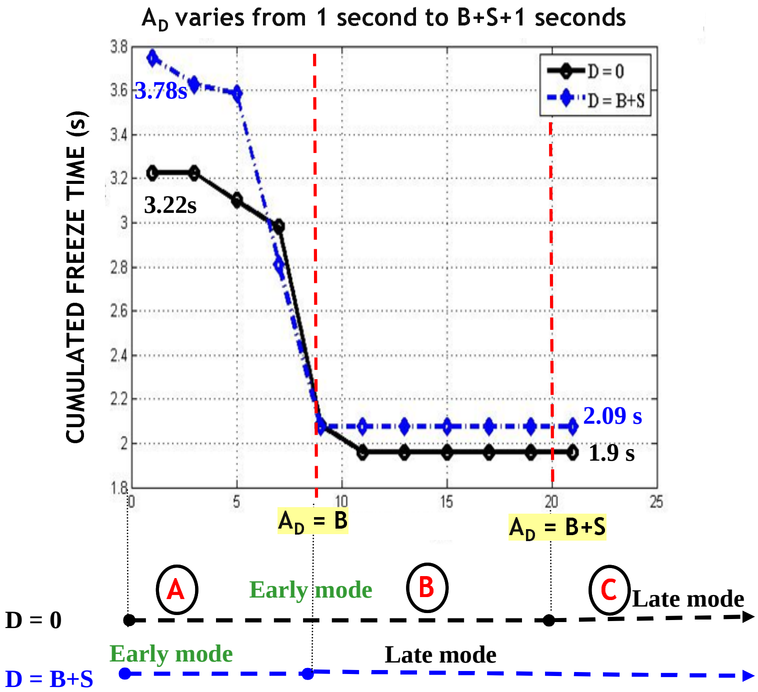


Figure 6.24: Video quality as a function of ADST delay in LMS-SU (B,S,D) = (9,11,0) and (9,11,20) and 90 % of cases

Conclusion:

Experimental analysis of the channel change performance with respect to the delay and the cumulated freeze time showed that for typical DVB-SH channels (TU and LMS) the value of the ADST delay can be chosen in order to minimize the channel change delay while maintaining an acceptable level of subjective video quality (minimal cumulated freeze time). However the quality level obtained is not optimal since MPE-IFEC decodes the data with a reduced correction capacity. In order to offer the lowest channel change delay with the better subjective video quality to the end-user, the [CDP IG] proposes a mechanism to enable a transitioning between an "immediate" play and a maximum error protection realized via a progressively switching between the early decoding mode and the late decoding mode of MPE-IFEC.

6.2.5.2.4.1 Zapping performance in class 2

Introduction:

The main focus of this chapter is the behavior of the DVB-SH class 2 physical layer. It is investigated what are typical delays between a switch-on or zapping command and the availability of a first I-Frame at the video decoder input buffer. Different events of switch-on and zapping are described. In addition to that, a reference composition of a DVB-SH MPEG-TS including time sliced services is introduced in a fairly abstract way, leaving exact numbers to be filled by experts on DVB-SH multiplex composition. Therefore, only ranges of values and their corresponding symbols are given.

Cases for switch-on and zapping:

S1) Acquisition of a new TS; this assumes that the parameters like bandwidth, FFT, etc. are known to the receiver, e.g. using the delivery descriptors available throughout any other DVB or DVB-SH link.

Z1) Decoding of a different elementary stream within the same TS, but within a different SHService.

Z2) Decoding of a different elementary stream within the same TS and within the same SHService.

Z3) Decoding of a different elementary stream within the same TS under the assumption of full multiplex decoding (including large de-interleaver memory to store the entire de-interleaver).

Cases S1 and Z1 might be similar as the de-interleaver memory is empty at the moment of the event, while cases Z2 and Z3 can benefit from previously-stored interleaver units within its deinterleaver memory. Special attention is therefore laid on cases S1 and Z1.

Zapping time simulation set up:

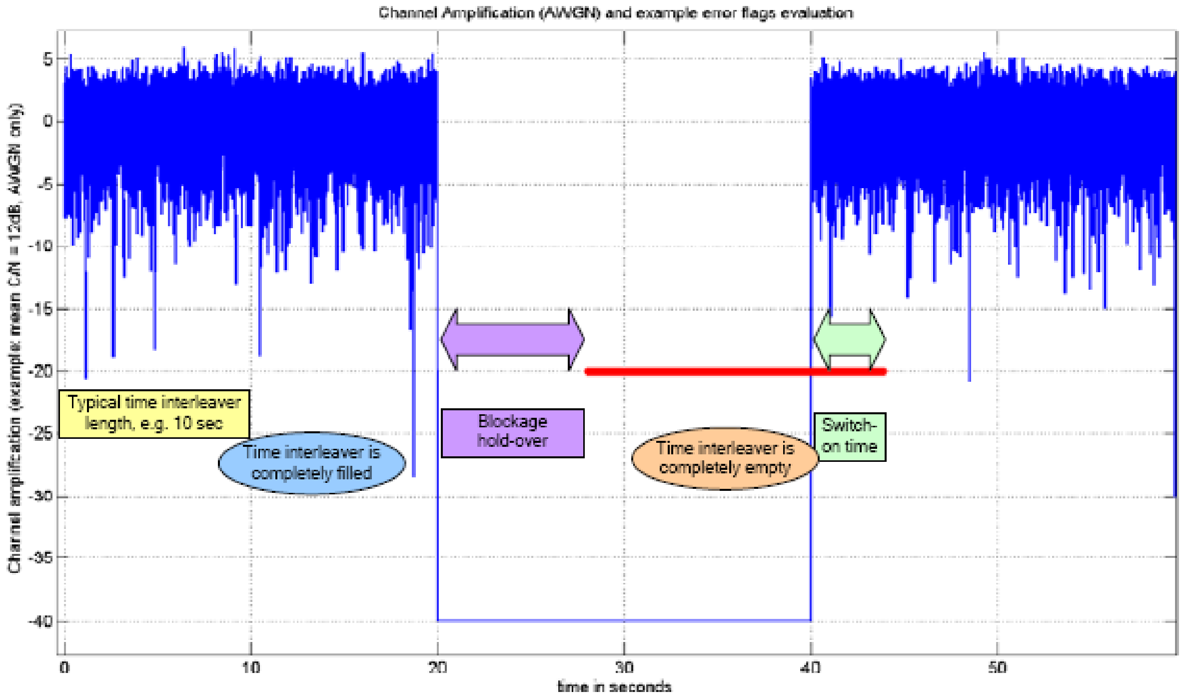


Figure 6.25: Channel amplification as simulated (here with C/N = 12 dB); channel off-time between seconds 20 and 40; blockage hold-over time (amber), switch-on time (green)

For simulation, an AWGN-channel with on-off characteristic is introduced. All on- and off-events are of larger duration than the time interleaver such that either the interleaver is completely filled (before the off-event to evaluate the blockage hold-over time) or completely empty (before the on-event to evaluate the switch-on time without any prior knowledge). The time between the off-event and the first error after the FEC decoder is measured (see amber arrow) - this value indicates how long the physical layer is able to deliver error-free output even after running into a blockage event. The time between the on-event and the last erroneous packet after the FEC decoder is measured (see green arrow) - this value indicates how long it takes for the physical layer to deliver error-free output again after switching-on.

Table 6.13: Class 2 zapping time cases

Configuration	Table	ID	Comments
2	Table A.10.11	1	-
3	Table A.10.12	1	10/12/10/8/4
4	Table A.10.11	1	40/0/12/8/4
5	Table A.10.16	1	10/18/11/8/4
6	Table A.10.12	1	5/18/11/4/4
8	Table A.10.26	7	

Zapping time simulation results

Configurations with 16QAM R=1/5.

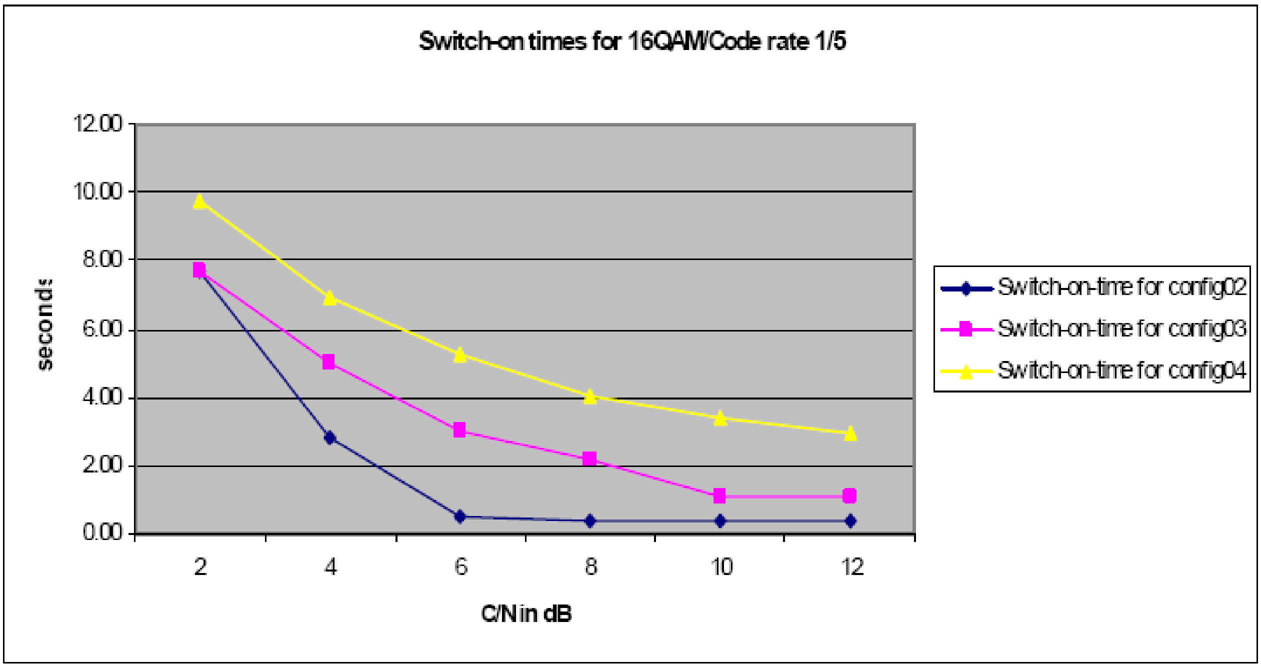


Figure 6.26: Switch-on times for 16QAM with R=1/5

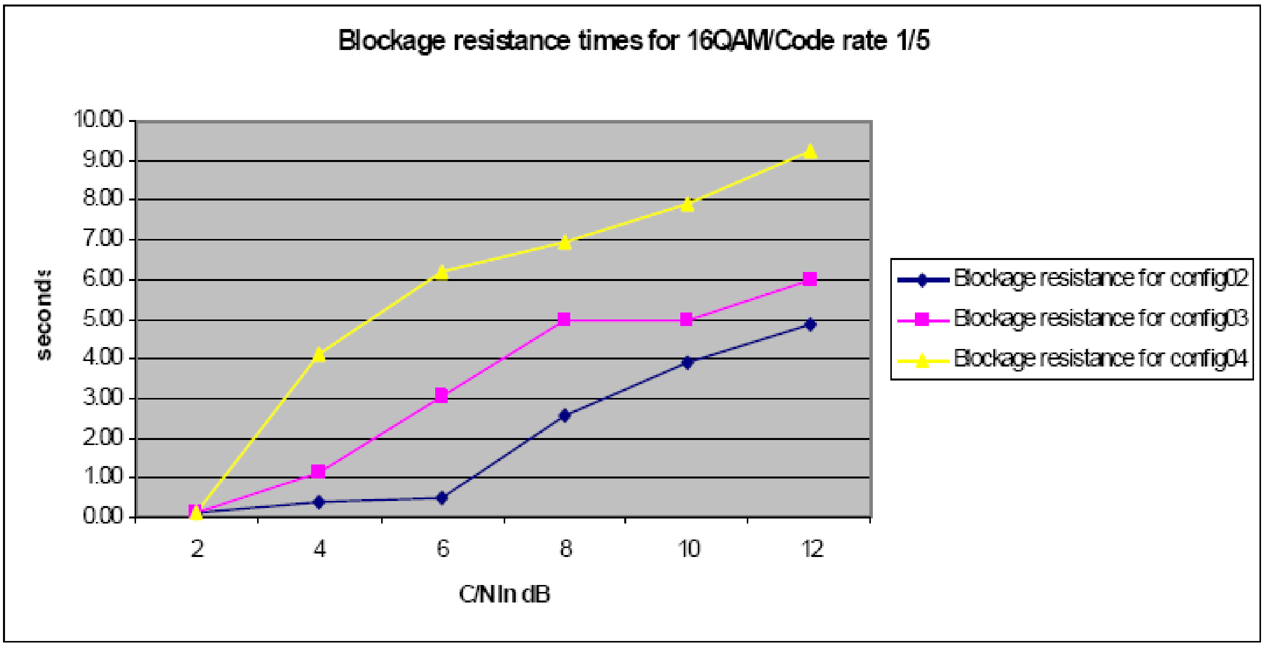


Figure 6.27: Blockage hold-over times for 16QAM with R=1/5

Configurations with different spectral efficiencies.

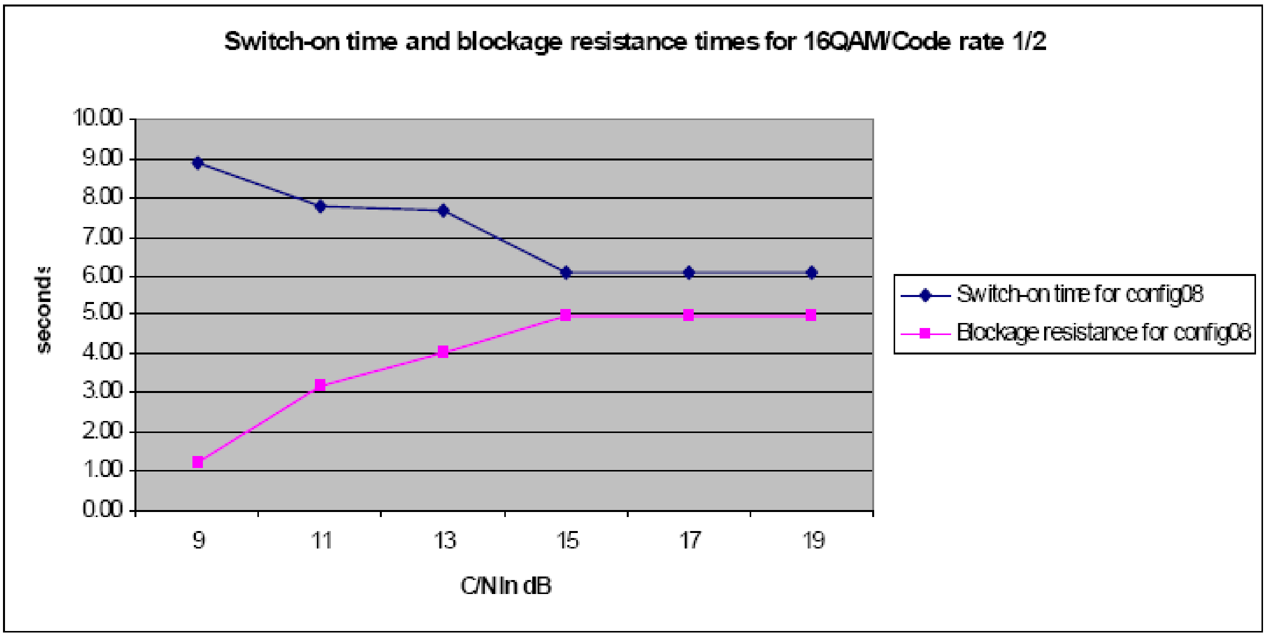


Figure 6.28: Switch-on and blockage hold-over times for spectral efficiency ~2 bps/Hz

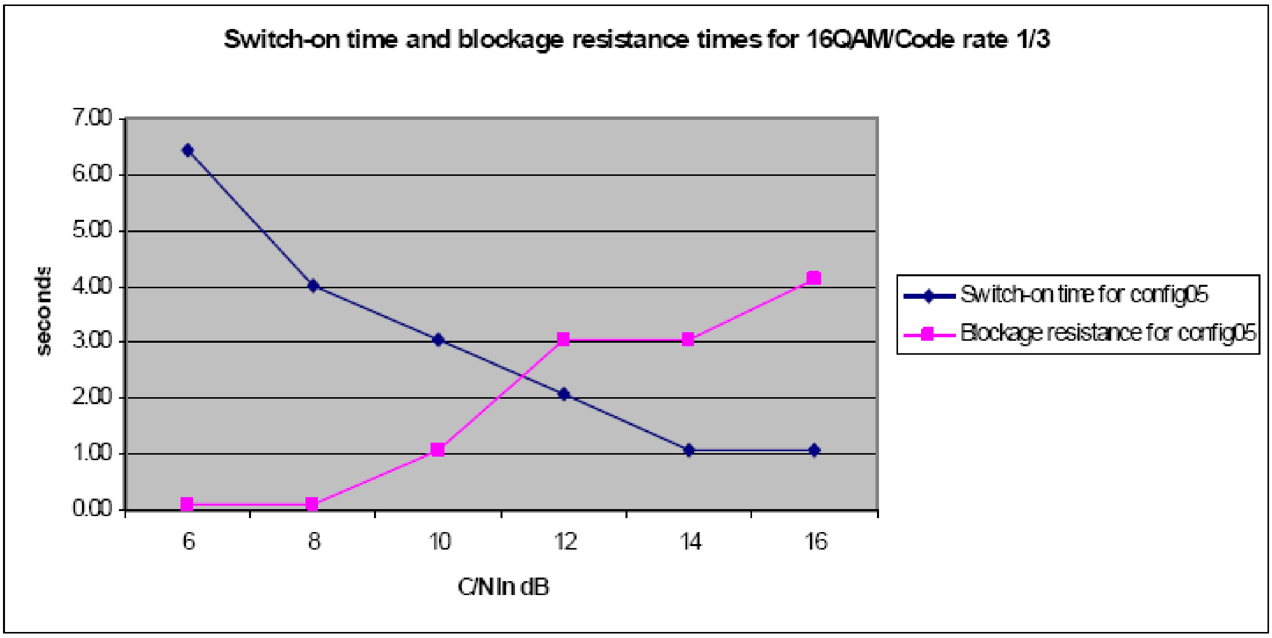


Figure 6.29: Switch-on and blockage hold-over times for spectral efficiency ~1,3 bps/Hz

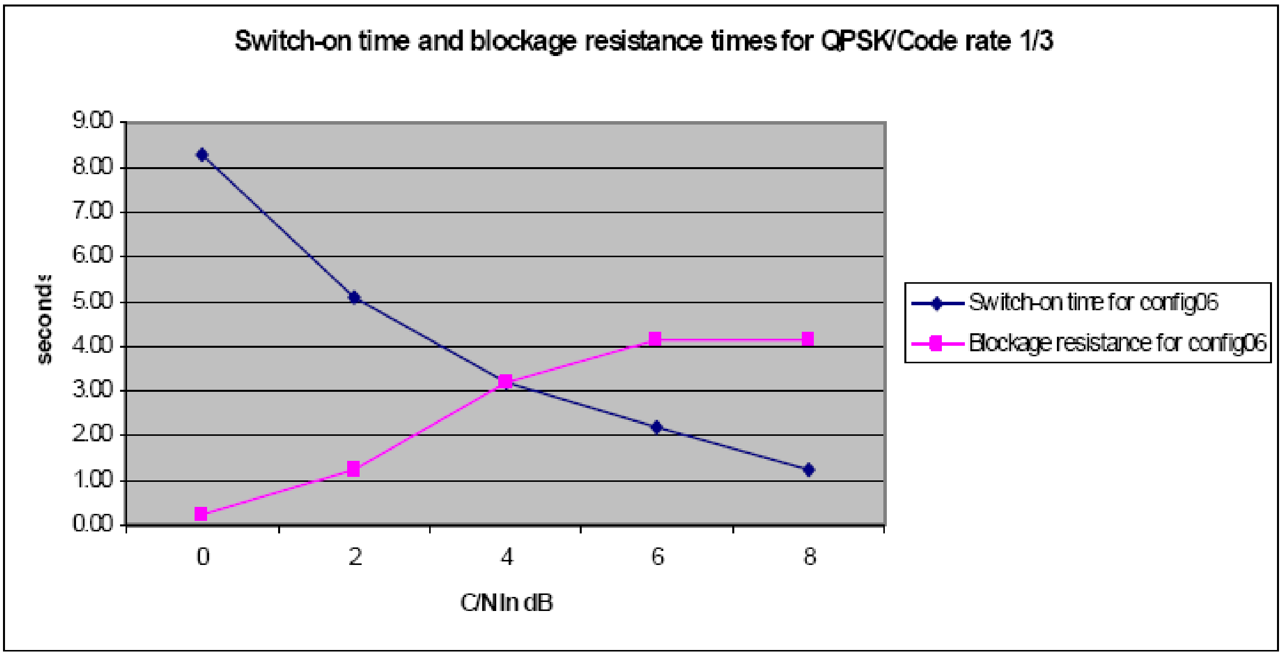


Figure 6.30: Switch-on and blockage hold-over times for spectral efficiency ~0,7 bps/Hz

Conclusions

Different Class-2 interleaver configurations and its behavior for different switch-on and zapping instants have been analyzed to determine the performance for a given spectral efficiency. The following conclusions can be derived:

1) For a given spectral efficiency, e.g. 0,8 bit/s/Hz resulting from 16QAM, R=1/5, different interleaver profiles result in significant performance differences.

   a) While the uniform interleaver has the highest performance to overcome blockages, its switch-on behavior is rather bad, even in high C/N conditions.

   b) The uniform/late interleaver as defined in IG-V1 has best switch-on performance but a lousy blockage resistance - therefore its use is discouraged also for IG-V2 simulations.

   c) A carefully designed uniform/late interleaver (with R=0,8 in the late part, config03) benefits highly from fast switch-on (e.g. 2,2 s at 8 dB C/N) while allowing long blockages (e.g. 5 s at 8 dB C/N).

2) Comparing at similar spectral efficiencies these 16QAM results with QPSK results, the performance of QPSK (here simulated with R=1/3 and a spectral efficiency of 0,7 bit/s/Hz) offering less throughput is comparable to the 16QAM, R=1/5 case:

   - For the carefully designed uniform/late interleaver (16QAM, config03), the switch-on times at 8 dB C/N is 2,2 s while overcoming 5 s of blockages.

   - For the carefully designed uniform/late interleaver (QPSK, config06), the switchon times at 8 dB C/N is lower (1,2 s) but the blockage performance is worse (4,1 s).

3) The statements made during the Call for technologies on the switch-on performance have been fully confirmed by these simulations:

   a) According to the instantaneous C/N, the switch-on time is reciprocal to the received C/N at switch-on.

4) For higher spectral efficiencies (e.g. 16QAM, R=1/3 offers 1,3 bit/s/Hz), it is shown that the uniform/late profile still offers good performance at switch-on and for blockage protection.

5) With even higher spectral efficiencies (e.g. 16QAM, R=1/2 offers 2 bit/s/Hz), the time interleaver shuffles the missing code bits (which are either lost in the blockage or have not been received before the terminal switch-on), such that:

   a) The switch-on performance is not acceptable.

b) The blockage hold-over performance is not acceptable.

c) The use of long physical layer interleaver with code rates 1/2 and higher is not recommended.

6) If using Class-2 concepts with uniform/late profile, it is recommended to keep the code rate below 2/5. If the targeted spectral efficiency cannot be achieved, a switch to a higher-order modulation scheme is recommended.

6.2.6 Memory requirements

6.2.6.1 Introduction

This clause gives the memory requirements for the receiver. We first explain in clause 6.2.6.2 Memory sizing function how the MPE-IFEC memory can be sized in a class 1 terminal by defining a "memory sizing function". We then introduce how this memory can be used in a typical implementation using the bijection between the ADST and ADT introduced in clause 6.2.3.5 ADT mapping, distinguishing between constant bit rate and variable bit rate IP flows. We describe an implementation applying for CBR. VBR can always be supported with this implementation but the memory may be significantly higher. Memory management optimization are possible that enable keeping the memory as low as for CBR traffic but this is for further study.

6.2.6.2 Memory sizing function

Hypothesis 1: we consider that ADST columns are stored in ADT via the adt_index and adt_colums functions (see clause 6.2.3.5 ADT mapping). As a consequence, there is no need for budgeting memory for ADST and memory is essentially budgeted for Encoding Matrices, ADTs and FDTs.

NOTE 1: One could say that at least 1 ADST buffer could be envisaged for absorbing current burst for which burst number is unknown and which cannot be mapped on ADT/FDT. This is true but, as the reader will see in the rest of the clause, optimizations could probably be possible to store this burst in ADT/FDT columns not already used without affecting quality of the decoding. So the conclusion remains the same, the budget is considered for the only ADT and FDT.

Since we budget only ADT and FDT, memory requirements are directly a function of the number of encoding matrices, which is a function of parameters B,S and D via the M function, and of ADT and FDT individual sizes:

$$\text{memory} = (\text{ADT\_size} + \text{FDT\_size}) * M(D,B,S) \text{ where } M(D,B,S) = B + \max(0,S-D) + \max(0,D-B).$$

Hypothesis 2: we assume that memory is the sum of all ADT and FDT sizes counted by their matrix weight in bytes and zeroing any pointer structure (that will be used in the document below) or ancillary information bits. This approach will give precise order of magnitude but actual implementation may require more memory capacity.

ADT and FDT memory is sized by their number of rows, T, and their number of columns, C for the ADST and ADT and R for the FDT. Since different optimizations are possible, for the parameter C we need to precise between ADT and ADST (C_{adst} and C_{adt}).

The list of parameters required for sizing the memory is given below:

- T is usually selected by the user but minimum values are fixed by the data volume per datagram burst since the maximum number of columns C is fixed to 255. T must be selected amongst {256, 512, 768 and 1 024} but T must be greater than ceil(datagram_burst_size/255). A default working value for T is 1 024.

- C is a function of the traffic profile and its instantaneous variations datagram burst by datagram burst: variable datagram burst will occupy an ADST that must be sized to accommodate its peak size:

  $\forall k \geq 0, T * C \geq \text{datagram\_burst\_size}(k) + 1$ => this sizes the C_{adst} parameter.

- Normally C_{adst} should be used for ADT sizing so that $C_{adt} = C_{adst}$. This is the natural option suggested by the specification. But terminal memory management optimizations can enable differentiation between both. These implementations are for further study. So in the following $C=C_{adt}=C_{adst}$.

- When the code rate is fixed, it is a direct function of C_{adt} => this sizes the R parameter such that:

$$R = C_{adt} * \frac{1-\text{code\_rate}}{\text{code\_rate}} \text{ according to clause 6.2.3.8.}$$

These C_{adt}, T and R parameters enable to derive memory requirements:

$$\text{Memory} = \frac{T * C_{adt} * M(D,B,S)}{\text{code\_rate}}.$$

NOTE 2: C_{adst} is not considered in the equation; this is extremely important since it enables to de-correlate the IFEC memory sizing from the actual instantaneous variations of the traffic which is measured through the C_{adst}. More considerations on the case where C_{adt} and C_{adst} are different is for further study.

NOTE 3: Memory will need to be sized for a minimum code rate since, the lower the code rate, the higher the memory requirement.

6.2.6.3 Implementation aspects

6.2.6.3.1 Introduction

Some implementation aspects of the memory are described in the following, in particular the memory requirements. Basically two opposite scenarios can be described:

- a first scenario will size the memory based on peak traffic ($C_{adt} = C_{adst}$);
- a second scenario will size the memory on averaged traffic (C_{adt} may differ from C_{adst}).

6.2.6.3.2 Approach based on peak traffic

In the first approach based on peak traffic, the memory requirement is to use M(D,B,S) times the memory required for a maximum burst size as given by the product $T*C_{adst}$. Implementation-wise, each ADST is constituted of a list of C_{adst} pointers, each pointer addressing one of the C columns of one of the M ADTs via the adt_index and adt_column functions. If one ADST column is padded, the memory is still reserved inside the corresponding ADT, leading to potential memory waste. In addition, there must be R columns for the FDT, R being equal at least to

$C_{adst} * \frac{1-\text{code\_rate}}{\text{code\_rate}}$, leading also to wastes on the FDT part. This is illustrated in figure 6.31.

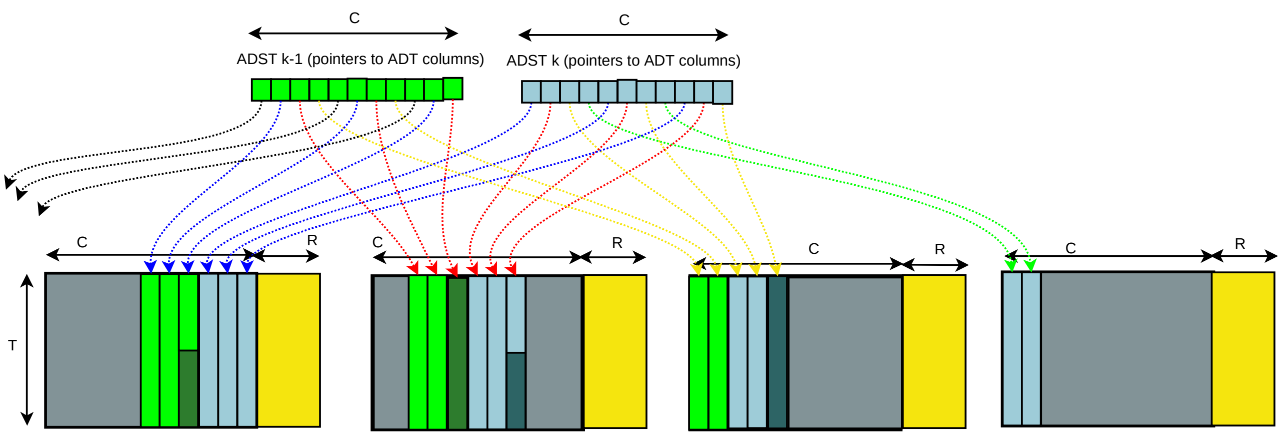


Figure 6.31: Implementation 1 (single pointer)

On figure 6.31, we can see that the padding column is greyed but its memory is "physically" reserved.

The main drawback of this implementation is particularly apparent in case of VBR where it can lead to excessive memory since, for a lot of datagram bursts, the ADST will be made of many padded columns. If we consider the typical example of clause 6.2.3.1 Introduction, we have on average 37 columns. If we reserve the full 191 columns, this means we will have a waste factor of 5!

6.2.6.3.3 Approach based on average traffic

Another approach is to use the averaging effect of ADT mapping.

Hypothesis 3: It is assumed that $\forall k \geq 0, \sum_{k}^{k+B-1} \mathrm{ADT\_columns}(k) \leq L$ where ADT_column(k) is the number of data columns in ADT(k) excluding the padding columns from the counting.

We then define $C_{ADT} = \frac{L}{B}$ and $R_{ADT} = \mathrm{ceil}\left(C_{adst} * \frac{1 - \mathrm{code\_rate}}{\mathrm{code\_rate}}\right)$. The memory can then be sized based on the averaged C_{ADT} rather than the peak C_{ADST}, using a double pointer approach illustrated in figure6.32: first pointer connect ADST columns to ADT columns (via standard ADST to ADT mapping functions adt_index and adt_column) whereas second pointer removed padding columns by pointing only the ADT data columns to the physical memory.

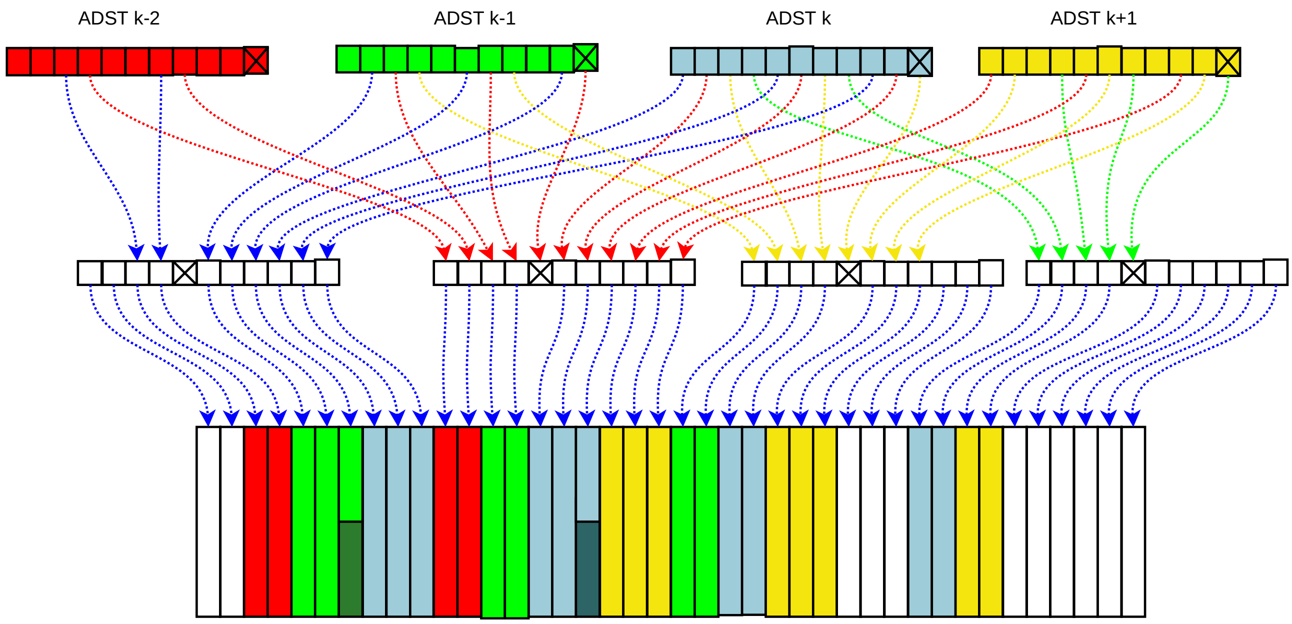


Figure 6.32: Implementation 2 (double pointer)

Therefore the real physical memory can be sized on the average bit rate and not on the peak bit rate. This information is given to the receiver via max_average_rate_over_B field found in the time_slice_fec_identifier. This field introduced in [i.33] with following definition: $C_B = \max\left(\frac{\sum_{i=n-n[EP]}^{i=(n-n[EP]+(EP-1))+(B-1)} B_i}{T_n}\right)_{\forall n \geq 0}$, where:

- B_i is the size of i[th] Time Slicing burst in MPE section payload bits.
- T_n is the time from the transport packet carrying the first byte of the first MPE section in burst/frame *n-n[EP]* to the transport packet carrying the first byte of the first MPE section in burst/frame *(n-n[EP]+(EP-1)+B)* within the same elementary stream.

It is then quite easy to cnvert this information into the number of columns.

6.2.6.3.4 Memory evaluation

We provide some memory quantitative evaluations. For the average MPE traffic, we use the parameter max_average_rate, given by the time_slice_fec_indentifier. In order to represent the burst traffic fluctuations, we introduce a parameter called traffic_variations_ratio which represents the burst traffic variations around this average value. $\forall k \geq 0, C_b(k) \leq C_{b\_ifec} * (1 + \text{traffic\_variation\_ratio})$. This parameter can be important for VBR traffic (more than 100 %). Even in the case of CBR, there are small variations of traffic around a medium value and the ratio is not null.

Hypothesis 4: we assume also that a repetition_interval (average time distance between two successive bursts) can be given. This can be compute by averaging over a sufficient number of bursts.

The number of rows nof_rows is given by T found in the time_slice_fec_identifier (we take 1 024 in the following).

C_{adst} is then sized by the average traffic plus the traffic_variations_ratio and given by following formula:

$$C_{adst} = \text{ceil}\left(\frac{\text{max\_average\_rate} * \text{repetition\_interval} * (1 + \text{traffic\_variations\_ratio})}{\text{nof\_rows} * 8}\right)$$

a) Implementation based on peak traffic: $C_{adt}=C_{adst}$.

b) Implementation based on averaged traffic:

$$C_{adt} = \text{ceil}\left(\frac{\text{ceil}\left(\frac{\text{max\_average\_rate} * \text{repetition\_interval} * \text{sizing}(B,S,D)}{\text{nof\_rows} * 8}\right)}{\text{sizing}(B,S,D)}\right)$$

c) The final C_{ADT} must be taken as the minimum of the two values - $\min(C_{adt}; C_{adst})$.

In any cases, $R = \text{ceil}\left(\frac{\text{fec\_ratio}}{1 - \text{fec\_ratio}} * C_{adt}\right)$.

For both cases, assuming for B and S the B_{opt} and S_{opt}, we can derive memory as a function of (B+S), code_rate and max_average_rate:

$$\left.\begin{array}{l} \left.\begin{array}{r} (B+S) \\ \text{code\_rate} \end{array}\right\} \Rightarrow (B_{opt}; S_{opt}) \Rightarrow M(D;B;S) \\ \left.\begin{array}{r} \text{max\_average\_rate} \\ \text{repetition\_interval} \end{array}\right\} \end{array}\right\} \Rightarrow (C_{adst}; C_{adt}; R)$$

The resulting bit rates are given in figure 6.33 for some typical maximum average bit rates values (128, 256, 512) and with a maximum value of code rate of 50 %.

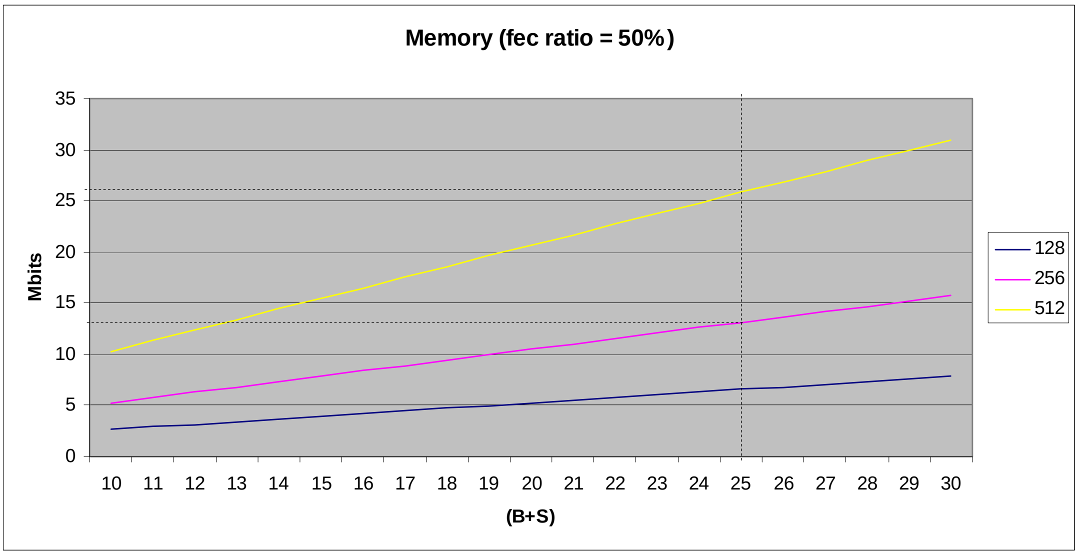


Figure 6.33: Memory requirements in CBR

For a 250 kbps CBR stream, based on simulated performance given in clause 6.2.5, we recommend values presented in table 6.14.

Table 6.14: Typical memory requirements for a CBR stream

	Modulation	B+S (bursts)	Code rate (%)	Memory (Mbits)
LMS-ITS	16QAM	25	40	13
	O-QPSK	20	55	10
	TDM	15	65	8
LMS-SU	all	10	66	5
NOTE 1: These values are excluding any memory for the MPE-FEC decoding.				
NOTE 2: These values are applicable for CBR. For VBR, one must ensure that the burst size does not exceed C*T. Memory optimizations for supporting efficiently VBR are for further study.				

6.2.7 MPE-IFEC usage scenarios

6.2.7.1 Definition

Equal Error Protection (EEP): case where a video stream made of several individual layers (e.g. SVC) is protected equally on all its individual layers, ID est each of the individual layer bears same FEC protection as the others.

NOTE 1: This case opposes to the UEP case.

MPE-IFEC sources: these are sets of elementary streams transporting MPE-IFEC sections coming from the same encoding matrices and having coherent signalling. These elementary streams may be located on same or different TS, same or different frequencies, same or different radio technologies.

Unequal Error Protection (UEP): case where a video stream made of several individual layers (e.g. SVC) is protected unequally on the different individual layers, ID est each individual layer bears a potentially different FEC protection than the others, more important layers being usually better protected than less important layers.

NOTE 2: This case opposes to the EEP case.

6.2.7.2 Introduction

This clause describes typical scenarios of MPE-IFEC usage in a DVB-SH environment, leveraging on the signalling flexibility. We distinguish two basic scenarios, one using a single source of MPE-IFEC and another one using two sources, the latter being for further study.

6.2.7.2.1 Single MPE-IFEC source

This is the basic scenario that has been used throughout the document. The system is delivering an IP flow sent in the format of MPE sections, protected by MPE-IFEC sections sent together with the original MPE data sections over the same frequency. This scenario applies for instance to SFN cases where the satellite and CGC are operating on the same modulation and frequency, forcing exactly the same content at the section level to be transmitted under the different areas (satellite only, hybrid satellite and CGC, CGC only).

In a constant bit rate situation, the number of MPE and MPE-IFEC sections is constant from an IFEC time-slice burst to another IFEC time-slice burst. However, when a variable bit rate traffic must be supported, the number of sections present in each burst can vary dramatically. This clause describe how these variations are handled by the sender and how the signalling helps the receiver coping with them.

Assuming the general case of variable bit rate traffic, the amount of MPE-IFEC section is limited by the following factors:

- the used code rate: as explained in clause 6.2.3 A practical example, maintaining a target code rate will enable a maximum number of MPE-IFEC sections in iFDT is equal to:

$$\mathrm{nof\_fec\_columns\_code\_rate}(i) = \mathrm{ceil}\left(\mathrm{nof\_data\_columns}(i) * \frac{1 - \mathrm{code\_rate}_{\mathrm{target}}}{\mathrm{code\_rate}_{\mathrm{target}}}\right);$$

- the capacity of the IFEC time-slice burst, usually computed in number of MPEG2 TP that turns into a number of MPE-IFEC sections once the capacity to send the MPE sections has been used. We assume this capacity to be nof_fec_columns_capacity(i).

It is assumed that the two variables will not differ significantly, in particular because the IP encapsulator will change nof_fec_columns_capacity(i) to follow the variations of nof_fec_columns_code_rate(i). However some adjustments are possible, especially if the nof_fec_columns_capacity(i) is slightly superior to nof_fec_columns_code_rate(i). It would then be useful to use the resource to send an additional MPE-IFEC section. For that purpose, the `max_iFDT_column` must be sent to a value superior or equal to the largest nof_fec_columns_capacity(i) plus additional margin multiple of S.

The procedure to constitute the IFEC burst can be the following:

- for each IFEC burst, list sections eligible for inclusion inside the burst according to the interleaving mechanism explained in clause 6.2.3.9 IFEC burst generation;
- for that purpose, pick in the S previous iFDT the required number of sections until one of the following conditions is reached:
    - no more IFEC sections are available (in that situation an MPE-IFEC section index discontinuity may happen as explained in clause 6.2.3.9 IFEC burst generation);
    - no more capacity is available in the MPE-IFEC time slice burst;
- once the list of MPE-IFEC sections has been established, insert them in the MPE-IFEC time slice burst and set their real-time information in the header:
    - set the burst number to current value;
    - set the section index to the current value;
    - set the iFDT index equal to the source iFDT;
    - set max_iFDT_column equal to the value computed from target_code_rate as explained in clause 6.2.3.8 FDT generation and code rate computation;
    - set IFEC_burst_size equal to total MPE-IFEC size;
    - compute real-time parameters, including Delta-t (the latter in exactly the same way as in MPE as explained in clause 6.2.3.11 Burst sending arrangement).

Typical algorithm supporting such procedure are for further study.

6.2.7.2.2 Multiple MPE-IFEC source

Scenarios allowing multiple sources of MPE-IFEC sections are also possible and presented as examples below:

- Scenario 1: MPE and MPE-IFEC sections are sent over the satellite link and complementary MPE and MPE-IFEC sections are sent on the terrestrial link. This complement is expected to provide enhanced performance at the border between the satellite and the terrestrial coverage. Synchronization between the satellite and the terrestrial links is required and already described in [21].

- Scenario 2: the hierarchical modulation is used to distribute complementary MPE and MPE-IFEC sections to the HP modulation. It is well known that some LMS channels like LMS-SU behave like on-off channels whereby receivers will be able to either receive error-free bursts or not receive at all the bursts with typically 2 consecutive bursts being erased. It would therefore be beneficial for receivers, during the "good state", to receive additional IFEC sections that would make the protection even stronger.

Following recommendation apply to multiple MPE-IFEC sources:

- the multiple MPE-IFEC sources must be mixed at the receiver before being processed by the MPE-IFEC decoder and this mixing is such that:

  - duplicate MPE-IFEC sections on HP and LP must be removed;

  - order is not necessarily maintained.

- concerning the MPE sections:

  - duplicate MPE sections on HP and LP must be removed;

  - order must be maintained.

- The consequence on the transmitting site is the following:

  - the two sources MPE-IFEC sections must be received at the mixer within within the signalled delta_t (from first MPE section on HP) + max_burst_duration;

  - the last MPE-IFEC having frame_boundary set to 1 transmitted on HP must be the last MPE-IFEC section received therefore all MPE-IFEC sections transmitted on LP must be transmitted before the last MPE-IFEC section of the HP;

  - repetition interval on both HP and LP shall be equal and there shall be the same number of time slice bursts on both modulations.

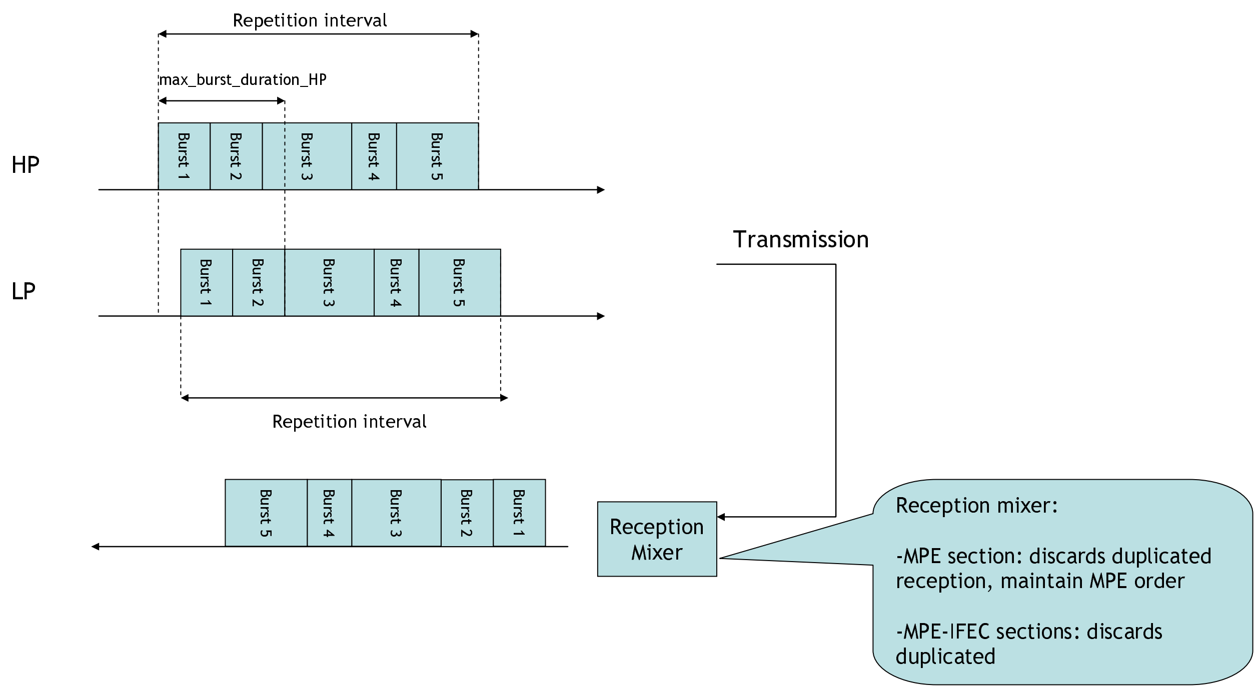


Figure 6.34: Synchronization principle in hierarchical multiple MPE-IFEC sources usage

6.2.7.2.3 Layered IFEC-encoding in SVC context

Introduction:

In this scenario, different layers compose a final video stream such as in SVC. MPE-IFEC enables to encode differently each layer and we want to check that the combined usage of SVC and UEP can be beneficial.

The practical scenario is targeting improvement of DVB-SH satellite coverage in rural outdoor environment. This is achieved by reducing the targeted minimal value of C/N (Carrier to Noise ratio), named C/N min required for having good quality of TV/video service reception for DVB-SH channels, on the main layer:

- The good quality of TV/video service reception is defined as the 1-ESR5 criteria of 90 %.
- The broadcast operator wants to extend the broadcast coverage on a specified area, without increasing transmission power and without impacting the number of broadcasted services (modulation and channel coding rate remain constant), and impacting the less service quality. The broadcast operator wants to offer access to a basic video quality on an extended coverage, keeping the nominal video quality over the nominal coverage, all others parameters (power, number of services) being equal.
- A nominal video service of bit rate Rs protected with a channeld coding of rate R_c is split into 2 complementary H.264 SVC services: a base service S_B of bit rate Rs_B and an SNR enhancement service S_E of bit rate Rs_E that increases the decoded video quality when combined with service S_B. S_B is protected using a channel coding rate Rc_B while S_E is protected using a channel coding rate Rc_E.
- Let α being the "base protection extension" factor, known and fixed by the operator with $\alpha > 1$. Base service radio coverage is extended if different code rates are used: ${Rc_B}^{-1} = \alpha \cdot Rc^{-1}$ with $\alpha > 1$ or different interleaving (B+S) depths are used. R_c is the code rate of the original AVC stream. Due to the disadvantage of using longer interleaver on the base service, thus increasing the overall zapping time and memory usage, the first solution is favored, having different code rates for the base and enhanced layers while keeping the overall bit rate.

Configuration setting:

IFEC configuration is configured such that ESR5 quality exceeds 90 %: in a channel with radio parameters QPSK 1/2 - C/N=11,9 dB, the selected IFEC parameters are CR = 2/3 & B+S = 9. This leads to a performance of 1 - ESR5 = 92,8 %.

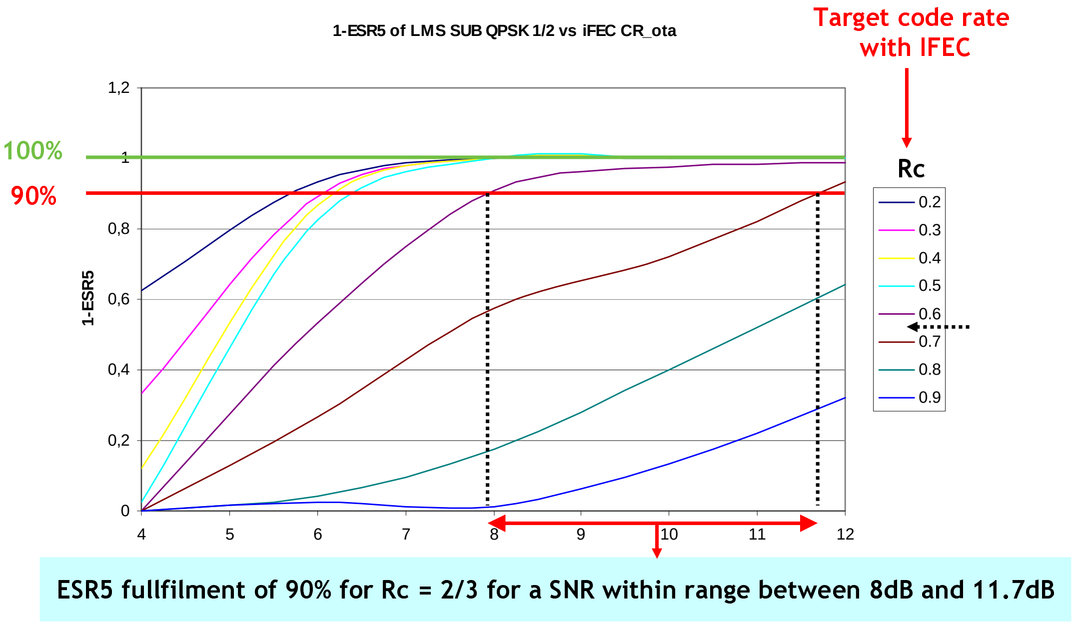


Figure 6.35: 1-ESR5 (%) w.r.t Rc vs. SNR (dB) for B+S=9

Then extension factor α is selected: Rs_B and α is fixed, Rc_B is computed and Rc_E is selected using the curves below.

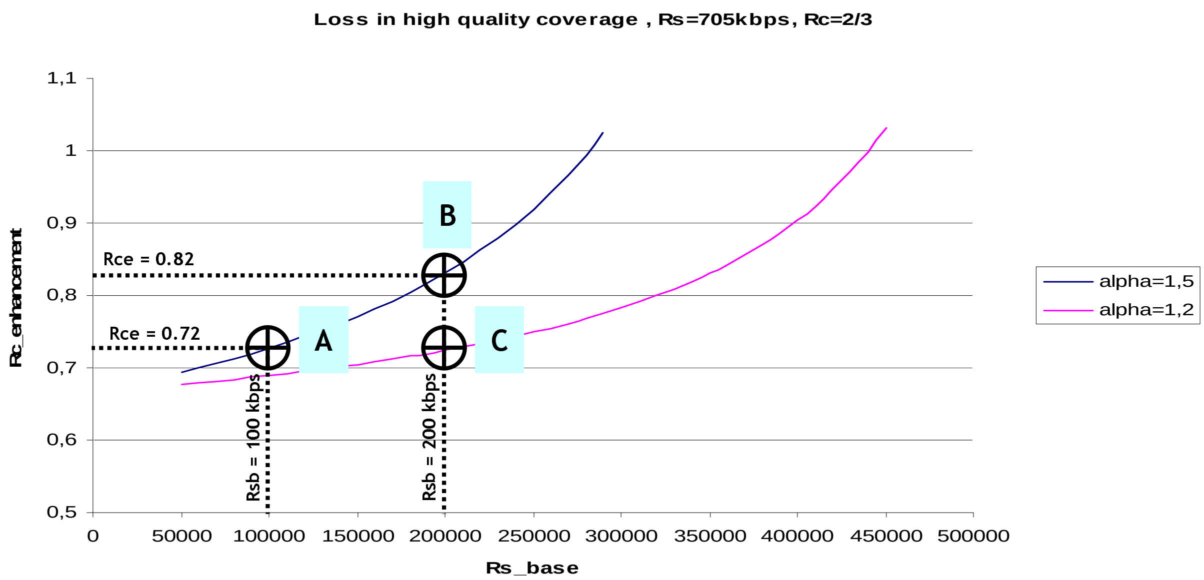


Figure 6.36: Sizing UEP with IFEC

Three configurations are tested:

a) R_{Sb} = 100 kbps, ext =1,5 ⇒ R_{CB} = 0,44, R_{ce} = 0,72, R_{SE} = 605 kbps, Burst_period = repetition interval = 1 s.

b) R_{Sb} = 200 kbps, ext = 1,5 ⇒ R_{CB} = 0,44 & R_{ce} = 0,82, R_{SE} = 505 kbps, Burst_period = repetition interval = 1 s.

c) R_{Sb} = 200 kbps, ext = 1,2 ⇒ R_{CB} = 0,55 & R_{ce} = 0,72, R_{SE} = 505 kbps,
Burst_period = repetition interval = 1 s.

Performance results:

In terms of video quality, the following results are given.

Table 6.15: LMS-SUB (Rs = 705 kbps) ⇒ ref AVC @ 38,1 dB PSNR - Rs_b = 100 kbps

Layer	Resolution	Framerate	Bitrate (kbps)	DTQ	PSNR
0 (AVC)	352x288	30	95,1	(0,0,0)	**28,9 dB**
1	352x288	30	446,6	(0,0,1)	
2	352x288	30	546,1	(0,0,2)	
3	352x288	30	649,0	(0,0,3)	
4	352x288	30	690,9	(0,0,4)	**35,9 dB**

Table 6.16: LMS-SUB (Rs = 705 kbps) ⇒ ref AVC @ 38,1 dB PSNR - Rs_b = 200 kbps

Layer	Resolution	Framerate	Bitrate (kbps)	DTQ	PSNR
0 (AVC)	352x288	30	196,8	(0,0,0)	**32,4 dB**
1	352x288	30	487,5	(0,0,1)	
2	352x288	30	581,2	(0,0,2)	
3	352x288	30	679,4	(0,0,3)	
4	352x288	30	720,6	(0,0,4)	**35,9 dB**

NOTE: We have only a light degradation of PSNR (less than 2,2 dB) that can be justified by the usage of more than 2 layers: in the example, 4 enhancement layers were encoded, which significantly reduces the SVC coding efficiency. These multiple layers are used for allowing a simple rate control in the video stream by means of dropping dynamically excessive layers, to compensate the absence in the SVC toolbox of means for providing at the encoder side CBR rate control that are needed for DVB-SH dumps usage. As it is well-known, the additional layers bring additional overhead and therefore PSNR degradation at equal bit rates. Therefore, the encoding performance could be significantly improved with adapted CBR control mechanisms and less enhancement layers as shown in [SW07-01]. We assume that this loss of 2,2 dB would significantly be lowered with the addition of efficient rate control mechanism with 2 layers.

In terms of C/Nmin we get following results.

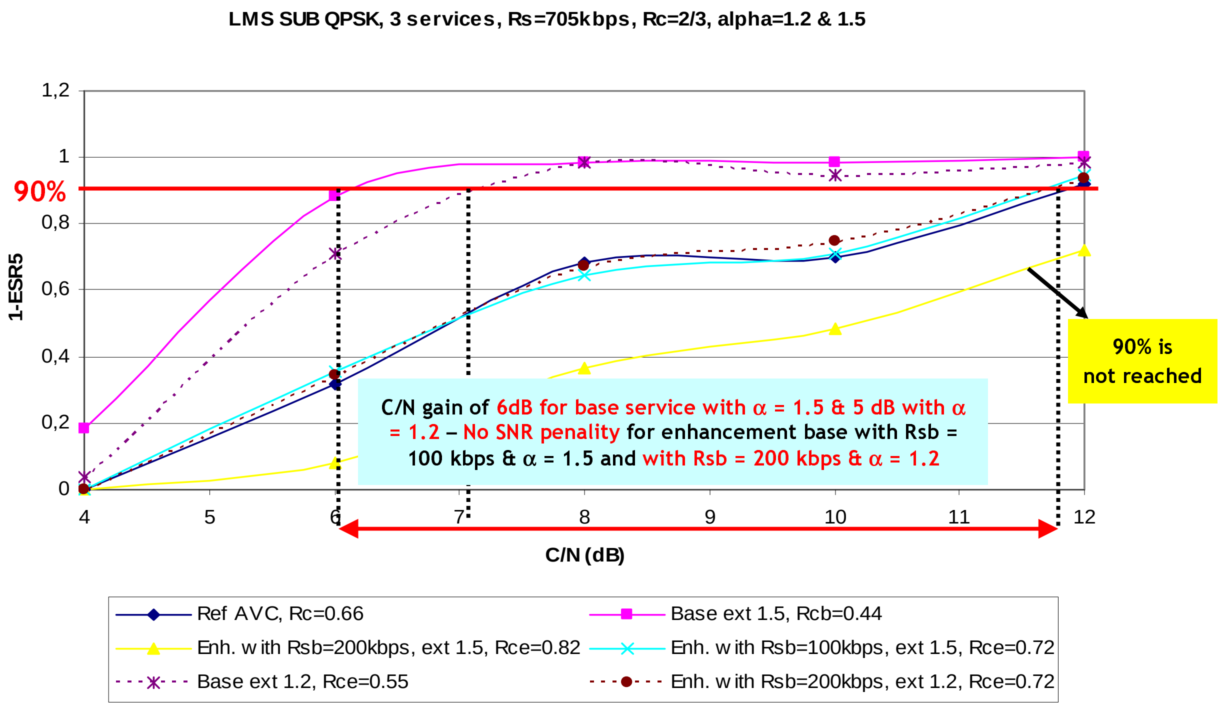


Figure 6.37: Curve of 1-ESR5 (%) vs. C/N (dB) for B+S = 9

Conclusion:

Therefore for Rsb = 200 kbps and α equal to 1,2, using Rcb = 0,55, Rce = 0,72, we observe a gain of ≈ 5 dB C/N gain in LMS sub urban environment on base service, while having a base service at 200 kbps of acceptable quality (above 30 dB PSNR for Foreman) and almost no loss on enhancement coverage.

6.3 Time-Slicing

Time slicing is one of the key features introduced by DVB-H. It enables three important features:

- to spare battery by powering off the receiver during intervals when no service is listened;
- to support fast zapping;
- to support variable bit rates and so statistical multiplexing in TDMA environment.

The DVB-SH makes full usage of this time slicing information in order to provide same kind of feature support. However the way this is handled depends on the choice of the long interleaver for protection against long impairments (class 1 or 2). In the following we first provide insight on the way Time-slicing signalling is handled in DVB-SH and the impacts of this handling on power saving and VBR. In each impact case, we differentiate between class 1 and class 2 long interleavers.

6.3.1 Signalling

For class 1, time slicing information is important since it enables the terminal to power off. Time slicing signalling is conveyed by data and FEC section headers in real-time parameters (taken from MAC destination bytes in MPE). Since the Time-slicing information is repeated in all sections, the probability to not receive this information is equal to the probability of losing all sections and, if ts_error_indicator is used, to the probability of having, for all section headers, at least one TS among the group conveying those header, erroneous. When such case happens, the receiver will loose Delta-t synchronization and will stay on until it can process a full section header and get a new Delta-t.

Table 6.17: LMS-ITS ratio of lost burst (%)

case	14	17	18	18	21	24
	its	its	its	its	its	its
#burst	3 691	3 691	3 691	3 691	3 595	3 586
#errburst	1 424	1 641	1 251	833	1 069	1 096
#lost burst	978	1 193	788	300	24	27
#lost bursts / #bursts (%)	26	32	21	8	1	1

Table 6.18: LMS-SUB ratio of lost burst (%)

case	24	24	27	30	31	34	37	77	80	80
	sub	sub	sub	sub	sub	sub	sub	sub	sub	sub
#burst	3 667	3 586	3 667	3 667	3 667	3 594	3 590	8 939	3 583	8 984
#errburst	278	1 096	278	300	262	344	342	2 237	139	2 214
#lost burst	190	27	190	223	166	9	12	1 061	2	953
#lost bursts / #bursts (%)	5	1	5	6	5	0	0	12	0	11

Assuming typical case of LMS-ITS, we have (case 18) 21 % of burst that are lost for which no Delta-t can be retrieved. During 20 % of bursts, the terminal, unless some different strategy is applied, will go back to power on, leading to a possible power saving impact. This can be approximated by saying that 80 % of the time, the normal power saving is used while during 20 % of the time no power saving is used. This leads to a degradation of 20 % of power saving.

For class 2 receivers, time slicing information is less important since the long interleaver makes the real-time parameters to be outdated. Other techniques such as the one presented in clause 7.2.3.3.1 enables to rely on other structures like the DVB-SH services signalled by SHIP.

6.3.2 Zapping time impact

This clause addresses how zapping time is impacted by the DVB-SH.

DVB-SH introduces long interleaving, at physical and link layer to counteract long fading experienced in LMS channels. Long interleaving can appear as contradictory with the zapping time since, to ensure protection of the stream, the receiver would need to wait for the full duration of the de-interleaving. Forcing such long zapping times is not acceptable from a user perspective, so different techniques are supported by DVB-SH to provide "fast zapping" while still ensuring good protection levels.

For class 2 receivers, the physical interleaver can be tuned to provide fast zapping using uniform late profiles as described in clause 7.3.3.5. Choosing e.g. a 50/50 uniform/late profile, the zapping time is the time duration of one time interleaver burst which corresponds to roughly 200 ms on the physical layer, plus link layer delays and I-frame searching. The price to pay is that this burst must be received with a few dB more than what is the C/N threshold for this code rate (e.g. for rate 1/4 now with 4,4 dB instead of -0,9 dB as presented in clause 7.3.2.6.4).

For class 1 receiver, selection of adequate link layer parameter (D) will also help reducing the zapping time and increase reception quality during zapping period. Such configurations enable to display the first well received burst immediately after physical interleaver latency assumed to be around 200 ms, plus the delay to receive the burst and search the I frame, so without waiting for the full redundancy to be received. As presented in this clause, the fast zapping provide immediate display in 80 % of the bursts in most stringent channels. The price to pay is that full redundancy has not yet been received and FEC recovery techniques such as the ones presented in clause 6.2.5.2 Zapping time performance must be applied to increase protection for following bursts.

6.3.3 Power saving impact

One of DVB-SH key elements is the introduction of a longer time interleaver at the physical level that has a negative impact on the power saving. Two cases are presented below depending on which type of interleaver is used.

6.3.3.1 "Terrestrial" physical interleaver

The linear part of the interleaver introduces a latency within the receiver, because of the convolutional nature, that adds a net time to the power on. The following formula can be used for deriving this information:

$$\text{time\_on} = \text{acquisition\_time} + \text{physical\_interleaver\_duration} + \text{burst\_duration}$$

- burst_duration can be approximated by $\frac{\text{repetition\_interval}}{\text{number\_of\_services}}$

$$\text{power\_saving} = 1 - \frac{\text{time\_on}}{\text{repetition\_interval}}$$

- depending on repetition_interval the power saving will evolve; typical values are:
  - 1 s for repetition interval and 9 programs => burst_duration~111 ms;
  - acquisition_time~50 ms;
  - physical interleaver~200 ms;
  - So time_on~361 ms and power_saving~64 % instead of 84 % in DVB-H.
- impact of loss pattern:
  - as referring to clause 6.3.1 Signalling we have power saving degradation of 20 % so 0,8*64 = 51 %.

6.3.3.2 "Long" physical interleaver

In that case, the involved durations are much longer than "terrestrial" case, in the order of multiples of SH-frames. In that conditions, time slicing signalling cannot really be used for providing the relevant off times since the decoding and MPE header processing happens after a time de-interleaving that lasts longer than the Time-slicing signalling itself. However, when service synchronization between DVB-H and DVB-SH is activated, it is possible to recover partially the power saving gain. In service synchronization, bursts are grouped in DVB-SH services signalled by SHIP packet. DVB-SH services are fixed-sized and their repetition interval does not vary. The terminal is then able to pre-determine the off periods without any knowledge on the MPE Delta-t: Time-slicing is actually managed at physical level. This approach is explained in clause 7.2.3.3.1.

- if we take the previous example and consider 3 DVB-SH services, we find:

$$\text{time\_on} = \text{acquisition\_time} + \text{physical\_interleaver\_duration} + \text{burst\_duration}$$

$$\text{time\_on} = \text{acquisition\_time} + \text{late\_tap\_interleaver} + \frac{\text{repetition\_interval}}{3} \qquad \text{power\_saving} = 1 - \frac{\text{time\_on}}{\text{repetition\_interval}}$$

- assuming same interleaver configuration as the one used for the class 2 terminal fast zapping in clause 6.3.2 Zapping time impact and same DVB-H service structure as in the short interleaver case in clause 6.3.3.1 "Terrestrial" physical interleaver, we have following values:
  - 1 s for repetition interval;
  - acquisition_time~50 ms;
  - physical interleaver (late taps only)~200 ms;
  - So time_on~583 ms and power_saving~42 % instead of 84 % in DVB-H.

- impact of loss pattern:
  - this degradation of power saving is compensated by the "constant" service structure: there is no need to switch on and search for the next burst and, in case of lossy channels, the power saving is not impacted by the low pattern contrarily to the class 1 (power saving maintained at 42 %).

6.3.3.3 Summary

For a typical configuration with 9 services of 111 ms duration each and a mapping of 3 services within 1 DVB-SH service, the achieved power savings are given in table 6.19. Please note that these values are given as examples, actual values will vary between configurations.

Table 6.19: DVB-SH impact on power saving

	DVB-H (reference)	DVB-SH class 1	DVB-SH class 2
Power saving clear	84 %	64 %	42 %
Power saving lossy (ITS)	N/A	51 %	42 %

6.3.4 VBR/statmux impact

6.3.4.1 Interest of statistical multiplexing with AVC and SVC

"Implementing Statistical Multiplexing in DVB-H"; International Journal of Digital Multimedia Broadcasting, Volume 2009 (2009), Article ID 261231, http://www.hindawi.com/journals/ijdmb/2009/261231.abs.html [i.45].

- [i.35]: http://telecom.esa.int/telecom/www/object/index.cfm?fobjectid=30341.

Figure 6.38 shows a comparison of variable bit rate (VBR) coding with respect to Constant Bit Rate (CBR) coding. CBR coding (dashed lines) results in a big fluctuation regarding video quality. VBR encoding (solid lines) allows for significant quality improvement of the most difficult scenes. Experiments indicate that when video is coded at a fixed quantizer step size (almost constant quality), the peak bit rate of a difficult picture frame may become more than 10 times the mean bit rate.

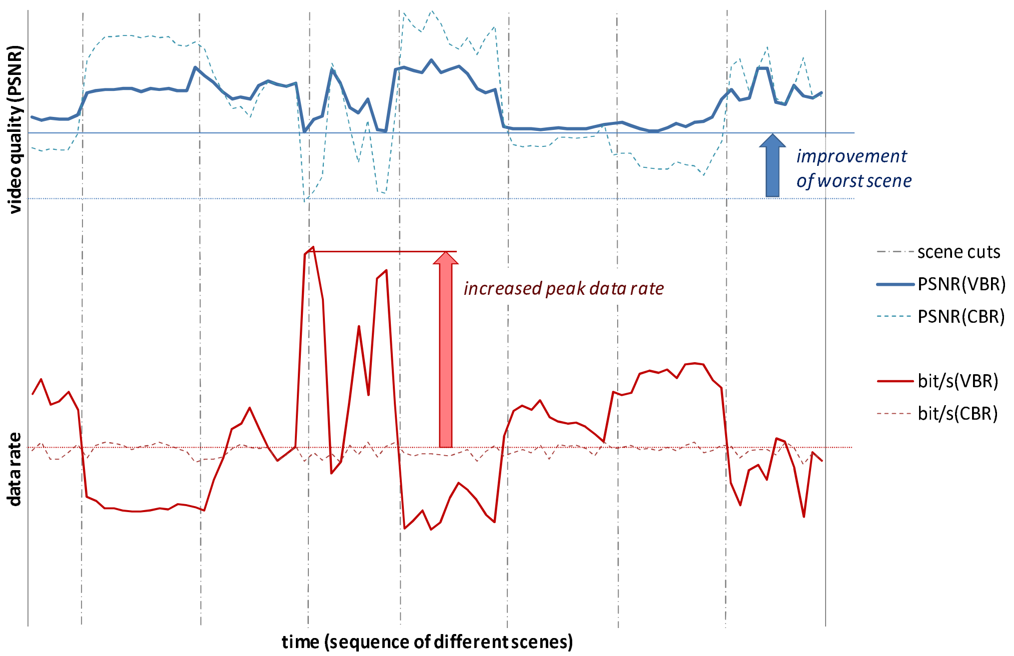


Figure 6.38: Comparison of CBR and VBR coding

A joint data rate control for all services within a multiplex can improve the utilisation of the whole multiplex as indicated in the right diagram of figure 6.39.

Although traffic variations may be shaped and turned into constant bit rate traffic, this would be done at the penalty of the receiver buffering and so zapping time. In order to avoid this buffering time, it is important to support "real-time" variations of the traffic bit rate. DVB-H supports such variations by letting the burst vary in size from frame to frame at the expense of some loss in power conservation (ref [Rezaei]). DVB-SH also supports these variations in a similar way as DVB-H although exact way of supporting depends on physical layer interleaver choice (class 1 and class 2) because different classes do not "mix" DVB-H services in the same way.

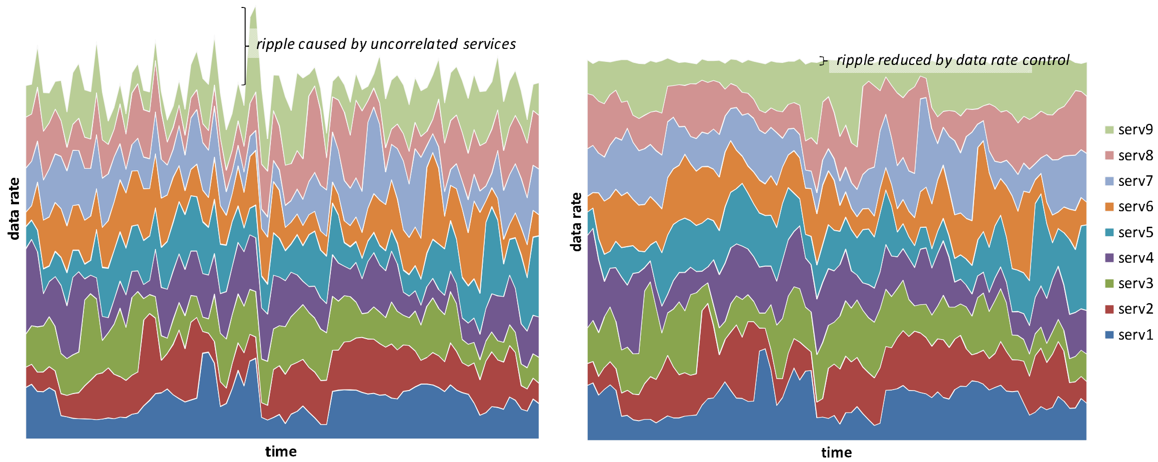


Figure 6.39: Uncorrelated services in statistical multiplex; left without, right with common data rate control

Experiments simulating all link layer processing steps for AVC coded video services (i.e. multi-protocol encapsulation, time slicing, multiplexing, and mapping onto MPEG2-TS packets) have shown that the statmux gain for DVB-SH is the same as reported for other already established broadcast systems like DVB-H. Moreover, results obtained in [SVCons] has shown that a better statmux gain can be achieved also for scalable video coding (SVC). Here, different layers of the scalable video should be transmitted with different error protection, e.g. using hierarchical modulation. Using SVC, statistical multiplex allows for fluctuation of the data rate within each layer, i.e. the available data rate in the high priority (HP) channel is shared among the Base Layer (BL) of all services. Exploitation of the features introduced by SVC, such as graceful degradation, only requires that the aggregated BL data rate for all services may not exceed the available capacity of the HP channel. Enhancement layers share the available data rate in the low priority (LP) channel(s), optionally also utilizing remaining data capacity of the HP channel. The typical statmux gain as a function of the number of SVC services within one multiplex compared to AVC services are shown in figure 6.40. It should be kept in mind that such results are dependent on the video material used, the video quality threshold adopted and the accepted relaxation of the power conservation in receivers.

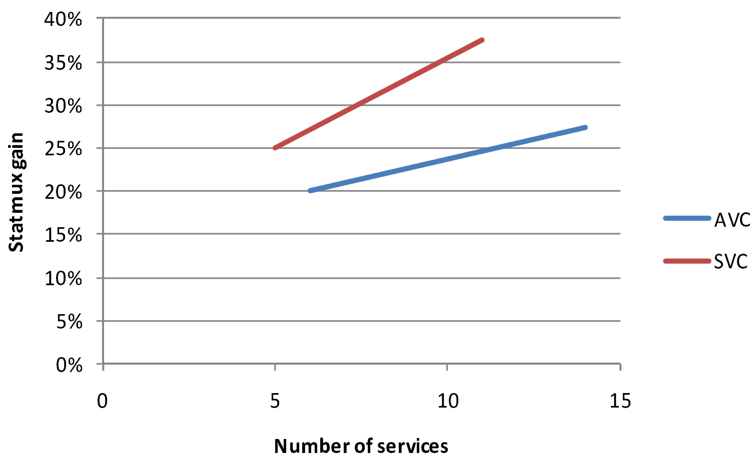


Figure 6.40: Statmux gain vs. number of services (from SVConS project)

6.3.4.2 Terrestrial (short) physical interleaver

With class 1 receives, the long interleaver is performed at the link layer. Since link layer also provides Delta-t information, the burst can freely vary in size and so VBR is fully supported: the addition of MPE-IFEC does not change or delay the reception and processing of the Delta-t signalling. VBR implies that useful traffic will vary from burst to burst, possibly in large proportions. For instance, the volume could multiply (or divide) by a factor of 2 between two successive bursts. Without any IFEC protection, a statistical multiplexing control would allocate dynamically the bandwidth to different streams so that compound of all streams stays within radio fixed capacity. This means that starting dates for burst (and Delta-t) will vary from burst to burst. The variable DATA burst sizes will make ADT more or less filled with DATA columns. Then different strategies can be applied to benefit from this gain of bandwidth:

- *FEC complement protection*: this strategy consists in varying the FEC so that, in each transmitted burst, the volume of DATA plus FEC is always the same. By this means, the freed volume by statmux is used to increase FEC protection and link quality. This leads also to varying code rates, with values lower than the minimum specified. Of course the capacity will not be increased. In the sender operation of MPE-IFEC specifications [MPE-IFEC], clause 3.5, the limitating factor becomes the fec_burst_size computed so that burst size is fixed, the number of FDT FEC columns being as large as required.

- *Fixed FEC code rate*: this strategy consists in keeping FEC code rate fixed while increasing the capacity. Depending on the actual ADT size in data columns, a varying number of FEC columns is created so that code_rate_FDT = nof_ADT_data_column / (nof_FDT_fec_column + nof_ADT_data_column). In the sender operation of EN 302 583 [1], clause 3.5, the limitating factor becomes the number of FDT columns dynamically set by the code rate, the fec_burst_size being as large as the maximum size.

Usual implementations maintain the bit rate of a group of individually services within a fixed setting. It is clear in that situation that the *D* parameter MUST be the same for all services, otherwise the different delay applied to services may lead to variable bit rate of the group of services and, possibly, exceed the setting as presented in figure 6.41. This does not prevent having, for instance, one group of statistically multiplexed services with a fixed setting and same D, and another group with another D.

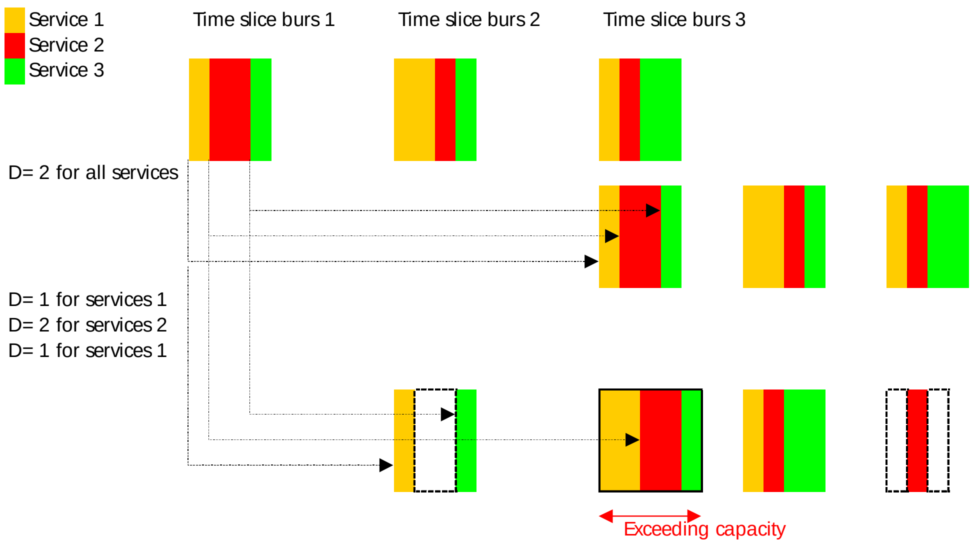


Figure 6.41: Influence of D in a group of statmuxed services

To summarize, the terrestrial physical interleaver combined with a link layer interleaver provides sufficient flexibility for supporting VBR and statmux in possibly different ways, some similar to the DVB-H way.

6.3.4.3 Uniform long interleaver

As expressed in clause 6.3.3.2 "Long" physical interleaver, the uniform long interleaver "breaks" the time slicing signalling. Unless power saving is not a key criteria of the system, in order to support VBR, DVB-H services must be grouped in "DVB-SH services" as presented in clause 7.2.3.3.1. Then it is possible to perform VBR among the services of a same DVB-SH service. The impact on the statistical multiplexing will depend on the size of the DVB-SH service and the number of DVB-H services grouped within, it will grow as the DVB-SH service becomes smaller in size. Actual performance of such a mapping is out of scope of this release.

In case power saving is not a key criteria, there is no need to group DVB-H services in DVB-SH services and each DVB-H service can freely vary in size: the system is then completely compatible with DVB-H statistical multiplexing.

6.3.5 Conclusion

Table 6.20 summarizes the impact of the DVB-SH system on the different features of the time slicing and related link layer features.

Table 6.20: Impact on link layer

	class 1	class 2
MPE Signalling	Ok	MPE Signalling is obsolete, rely rather on SHIP
Power saving	ok but depends on repetition_interval Degradation in erasure channel according to % of completely_lost_burst	Depends on mapping between DVB-H/SH services rather independent of channel burst loss statistics due to fixed structure
Statistical multiplexing with time slicing	full support	Depends on mapping between DVB-H/SH services

Class 1 has less impact / better support of the time slicing and related features: all key features of the link layer can be satisfied at the same time. The class 1 receiver can fully exploit the time slicing and its related features but bandwidth efficiency and power saving may be degraded in certain LMS channels.

Class 2 can compensate the "obsolescence" of the time slicing via the mapping of DVB-H over DVB-SH services, but the mapping optimization goes in two different directions depending if power saving or statistical multiplexing is sought. Actual positioning will then depend on system constraints and in particular on terminal capacity to sustain more memory and more battery requirements: a class 2 terminal having no battery restriction and being always on will be as efficient with statistical multiplexing as a class 1. For other class 2 receiver, a proper balance has to be found between power saving and the gain of statistical multiplexing, dependent on the application scenario.

6.4 Mobility

Mobility is supported in a similar way as DVB-H and subject of a specific implementation guideline. Specificities of DVB-SH mobility is the usage of hybrid_delivery_descriptor that is similar to a terrestrial and/or satellite delivery descriptor and usage of SDT service_availability. Such specification is for further study.

7 Physical Layer elements

This clause summarizes the different building blocks used in DVB-SH. Most of the blocks have already been used within other DVB standards EN 302 304 [3] and EN 302 307 [6] and are considered to be sufficiently well described in the references. Those elements which have been introduced to DVB within the DVB-SH waveform are described with a high degree of detail.

This clause is normative unless stated otherwise.

7.1 Overview to the physical layer elements

An overview on the physical layer technologies of the DVB-SH system is given in tables 7.1 and 7.2.

The different technology submodules are grouped as follows:

(A) encapsulation, forward error protection, interleaving and frame adaptation;

(B) OFDM modulation including TPS and reference signal insertion as well as Fourier Transform processing;

(C) TDM modulation including Pilot field insertion and roll-off filtering.

Table 7.1: Technology sub-modules and their descriptions (part 1)

Category	Technology sub-module	Description	Related features
(A)	Turbo code with block length of 12 282 bits	Subset of 3GPP2 turbo code has been selected as FEC scheme.	High power efficiency. Block length of 12 282 bits offers also high flexibility for time interleaver design.
(A)	Turbo code with code rates between 1/5 and 2/3	Wide range of code rates with stepping of approx. 1…1,5 dB in terms of required energy per code bit over $N_{0.}$	High receiver sensitivity for low code rates. Low code rates allow powerful time interleaver design and outage protection at physical layer. Higher code rates are selected when link layer protection is used.
(A)	Block code structure suitable for MPEG-TS encapsulation	Block structure allows to encode 8 MPEG-TS packets in one turbo encoded word. Block code framing is aligned to the SH framing.	Diversity combining is simplified. Synchronization information for hand-over and combining can be derived from the framing.
(A)	CRC for each MPEG-TS packet	Additional error detection mechanism.	Allows support of error mitigation techniques or link layer protection.
(A)	Flexible time interleaving	Time interleaving of different length is applied at physical layer.	Flexible exchange of fading protection between physical layer and link layer.
(A)	Short time interleaving (approx. 300 ms)	Interleaver length is selected accordingly to available memory. class 1 receivers with reduced physical layer memory size are supported.	Additional link layer protection can be selected to cope with fading channels, especially for the satellite.

Table 7.2: Technology sub-modules and their descriptions (part 2)

Category	Technology sub-module	Description	Related features
(A)	Long uniform time interleaving (e.g. 10 s to 15 s, or longer)	Best "channel averaging" for fading channels and short random signal blockages. class 2 receivers with extended physical layer memory size are addressed.	Combined with low code rates, receiver provides high sensitivity. All signal blockages are handled by the physical layer. Profile may have impacts on the access time after switch-on or recovery after blockage.
(A)	Long uniform/late time interleaving (e.g. 10 s to 15 s, or longer)	Interleaver profile optimized for short zapping time and fast recovery after temporary longer signal blockages. Reduced "channel averaging" for fading channels. class 2 receivers with extended physical layer memory size are addressed.	Combined with low code rates, receiver provides high sensitivity. All signal blockages are handled by the physical layer. Profile allows fast access time after switch-on or recovery after blockage due to the "late" burst contribution.
(B)	Pilot symbol aided OFDM	Waveform identical to DVB-T with changes in the TPS bit description.	Allows reuse of existing OFDM demodulators for DVB-T or DVB-H, with some adaptations.
(B)	Addition of 1 k mode	Waveform for 1 k FFT length added to support higher speeds and/or smaller bandwidths.	Allows reuse of existing OFDM demodulators for DVB-T or DVB-H, with some adaptations.
(B)	Addition of 1,7 MHz channelization	Waveform for L-Band channelization added.	Allows reuse of existing OFDM demodulators for DVB-T or DVB-H, with some adaptations.
(C)	Pilot symbol aided TDM	Waveform derived from DVB-S2 with fixed framing and preamble distance.	Pilot symbol pattern is designed to support synchronization and tracking also at very low C/N values. Regular pilot scheme allows the prediction of framing.
(C)	Modulations QPSK, 8PSK and 16APSK	Waveform allows flexible choice of modulation independent of the physical layer code rate. Modulations are derived from DVB-S2.	Different modulation orders allow to efficiently use available satellite power, independent selection of parameters for satellite and CGC and support for local content insertion.
(B), (C)	Different bandwidth for TDM and OFDM	Usually TDM and OFDM bandwidth is considered equal. In DVB-SH no such restriction applies, TDM and OFDM bandwidth can be selected independently from the specified set. For code-combining the same SH framing applies to both modulations.	Different channel bandwidths for TDM and OFDM allow a higher flexibility using the available bandwidth resources.

7.2 Turbo code, time interleaver and SH framing

7.2.1 Introduction

This clause gives an overview on features which comprises of the encapsulation and encoding of MPEG-TS packets into turbo encoded words. Additionally, the time interleaving operating on turbo encoded words is introduced.

In this clause the description is only dealing with the regular latency DVB-SH content. The specialties related to the low-latency extension are handled in Annex D.

7.2.2 Turbo code

7.2.2.1 Introduction

This clause will explain the characteristics of this FEC solution from the receiver side. Details are also given for combining received code bits from two reception chains (OFDM/OFDM non-SFN or OFDM/TDM) with identical or different puncturing pattern and/or code rates.

The transmission side is described in the waveform document [1], therefore the focus of this clause will be (after some short introduction on the encoder) on the implementation of the turbo decoder.

7.2.2.2 Overview of the key elements

7.2.2.2.1 Top level description

The turbo encoder is a block encoder, which works on a block length of $L_{TC\text{-}input}$ bits. The parameter $L_{TC\text{-}input}$ can be either 1 146 bits (signalling field) or 12 282 bits (payload).

The turbo encoder consists of two Recursive Systematic Convolutional (RSC) encoders, each of the encoders producing an output of three bits. The first RSC produces the bits X, Y_0 and Y_1. The output of the second RSC encoder is called X', Y_0' and Y_1', respectively.

The two encoders are connected by an intra-codeword interleaver (called *3GPP2 interleaver* in figure 7.1). The interleaving instruction is given in EN 302 583 [1], clause 5.3.3. For each block length $L_{TC\text{-}input}$, the instructions to shuffle the input bits are different; however this can be described by selection of one parameter and another small lookup table.

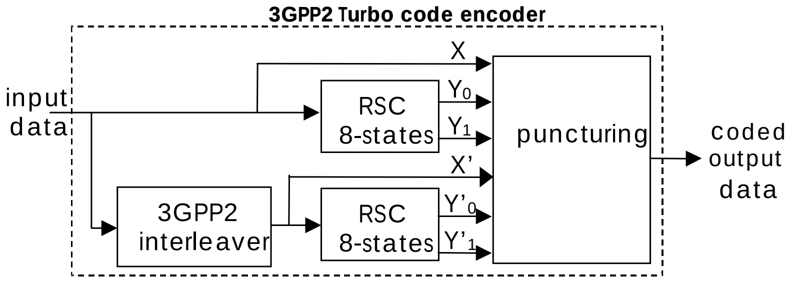


Figure 7.1: Turbo encoder schematic

7.2.2.2.2 Parameters and numbers

The number of output bits at the *internal* interface [X Y_0 Y_1 X' Y_0' Y_1'] is (independently of the selected code rate or puncturing pattern):

$$N_{Outputs}*(L_{TC\text{-}input}+L_{TailBits})$$

This results in the following parameters:

- $N_{Outputs}$ = 6 (size of the internal interface [X Y_0 Y_1 X' Y_0' Y_1'])
- $L_{TC\text{-}input}$ = 1 146 or $L_{TC\text{-}input}$ = 12 282
- $L_{TailBits}$ = 6

Using the above formula, it can be derived that for the payload block length $L_{TC\text{-}input}$ = 12 282, the number of bits at this internal interface is 73 728 bits. From these bits, only a fraction is used for transmission. This fraction is derived by the puncturing pattern for the data and the tail bit periods. An example is given in the next clause.

7.2.2.2.3 Examples for puncturing the [X Y0 Y1 X' Y0' Y1'] vector

The method of puncturing the data and the tail periods of the turbo encoder is described in the waveform document [1] for code rates 1/N with 1<N<5. For other code rates like 2/5, the generation of the data and tail periods is more difficult, therefore an additional example is given here.

- Puncturing pattern ID = 6: code rate R=2/5, standard
- Data puncturing pattern:
  1;0;0;0;0;0; 1;0;1;0;0;1; 0;0;1;0;0;1;
  1;0;1;0;0;1; 1;0;1;0;0;1; 0;0;1;0;0;1;
  1;0;1;0;0;1; 1;0;1;0;0;1; 0;0;1;0;0;1;
  1;0;1;0;0;1; 1;0;1;0;0;1; 0;0;1;0;0;1
- Tail puncturing pattern:
  1;1;1;0;0;0; 1;1;1;0;0;0; 1;0;1;0;0;0;
  0;0;0;1;1;1; 0;0;0;1;1;1; 0;0;0;1;0;1

The puncturing patterns are organized in groups of 6 elements, each such group specifying the puncturing of the 6 internal output bit vector [X Y_0 Y_1 X' Y_0' Y_1'] for one data or tail bit.

The repetition period of the data puncturing pattern is 12 input bits. For the turbo input word length of 12 282 bits, the first 12 276 input bits are processed by repeating the data puncturing pattern 1 023 times (floor(12 282/12)). For the remaining 6 input bits (the 12 277th to the 12 282th input bit), only the first 6 groups of the data puncturing pattern are used, before the tail puncturing pattern is applied. The number of generated bits using the data puncturing pattern is 30 704 bits (selecting punct_Pat_ID = 6).

The tail puncturing pattern is used for the 6 tail bits only, and it is not repeated. The number of generated bits using the tail puncturing pattern is 16 bits (selecting punct_Pat_ID = 6).

In total, the turbo encoder generates 12 288/R = 30 720 bits.

Please note that the number of bits generated using the data puncturing pattern is *not always* 12 282/R and the number of bits generated using the tail puncturing pattern is *not always* 6/R for all selections of the puncturing pattern ID, however the overall number of generated bits using the data and the tail puncturing pattern is *always* 12 288/R.

The short block length of 1 146 input bits is only used together with puncturing pattern ID = 0, such that the statements above only apply to the block length of 12 282 bits.

7.2.2.2.4 Bit-wise interleaver

The bit-wise interleaver is an intra-codeword interleaver which operates on the punctured output of the turbo encoder. The interleaver increment is selected according to the code rate and therefore according the turbo code word length at the output of the puncturing. Its task is to prepare the code word for the use together with the convolutional time interleaver and the transmission over channels with burst erasure behaviour.

Taking into account the interleaver profiles, it can be seen that - what concerns one code word - the distribution of the encoded bits in time is usually not equally spaced but somehow grouped:

- first of all, the minimum unit to be processed by the time interleaver is one interleaving unit (IU) of 126 bits each;
- additionally, the use of time slicing groups several of these IU onto one relatively small burst.

If one or more IUs or time slices are lost, the performance of the turbo decoder may suffer from such bursty data losses. The approach of the bit-wise interleaver is to transform bursty losses on the transmission channel into "approximately evenly distributed" bitwise erasures at the input of the turbo decoder in the receiver. This loss pattern can be recovered with higher probabilities by the turbo decoder, dependent on the reception condition of those bits which have not been erased or exhibited deep fades.

The bit-wise interleaver is designed such that the distance between erased bits of a burst is maximized; this is done by choosing an interleaver increment which is relatively prime to the codeword length after puncturing.

One example for code rate 1/5 is given in the figure 7.2. The parameters are:

- block length N_{TCB} after turbo encoder and puncturing: 61 440;
- bit-wise interleaver increment: 247.

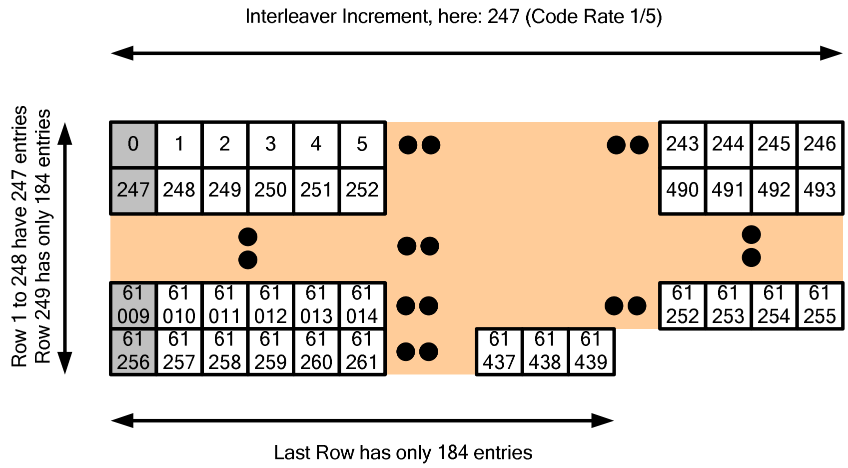


Figure 7.2: Bit-wise interleaver

The first 300 index positions are given in the following list:

$b_0 \ldots b_{299} = a_{H(w)}$ with H(w) = {

0	247	494	741	988	1235	1482	1729	1976	2223	2470	2717	2964	3211	3458	3705
3952	4199	4446	4693	4940	5187	5434	5681	5928	6175	6422	6669	6916	7163	7410	7657
7904	8151	8398	8645	8892	9139	9386	9633	9880	10127	10374	10621	10868	11115	11362	11609
11856	12103	12350	12597	12844	13091	13338	13585	13832	14079	14326	14573	14820	15067	15314	15561
15808	16055	16302	16549	16796	17043	17290	17537	17784	18031	18278	18525	18772	19019	19266	19513
19760	20007	20254	20501	20748	20995	21242	21489	21736	21983	22230	22477	22724	22971	23218	23465
23712	23959	24206	24453	24700	24947	25194	25441	25688	25935	26182	26429	26676	26923	27170	27417
27664	27911	28158	28405	28652	28899	29146	29393	29640	29887	30134	30381	30628	30875	31122	31369
31616	31863	32110	32357	32604	32851	33098	33345	33592	33839	34086	34333	34580	34827	35074	35321
35568	35815	36062	36309	36556	36803	37050	37297	37544	37791	38038	38285	38532	38779	39026	39273
39520	39767	40014	40261	40508	40755	41002	41249	41496	41743	41990	42237	42484	42731	42978	43225
43472	43719	43966	44213	44460	44707	44954	45201	45448	45695	45942	46189	46436	46683	46930	47177
47424	47671	47918	48165	48412	48659	48906	49153	49400	49647	49894	50141	50388	50635	50882	51129
51376	51623	51870	52117	52364	52611	52858	53105	53352	53599	53846	54093	54340	54587	54834	55081
55328	55575	55822	56069	56316	56563	56810	57057	57304	57551	57798	58045	58292	58539	58786	59033
59280	59527	59774	60021	60268	60515	60762	61009	61256	63	310	557	804	1051	1298	1545
1792	2039	2286	2533	2780	3027	3274	3521	3768	4015	4262	4509	4756	5003	5250	5497
5744	5991	6238	6485	6732	6979	7226	7473	7720	7967	8214	8461	8708	8955	9202	9449
9696	9943	10190	10437	10684	10931	11178	11425	11672	11919	12166	12413}				

7.2.2.3 Combining at the input of the turbo decoder

This clause explains decoding strategies in the presence of more than one demodulator and received signal.

7.2.2.3.1 Overview

Figure 7.3 gives a first overview on the combining possibilities. The architecture shown is a placeholder for real implementations but tries to sketch the principles behind combining at the turbo decoder input.

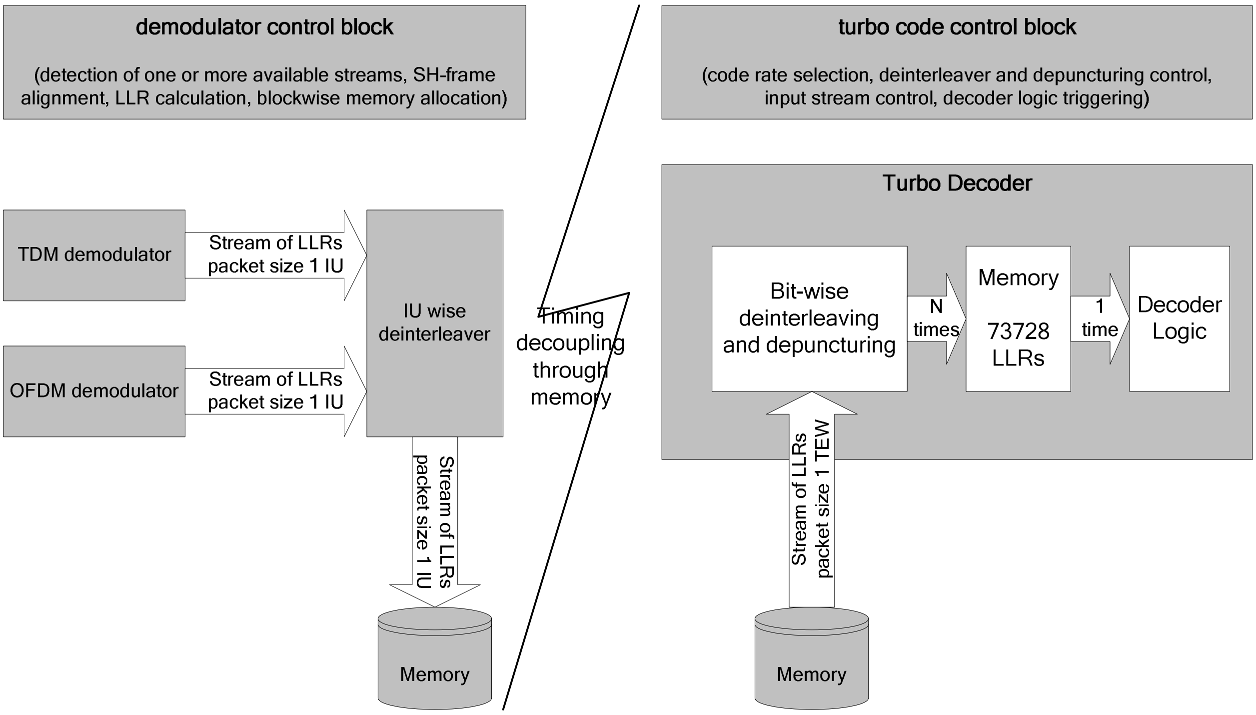


Figure 7.3: Possible turbo decoder integration into the receiver

Even in the presence of N>1 demodulated signals, the turbo decoder logic must be run only once after the combining of demodulated streams has taken place. In DVB-SH scenarios, either N = 1 or N = 2 demodulated signals are typically available at the receiver input:

- TDM for satellite transmission, OFDM for terrestrial transmission;
- OFDM for satellite and terrestrial transmission (in MFN mode).

As far as antenna diversity is considered, the combining of more than one antenna may take place before or after the demodulator stage, resulting in one or two demodulators from the turbo decoders' perspective.

7.2.2.3.2 Implementation

The turbo decoder itself can be considered to be made-up by various internal blocks. Their functionalities are:

- **Bit-wise deinterleaving:** the bit-wise deinterleaving reverts the bit-wise interleaving of the transmitter. Dependent on the selected puncturing pattern ID, this stage works on block sizes between 18 432 bits and 61 440 bits which can be directly read from the deinterleaver memory. This bit-wise deinterleaving is invoked several times per turbo decoding process, dependent on the number N of demodulators active for the same code word.
- **Depuncturing:** the depuncturing reverts the puncturing process of the transmitter. Dependent on the selected puncturing pattern ID, this stage works on the same input block size as the bit-wise deinterleaving. It is also invoked several times per turbo decoding process, if more than one demodulator is active.

- **Combining:** the combining reads the LLR entries already present in the decoder input memory (corresponding the codewords from those demodulators that have already been combined) and adds the LLRs just read from the interleaver memory for one codeword received by one of the N demodulators. Hence a codeword from another demodulator is combined with the codewords of those demodulators that have been combined by this block before for the current turbo info word. The combined LLRs are then again stored in the decoder input memory. This process is repeated until the N codewords from the demodulators have been combined that are associated with one turbo info word. Note that this process can even be carried out without performance loss when one of the demodulators does not receive a signal: in this case, the LLRs from this demodulator are zero, hence the LLRs from the other demodulators are not changed by adding the zero-codeword from the inactive demodulator.

- **Decoder Input Memory (either input and working memory or both memories combined):** this memory has the size of the non-punctured turbo encoder code word of 12 288*6 = 73 728 LLR values. Dependent on the architecture and the speed of the decoder logic, this memory has either to be instantiated twice (double buffering) or only once (see figure 7.4). This memory is read and written by the depuncturing stage several times per turbo decoding process, but read only once by the decoder logic when no other input is available and the decoder logic can start.

- **Decoder logic:** the decoder logic is called once all available demodulated streams have been processed and combined on the memory. The output is forwarded to the next stage in the receiver, e.g. the CRC16 check and the decapsulation.

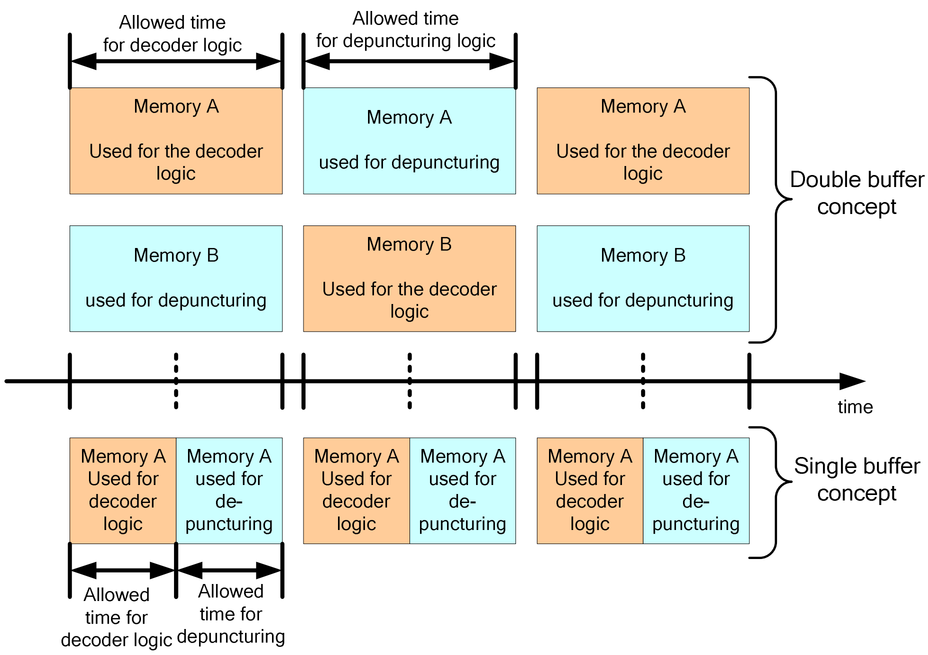


Figure 7.4: Possible turbo decoder memory architecture

7.2.2.3.3 Maximum ratio combining and complementary code combining

Combining of two signals is done by the accumulation of all LLRs available for one bit; it is a simple addition of LLR values. This is performed as depicted in figure 7.5. Please note that the figures do not represent exactly the puncturing patterns nor the code word structures introduced in DVB-SH, but should give an idea on the way combining works.

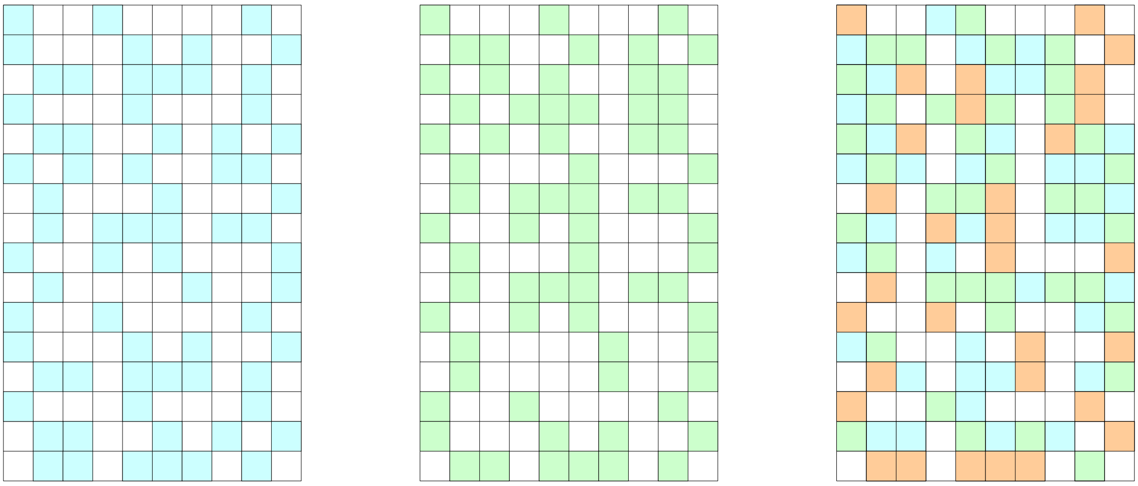


Figure 7.5: Received LLRs from demodulator 1 (blue), received LLRs from demodulator 2 (green) Combined LLRs from both modulators (blue, green, orange)

The different colours represent the LLRs received from different demodulators:

- **figure on the left hand side:** the blue squares represent LLRs received from demodulator 1. This codeword is a valid codeword which can be decoded at the threshold $(C/N)_1$ for the selected code rate (e.g. 1/2);
- **figure in the middle:** the green squares represent LLRs received from demodulator 2. This codeword is a valid codeword which can be decoded at the threshold $(C/N)_2$ for the selected code rate (e.g. 2/5);
- **figure on the right hand side:** all squares in colour represent the combined LLRs available from both demodulators. This codeword is a valid code word which can be decoded at the threshold $(C/N)_3$ for the combined code rate. The colours represent the following:
  - **orange:** these LLRs have been received by **both** demodulators and have been combined by pure addition of the LLRs. Addition of these LLRs is equivalent to **maximum ratio combining** of the demodulated signals;
  - **green/blue:** these LLRs have either been received through demodulator 1 or 2. Each green or blue square complements the other received code word. These LLRs are **complementary code combined**.

Dependent on the choice of puncturing pattern IDs on the transmit side, the percentage of LLRs being maximum ratio combined or complementary code combined can vary. In general, maximum ratio combining of the complete codeword is a special case of combining codewords for the following configuration:

- identical code rates on the different transmission paths;
- identical puncturing pattern IDs on the different transmission paths.

7.2.2.4 Selection of turbo code rate together with link layer protection

The DVB-SH standard offers a wide range of combinations for code rates both on the physical layer and the link layer. The overall spectral efficiency depends on both protection mechanisms and the selection of modulation order. A short computation example for the code rate selection is given here:

- physical layer code rate: $R_{PHY} = 1/2$;
- link layer code rate: $R_{IFEC} = 2/3$;
- overall code rate: $R_{OVA} = 1/2 * 2/3 = 1/3$.

The proper choice of code rates and modulation orders is the key for the solution of the trade-off between:

- high spectral efficiency (by selection of high code rates with lower protection);
- high robustness in typical DVB-SH environments (by selection of low code rates with high protection).

This choice is further complicated by the fact that the redundancy can be assigned seamlessly between physical layer and link layer. Some recommendations (for QPSK modulation order) are given here:

- assignment of **all redundancy** to the physical layer (typical R_{PHY} = 2/5 or lower):
  - should be chosen if long physical layer interleavers are used;
  - should be chosen if higher values of Doppler spread in OFDM have to be supported (see clause 7.3.1.3);
- assignment of **the larger part** of the overall redundancy to the physical layer (typical R_{PHY} = 1/2):
  - should be chosen if the link layer FEC protection is relatively low (typical R_{IFEC} around 2/3);
  - should be chosen if long physical layer interleavers cannot be used, e.g. due to memory constraints;
  - may degrade the C/N thresholds for error-free reception as higher C/N values for static and mobile reception are necessary;
  - impacts the satellite link budgets and terrestrial coverage planning;
- assignment of **the smaller part** of the overall redundancy to the physical layer (typical R_{PHY} = 2/3):
  - should be chosen if the link layer FEC code rates is relatively high (typical R_{IFEC} around 1/2);
  - should be chosen if long physical layer interleavers cannot be used, e.g. due to memory constraint;
  - severely degrades the C/N thresholds for error-free reception as higher C/N for static and mobile reception are necessary;
  - impacts the satellite link budgets and terrestrial coverage planning.

According to the results presented in clause A.12, the selection of code rates on the physical layer has to be made according to the following criteria:

- available satellite link budgets (given EIRP) and terrestrial coverage planning (given repeater EIRP and density);
- physical layer margins necessary for certain reception scenarios and receiver classes;
- desired spectral efficiency and desired reception quality.

The choice to select the split of redundancy between the physical layer and the link layer will mainly be driven by the terminal classes addressed. The selection guide between solutions is for further study.

7.2.2.5 Processing at PHY to support Erasure decoding in UL

Being the DVB-SH interface between Physical and Link layer, the MPEG-TS is the only mean to transport reliable error information (synchronized with the data payload) between these two layers.

The DVB-SH frame supports an additional CRC16 error detection mechanism at the level of each individual TS. By checking this CRC16, the physical layer can detect erroneous TP and set accordingly the transport_error_indicator bit. This bit can then be used by following layers, in particular the link layer to optimize decoding process. Therefore, the mechanisms defined in ISO/IEC 13818-1 [8] for MPEG-TS packets should be used in order to signal the integrity of the MPEG-TS payload. Different algorithms are possible and one is proposed hereunder:

The algorithm depicted in figure 7.6 takes both the CRC16 in the EHEADER and the various CRC16 over the User Packets (UP) into account.

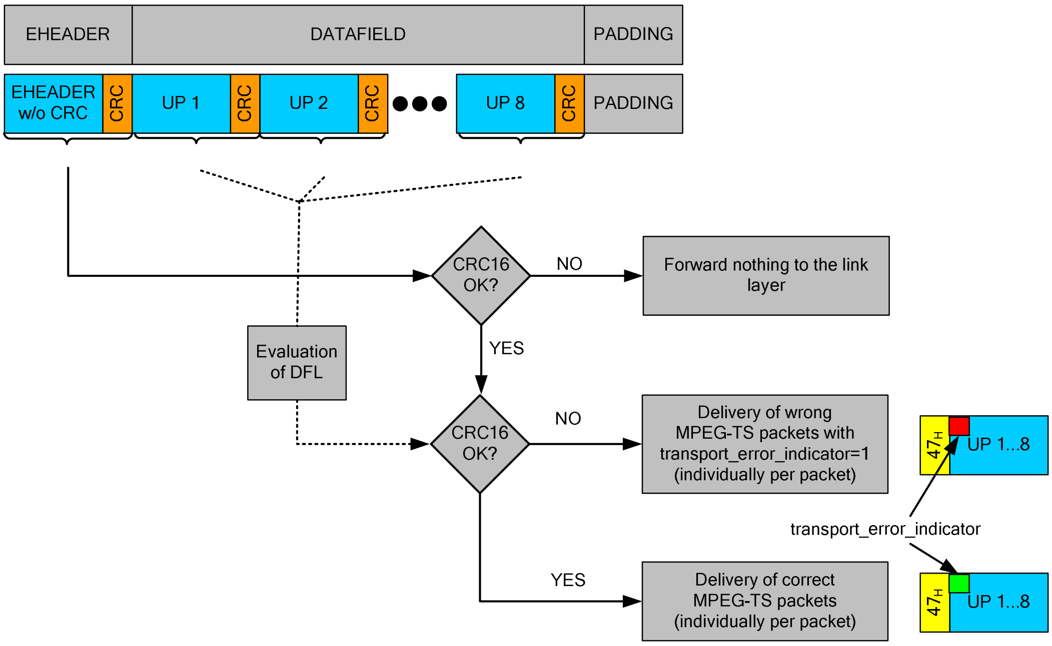


Figure 7.6: Proposed algorithm for decapsulation to support erasure decoding in UL

7.2.2.5.1 Overview on the proposed algorithm

In total, the payload of one turbo code word consists of up to 9 (nine) CRC-checks, each of length 16 bits:

- The first one to be considered is the CRC16 over the first 98 bits of the EHEADER. If this check fails, the EHEADER must be considered to be corrupted. No information on DataField Length (DFL) is available for this turbo code packet, therefore it cannot be derived how many MPEG-TS packets had been encapsulated. No information can be forwarded to the link layer.
- Up to 8 following CRCs16 have to be evaluated next, if the CRC16 over the EHEADER is correct and the SYNC byte has the expected value 47_H. The DFL indicates the number of MPEG-TS packets in the DATAFIELD section. Over each packet of 187 bytes (UP), the CRC16 is calculated. The following applies:
  - if the CRC16 is wrong, the transport_error_indicator has to be set to 1, and the packet has to be delivered to the link layer processing;
  - if the CRC16 is correct, the transport_error_indicator remains untouched (it may have been set already by any other TS handler on the transmit side).

7.2.2.5.2 MPEG-TS packet format

The format of the MPEG-TS transport packet is given in table 7.3.

Table 7.3: MPEG-TS transport packet format

Syntax	No. of bits	Mnemonic
transport_packet(){		
sync_byte	**8**	**bslbf**
transport_error_indicator	**1**	**bslbf**
payload_unit_start_indicator	**1**	**bslbf**
transport_priority	**1**	**bslbf**
PID	**13**	**uimsbf**
transport_scrambling_control	**2**	**bslbf**
adaptation_field_control	**2**	**bslbf**
continuity_counter	**4**	**uimsbf**
if(adaptation_field_control == '10' || adaptation_field_control == '11'){		
adaptation_field()		
}		
if(adaptation_field_control == '01' || adaptation_field_control == '11') {		
for (i = 0; i < N; i++){		
data_byte	**8**	**bslbf**
}		
}		
}		

The relevant flag to be set in case of a CRC16 failure on the user packet (UP) is the transport_error_indicator.

7.2.2.5.3 Generation of MPEG-TS null-Packet

In case of a non-recoverable EHEADER, it is proposed to transmit a so-called MPEG-TS null packet. Although indicated by the name, the payload of a null-packet is not the all-zero sequence but defined as follows:

Byte[0]	Byte[1]	Byte[2]	Byte[3]	Byte[4]	Byte[5] …	Byte[187]
47_H	$1F_H$	FF_H	10_H	FF_H	FF_H …	FF_H

Please note that all bytes between Byte [4] and Byte [187] are set to FF_H.

7.2.2.5.4 Signalling of wrong MPEG-TS packet

In case of an erroneous CRC16 over one of the user packets, the transport_error_indicator has to be set to 1. This is done by an OR-combination of the value 80_H with the MSB (most significant bit) of Byte 1. This helps to preserve any transport_error_indicator that has been set along the transmission chain. The way to set the transport_error_indicator is depicted in figure 7.7.

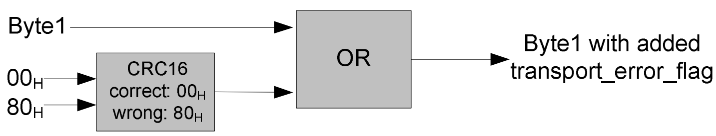


Figure 7.7: Setting the transport_error_indicator flag

7.2.2.6 C/N performance values

7.2.2.6.1 Ideal performance in AWGN channel

For AWGN channel an ideal receiver should have the theoretical performance given in table 7.4 for OFDM and table 7.5 for TDM. The values reported in tables 7.4 and 7.5 exclude any LL-FEC. An ideal transmitter is assumed, as well as an ideal receiver with perfect synchronization and ideal channel knowledge.

Pilot overhead in C/N is only taken into account what concerns the boosting of pilots in OFDM (0,3 dB) which reduces the Es/N_0 of the data carriers with respect to the C/N. Any other overhead (signalling, guard intervals) is addressed in the spectral efficiency curves.

Table 7.4: Theoretical C/N (dB) in AWGN channel for OFDM @ BER = 10^{-5}

Code rate	QPSK	16QAM
1/5	-3,6	0,7
2/9	-3,1	1,3
1/4	-2,5	1,9
2/7	-1,8	2,8
1/3	-0,9	3,7
2/5	0,1	5,0
1/2	1,4	6,8
2/3	3,5	9,7

Table 7.5: Theoretical C/N (dB) in AWGN channel for TDM @ BER = 10^{-5}

Code rate	QPSK	8PSK	16APSK
1/5	-3,9	-1,3	0,4
2/9	-3,4	-0,7	1,0
1/4	-2,8	-0,1	1,6
2/7	-2,1	0,7	2,5
1/3	-1,2	1,6	3,4
2/5	-0,2	2,7	4,7
1/2	1,1	4,4	6,5
2/3	3,2	6,9	9,4

7.2.2.6.2 Ideal performance in Rice channel

Only results for TDM (QPSK) over Rice channel are available and listed in table 7.6. The simulation assumes ideal channel estimation, perfect interleaving/deinterleaving (i.e. uncorrelated fading), the Rice factor K used is 3 dB.

Table 7.6: Theoretical C/N (dB) in Rice channel (K=3 dB) for TDM @ BER = 10^{-5}

Code rate	QPSK
1/5	-3,4
1/4	-2,2
1/3	-0,4
1/2	2,2

7.2.2.6.3 Ideal performance in Rayleigh channel

Only results for TDM (QPSK) over Rayleigh channel are available and listed in table 7.7. The simulation assumes ideal channel estimation and perfect interleaving/interleaving (i.e. uncorrelated fading), the Rice factor used is minus infinity dB (equivalent to a Rayleigh distribution).

Table 7.7: Theoretical C/N (dB) in Rayleigh channel for TDM @ BER = 10^{-5}

Code rate	QPSK
1/5	-3,2
1/4	-2,1
1/3	-0,2
1/2	2,9

7.2.2.6.4 Ideal performance in burst erasure channel

Only results for OFDM (QPSK) in burst erasure are available. The simulation assumes ideal channel estimation. Table 7.8 gives an example for the OFDM performance for the burst erasure channel. This channel removes parts of the redundancy directly at the input of the deinterleaver. One modulation (QPSK) and one code rate (1/3) has been selected to demonstrate the performance of the turbo code. The 1/3 FEC code rate can cope with a maximum erasure rate of (1-1/3)*100= 66 %. Approaching this erasure rate the required C/N to achieve the wanted Word Error Rate (WER) is rapidly increasing to the C/N required for the uncoded case. Each turbo code word of rate 1/3 consists of 18 capacity units, which equals 288 interleaver units.

The table has to be read as follows:

- the first and second rows show the selected erasure rate;
- the third row calculates the remaining effective code rate, i.e. the ratio of transmitted information bits over the received (non-erased) code bits: code rate / (1-erasure rate), after the applied burst erasure;
- the fourth row displays the minimum C/N necessary to reach a word error rate (WER) of 10^{-3};
- the fifth row displays the additional C/N necessary compared to AWGN performance (= no erasures).

Table 7.8: Performance of turbo codes on burst erasure channel (ideal interleaving)

Erased interleaver units	0/288	24/288	48/288	72/288	96/288	120/288	144/288
Erasure rate in per cent	0 %	8,3 %	16,6 %	25 %	33,3 %	41,6 %	50 %
Effective Code rate	0,333	0,3636	0,4000	0,4444	0,5000	0,5714	0,6667
Minimum C/N req. for WER=10^{-3}	-0,9 dB	0,0 dB	0,6 dB	1,4 dB	1,8 dB	3,0 dB	4,4 dB
Additional C/N req. compared to AWGN	0,0 dB	0,9 dB	1,5 dB	2,3 dB	2,7 dB	3,9 dB	5,3 dB

This table is a good indicator of the receiver performance in the AWGN and a channel experiencing a certain percentage of time signal blockage and ideal interleaving. For AWGN channel, the actual required C/N to achieve a certain WER results in the implementation loss achieved, whereas the block erasure channel allows to derive the implementation loss in burst fading channels.

The latter also helps to evaluate the performance of the time interleaver described in clause 7.2.3 Time Interleaver. The interleaver profiles are designed such that - depending on the current reception condition and/or receiver memory - not all interleaver units are necessary to successfully decode the turbo code word. This allows to rapidly decode data on the physical layer in case of receiver switch-on or change of service, without waiting for a full interleaver delay before starting to decode. The same argument can be used to evaluate the performance degradation of memory-limited receivers that are forced to drop part of the transmitted interleaver units.

The performance of the receiver in this burst erasure channel scenario is crucial for the switch-on time (zapping time) of the receiver; the expected behaviour is explained further in details in clause 7.2.3.5.4 Late decoding.

7.2.3 Time Interleaver

7.2.3.1 Introduction

Time interleaving in general helps to counteract the effects of signal fading and shadowing/blockage on the transmission channel. Reception instants with bad signal quality can be counteracted with periods of time with good signal reception as their time distribution is over the FEC frame "randomized" by the time interleaver. To have successful FEC frame reception the % of erased (bad) symbols shall be lower than the FEC erasure correction capabilities (see clause 7.2.2.6.4).

Interleaving always has to be aligned with the assigned code rate and modulation scheme used for transmission, as well as with the expected reception field strength:

- if a system is working with relatively low link margins (difference between the received and the required signal power strength for error-free reception is lower than 4 dB), interleaving over a long time span (e.g. 10 s) normally will improve reception. Even though in special cases it may worsen the reception as it spreads short errors over a long time span. See the top drawings of figure 7.8 for details;

- if indeed a system works with higher link margins (difference between the received and the required signal power strength for error-free reception is higher than 7 dB), interleaving over a long time span (e.g. 10 s) can strongly improve the reception as it is able to recover many bad reception conditions. See the bottom drawings of figure 7.8 for details.

Interleaving can be seen as time-averaging of the received time variant C/N. It averages out good and bad reception conditions over the duration of the time interleaver., In practice the C/N does not follow an on-off characteristics, but has a more complex time dependency related to the environment, user speed and associated channel state, shadowing and fading parameters. These channel first and second order statistics are reflected by the channel models described in the clause A.7. As shown in the previous clause, the FEC is designed to cope with a certain erasure percentage, dependent on the code rate selected for the physical layer FEC.

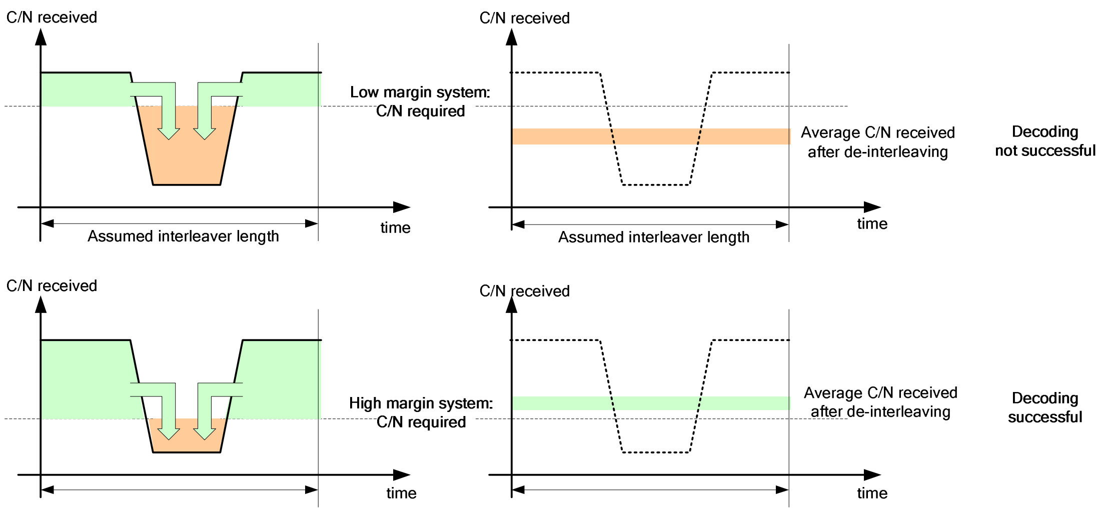


Figure 7.8: Interleaving with different margins; top: not sufficient margin; bottom: sufficient margin

Physical layer interleaving (which is addressed here) makes use of the fact that one turbo code word is cut into various pieces (up to 48, identical to the number of interleaver taps) and spread over many distinct time instants. In the receiver, these pieces are again collected, reordered into the original sequence and forwarded to the turbo decoder.

Dependent on the capabilities of the receiver (refer to the receiver Class definition), a decision has to be taken whether the receiver is able or not to successfully deinterleave the selected SH signal. Details on typical interleaver profiles are given in clause 7.2.3.3.5 Typical interleaver profiles, deinterleaving strategies are given in clause 7.2.3.4.3 Interleaver synchronization on the receiver side, and details on the compatibility of the receiver with the transmitted signal is given in clause 7.2.3.4.

7.2.3.2 Selection of receiver classes

7.2.3.2.1 Overview

Two different types of receivers are addressed in TS 102 585 [2]:

- class 1 receivers can work with a number of interleaver units up to 6 528, each of 126 bits, which is the equivalent to 1/2 (half) of the SH-frame capacity. Two types of interleaver configurations in the transmit signal are compatible with this type of receiver:
    - short uniform interleaver;
    - the late burst of a long uniform-late interleaver;
    - the late burst of an early-late interleaver.
- class 2 receivers can work with a number of interleaver units up to 417 792, each of 126 bits, which is the equivalent to the capacity of 32 SH-frames. All types of interleaver configurations in the transmit signal (which are within the range of the memory storage capabilities of the receiver) are compatible with this type of receiver:
    - short uniform interleaver;
    - long uniform-late interleaver;
    - long uniform interleaver;
    - long early-late interleaver.

7.2.3.2.2 Class 1/Class 2 compatible interleaver profile selection

As discussed within clause 7.2.2.6.4 Ideal performance in burst erasure channel, class 1 receivers may still operate with interleaver profiles intended primarily for class 2 receivers. Therefore, it was introduced as a mandatory requirement that any receiver not only derives the full list of interleaver taps, but cross-checks this with its own memory and decoding capabilities. The expected behaviour of any terminal (class 1 / class 2) is the following:

- calculate all interleaver taps as signalled in either TPS (OFDM) or signalling field (TDM);
- drop `k` taps which exceed the memory capabilities of this receiver type and keep only `L[0]` to `L[47-k]`;
- check whether the remaining code rate is still decodable (remaining code rate < 1,0);
- decode the remaining fragment of turbo code word if both the memory and the remaining code rate allow this.

One example for the performance penalty of such a configuration has been given in table 7.8. Assigning e.g. a late burst contribution of 50 % of all taps (*nof_late_taps* = 24), which can all be decoded by a class 1 receiver, the performance loss would be 5,3 dB (see clause 7.2.2.6.4 Ideal performance in burst erasure channel) for an effective code rate of 2/3 instead of a total code rate of 1/3. These values may differ for the selection of other code rates and/or assignment of taps to the late part, and depend on the channel conditions.

The benefit of such a network configuration is that both class 1 and class 2 receivers can co-exist in a network primarily intended for class 2 receivers, allowing also a soft-transition between the classes.

7.2.3.3 Interleaver profile description using the waveform parameters

7.2.3.3.1 Overview

For all transmitted signals in a DVB-SH system, the interleaver parameters can be chosen and signalled independently. Certain rules apply when setting the interleaver parameters for signals that are intended to be combined in the receiver prior to FEC decoding, see clause 7.2.2.3.3 Maximum ratio combining and complementary code combining for details.

The difference in the signalling also results from the different SH frame lengths in OFDM and TDM mode, which are defined in multiple of CU (capacity units) and IU (interleaver units) and not in units of time. Defining identical interleaver profiles for OFDM and TDM will typically lead to slightly different interleaver length (in time). However, the maximum memory capabilities of the receiver classes have to be respected.

The interleaver parameters are always defined from the receivers perspective, giving clear guidance on the selection of parameters for the de-interleaving process.

Long physical layer interleaver contradicts with variable burst sizes and distances of DVB-H when power saving is addressed. However, mapping techniques of DVB-H over DVB-SH can partially compensate the contradiction. In general, time slicing (as introduced by DVB-H and preserved in DVB-SH) and long time interleaving (as introduced within DVB-SH) seem to be contradictory:

- *time slicing* tries to concentrate the data to reduce the on-time of a receiver, thus lowering significantly the power consumption;
- *time interleaving* tries to spread the data over a large amount of time to benefit from time-variant channels.

The tools provided in DVB-SH provide the necessary flexibility in the choice of time slicing and time interleaving jointly. Therefore, the concept of *SH-service* was introduced, signalled by the *service_synchronization_function*. Each SH-service has a constant repetition period RP_{SH} and may cover a number of time-sliced services. These SH-services are then fed into the time interleaver who's repetition period RP_{TI} should be identical to the repetition period of the SH-service.

From the transmitter's perspective, this categorization of services looks like n virtual transmission channels with reduced throughput. By selecting a reasonable number of SH-services, this SH-service category aggregates in the transmitter:

- a small number (2 to 3) of DVB-H services within one SH-service.

By choosing 1 out of n present SH-services, this SH-service category allows the receiver:

- to decode only parts of the whole multiplex (reducing significantly the throughput requirements and deinterleaver memory necessary);
- to benefit from time slicing on an SH-service by SH-service basis (less but still considerable power savings when compared to DVB-H time slicing mode).

Figure 7.9 displays the relationship between time slicing services and SH-services, between SH-services and SH frames, and the output of the time interleaver. The mapping and parameter selection is the following:

- n_1 DVB-H services map onto *one* SH-service; n_1 may be different for each SH-service.
- *One* SH-service maps onto n_2 EEFRAMEs; n_2 may be different for each SH-service.
- The repetition period RP_{SH} for each SH-service is **identical** in the number n_3 of SH frames.
- The time interleaver works with the repetition period RP_{TI}. The repetition period is given in SH frames with the following condition: $RP_{TI} = RP_{SH}$ = slice_distance (with slice_distance being defined in the next clause).

The usage of the parameter set as described above ensures that features like VBR support (at the transmitter) and time slicing (at the receiver) can be, at least partially, exploited jointly with (long) physical layer interleaving.

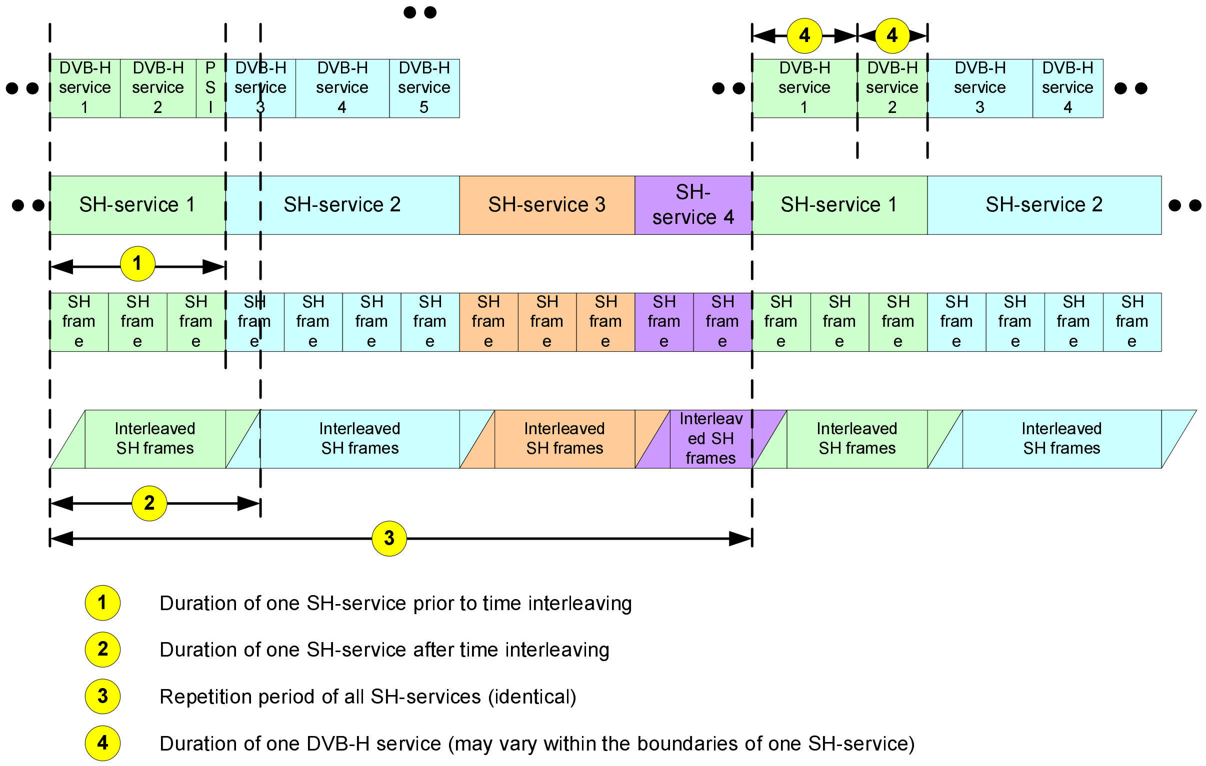


Figure 7.9: Timing relationships between Time-Slice, SH-Service and SH-frame, prior to and after interleaver (short interleaver)

Figure 7.10 displays the relationship between time slicing services and SH-services, between SH-services and SH frames, and the output of the time interleaver for the case where the time interleaver is not a short interleaver. In the example an early-late interleaver is shown. This interleaver spreads a burst into two bursts with distance RP_{SH}.

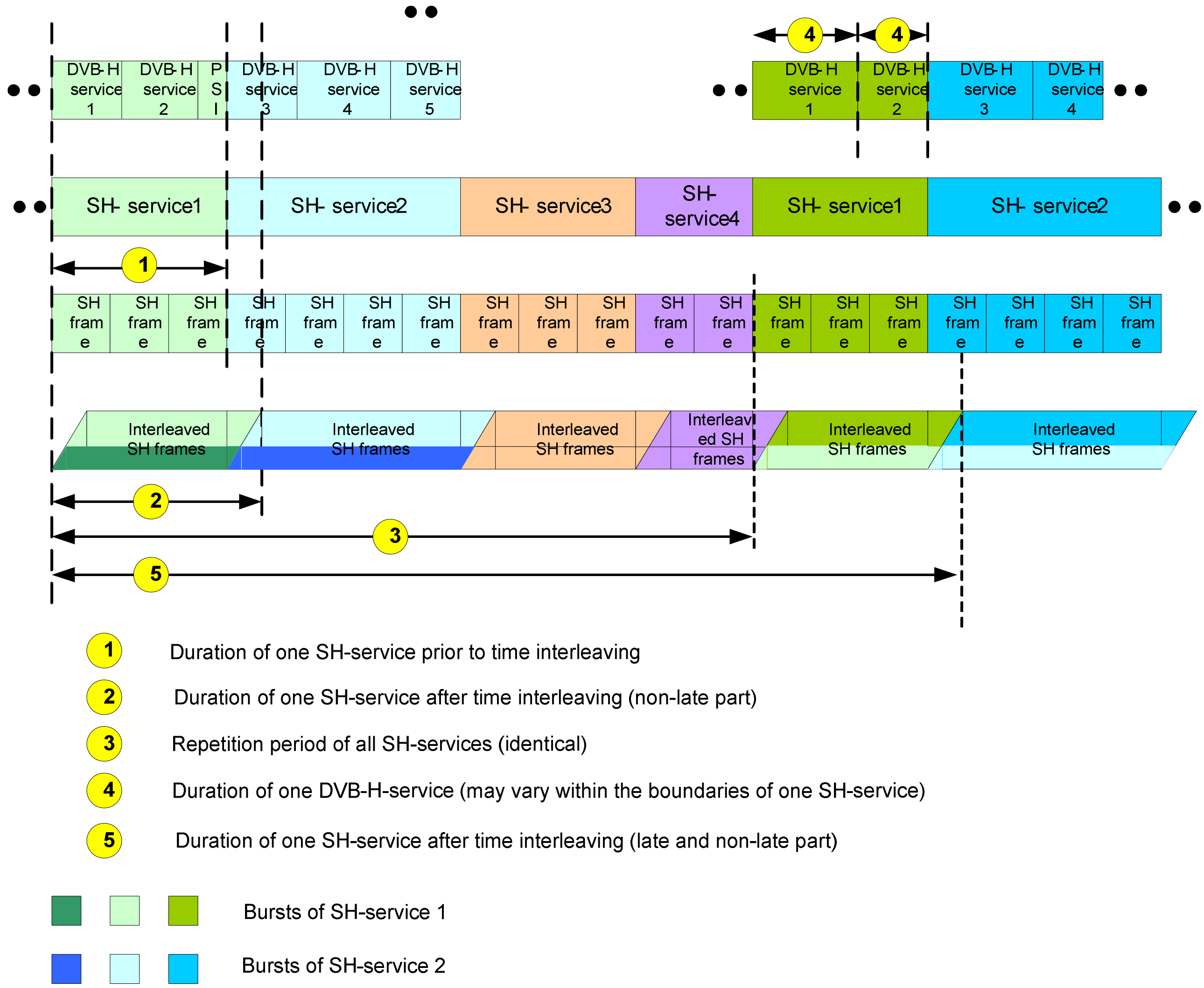


Figure 7.10: Timing relationships between Time-Slice, SH-Service and SH-frame, prior to and after interleaver (e.g. early-late interleaver)

7.2.3.3.2 Parameters and description

The waveform document [1] specifies the following table of parameters for the description of the interleaver.

- **Common_multiplier:** this parameter is a multiplier of the number of interleaver units for all interleaver taps *within one slice*. The minimum value is 1, the maximum value is 63.
- **Nof_late_taps:** this parameter partitions the interleaver into a *late* and a *non-late* part. The border between the two different parts is somewhere between 0 (no late part available) and 48 (only late part available).
- **Nof_slices:** this parameter defines the number of interleaver slices over which the interleaver distributes the data. This parameter has to be set together with the next parameter slice_distance to determine the interleaver length in time, and has to be aligned to time slicing as described in clause 7.2.3.3.1 Overview Overview. In case of selection of a late part only, this parameter is set to 1, and all subsequent parameters can be set to 0. Also in case of selection of no late part this parameter is set to 1, otherwise this parameter is set > 1 (refer to figure 5.10 and table 5.39 of [1]).

- **Slice_distance:** this parameter defines the distance between two interleaver slices in SH-frames. The capacity (in CU) of an SH-frame depends on the modulation type, modulation order (TDM), and roll-off (TDM) selection. The TDM parameters itself of course depend from modulation order (OFDM), guard interval length (OFDM).To transform this value in an increment in IU, the parameter `nof_Data_CU` has to be calculated according the following clause.

- **Non-late increment:** this parameter is an additional multiplier for all interleaver taps *within all non-late slices*. It has to be multiplied with the common_multiplier to determine the increment in all slices which are *not* the late part of the interleaver.

In order to ease modulator design, it is recommended to have the following constrains applied to the selection of the common_multiplier:

- using OFDM with 16QAM modulation, the common_multiplier must be a multiple of 2 (in 4 k modes) or 4 (in 8 k modes); for all other modes, no constraints apply;

- using OFDM with QPSK modulation, the common_multiplier must be a multiple of 2 (in 8 k modes); for all other modes, no constraints apply.

For alignment of tables 5.9 and 5.39 of [1] it shall be noted that there is no real reason to set parameter non_late_increment = 0 in case where the receiver configuration only applies one slice. Configurations where this is allowed are shown in table 5.39 of [1] and in other available interleaver configurations (see clause A.10).

For better understanding the example in the subsequent section a dependent parameter is introduced.

- **Nof_nonlate slices:** this parameter defines the number of interleaver slices over which the **non-late part** interleaver distributes the data. It is calculated as follows:

  - **Nof_late_taps = 48: then Nof_nonlate_slices = 0.**

  - **Nof_late_taps = 0: then Nof_nonlate_slices = Nof_slices.**

  - **Nof_late_taps = 1..47: then Nof_nonlate_slices = Nof_slices-1.**

7.2.3.3.3 Calculation examples

Figure 7.11 gives an overview on how to derive exactly the tap delays for the time interleaver.

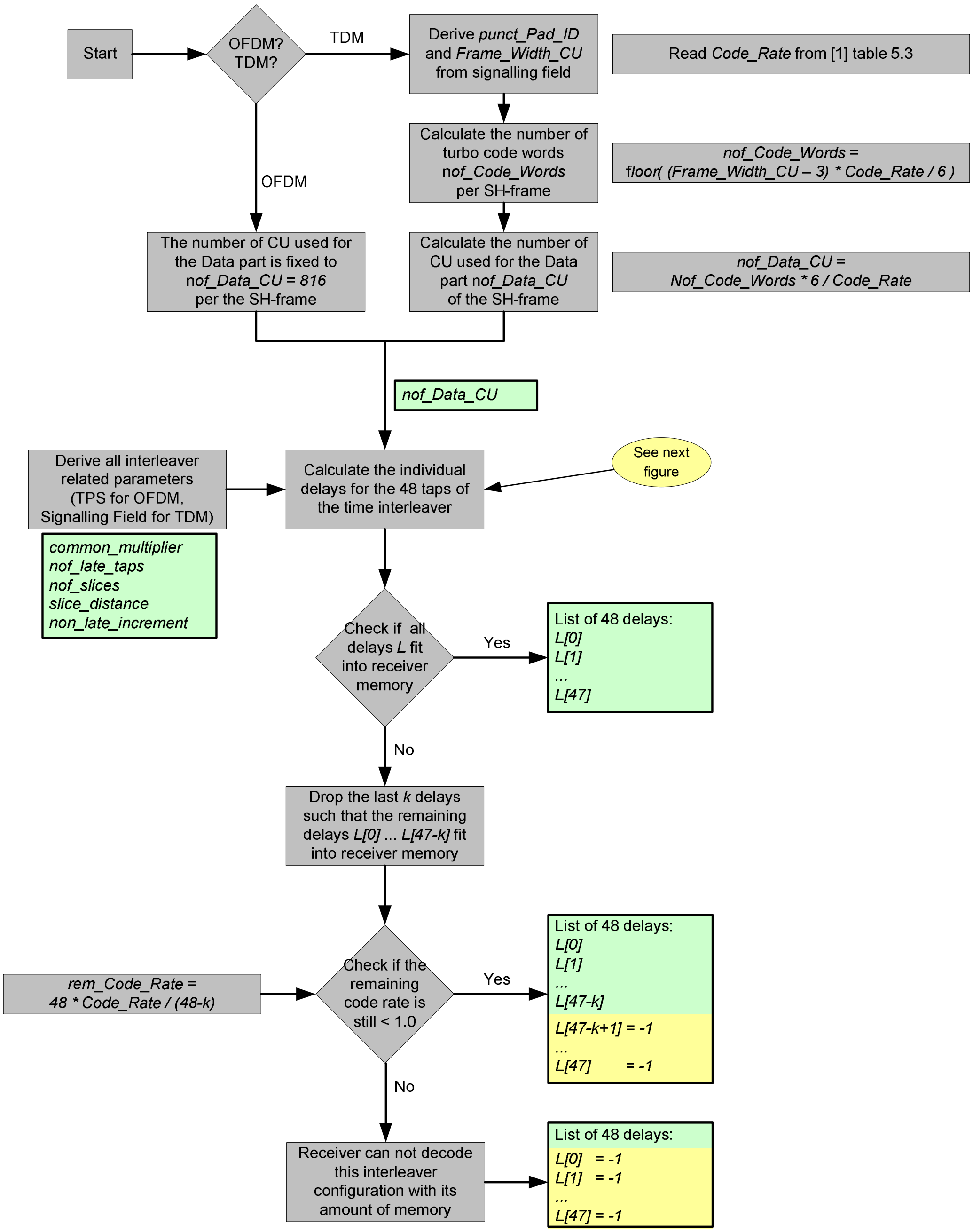


Figure 7.11: Process description for deriving the interleaver parameters

Figure 7.12 explains the derivation of the list of the 48 interleaver delays *L[0]* to *L[47]* from the parameters as derived above.

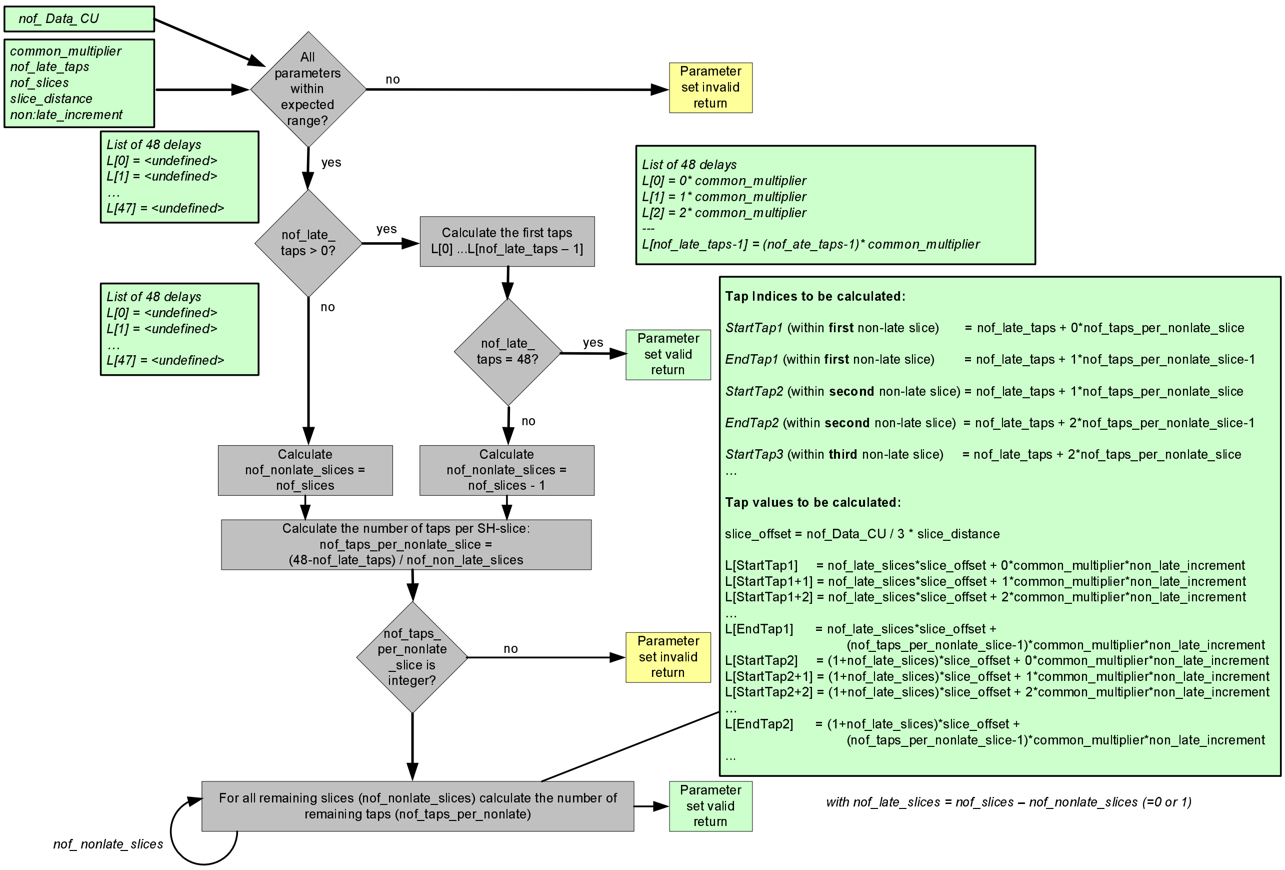


Figure 7.12: Process description for calculation of interleaver parameters

7.2.3.3.4 Calculating the interleaver transit delay

The interleaver transit delay in time can be calculated from the interleaver parameters and the waveform parameters. The interleaver transit delay can be considered as length of the interleaver, or as spreading, as it is the difference between shortest (=0) and longest tap delay.

The interleaver transit delay is calculated from the delay of the latest tap. It is of no matter whether it is calculated from transmitter or receiver side. The different views lead to different formulas, but provide the same results. As it is common practice, in the present document the calculation is done from the receiver point of view.

In a first section the interleaver transit delay is calculated in terms of IUs, later the delay is calculated in terms of time.

7.2.3.3.4.1 Calculation in terms of integer number of IU

For detailed calculation different scenarios have to be considered, the detailed formulas are given here:

Only late taps (or spreading inside late taps):

$$\text{largest\_IU\_per\_tap\_value} = (nof\_late\_taps - 1) * common\_multiplier$$

1) No late taps:

$$\text{largest\_IU\_per\_tap\_value} = (nof\_slices - 1) * slice\_dist * DataIUperSHframePerTap +$$
$$(48/nof\_slices - 1) * common\_multiplier * non\_late\_incr$$

2) Late taps and Non-late taps:

$$\text{largest\_IU\_per\_tap\_value} = (nof\_slices - 1) * slice\_dist * DataIUperSHframePerTap +$$

$$((48 - nof\_late\_taps)/(nof\_slices - 1) - 1) * common\_multiplier * non\_late\_incr$$

With:

$\text{IUperSHframePerTap} = 816*16$, for OFDM, where PAD IUs are also interleaved; and

$\text{IUperSHframePerTap} = \text{DataIUperSHframePerTap}$, for TDM, where PAD IUs are not interleaved.

NOTE 1: The formula for only late taps also gives the transit delay of the late part in case of a uniform-late profile.

NOTE 2: According to clause 7.2.3.3.2 parameter nof_nonlate_slices can be used.
In 2) nof_nonlate_slices = nof_slices, in 3) nof_nonlate_slices = nof_slices - 1.

7.2.3.3.4.2 Calculation in terms of time

The interleaver transit delay in time can be calculated from the interleaver length in terms of integer number of IU and the waveform parameters. For a given configuration in OFDM, this delay is always constant for all IUs, because all IUs are interleaved. Therefore the interleaver transit delay in time given below is unique. However in TDM, since some parts of the SH frame are not interleaved (SF and PADDING), the interleaver delay may vary depending on the position of the IUs in the frame. Therefore, in the following, the transit time is calculated for the IU located at the beginning of the SH frame since this is the one used for synchronization purposes (see clause 7.5).

a) OFDM case

The OFDM time interleaver transit delay can be expressed as a function of the SH frame duration (noted T_{SHF}):

$$\text{ofdm\_interleaver\_transit\_delay} = \frac{\text{largest\_IU\_per\_tap\_value} * 48}{816*16} * T_{SHF}$$

In this formula, $T_{SHF}/(816*16)$ is the IU duration at the interleaver output.

This transit delay can be expressed as:

- a multiple of the SH frame duration; plus
- a fractional SH frame duration;

as follows:

$$\text{ofdm\_interleaver\_transit\_delay} = N_{OFDM} * T_{SHF} + \text{ofdm\_frac} * T_{SHF}$$

with:

$$N_{OFDM} = \text{Floor}\left(\frac{\text{largest\_IU\_per\_tap\_value} * 48}{816*16}\right)$$

and

$$\text{ofdm\_frac} = \frac{\text{largest\_IU\_per\_tap\_value} * 48 - N_{OFDM} * 816 * 16}{816*16}$$

Assuming that T_{SHF} is expressed in units o 100 ns, i.e.

$$T_{SHF} = \text{int}\left(\frac{2176 * 896 * (1 + GI)}{\text{Bps\_OFDM} * \text{BW\_OFDM}} * 10\right),$$

where BW_OFDM is expressed in Hz (recall: for the 1,7 MHz bandwidth, the value 1,6 MHz must be used in this formula).

To express the interleaver transit delay in units of 100 ns the following rule applies:

$$\text{ofdm\_interleaver\_transit\_delay} = N_{OFDM} * T_{SHF} + \text{T\_frac\_ofdm} \qquad (1)$$

where:

- T_frac_ofdm = Floor (ofdm_frac*T_{SHF}) is a fraction of the SH frame duration expressed in 100 ns.

b) TDM case

The transit delay of the TDM time interleaver can be also be expressed as:

- a multiple of the SH frame duration; plus
- a fractional SH frame duration.

But the computation of this transit delay is different from the OFDM case, as only DATA IUs are passed through the interleaver: signalling IUs and PADDING IUs are inserted in the SH frame after the interleaver.

$$\text{tdm\_interleaver\_transit\_delay} = N_{TDM} * T_{SHF} + \text{tdm\_frac} * T_{SHF}$$

with

$$N_{TDM} = \text{Floor}\left(\frac{\text{largest\_IU\_per\_tap\_value} * 48}{\text{IU\_DATA\_SHF\_TDM}}\right)$$

Where IU_DATA_SHF_TDM is the number of DATA IUs in the TDM SH frame, and is given by:

$$\text{IU\_DATA\_SHF\_TDM} = \frac{\text{Num\_Codewords\_per\_SHfr}}{\text{tdm\_ code\_rate/6}} * 16$$

where

$$\text{Num\_Codewords\_per\_SHfr} = \text{floor}[(\text{Frame\_Width\_CUs} - 3) * \text{code\_rate} / 6]$$

Exact expression of the fractional part

The exact expression of the fractional part is given by:

$$\text{tdm\_frac} = \frac{\text{largest\_IU\_per\_tap\_value} * 48 - N_{TDM} * \text{IU\_DATA\_SHF\_TDM}}{\text{Frame\_With\_CUs} * 16}$$

This formula takes into account the insertion of the signalling IUs and PADDING IUs after the time interleaver.

Again, assuming that T_{SHF} is expressed in units of 100 ns, to express the interleaver transit delay in units of 100 ns the following rule applies:

$$\text{tdm\_interleaver\_transit\_delay} = N_{TDM} * T_{SHF} + \text{T\_frac\_tdm} \qquad (2)$$

where

- T_frac_tdm = Floor (tdm_frac*T_{SHF}) is a fraction of the SH frame duration expressed in 100 ns.

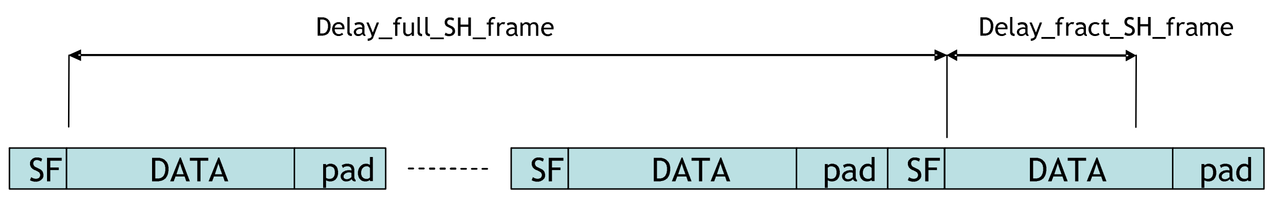


Figure 7.13: TDM interleaver delay

Approximation of the fractional part

For special interleaver configurations that have been used for simulation, lab measurements and/or field trials later in the present document an average calculation formula for tdm_frac has been used:

$$\text{tdm\_frac\_approx} = \frac{\text{largest\_IU\_per\_tap\_value} * 48}{\text{IU\_DATA\_SHF\_TDM}} - \text{Floor}\left(\frac{\text{largest\_IU\_per\_tap\_value} * 48}{\text{IU\_DATA\_SHF\_TDM}}\right)$$

c) Numerical applications

The examples here are for information.

EXAMPLE 1: SH-B network:

- OFDM parameters: BW = 5 MHz, GI = 1/8, 16QAM (Bps_OFDM = 4), time interleaver 8/48/1/0/0.
- TDM parameters: BW = 5 MHz, roll-off= 0,15, QPSK (Bps_TDM =2), time interleaver 16/0/1/0/40.

The largest_IU_per_tap_value of the OFDM interleaver is equal to 376, and we obtain:

T_{SHF} (x100 ns)	N_{OFDM}	ofdm_frac * TSHF (x100 ns)	ofdm_interleaver_transit_delay (x100 ns)
1 096 704	1	419 328	1 516 032

The largest_IU_per_tap_value of the TDM interleaver is equal to 30 080, and we obtain:

TDM code rate	Frame_Width_Cus	IU_DATA_SH_TDM	N_{TDM}	tdm_frac * TSHF (x100 ns)	tdm_interleaver_transit_delay (x100 ns)
1/5	434	6 720	214	909 708	235 604 364
2/9	434	6 480	222	833 899	244 302 187
1/4	434	6 528	221	181 941	242 553 525
2/7	434	6 720	214	909 708	235 604 364
1/3	434	6 624	217	1 015 841	239 000 609
2/5	434	6 720	214	909 708	235 604 364
1/2	434	6 720	214	909 708	235 604 364
2/3	434	6 768	213	356 302	233 954 254

EXAMPLE 2: SH-B network:

- OFDM parameters: BW = 5 MHz, GI = 1/4, 16QAM (Bps_OFDM = 4), time interleaver 8/0/1/0/40.
- TDM parameters: BW = 5 MHz, roll-off = 0,15, 16-APSK (Bps_TDM =4), time interleaver 8/0/1/0/40.

The largest_IU_per_tap_value of the OFDM interleaver is equal to 15 040, and we obtain:

T_{SHF} (x100 ns)	N_{OFDM}	ofdm_frac * TSHF (x100 ns)	ofdm_interleaver_transit_delay (x100 ns)
1 218 560	55	358 400	67 379 200

The largest_IU_per_tap_value of the TDM interleaver is equal to 15 040, and we obtain:

TDM code rate	Frame_Width_Cus	IU_DATA_SH_TDM	N_{TDM}	tdm_frac * T_{SHF} (x100 ns)	tdm_interleaver_transit_delay (x100 ns)
1/5	952	14 880	48	614 400	59 105 280
2/9	952	15 120	47	902 400	58 174 720
1/4	952	14 976	48	245 760	58 736 640
2/7	952	15 120	47	902 400	58 174 720
1/3	952	14 976	48	245 760	58 736 640
2/5	952	15 120	47	902 400	58 174 720
1/2	952	15 168	47	721 920	57 994 240
2/3	952	15 120	47	902 400	58 174 720

7.2.3.3.5 Typical interleaver profiles

In this clause, typical interleaver profiles are presented. These interleaver profiles have been evaluated to be compliant to the receiver classes as mentioned above and taken as reference for simulation and evaluation. In clause A.12 the performance characteristics for these profiles are given over the reference channels selected.

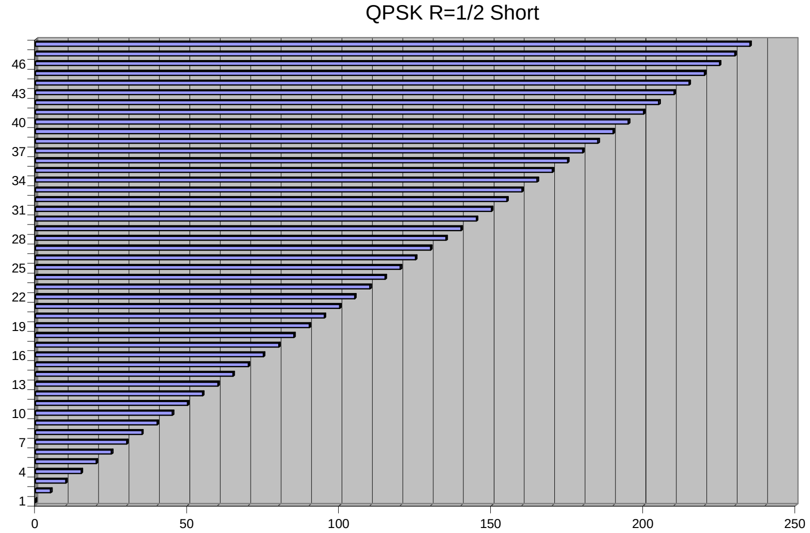
The support of time slicing is incorporated into the interleaver profiles, which can be parameterized according to the rules applied for the generation of time slicing on the link layer. To preserve the power-saving effect of time slicing, the parameters for a long interleaver and time slicing need to be selected jointly, otherwise reduction in the gain of power-saving through time-slicing may occur.

7.2.3.3.5.1 Short uniform interleaver profile (reference Terr48)

The *short uniform* interleaver profile is intended for class 1 receivers capable of saving the equivalent of 1/2 (half) SH-frame in OFDM mode. The resulting maximum interleaver length for this *short uniform* interleaver profile is therefore 1 (one) SH-frame in OFDM mode.

Depending on modulation order, channel bandwidth and guard interval selection, this is equivalent to time spans between 100 ms (bandwidth 5 MHz, guard interval 1/32, modulation 16QAM) and 240 ms (bandwidth 5 MHz, guard interval 1/4, modulation QPSK). For other bandwidths the SH-frame length (with it the interleaver length) scales accordingly.

The *short uniform* interleaver profile used to derive simulation results is defined in clause A.10. A view of this interleaver path is given in figure 7.14.



NOTE: Interleaver parameters are 5/48/1/0/0 corresponding respectively to common_multiplier/nof_late_taps/nof_slices/slice_distance/non_late_increment.

Figure 7.14: Interleaver delays for taps L[0] to L[47]

7.2.3.3.5.2 Long uniform interleaver profile (reference Uni10)

The *long uniform* interleaver profile is intended for class 2 receivers capable of saving the equivalent of 32 SH-frames in OFDM mode. The resulting maximum interleaver length for this *long uniform* interleaver profile is therefore in the range of 64 SH-Frames in OFDM mode.

Depending on modulation order, channel bandwidth and guard interval selection, this is equivalent to time spans between 6,4 s (bandwidth 5 MHz, guard interval 1/32, modulation 16QAM) and 15,6 s (bandwidth 5 MHz, guard interval 1/4, modulation QPSK).

The *long uniform* interleaver profile used to derive simulation results is defined in clause A.10. A view of this interleaver path is given in figure 7.15.

The use of MPE-IFEC "on top" of the *long uniform* interleaver profile is not recommended, but not strictly forbidden as both classes of terminals (class 1 and class 2) have to support MPE-IFEC.

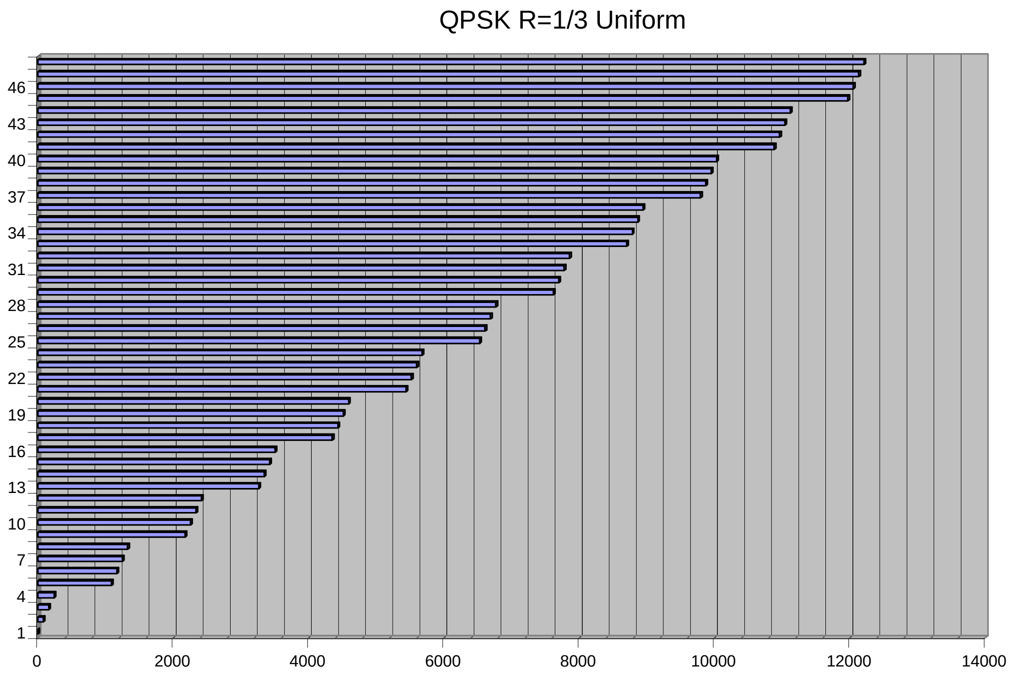


Figure 7.15: Interleaver delays for taps L[0] to L[47] (long uniform, 40/0/12/4/2)

7.2.3.3.5.3 Long uniform-late interleaver profile (reference ULate10)

The *long uniform-late* interleaver profile is intended for class 2 receivers capable of saving the equivalent of 32 SH-frames in OFDM mode. The resulting maximum interleaver length for this *long uniform-late* interleaver profile is typically larger than for the *long uniform* interleaver profile, and can be in the range of up to 128 SH-Frames in OFDM mode.

One special case occurs when considering class 1 terminals in networks transmitting signals intended primarily for class 2 terminals. Depending on the detailed parameter selection, a class 1 terminal may still - with penalty in mobile performance - be able to successfully decode the received signal, even when the full diversity is not exploited. The recommended class 1 receiver behaviour is explained in clause 7.2.3.2.2 Class 1/Class 2 compatible interleaver profile selection Class 1/Class 2 compatible interleaver profile selection. A proper parameter selection of this *long uniform-late* interleaver profile is mandatory for this coexistence to work. Two examples of such a *long optimized-uniform-late* interleaver profile are given below.

Depending on modulation order, channel bandwidth and guard interval selection, this is equivalent to time spans between 12,8 s (bandwidth 5 MHz, guard interval 1/32, modulation 16QAM) and 31,2 s (bandwidth 5 MHz, guard interval 1/4, modulation QPSK).

The *long uniform-late* interleaver profile has been used to derive simulation results is defined in clause A.10. A view of this interleaver path is given in figure 7.16.

The use of MPE-IFEC "on top" of the *long uniform-late* interleaver profile is not recommended, but not strictly forbidden as both classes of terminals (class 1 and class 2) have to support MPE-IFEC.

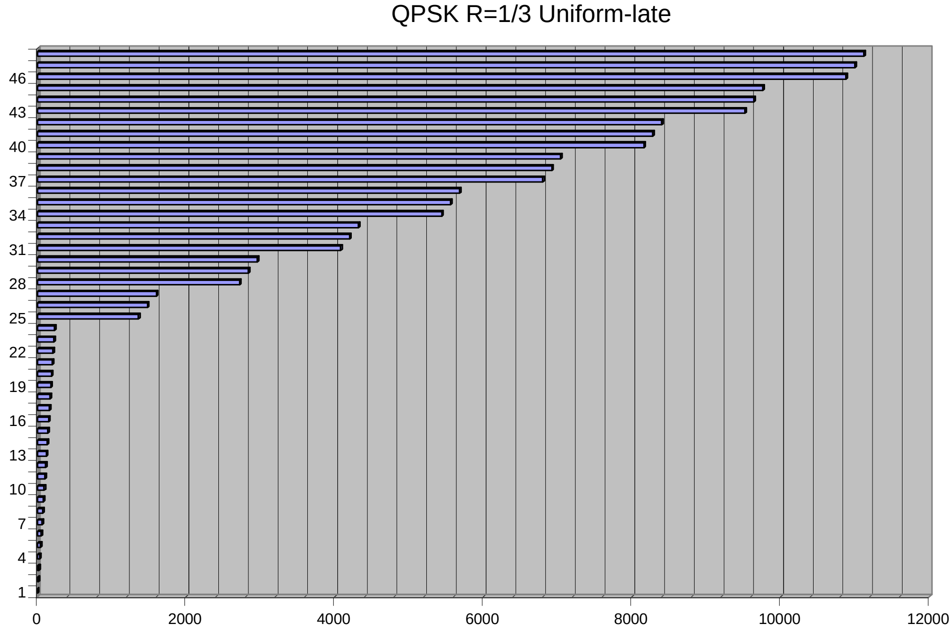


Figure 7.16: Interleaver delays for taps L[0] to L[47] (long uniform-late, 10/24/9/5/12)

7.2.3.3.5.4 Long early-late interleaver profile

The *long early-late* interleaver profile is intended for class 2 receivers capable of saving the equivalent of 32 SH-frames in OFDM mode. The resulting maximum interleaver length for this *long early-late* interleaver profile can be set to the same value as for the *long uniform* interleaver profile, and can be in the range of up to 128 SH-Frames in OFDM mode. Given the same interleaver length the *long early-late* interleaver profile requires the same memory than the *long uniform* interleaver profile.

One special case occurs when considering class 1 terminals in networks transmitting signals intended primarily for class 2. Depending on the detailed parameter selection, a class 1 terminal may still - with performance penalty in mobile condition - be able to successfully decode the received signal, even when the full diversity is not exploited. The recommended class 1 receiver behaviour is explained in clause 7.2.3.2.2 Class 1/Class 2 compatible interleaver profile selection Class 1/Class 2 compatible interleaver profile selection. A proper parameter selection of this *long early-late* interleaver profile is mandatory to ensure that this class 1/class 2 coexistence can work.

Depending on modulation order, channel bandwidth and guard interval selection, this is equivalent to time spans between 12,8 s (bandwidth 5 MHz, guard interval 1/32, modulation 16QAM) and 31,2 s (bandwidth 5 MHz, guard interval 1/4, modulation QPSK).

An example of the *long early-late* interleaver is shown in figure 7.17. The use of MPE-IFEC "on top" of the *long early-late* interleaver profile is not recommended, but also not forbidden as both classes of terminals (class 1 and class 2) have to support MPE-IFEC.

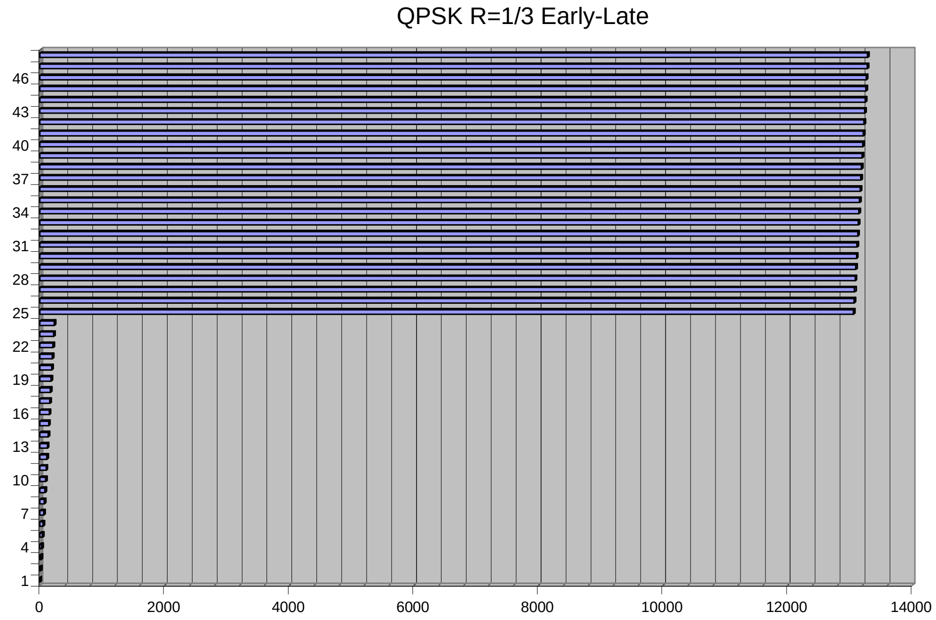


Figure 7.17: Interleaver delays for taps L[0] to L[47] (long early-late, 10/24/2/x/1)

7.2.3.3.5.5 Long optimized-uniform-late interleaver profile

The *long optimized-uniform-late* interleaver profile is intended for class 2 receivers capable of saving the equivalent of 32 SH-frames in OFDM mode. The resulting maximum interleaver length for this *long optimized-uniform-late* interleaver profile can be set to the same value as for the *long uniform* interleaver profile, and can be in the range of up to 128 SH-Frames in OFDM mode. Given the same interleaver length the *long optimized-uniform-late* interleaver profile requires the same memory than the *long uniform* interleaver profile.

One special case occurs when considering class 1 terminals in networks transmitting signals intended primarily for class 2. Depending on the detailed parameter selection, a class 1 terminal may still - with performance penalty in mobile conditions - be able to successfully decode the received signal, even when the full diversity is not exploited. The recommended class 1 receiver behaviour is explained in clause 7.2.3.2.2 Class 1/Class 2 compatible interleaver profile selection. A proper parameter selection of this *long optimized-uniform-late* interleaver profile is mandatory to ensure that this class 1/class 2 coexistence can work.
In the same manner as class 1 terminals can exploit the late part of the interleaver profile for successful decoding also class 2 terminals can exploit the late part at startup for "fast access".
To enable this class 1 decoding or class 2 "fast access" the coding rate included in the late interleaver taps has to be properly selected. For the two examples of the *long optimized-uniform-late* interleaver profile given in figures 7.18 and 7.19 the code-rate of the late part is set to 4/5.
To achieve this code-rate with the late part it is clear that the available capacity to be put on the uniform part depends on the overall code rate. For a low code-rate (e.g. 1/5) there is a larger amount of capacity available for the uniform part than there is available in case of a high overall code-rate. In the example (with code-rate 4/5 in the late part) for code-rate 1/5 there is free capacity to fill 36 uniform taps, whereas for code-rate ½ only 18 uniform taps can be used. Thus for the *long optimized-uniform-late* interleaver profile the interleaver performance for the uniform part increases with lower code-rate.
Selecting a target code-rate 4/5 in the late part is a rule-of-thumb to achieve successful decoding / fast access valid for high C/N. For accurate parameterization the target code-rate for the late part depends on the C/N operating point, the C/N threshold for the waveform configuration (the delta is the C/N margin) and the overall code-rate. According to this input and the acceptable erasure rate for this configuration (refer to table 7.8, clause 7.2.2.6.4) can be calculated, thus assuming the erasures are the part included in the uniform part of the interleaver the target code-rate for the late part of the interleaver can be calculated for the scenario.

Depending on modulation order, channel bandwidth and guard interval selection, for R=1/5 this is equivalent to time spans between 7,3 s (bandwidth 5 MHz, guard interval 1/32, modulation 16QAM) and 17,7 s (bandwidth 5 MHz, guard interval 1/4, modulation QPSK), for R=1/2 this is equivalent to time spans between 15,4 s (bandwidth 5 MHz, guard interval 1/32, modulation 16QAM) and 37,3 s (bandwidth 5 MHz, guard interval 1/4, modulation QPSK).

The use of MPE-IFEC "on top" of the *long optimized-uniform-late* interleaver profile is not recommended, but also not forbidden as both classes of terminals (class 1 and class 2) have to support MPE-IFEC.

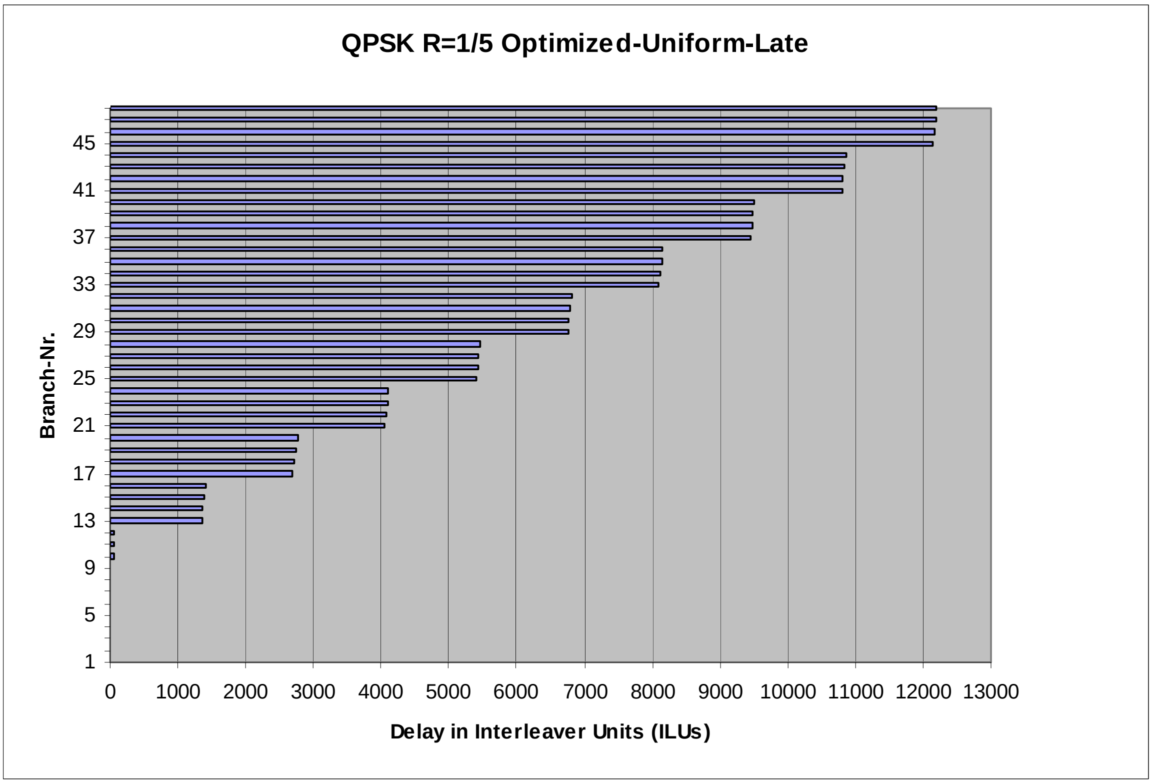


Figure 7.18: Interleaver delays for taps L[0] to L[47] (Long optimized-uniform-late, 5/12/10/5/4)

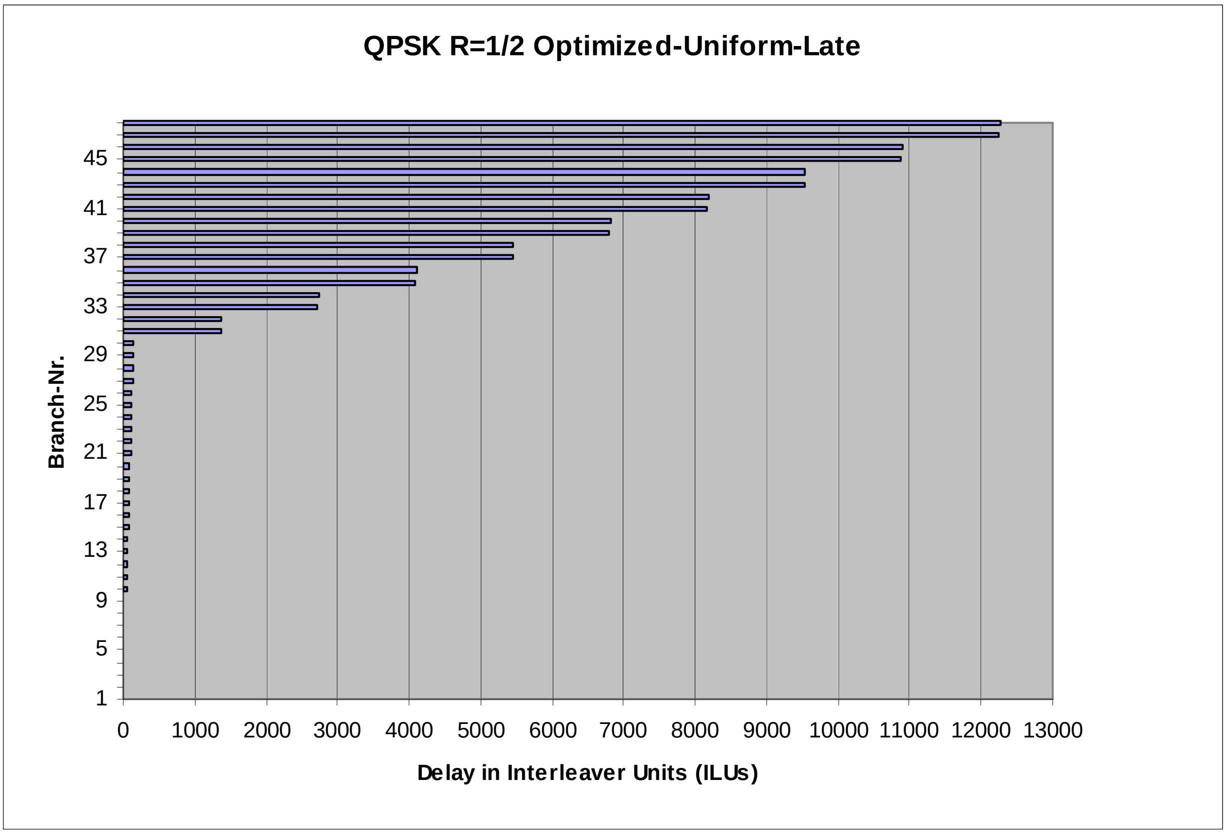


Figure 7.19: Interleaver delays for taps L[0] to L[47] (Long optimized-uniform-late, 5/30/10/5/4)

7.2.3.4 Interleaver profile alignment between satellite and terrestrial / different carriers in non-SFN configurations

This clause is dedicated to give the background on the alignment of interleaver profiles between satellite and terrestrial transmission, if different carrier frequencies are used. This clause is not applicable to SH-A in SFN mode. The details on transmitter delay alignment (in terms of interleaver delay compensation) are given in clause 7.5 Synchronization.

The synchronization aspects which are not interleaver-related are also given in clause 7.5 Synchronization for all types of DVB-SH networks. There, the following items are addressed:

- distribution network delays and the use of the SHIP packet for synchronization;
- compensation of the satellite round-trip-delay using the SHIP mechanisms.

7.2.3.4.1 Overview

Certain rules apply when the interleaver on the satellite and the terrestrial transmission are different, this can be the case if the interleaving parameters are selected differently, but can also be the case if the same interleaver parameters are selected for different waveform parameters. This is possible in the following network configurations:

- SH-A in MFN mode;
- SH-B.

Thus, the interleaver profile selected will in most cases not be identical, due to the possibility to choose the parameters individually for both transmission paths in terms of:

- modulation and modulation order (OFDM/TDM, QPSK/8PSK/16QAM/16APSK);
- code rates (between 1/5 and 2/3);
- interleaver length (between 100 ms and 30 s);
- throughput.

To be able to combine both demodulated signals, and to minimize the penalty on the receiver in terms of de-interleaver memory consumption, the transmit signals are constructed such that the difference in delay is compensated already in the transmitter. Figures 7.20 and 7.21 introduce the underlying concept.

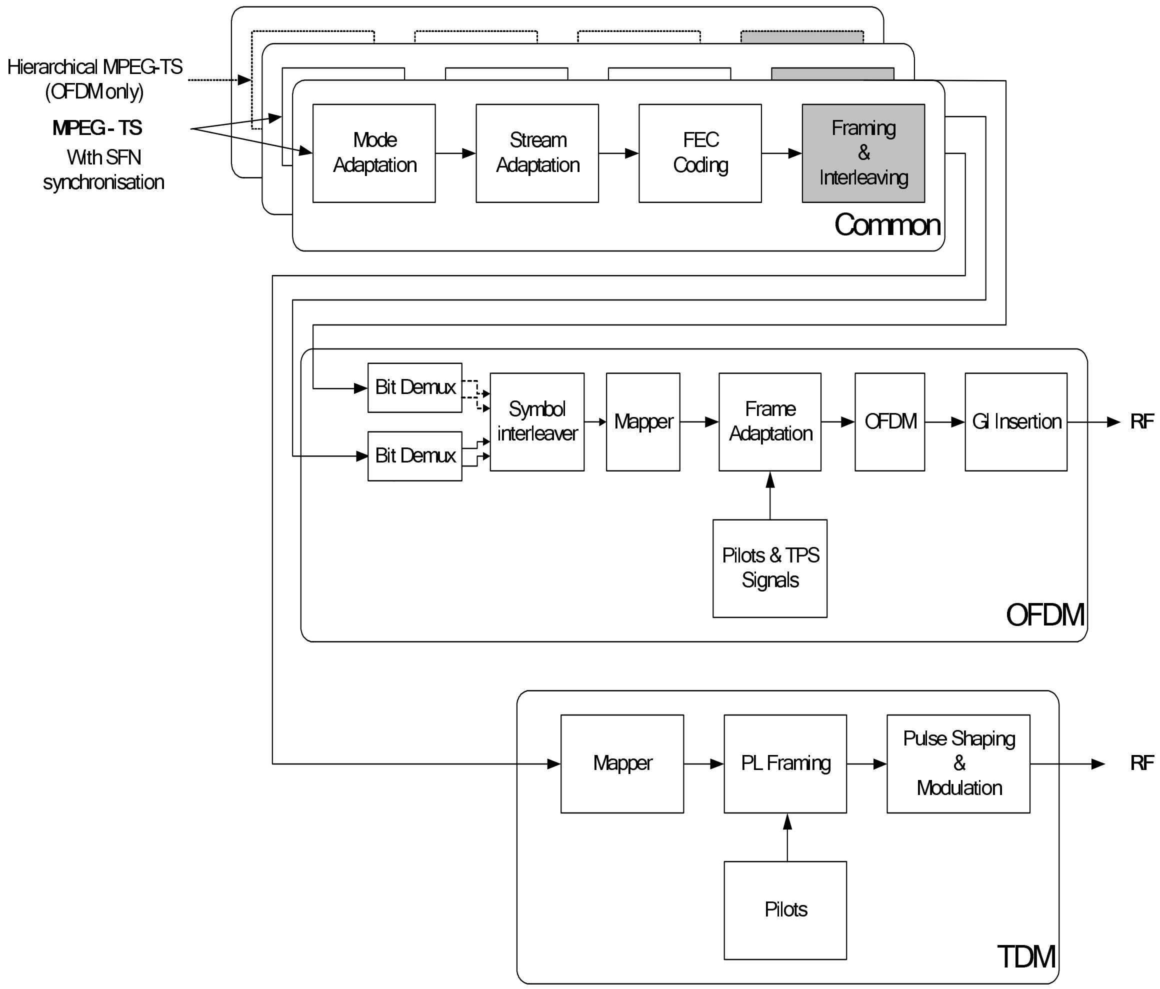


Figure 7.20: Reference system block diagram (physical layer)

The blocks marked in grey (framing and interleaving) have to be considered for each input stream individually, but with knowledge of the other path if combining is intended. This is explained below.

To minimize the memory impact on the receiver, the transmitters are requested to align the transmission latency between all the different time interleaver transit delays used in the DVB-SH system. Figure 7.21 shows the concept.

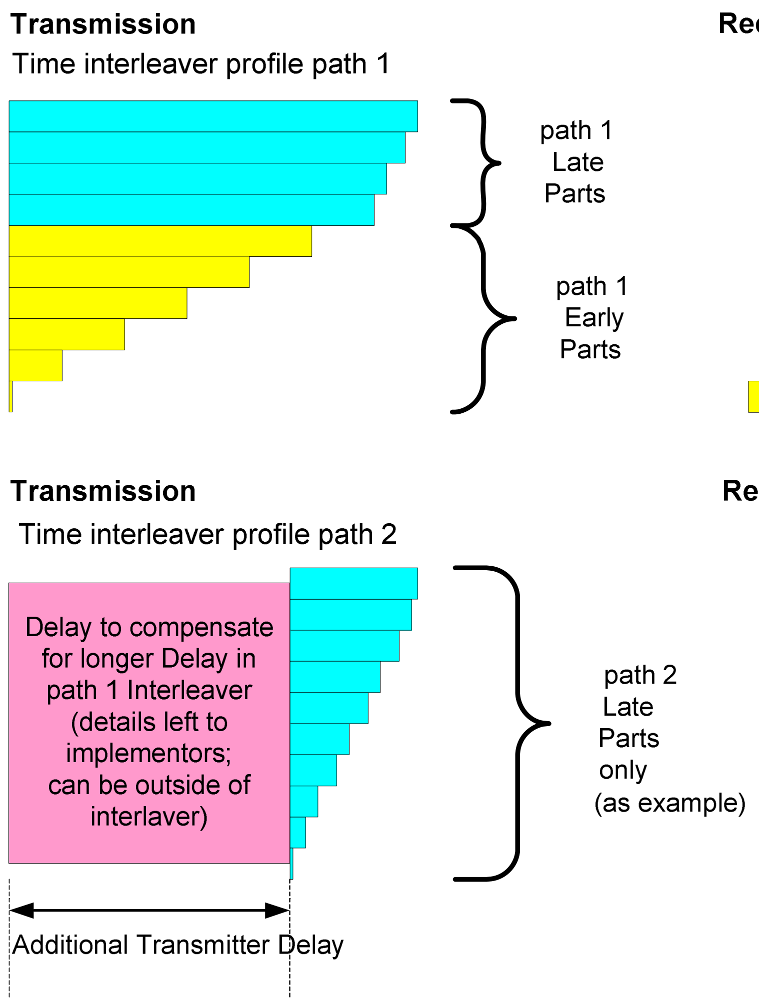


Figure 7.21: Conceptual Deployment of interleaver profiles with different length

In this example, the "effective" time interleaving is shorter for the path 2 part than for the path 1 part (see both drawings on the right hand side, which displays the de-interleaver required on the receiver). One transmitter (in case of SFN the complete set of transmitters in the network) therefore has to compensate the different end-to-end delays on the shorter transmission path, in this case the terrestrial one. In general DVB-SH is flexible on which interleaver profile is used for terrestrial or satellite transmission, to reflect this in the figure the transmission is split into part 1 and 2, rather than terrestrial and satellite.

NOTE 1: By definition the delay of the shortest receiver tap (L[0]) is always 0, for both TDM and OFDM.

NOTE 2: The figure only provides a conceptual view on the interleaver transition delay compensation that needs to be done at the transmitter. The detailed procedure and how to achieve interleaver delay compensation (in front of the interleaver, inside the interleaver, or after the interleaver) is left to open implementors, some options are presented in clause 7.5 Synchronization.

7.2.3.4.2 void

The content of this section is moved to clause 7.5 Synchronization.

7.2.3.4.3 Interleaver synchronization on the receiver side

Due to interleaver delay compensation in the transmitter, the receiver can assume to receive synchronized OFDM and TDM SH-frames (in SH-B) and synchronized OFDM1 and OFDM2 SH-frames (in SH-A-MFN). The delay between the two signals is limited on air as given in clause 7.5.2.

According to the different interleaver profiles for TDM and OFDM as given in the TDM signalling field, or the OFDM TPS bits respectively, the receiver aligns the SH-frames received in parallel, performs de-interleaving and combining/decoding without further consistency checks. The basic fact is that the delay of interleaver tap L[0]=0 at the receiver side, therefore the output of the interleaver for TDM and OFDM is directly aligned.

Once TDM and OFDM framing is detected the receiver directly is able to start de-interleaving and decoding task on the synchronously received SH-frames.

The de-interleaving/decoding task needs to be done according to clause 5.4.3 of [1].

7.2.3.5 Fast interleaver synchronization strategies

This section is informative.

7.2.3.5.1 Motivation

When long interleaver profiles are used, minimization of the access time to successfully decode a service is crucial. Independently whether it is the first access to a service ("zapping time") or the access to a service after a signal dropout ("recovery time"), it is preferable that the user gets fast feedback on whether the current location is suited for reception.

The choice of a proper decoding strategy is essential for streaming services which are protected by long interleavers. Independently whether the long interleaving is done on the physical or the link layer, different decoding strategies as presented in the following are possible.

7.2.3.5.2 Definition of jitter and delay

The major impact of the choice of decoding strategy is related with jitter and delay. The following definitions apply:

- **jitter:** variation of decoding instant or variation of decoder output rate;
- **delay:** time span between injection (transmitter side) and decoding (receiver side) of payload;
- **reduced diversity decoding:** makes use of the fact that the full diversity (redundancy) is not needed in all reception scenarios. As soon as the receiver has successfully gathered sufficient number of packets to decode, it starts the decoding process immediately. The decision is typically made on the effective code rate:
    - choose the net code rate (e.g. 1/3);
    - derive the ratio of packets received for one code word (e.g. 40 %);
    - calculate the effective code rate (60 % erased bits from code rate 1/3 correspond to effective code rate of 5/6);
    - check whether the receiver is capable of decoding (effective code rate < 1,0 or any other limit < 1,0).

7.2.3.5.3 Early decoding

Early decoding tries to minimize the end-to-end delay of a time-interleaved signal. Figure 7.22 shows the typical delays which are in many cases smaller than the full time span of packets.

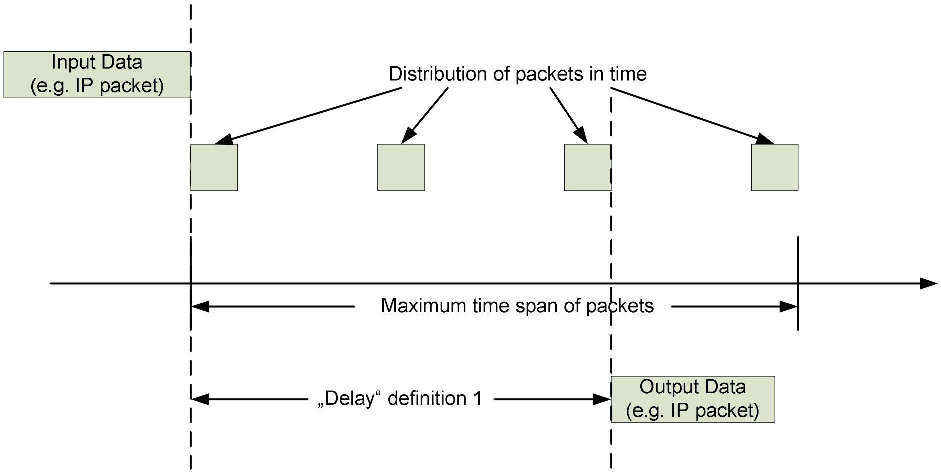


Figure 7.22: Principle of early decoding

However, assuming that not all injected packets are necessary for decoding purposes, the jitter at the output of the physical layer may increase as soon as the receiver runs into a channel situation that is worse than before: it needs now the full redundancy transmitted, such that for a certain amount of time, no or only very few packets will be processed.

The summary is the following:

- early decoding does not always need the full end-to-end delay;
- early decoding can provide short zapping times in good reception conditions;
- video rebuffering is necessary in fading environments.

7.2.3.5.4 Late decoding

Late decoding tries to minimize the jitter at the output of the physical layer decoder. It always waits for the full redundancy transmitted; the decoding instant is almost jitter-free. Figure 7.23 shows the constant delay which is the full time span of redundancy transmitted.

For regular physical layer interleaving late-decoding is assumed to be baseline, as data can immediately be fed to the decoder. The decoding capability at startup depends on the margin and increases over time (with the interleaver length as maximum duration).

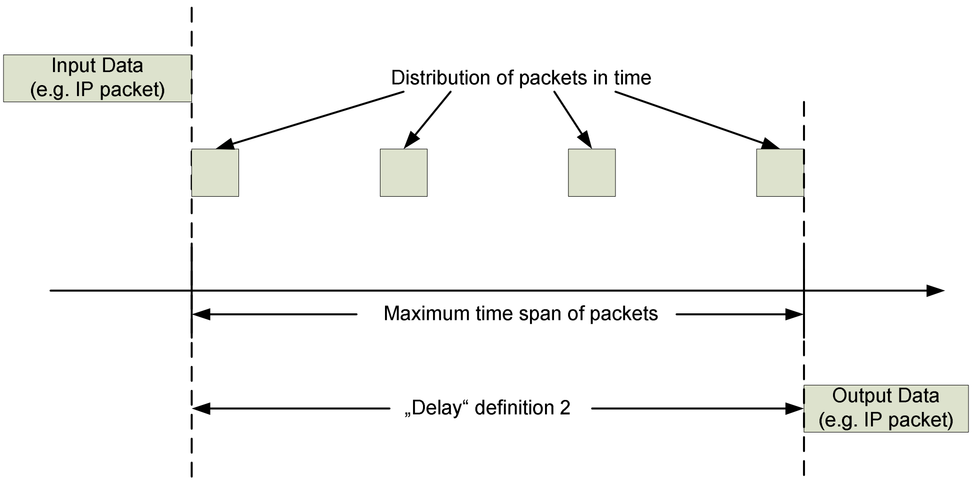


Figure 7.23: Principle of late decoding

The interleaver profiles defined in the waveform document is designed for the late decoding principle: it is possible to transmit a "late" burst which is delayed very long in the transmitter but has only a very short delay in the receiver.

When access is requested to a service which is interleaved over a long time span, this late burst enables the receiver to reconstruct the data transmitted without having received all parts before. Then, as time advances, more "uniform" parts are available, making the reception more robust, until the full interleaver length is reached and full diversity is achieved. This principle of increased redundancy as time advances in presence of a "late" part in the interleaver profile can be understood as "fast access".

In case of late decoding the "delay" of the transmitted data is located the transmitter, as it is calculated from the late burst, which is not further delayed inside the receiver.

The summary is the following:

- late decoding allows jitter-free output of the physical layer;
- short zapping times can be achieved in good reception conditions;
- the waveform is designed to support late decoding;
- the full end-to-end delay is always experienced even in good reception conditions;
- the delay driving the end-to-end delay is located in the transmitter.

7.2.3.5.5 Zapping time in case of long physical layer interleaver

The use of low code rates provides a high coding gain, a high performance in case of fading and offers short zapping time. In theory only R (code rate as ratio) of the data has to be received to allow a decoding, if the received data are error free. If not yet all data of one code word are received this is equivalent to a higher code rate. If for example for a code rate of R=1/3 40 % of the interleaved data are received this can be considered as a code rate of R=5/6. For this code rate a higher C/N is required for a correct FEC frame decoding. If the available C/N is low a lower code rate may be required for error free decoding. In this case the receiver has to wait for the next burst to progressively lower the required C/N. The interleaver profiles are configurable. The Uniform-Late profile ensures that a big part of the encoded code word is available already with the first burst. In case of good reception conditions this part is already sufficient for successful decoding. Thus the Uniform-Late interleaver profile is recommended, if a short zapping time is important but some C/N performance loss is acceptable compared to an uniform long interleaver. The late decoding principle ensures a jitter free decoding (jitter introduced by the multiplexing, variable bit rate coding and the time slicing are not considered here). The principle of the zapping behaviour of the physical layer is summarized in figure 7.24:

- An input burst is subdivided in several small bursts by the time interleaver. To simplify the drawing one burst is split in 6 short bursts.
- In case of uniform-late profile 50 % of the data are send as "late burst". In the example this is equivalent to 3 bursts are spread over a long time, whereas the remaining 3 bursts are send a one bigger burst.
- From a FEC coding point of view the partitioning of the data in short burst is equivalent to sending complementary punctured subsets of the Turbo encoder output. Assuming a code rate of R=1/3 the subset send as late burst is equivalent to a code word encoded with a code rate of R=2/3.
- If the receiver is tuned to the channel at the time t0 for the burst n the late burst is available at the time t1. This defines the zapping time for good reception conditions. The remaining data are considered as "erasures". According to table 7.8, 50 % erasure are allowed if the C/N is 5,3 dB above the threshold. This is equivalent to a nearly instantaneous zapping. In this case the latency caused by time slicing and the AVC encoder is dominating.
- In bad reception conditions a lower code rate is required to decode the data. If for example 4 (3 are transmitted as late burst, 1 out of 3 from the uniform part is received) are received the equivalent code rate is R=1/2 or the not yet received data are considered as "33 % of the data are erasure". According to table 7.8 the required additional C/N is 2,7 dB.

- For very bad reception condition when all of the transmitted redundancy is required to receive the data the zapping time may be up to the full interleaver length.

NOTE: If all redundancy is assigned to the Turbo code the physical layer uses a lower code rate resulting in a higher receiver sensitivity compared to a configuration where the parts of the redundancy are assigned to the IFEC.

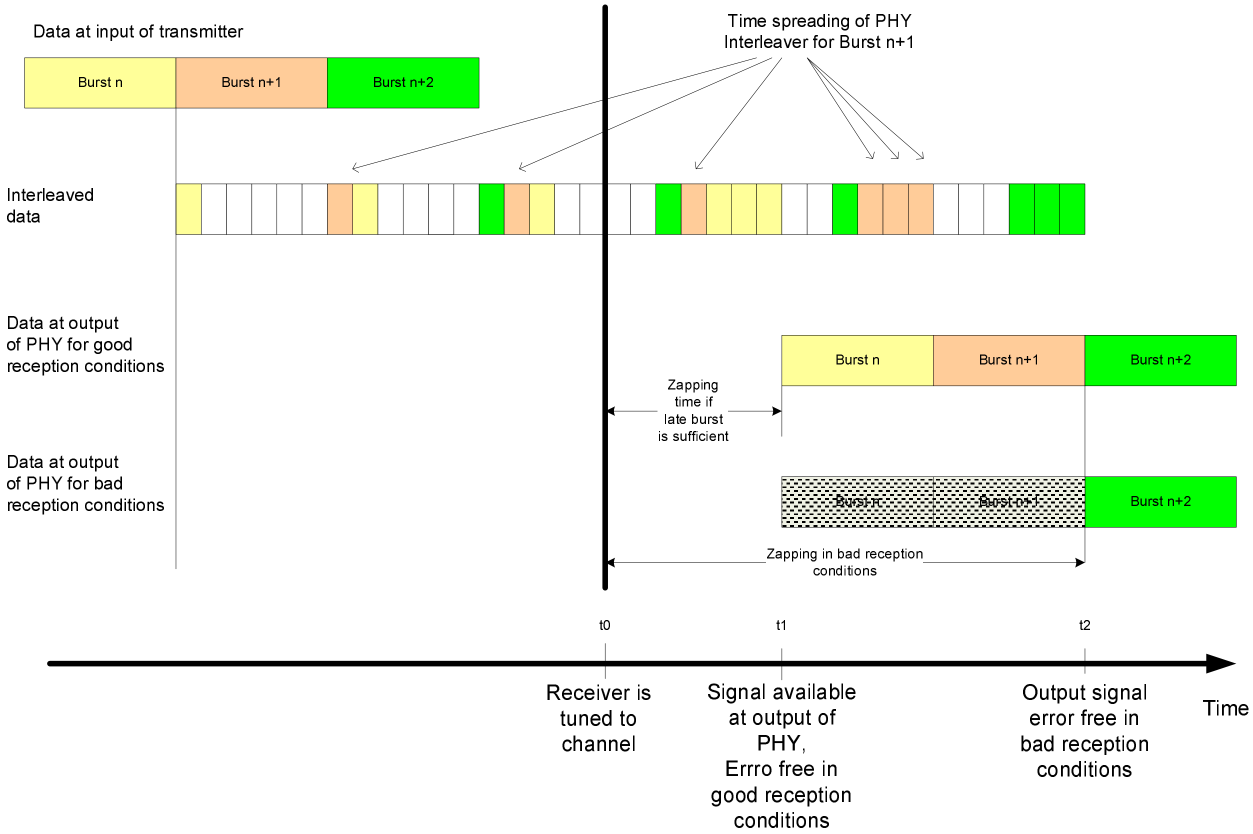


Figure 7.24: Principle of fast zapping

The numbers used for the example should be compared to the link budget. Typically a link margin (difference between available C/N for LOS reception and required C/N for AWGN channel) in the range of 6 dB to 10 dB is recommended (see clause A.12). With the given numbers the following conclusions can be derived:

- for LOS reception (very good reception condition) the available link margin is typically sufficient to support instantaneous zapping;
- if the receiver is in a bad reception condition when the channel is selected a longer zapping time may result;
- receiver offering a lower link margin (e.g. receivers with low gain antenna) main have a higher zapping time;
- for terrestrial reception the reception area can be split in three parts:
  - area with high C/N: short zapping time is provided;
  - area with medium/low C/N: a longer zapping time has to be accepted;
  - area with very low C/N: if all redundancy is assigned to the physical layer (no IFEC is used) the receiver offers a higher sensitivity.

7.2.4 SH framing

This clause is dedicated to give advice on the alignment of the start of an SH-frame after "MAPPING ON CU and SH-FRAMING" (according to figures 5.11 and 5.12 of [1]) in OFDM/TDM mode with respect to the framing in front of the convolutional time interleaver. Without time interleaver the alignment is straight forward as given in [1]. In this clause it is described how the time interleaver affects the SH-frame generation in OFDM/TDM configuration.

This clause in detail describes the process of mapping the sequence of CUs at the output of the time interleaver onto OFDM symbols, TDM PL slots respectively, according to figure 7.20.

7.2.4.1 OFDM

Definition:

For OFDM the "start of an SH-frame" shall be understood as the first sample of the guard interval of the OFDM symbol that carries the first IU of the SH-frame, the first IU being considered after the rate adaptation and the padding, but before the convolutional interleaver according to [1], figure 5.11.

This first OFDM symbol is built when this same first IU of the first CW of the first SH-frame exits from the interleaver after the convolutional interleaver, but before the mapping on CU and SH framing according to [1], figure 5.11 (see figure 7.25 for a schematic view of this process).

Considering the start of SH frame in OFDM the following facts shall be noted:

- There is always an integer nof_CW_per_SH_frame.
- The nof_IU_per_CW is a multiple of nof_taps.
- Therefore the first IU of a codeword is always introduced in tap 0 of the interleaver.
- The last IU of an interelaver cycle introduced in the last tap 47 stays at the same location within the interleaver cycle also at the output of the interleaver.
- The first IU of an SH-frame is introduced in the first tap 0 is also always at the same position in an OFDM symbol.

NOTE: Figure 7.25 shows a schematic view of how the start-of-SH frame is identified for OFDM. For visualization purposes some parameters are set to special values that are not available in DVB-SH, these modified settings are:

- nof_IU_per_CW 96
- nof_CW_per_SH_frame 6
- nof_IUs_per_OFDM_symbol 50
- nof_taps 48
- largest_tap_length 47
- nof_interleaver_cycles_before_IU_0 47
- nof_IUs_before_IU_0 2 256
- nof_symbols_before_IU_0 45

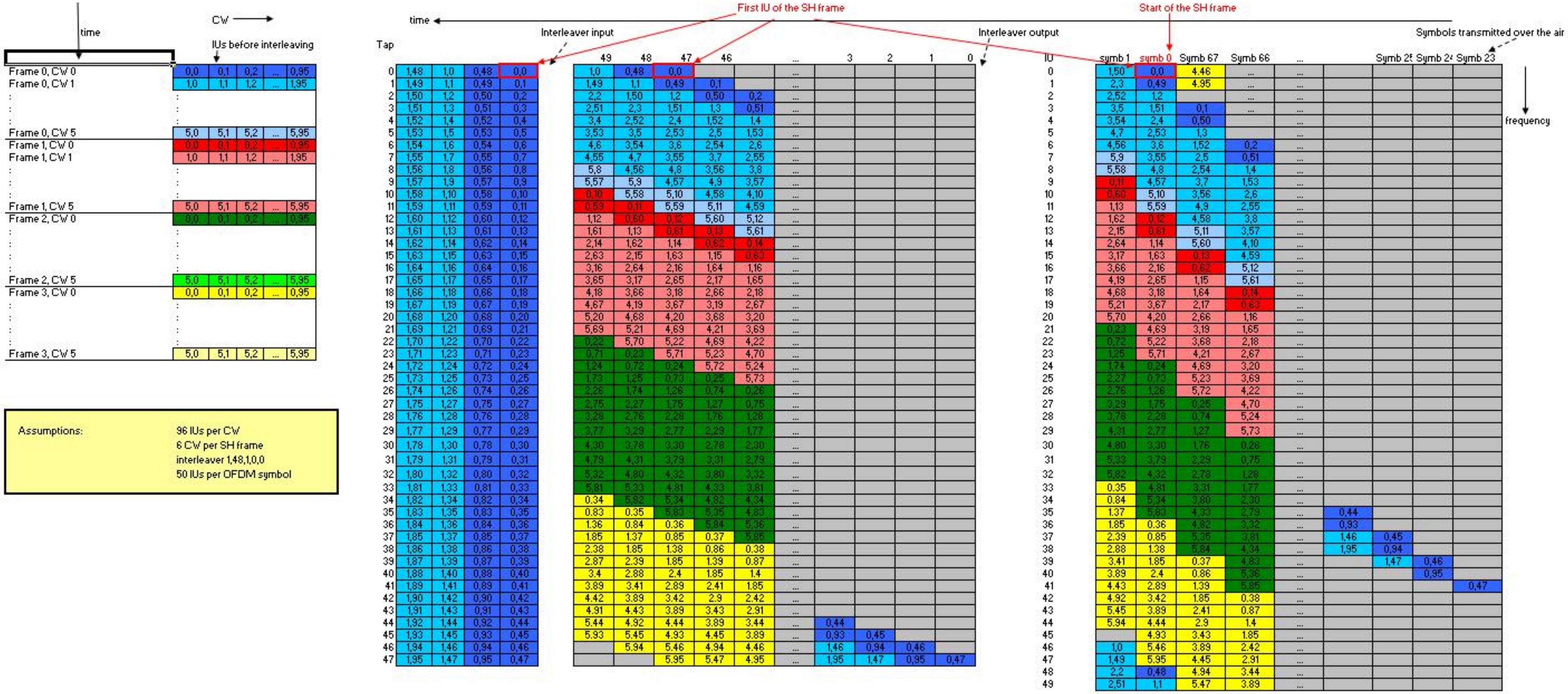


Figure 7.25: Start of SH frame / first OFDM symbol of SH frame for OFDM (for visualization some parameter values differ from standard DVB-SH, see assumptions box)

For information:

With the 2 given preceding constraints, the first IU of an SH-frame inserted in tap 0, is always at the same position within an OFDM symbol. In the example this is IU(0,0).

The demonstration of Figure 7.25 is the following:

1) we demonstrate, at cold start, that if we introduce IU(0,0) in tap 0, it will always exit at same position within the symbol (at start of symbol 0);

2) we demonstrate that, on a continuous basis, the next IU(0,0) will always be introduced in tap 0 and will also exit at same position within the symbol.

To 1) Demonstration of the unique position of IU(0,0) at cold start:

- Let us assume the synchronization on the time is the only valid (no synchronization of the first IU inside the symbol), this means that the IU introduced at tap 47 of delay 0 is immediately introduced in the symbol at position 47 out of [0;49] possible; in the example, this is the IU(0,47).
- The number of IU to be inserted between the IU(0,0) in the symbols is computed at the exit of the interleaver and equal to nof_interleaver_cycles_before_IU_0*nof_taps.
  Since there are 47 interleaver_cycles_before_IU_0, this lead to 47 x 48 = 2 256 IUs.
- Given the size of the symbol equal to 50 IUs, it is easy to compute the number of symbols transmitted before the one carrying IU(0,0) and the position of the IU(0,0) inside his symbol (0):
  - nof_symbols_before_IU_0 = floor(nof_IUs_before_IU_0/IUs_per_OFDM_symbol) = 45.
  - nof_IUs_before_IU_0_in_symbol = nof_IUs_before_IU_0 - IUs_per_OFDM_symbol * nof_symb_before_IU_0 = 6.
  - Therefore the IU(0,0), introduced on tap 0 with a symbol build on the exit of the tap 47 will always be positioned on the 6th IU in the symbol (using the given parameters).
- Now the second constraint of building a symbol on IU(0,0) start implies a constant shift of 6 IU before:
  - Therefore, the position of the IU(0,0) shall always be the same within the symbol if inserted at the tap 0.

To 2) Demonstration of maintenance of that position for following SH frames IU(0,0) after cold start:

For this we must demonstrate:

2a) that following IU(0,0) are also introduced in tap 0;

2b) that an SH frame always requires an integer number of symbols to carry the SH frame.

To 2a) Following IU(0,0) are always introduced in tap 0:

- In [1], clause 5.4.3, it is said that "for each cycle of the interleaver, 48 non interleaved IUs are read sequentially (starting on a coded word) and fed into the branches. The output of the interleaver is the 48 interleaved IU. Output is read synchronously with the input". Therefore the IU(0,0) shall always be inserted at tap 0 because it is the first IU of the first CW that starts the cycle at the input. The SH frame being equal to 816 CU, the number of interleaver cycles to feed a complete SH frame to the interleaver is equal to the integer 272 and each IU(0,0) finds naturally the same tap 0:
  - 816 CU * (2 016 bits_per_CU / 126 bits_per_IU) / 48 taps = 272 cycles.

To 2b) There are always an integer number of symbols required to carry an SH frame:

- The number of IU per symbol actually depends on modulation parameters:
  - Nof_IU_per_sybol = FFT_size * 756 * mod_order / 126 = FFT_size * modulation_order * 6 where FFT_size belongs to {1;2;4;8} and modulation_order belongs to {2;4}.
  - Nof_IU_per_symbol belongs to {12; 24; 48; 96; 192}.

- The total number of IU per SH frame is equal to 13056 IUs that makes 1088, 544, 272, 136 or 68 symbols depending on the nof_IU_per_symbol.

Therefore there is always an integer number of symbols required to carry the SH frame.

From the receivers point of view it is obvious that the first IU of an SH frame IU(0,0) must be located at start of the first OFDM symbol of that SH frame. As the delay of tap 0 on the receiver is 0, the first IU that is fed into the receiver interleaver is also the first IU that exits from the interleaver, on both sides this IU corresponds to the start of an SH-frame. From this it is clear that IU(0,0) needs to be at the start of the first OFDM symbol of an SH frame.

7.2.4.2 TDM

Definition:

For TDM the "start of an SH-frame" shall be understood as the first sample of the PL slot that carries the SF right in front the first data IU of the SH-frame, the first data IU being considered after the rate adaptation, but before the convolutional interleaver according to [1], figure 5.12.

This first PL slot is built with the SF when this same first data IU of the first CW of the first SH-frame exits from the interleaver after the convolutional interleaver, but before the mapping on CU and SH framing according to [1], figure 5.12 (see figure 7.26 for a schematic view of this process). Taking the first sample of the first PL slot as start of SH-frame is directly related to the first sample of the pilot field at start of this first PL slot.

Considering the start of SH frame in TDM (same as for OFDM) the following facts shall be noted:

- There is always an integer nof_CW_per_SH_frame.
- If a SH frame has more capacity than required for nof_CW_per_SH_frame and SF, then PAD CUs are inserted, PAD CUs are not interleaved.
- The nof_IU_per_CW is a multiple of nof_taps.
- Therefore the first IU of the SH frame is always introduced in tap 0 of the interleaver.
- The last IU of an interelaver cycle introduced in the last tap 47 stays at the same location within the interleaver cycle also at the output of the interleaver.
- The first IU of an SH-frame introduced in the first tap 0 is also always at the same position in a PL slot.

NOTE: Figure 7.26 shows a schematic view of how the start-of-SH frame is identified for TDM. For visualization purposes some parameters are set to special values that are not available in DVB-SH, these modified settings are:

▪ nof_IU_per_CW	96
▪ nof_CW_per_SH_frame	6
▪ nof_taps	48
▪ largest_tap_length	47
▪ nof_interleaver_cycles_before_IU_0	47
▪ nof_IUs_before_IU_0	2 256
▪ CUs per_PL_slot (=modulation_order): QPSK	2
▪ nof_PAD_CUs	1
▪ nof_PL_slots_before_IU_0 (at start)	73

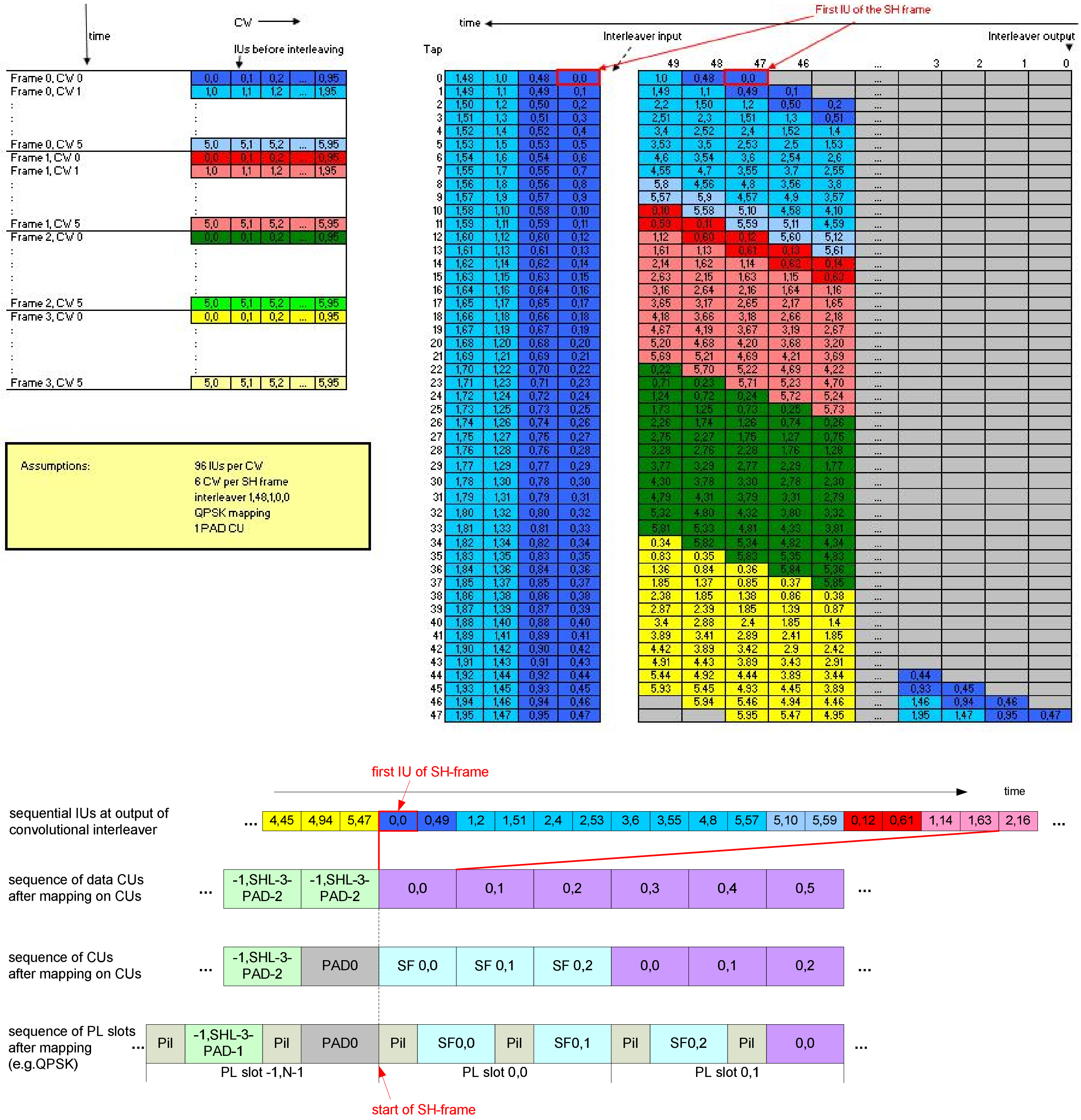


Figure 7.26: Start of SH frame / first PL slot of SH frame for TDM (for visualization some parameter values differ from standard DVB-SH, see assumptions box)

From the receivers point of view it is obvious that the first IU of an SH frame IU(0,0) must be located as first data IU after the SF in the according PL slot that SH frame. As delay of tap 0 on the receiver is 0, the first IU that is fed into the receiver interleaver is also the first IU that exits from the interleaver, on both sides this IU corresponds to the start of an SH frame. From this it is clear that IU(0,0) needs to be the first data IU of an SH frame.

7.3 OFDM and TDM elements

7.3.1 OFDM Elements

This clause is informative.

7.3.1.1 Overview

The OFDM modulation part is a full reuse of the functions of DVB-H with reasonable extensions necessary for the application "DVB-SH". The main new element is the introduction of a 1 k-FFT mode suitable for all channelizations. Furthermore, the TPS signalling has been extended to support the full signalling necessary in DVB-SH.

All design guidelines given in the Implementation Guidelines for DVB-H [3] are still valid and are explicitly referred here.

The main difference in the use of the OFDM part with respect to DVB-H is its use also at lower spectral efficiencies, down to QPSK modulation with code rate 1/5. This imposes higher requirements on the demodulator which should be designed and tested to comply with low C/N, satellite large-scale fading channels and frequency-selective reception scenarios.

In the next clause, one example for the performance of OFDM in a mobile channel is given to demonstrate the interaction between demodulator and decoder. Further OFDM specific simulation results are given in clauses A.11 and A.12, measurement results are given in clause A.13.

7.3.1.2 Performance in mobile TU6 Channel

7.3.1.2.1 Simulated capacity performance

Figures 7.27 and 7.28 summarize the simulated TU6 performance for various configurations assuming ideal channel estimation. The simulation conditions and the results are taken from clause A.12. The quality of service criterion is FER=5 %, to be in line with the measured results. The results are represented in the plot as capacity versus *C/N* for different waveform configurations:

a) 16QAM modulation with FEC coding rates R={1/5, 2/9, 1/4, 2/7, 1/3, 2/5, 1/2, 2/3} and two different physical layer interleaving;

b) QPSK modulation with FEC coding rates R={1/5, 2/9, 1/4, 2/7, 1/3, 2/5, 1/2, 2/3} and two different physical layer interleaving profiles.

The physical layer interleaving profiles are: short =200 ms class 1 interleaver, long=10 sec class 2 interleaver.

Figures 7.27 and 7.28 highlight the additional interleaving gain of long physical interleaving. For 50 km/h already a short interleaving is sufficient and a longer interleaving provides only a small additional gain for 16QAM almost zero for QPSK, whereas for lower speed the gain is significant i.e. 1 dB for 16QAM and 0,5 dB for QPSK.

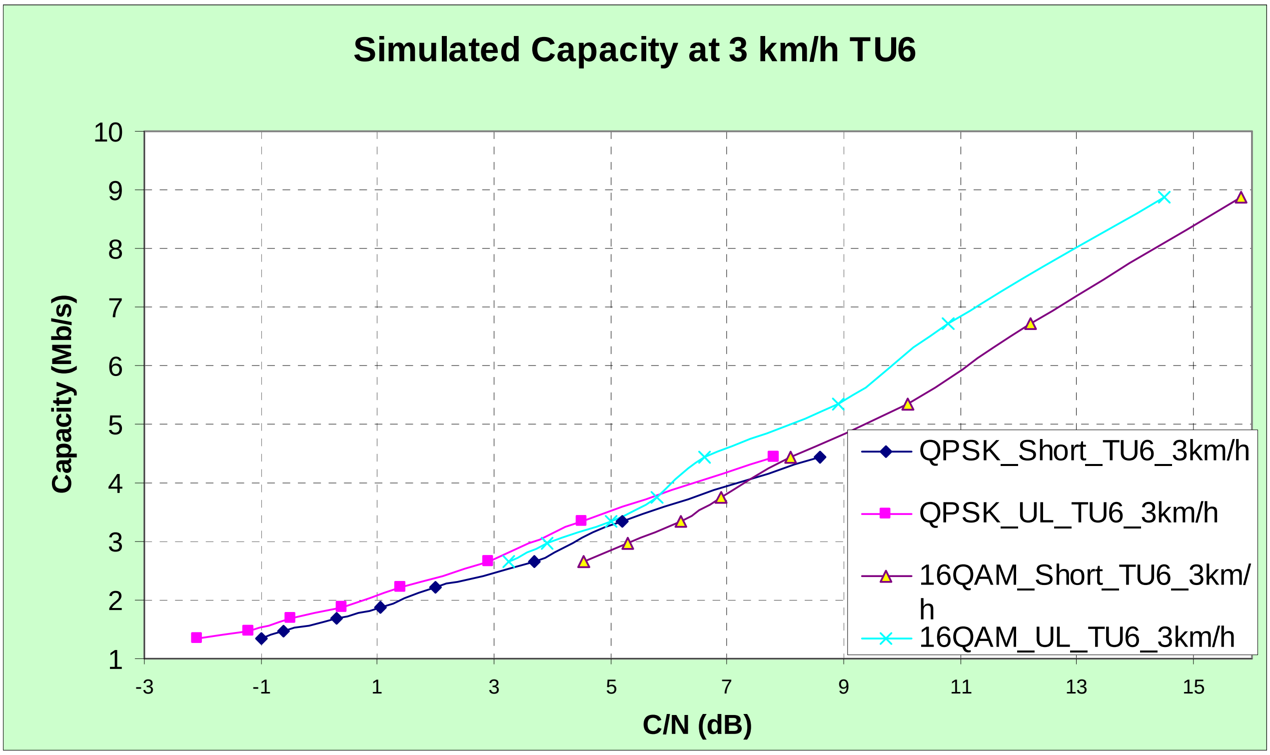


Figure 7.27: Simulated Capacity vs. C/N results for OFDM, TU6 channels at 3 km/h

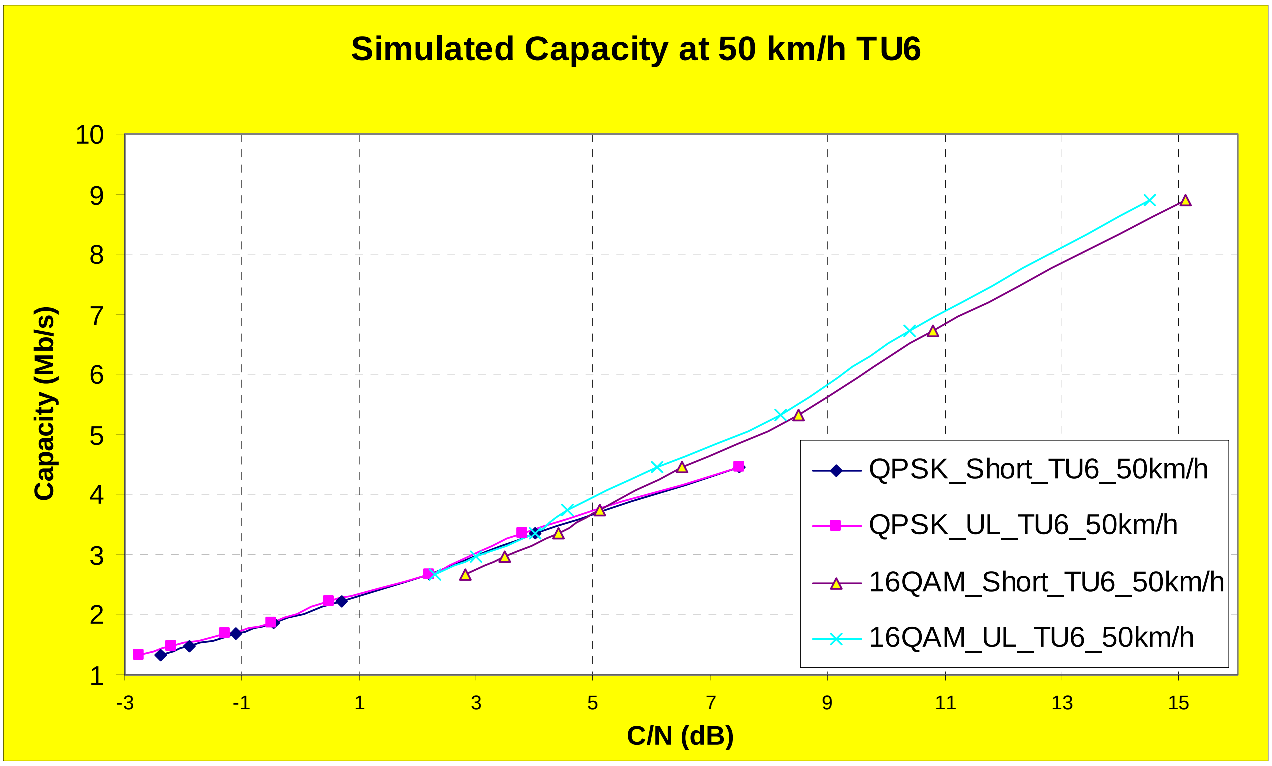


Figure 7.28: Simulated Capacity vs. C/N results for OFDM, TU6 channel at 50 km/h

7.3.1.2.2 Measured capacity performance

Figures 7.29, 7.30, 7.31 and 7.32 summarize the TU6 performance measured in laboratory using a real receiver for various configurations. The tests conditions and the results are taken from clause A.12. The quality of service criterion is FER=5 %. The results are represented in the plot as capacity versus *C/N* for different waveform configurations:

a) 16QAM modulation with FEC coding rates R={1/5, 2/9, 1/4, 2/7, 1/3, 2/5, 1/2, 2/3}, two GI values and two different physical layer interleaving profiles;

b) QPSK modulation with FEC coding rates R={1/5, 2/9, 1/4, 2/7, 1/3, 2/5, 1/2, 2/3}, two GI values and two different physical layer interleaving profiles.

The physical layer interleaving profiles are: short = 200 ms class 1 interleaver, long = 10 s class 2 interleaver. Two cases are provided: with GI of 1/8 and with GI of 1/4.

NOTE 1: The simulated performance is using a uniform-late interleaver profile, for the measurements a uniform interleaver profile is used.

NOTE 2: Some of the previous curves have been computed using GI = 1/8, which is not the baseline in clause A.12. Results reported in figure 7.33 show that for required C/N at low and medium speed, GI has no impact on the required C/N values.

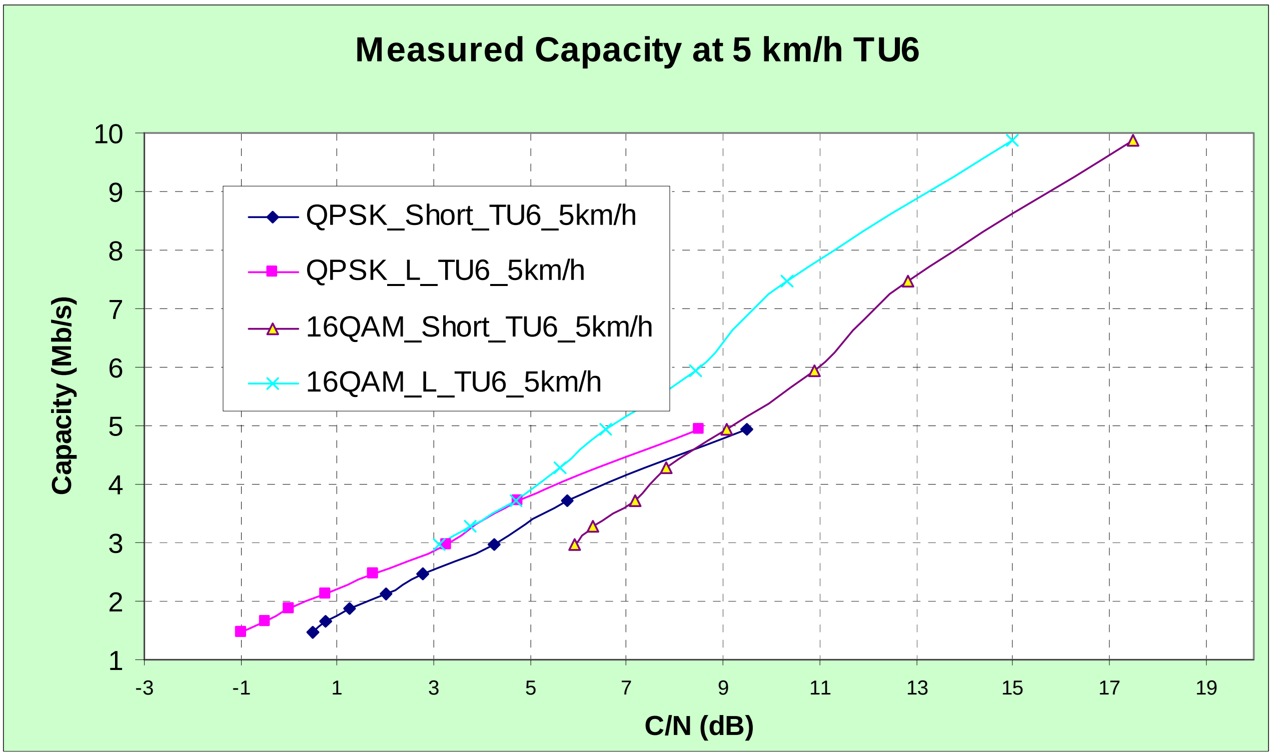


Figure 7.29: Measured Capacity vs. C/N results for OFDM, TU6 channels at 5 km/h (GI = 1/8)

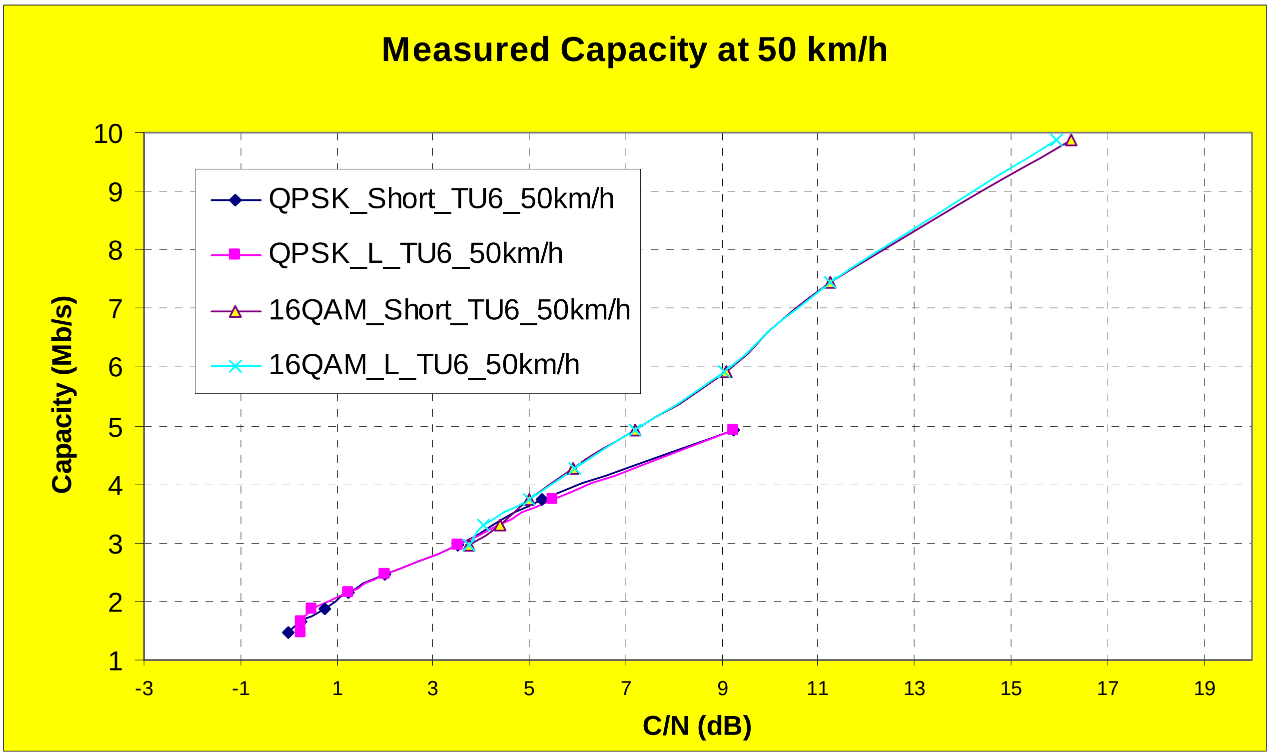


Figure 7.30: Measured Capacity vs. C/N results for OFDM, TU6 channels at 50 km/h (GI = 1/8)

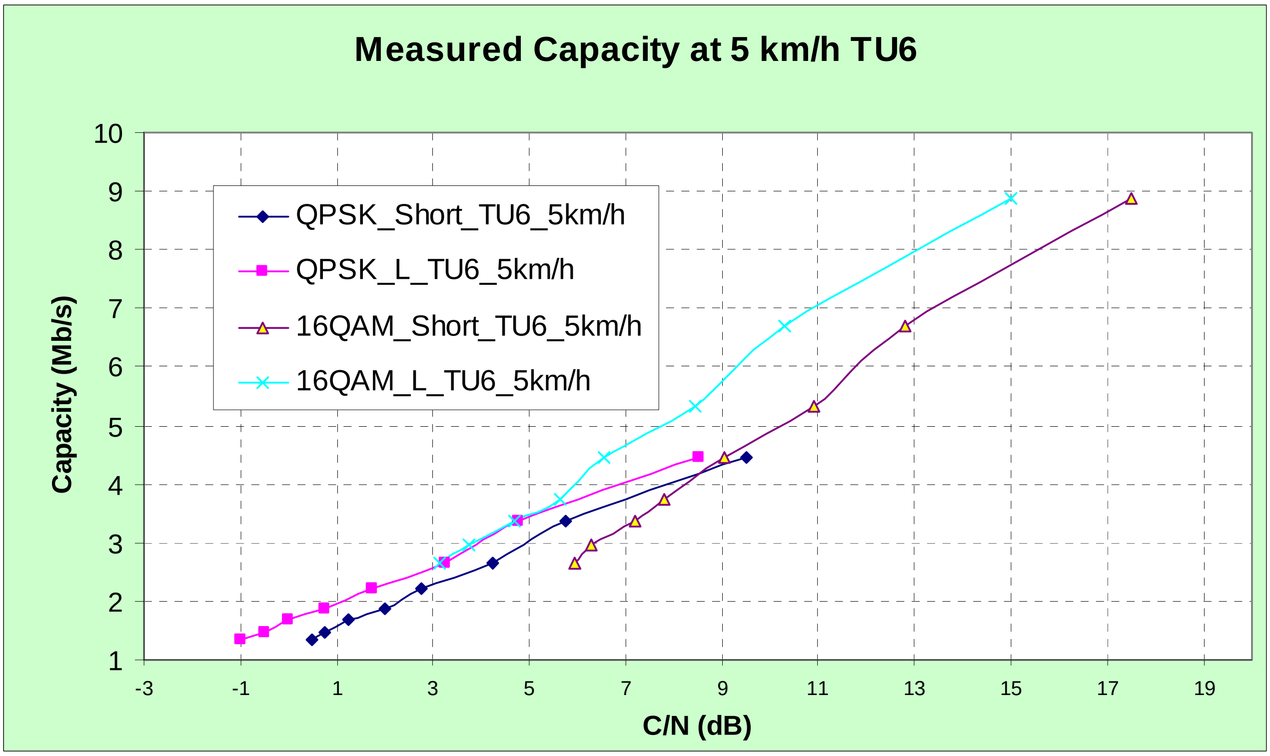


Figure 7.31: Measured Capacity vs. C/N results for OFDM, TU6 channels at 5 km/h (GI = 1/4)

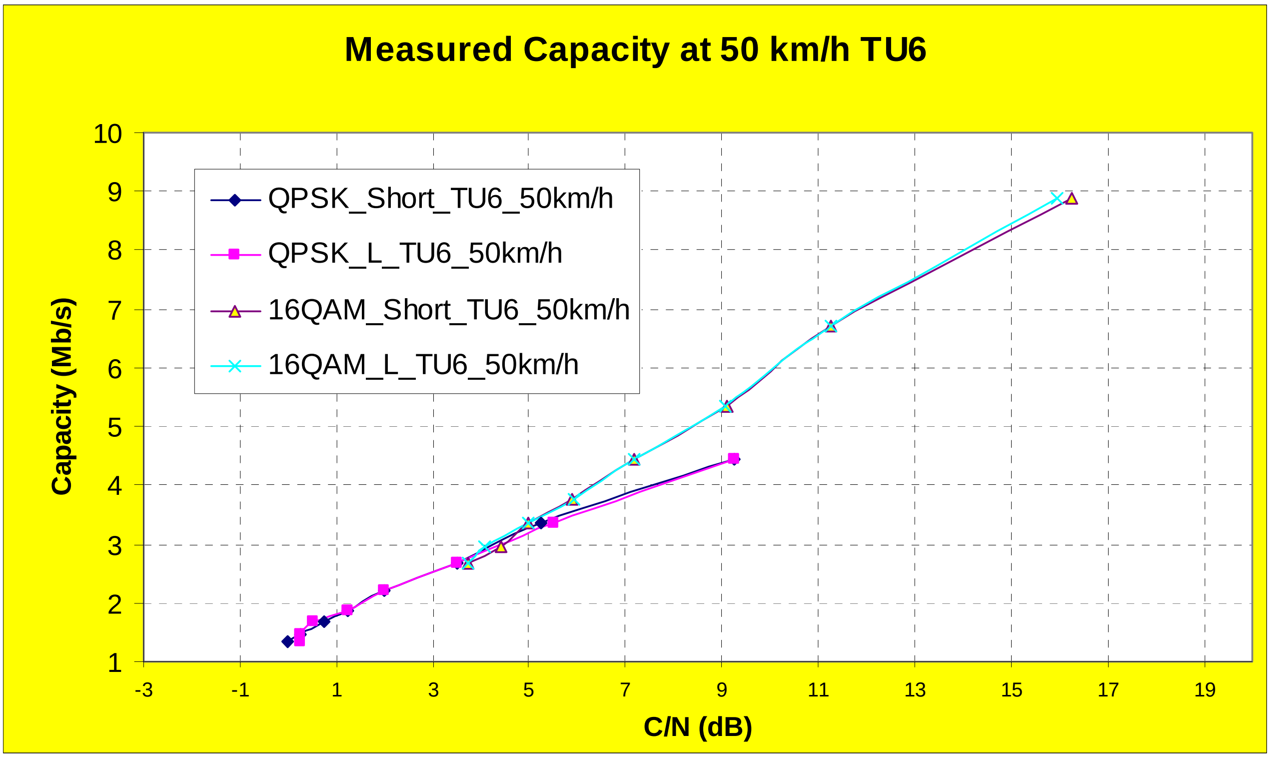


Figure 7.32: Measured Capacity vs. C/N results for OFDM, TU6 channels at 50 km/h (GI = 1/4)

The previous curves confirm that long interleaving does not yield much improvement at medium speed in TU6 channel condition. At lower speed the performance improvement is significant. (1 dB for QPSK and 2 dB for 16QAM). This is due to the fact that for measurement a uniform late interleaver has been used instead of a uniform long interleaver exploited in the simulations.

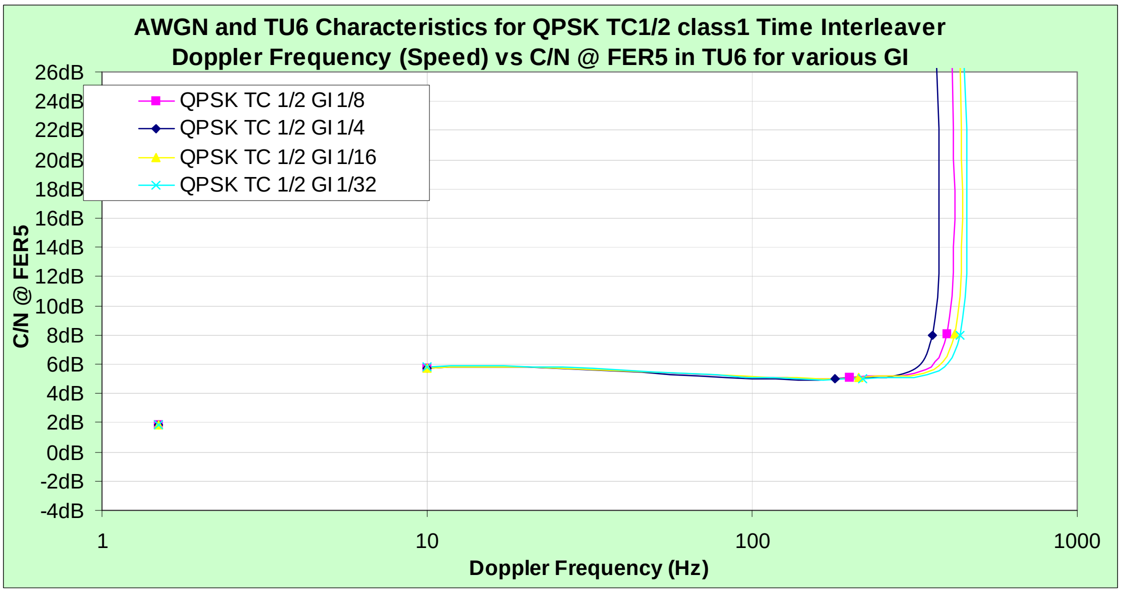


Figure 7.33: Impact of GI on the required C/N for QPSK

Some conclusions:

- There is a good matching between the simulated and the measured capacity in TU6 at various speeds.
- In most of the cases, 1 dB implementation loss can be achieved for both QPSK and 16QAM.

- Long physical layer interleaving shows its benefit on low speed and it is higher for 16QAM than QPSK and also more marked for a real demodulator than an ideal one. This may be due to the long interleaver mitigation effect of channel estimation errors.

7.3.1.2.3 Simulated required C/N as a function of the mobile speed

The used reference receiver model characterizes a DVB-SH receiver performance in an ideal way using two numbers, C/N_{ain} and Fd_{3dB}. The C/N_{ain} gives the minimum required C/N for BER = 10^{-5}. As shown in figure The C/N curve is flat up to high Doppler frequencies. Fd_{3dB} gives the Doppler frequency, where the C/N requirement has raised by 3 dB from the C/N_{ain} value.

Simulation results are given for QPSK and different code rates in figure 7.34. This clause gives a first indication on the performance of the DVB-SH waveform using the reference demodulator of clause A.11.2.1) instead of ideal channel estimation. The values for ideal and real channel estimation are summarized in table 7.9. The interleaver configuration used for this simulation is the short uniform interleaver.

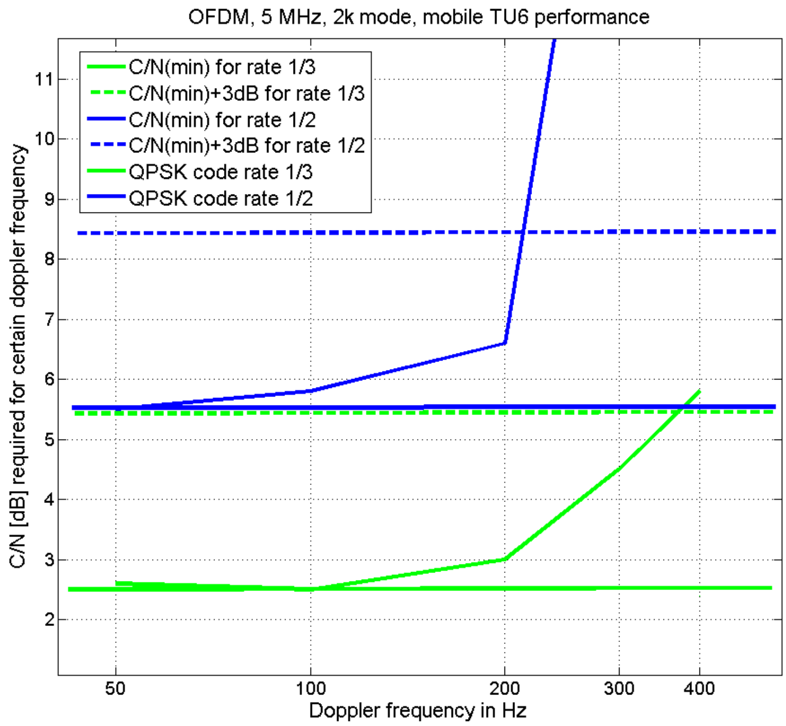


Figure 7.34: Simulated DVB-SH OFDM reference receiver C/N behaviour in mobile TU6 channel

From figure 7.34, the relevant parameters C/N_{ain} and Fd_{3dB} can be extracted for this case.

Table 7.9: Simulated DVB-SH OFDM reference receiver C/N behaviour in mobile TU6 channel

Code rate	Ideal estimation: C/N_{ain}	Real estimation: C/N_{ain}	Fd_{3dB}
1/2	4,0 dB	5,5 dB	230 Hz
1/3	1,0 dB	2,4 dB	370 Hz

Further results on the Doppler performance with DVB-SH are given by measurement results in the following clause.

7.3.1.2.4 Measured required C/N as a function of the mobile speed

Different measurements have been made with all the different modulation/coding cases, and at different speeds up to the maximum sustainable Doppler spread. The results in figures 7.35 and 7.36 represent the variation of required C/N versus the applied maximum Doppler frequency corresponding to a certain speed, depending on the used central frequency.

Different *C/N* values are represented in the following plots:

a) AWGN required *C/N* : "isolated" points on the left of the graph.

b) F_{dmax}, corresponding to the vertical part of the curves, and maximum Doppler frequency.

c) Required *C/N* at $F_{dmax}/2$, corresponding to half of the maximum speed. One cannotice that the required C/N value is quasi constant from 100 Hz to $F_{dmax}/2$.

d) Required *C/N* at $F_{dmax}/2$ + 3 dB corresponding at the previous C/N plus 3 dB and with a corresponding speed.

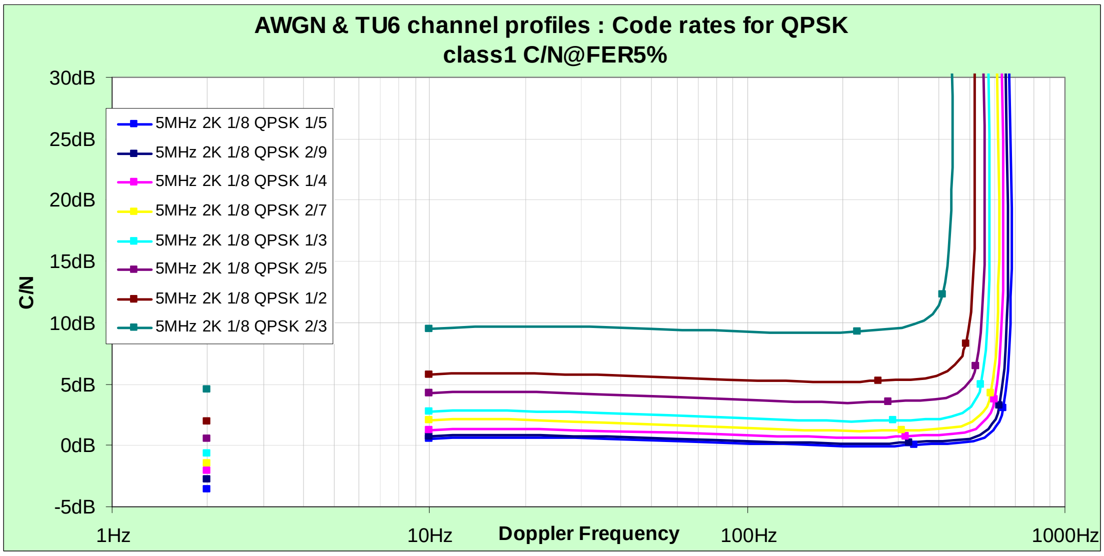


Figure 7.35: QPSK terrestrial performances

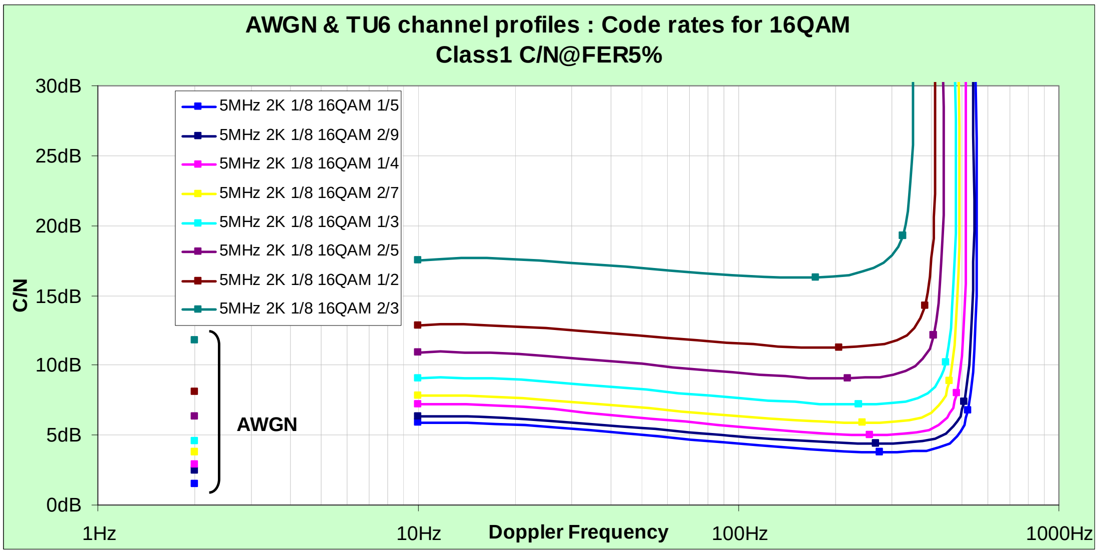


Figure 7.36: 16QAM terrestrial performances

Figure 7.37 provides a summary of the maximum Doppler frequency (F_{dmax}) in S band versus the bit rate.

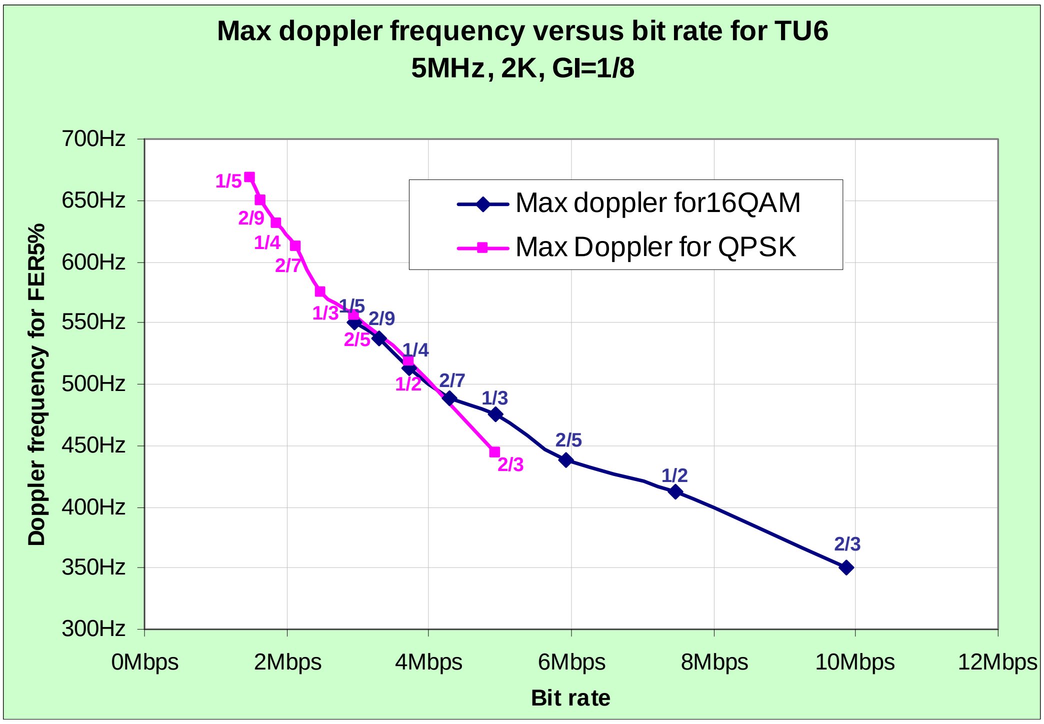


Figure 7.37: Maximum Doppler frequency with respect to bit rate (class 1)

Finally, figure 7.38 shows the expected impact of FFT size on the maximum Doppler frequency.

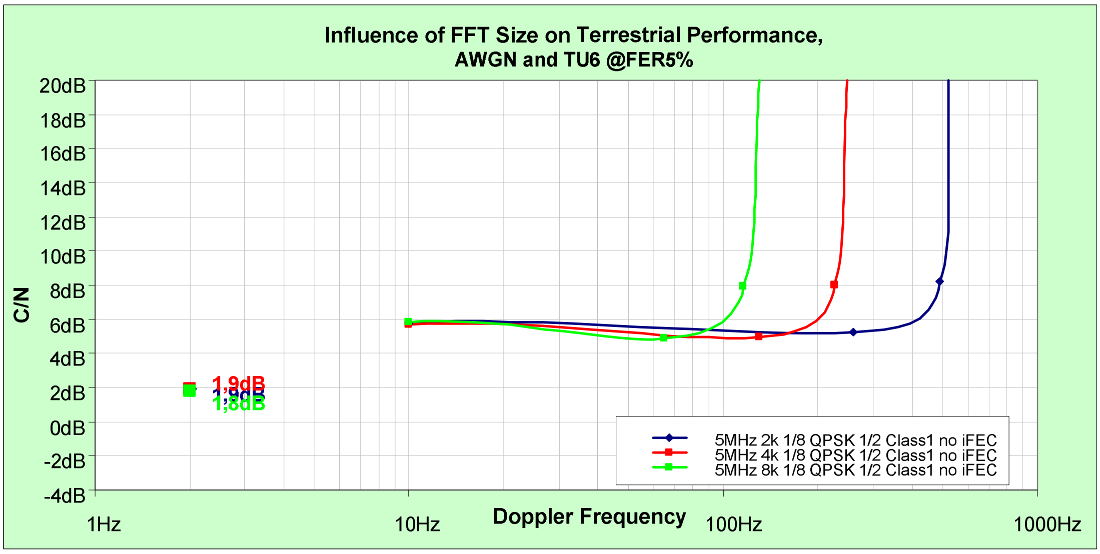


Figure 7.38: Influence of FFT size on maximum Doppler Frequency

Overall these results can be summarised as follows: $F_{dmax,2K} = 2 \times F_{dmax,4K} = 4 \times F_{dmax,8K}$, i.e. F_{dmax} is inversely proportional to the FFT length.

7.3.1.3 Hierarchical terrestrial modulation performance

7.3.1.3.1 Hierarchical modulation implementation

In this clause we provide results based on a simulation campaign for the following simulated radio scenarios for the AWGN environment (hierarchical 16QAM - coding rate = 1/5, 2/9, 1/4, 2/7, 1/3, 1/2, 2/3).

The DVB-SH modulator achieving HM implements two parallel TS data stream of different priority, coded, interleaved and framed (at bit level) independently (figure 7.39):

- High-Priority data stream (HP) with a coding rate R_{HP}.
- Low-Priority data stream (LP) with a coding rate R_{LP}.

The hierarchical mapping consists in mapping each data stream (HP and LP) hierarchically onto the bits word of the 16 constellation points (or symbols) of a standard 16QAM. For each constellation symbol, the two Most Significant Bits (MSB) code the HP data stream and identify the quadrant of the complex signal plane. The two Low Significant Bits (LSB) code the LP data stream and identify the position of the symbol in the quadrant.

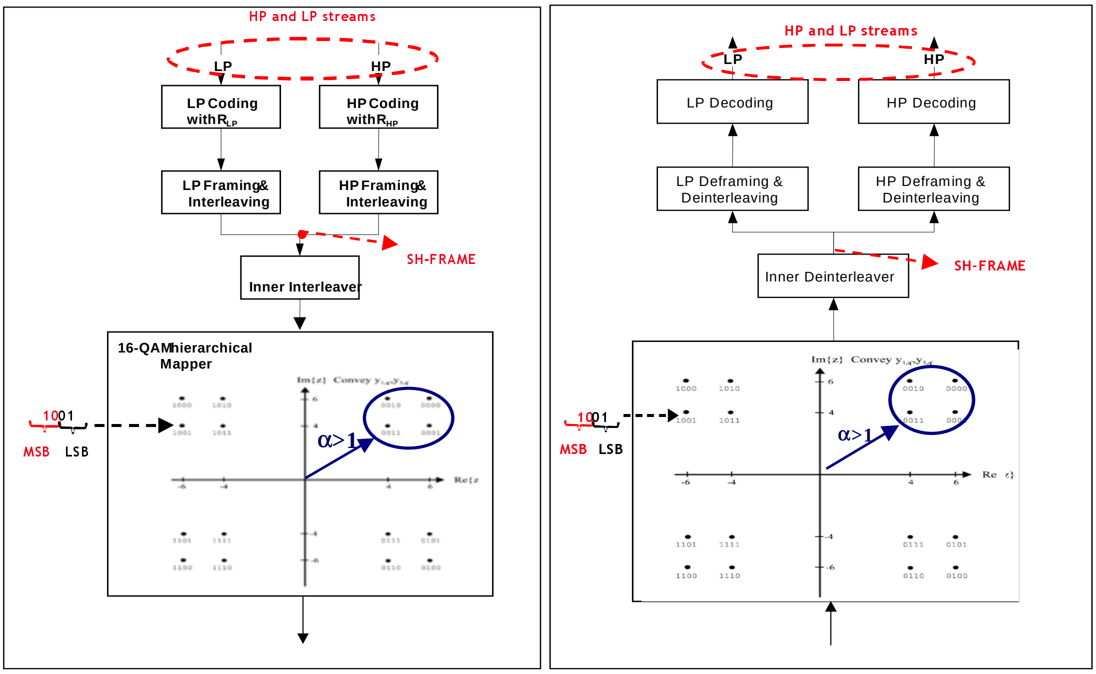
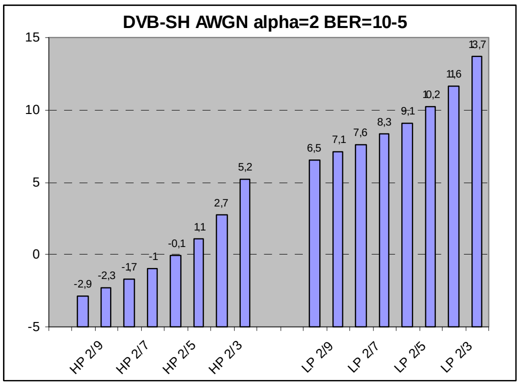


Figure 7.39: Architecture scheme of SH-A receiver implementing hierarchical modulation

At the receiver side, the demapper module receives both TS data streams HP and LP completely de-correlated as inputs. The receiver is basic in the sense that it does not use HP to help of LP decoding, which gives worst-case results. The two parallel independent TS data stream transport independent contents.

7.3.1.3.2 Raw performance results

The performance of HM in AWGN is characterized by SNR operating points leading to BER = 10^{-5} after decoding of the turbo-code and to a maximal MFER 5 %. This targeted performance at physical layer level is assumed to guarantee good video and audio or data decoding at application layer level. Figure 7.40 gives SNR operating points in the hierarchical mode for α = 2 and α =4.



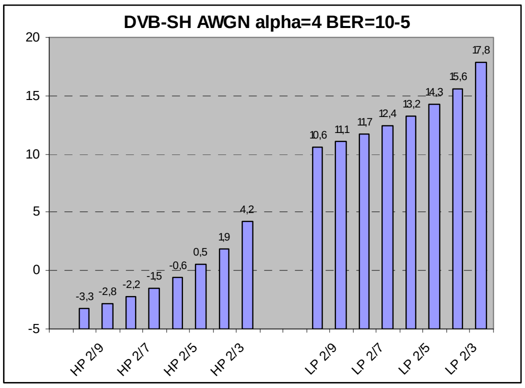


Figure 7.40: Hierarchical reception in DVB-SH versus coding rates - α = 2 and α = 4

Comparing the results obtained in the non-hierarchical mode (baseline QPSK or 16QAM) we note SNR gaps with the SNR operating points obtained in the hierarchical mode for the both values of α. They are given in tables 7.10, 7.11 and 7.12.

Table 7.10: SNR gap for HP hierarchical reception in SH (α = 2 and α = 4) vs. QPSK

HP alpha=2					Penalty	HP alpha=4					Penalty
Coding rate		Label	QPSK	C/N	/ QPSK	Coding rate		Label	QPSK	C/N	/ QPSK
0,2	1/5	HP 1/5	-3,6	-2,9	0,7	0,2	1/5	HP 1/5	-3,6	-3,3	0,3
0,22	2/9	HP 2/9	-3,1	-2,3	0,8	0,22	2/9	HP 2/9	-3,1	-2,8	0,3
0,25	1/4	HP 1/4	-2,5	-1,7	0,8	0,25	1/4	HP 1/4	-2,5	-2,2	0,3
0,29	2/7	HP 2/7	-1,8	-1,0	0,8	0,29	2/7	HP 2/7	-1,8	-1,5	0,3
0,33	1/3	HP 1/3	-0,9	-0,1	0,8	0,33	1/3	HP 1/3	-0,9	-0,6	0,3
0,4	2/5	HP 2/5	0,1	1,1	1,0	0,4	2/5	HP 2/5	0,1	0,5	0,4
0,5	1/2	HP 1/2	1,4	2,7	1,3	0,5	1/2	HP 1/2	1,4	1,9	0,5
0,67	2/3	HP 2/3	3,5	5,2	1,7	0,67	2/3	HP 2/3	3,5	4,2	0,7

Table 7.11: SNR gap for HP hierarchical reception in SH (α = 2 and α = 4) vs. 16QAM

Coding rate		**16QAM**	α = 2		α = 4	
			C/N required	**SNR gap vs. 16QAM**	**C/N required**	**SNR gap vs. 16QAM**
0,20	1/5	0,7	-2,9	**-3,6**	-3,3	**-4,0**
0,22	2/9	1,3	-2,3	**-3,6**	-2,8	**-4,1**
0,25	1/4	1,9	-1,7	**-3,6**	-2,2	**-4,1**
0,29	2/7	2,8	-1,0	**-3,8**	-1,5	**-4,3**
0,33	1/3	3,7	-0,1	**-3,8**	-0,6	**-4,3**
0,40	2/5	5,0	1,1	**-3,9**	0,5	**-4,5**
0,50	1/2	6,8	2,7	**-4,1**	1,9	**-4,9**
0,67	2/3	9,7	5,2	**-4,5**	4,2	**-5,5**

HP decoding at same coding rate:

- maintains almost same performance as QPSK (loss limited to 0,3 dB to 1,7 dB);
- increases hugely performance with regards to non-hierarchical 16QAM (up to 5,5 dB).

When α increases and/or the coding rate decreases:

- C/N penalty with regards to QPSK decreases (at the same coding rate between QPSK and HP);

- C/N gap with regards to 16QAM increases, (at the same coding rate between QPSK and LP).

Table 7.12: SNR gap for LP hierarchical reception in SH (α = 2 and α = 4) vs. QPSK and 16QAM

LP alpha 2						Penalty	
Coding rate		Label	16QAM	QPSK	C/N	/ QPSK	/ 16QAM
0,2	1/5	HP 1/5	0,7	-3,6	6,5	10,1	5,8
0,22	2/9	HP 2/9	1,3	-3,1	7,1	10,2	5,8
0,25	1/4	HP 1/4	1,9	-2,5	7,6	10,1	5,7
0,29	2/7	HP 2/7	2,8	-1,8	8,3	10,1	5,5
0,33	1/3	HP 1/3	3,7	-0,9	9,1	10,0	5,4
0,4	2/5	HP 2/5	5,0	0,1	10,2	10,1	5,2
0,5	1/2	HP 1/2	6,8	1,4	11,6	10,2	4,8
0,67	2/3	HP 2/3	9,7	3,5	13,7	10,2	4,0

LP alpha 4						Penalty	
Coding rate		Label	16QAM	QPSK	C/N	/ QPSK	/ 16QAM
0,2	1/5	HP 1/5	0,7	-3,6	10,6	14,2	9,9
0,22	2/9	HP 2/9	1,3	-3,1	11,1	14,2	9,8
0,25	1/4	HP 1/4	1,9	-2,5	11,7	14,2	9,8
0,29	2/7	HP 2/7	2,8	-1,8	12,4	14,2	9,6
0,33	1/3	HP 1/3	3,7	-0,9	13,2	14,1	9,5
0,4	2/5	HP 2/5	5,0	0,1	14,3	14,2	9,3
0,5	1/2	HP 1/2	6,8	1,4	15,6	14,2	8,8
0,67	2/3	HP 2/3	9,7	3,5	17,8	14,3	8,1

LP decoding at the same coding rate is affected by an important performance degradation compared to non-HM:

- Loss versus non-HM 16QAM: 4 - 5,8 dB penalty for alpha = 2 and between 8 dB to 9,9 dB for α = 4;
- Loss versus non-HM QPSK: 10 dB penalty for α = 2 and 14 dB penalty for α = 2.

This penalty increases by 4 dB with alpha=4 (alpha = 2 is better than alpha = 4).

Decreasing the coding rate has little effect on QPSK but degrades (up to 1,8 dB) 16QAM performance.

7.3.1.3.3 Summary:

Based on AWGN simulation results, the following conclusions can be drawn.

For HP stream an α value of 4 represents the best choice. However this value would cause a major (4 dB compared to α = 2) performance loss increase for the LP stream. Therefore α = 2 seems to be a good trade-off. Using this α value, in an AWGN channel we can see that a "basic encoding and decoding" can provide two types of usages:

- Starting from a non-HM QPSK reference point:
  - HM can provide an increased throughput on the LP:
    - At marginal cost on HP: 0,7 dB to 1,7 dB penalty with regard to QPSK at the same coding rate.
    - The penalty of LP with regard to non-HM QPSK is 10 dB at same code rate, which may require to use lower code rates (and then different bit rates) on this TS, reducing the penalty to as little as 3 dB.
- Starting from a non-HM 16QAM reference point:
  - HM can provide an important SNR gain on the HP:
    - This gain of HP with regards to 16QAM is between 3,6 dB to 4,5 dB for the same code rate; Using lower code rates on HP stream (and thus different bit rates regarding to 16QAM code rate) is increasing the gap up to 12,6 dB.

The reception area of both LP and HP would be impacted by a loss of 4,0 dB to 5,8 dB for the same code rates. Different code rates could be used to reduce this penalty down to 1,5 dB for an identical bit rate by increasing HP and decreasing LP. In such condition, the HP gain is also lowered. Some trade off can be found at 1/3 and 1/2 code rates where there are gains and penalties of the same order as illustrated below in figure 7.41.

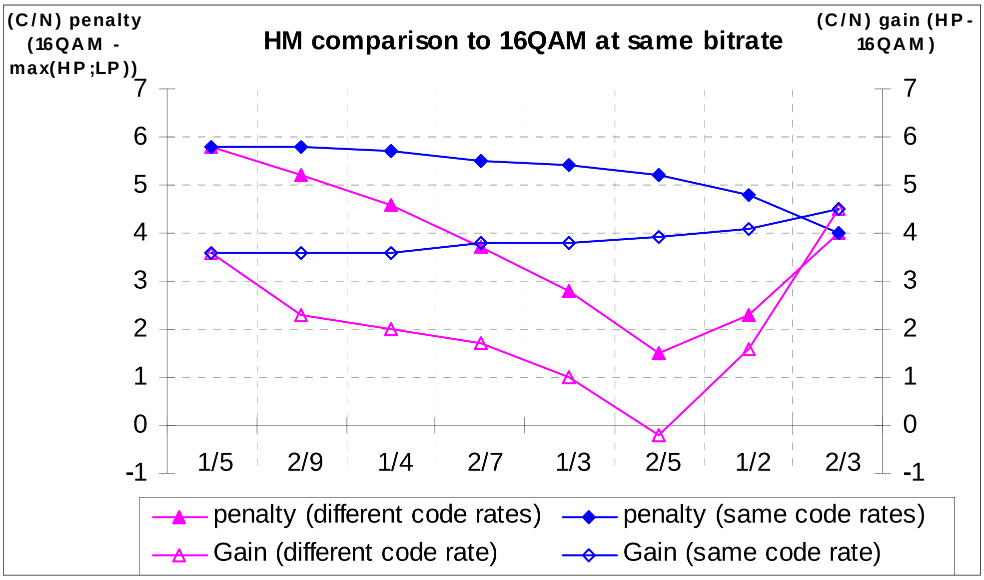


Figure 7.41: C/N penalty and gain of HM versus 16QAM as a function of code rate

7.3.2 TDM Overview

The TDM modulation of DVB-SH is derived from DVB-S2 with reasonable adaptations necessary. An important new element is the constant distance of the pilot fields, independent from code rate choice, plus an addition of a signalling field incorporating all relevant parameters for FEC and interleaving.

The guidelines concerning modem algorithms design are given in the clause A.11.

TDM-specific simulation results are presented in clause A.11.

7.3.2.1 TDM Bandwidth Selection

The DVB-SH waveform specification makes no assumption on the alignment of TDM and OFDM bandwidth in the SH-B case although the examples given are all considering equal bandwidth. Therefore, although unequal bandwidth configurations for TDM and OFDM are made possible by the waveform specification, the unequal bandwidth configurations are not considered for this release of the implementation guidelines that consider equal bandwidth in the TDM and OFDM modulations.

7.4 Combining techniques

Only a short overview of combining techniques is given below.

7.4.1 Overview of diversity combining techniques

In general the diversity combining techniques can be grouped in:

a) antenna diversity;

b) satellite-to-terrestrial selective combining (incl. hand-over);

c) diversity combining between different reception path (including hand-over to other frequency bands).

Different methods of combining techniques are defined:

- **selective combining:** the signal, which provides better quality, is chosen;
- **maximum ratio combining:** the signals are combined, weighted accordingly to its specific reception quality;

- **complementary code combining:** different code bits are combined, chosen accordingly to its occurrence in the puncturing pattern.

For the satellite-to-terrestrial combining and hand-over different scenarios have to be considered.

- SH-A, SFN
  In this case no special considerations have to be made for the combining inside the demodulator as combining occurs directly at the receiver antenna (see note). The satellite signal is considered as additional repeater in a SFN. But significantly higher delay spread may result and new characteristics of the channel impulse response have to be taken into account.

NOTE: It should be remarked that in case of SFN operations for both satellite and terrestrial repeaters the combining takes place at the antenna thus the signals are summing up with their own phase. This may create temporary constructive or destructive signal combining.

- SH-A, MFN
  In this case the same content may be available over different carrier frequencies. As a consequence, it may be worthwhile that the receiver scans in the background for alternative (better) sources for the program to support a seamless hand-over.

- SH-B
  In case of SH-B architecture the satellite and terrestrial signal is demodulated by separate demodulators. Three combining solutions are possible in this case:

  - the signal is selected after the FEC decoding (selective combining). This method is not recommended, as it provides the poorest performance;

  - the combining is done after de-interleaving and before FEC decoding. This allows that different interleaver profiles and even different code rates are used for the satellite and terrestrial signal. This method is called code combining. Code combining is very similar to maximum ratio combining. Further details on code combining and maximum ratio combining are given in clause 7.2.2.3.3;

  - combining before the de-interleaving (maximum ratio combining or selective combining).
    This method will only work if the terrestrial branch and the satellite branch use the same code rate and interleaver parameter. For a fully SH-B compliant receiver this method is therefore not recommended. This combining point is attractive for antenna diversity, where the TDM and OFDM streams are combined separately;

  - combining before de-interleaving and code combining can be applied together at the same time, enabling for a very powerful diversity reception.

In case of SH-B other strategies may be applicable also. For example it may be possible to completely switch off a receiver branch completely to reduce the power consumption. The evaluation of these techniques requires joint channel models covering the satellite and terrestrial propagation characteristics in one model. These models have been developed under the framework of the DVB-SH validation task force and are presented in TS 102 591-2 [34].

7.4.2 Void

The content of this clause has been integrated into clause 7.4.1.

7.4.3 Void

The content of this clause has been moved to clause 7.7.

7.4.4 Experimental results for hybrid networks

This clause is informative.

7.4.4.1 Introduction

As explained above, there are many combining techniques. Amongst these combining techniques , there are those related to the Network Architecture, typically the Hybrid Network architecture, where satellite and terrestrial signals use the same frequency band and use the same modulation namely SHA-SFN mode, where signal combining is performed naturally at the user terminal antenna. The Hybrid Network description is included in clause 11 and this section gives the summary of following classes of results:

- Simulations results.
- Laboratory measurements results.
- Field trial results.

The Hybrid Channel tests and simulations aim to address hybrid situations when the on-air signal is a combination of terrestrial and satellite signals. This situation is particularly interesting when the satellite and the terrestrial signals are both using OFDM at the same frequency. This mode is called DVB-SHA in hybrid SFN in the standard.

More generally, hybrid channel is defined by the combination at the receiver of:

- One satellite link under ITS or SU conditions.
- One or several terrestrial links under TU6 conditions.

In the case of Hybrid SFN, the two signals are at the same frequency, using the same modulation/coding/interleaving schemes, and carrying exactly the same data. In this section, only SHA-SFN configurations have been retained from simulations and laboratory tests:

- OFDM over satellite and terrestrial at same frequency bands.
- A single terrestrial repeater is emulated.

The objectives of the measurements are:

- Assess the improvement of system coverage beyond the edge of the terrestrial cell coverage where the received C/N is below the required threshold.
- Assess the possible existence of interference when satellite and terrestrial signals are not in the GI window and the impact of these interferences.
- Assess the influence of the GI on the interference reduction when the interferences are existing.

Taking into account the above objectives it will not be necessary to test the following "weak" hybrid situations:

a) A very high terrestrial C/N (for instance 20 dB): this terrestrial signal will be very "dominant" and no satellite component impact will be observed.

b) A very low terrestrial C/N as also no terrestrial component impact will be observed being the satellite signal dominant.

The different tests conditions are the following:

- Waveform baseline configurations:
  - 5 MHz/2k/GI 1/8 or 1/4.
  - Long Interleaving : class 2 and class 1 plus IFEC.
- Satellite channels configurations:
  - ITS 50 km/h.

- SU 50 km/h.

- Terrestrial network configurations:
  - 2 kW EIRP single repeater.
  - Different repeater heights: 30 m, 45 m and 60 m.
  - TU 6 channel.
  - Use of Costa Hata model in SU and Rural conditions.

- The Receiver is a car receiver, single antenna with satellite oriented receiver (circular polarized antenna) as most of the satellite reception area will be on roads and highways, and is located at the point corresponding to the terrestrial C/N received value.

The operating mode for the simulations is the following:

- Based on the terminal location and repeater characteristics, terrestrial C/N is settled at the wanted value, which corresponds to a certain differential delay.
- The satellite received C/N is settled at the required value to comply with the ESR5(20) fulfilment ratio of 90 %.
- Then compute the ESR5 with the CGC repeater on.
- Observe if there is a performance improvement or degradation.

The basic principles of the laboratory tests are the following:

- Take the reference curve in LMS (ITS or SU) satellite reception with class 2 and/or IFEC.
- Set the terrestrial C/N to the wanted value, which corresponds to a certain differential delay.
- Perform the hybrid configuration. tests for different satellite received C/N.
- Compare the ESR5 fulfilment curve with the LMS reference curve mentioned before.
- Observe if there is a performance improvement or degradation.

As said above, the tests will be performed with class 2 and class 1 plus IFEC receiver configurations. In class 2, only QPSK 1/3 and 16QAM 1/3 configurations have been tested. For class 1 plus IFEC, the following configuration has been tested: 16QAM 1/2 at physical layer plus MPE-IFEC 70 % with GI = 1/8.

In a hybrid channel, there are three main parameters to consider:

- The received terrestrial C/N.
- The received satellite C/N.
- The difference of time of arrival of satellite and terrestrial signals, called here Delta T (ΔT). In the test, the satellite signal C/N is varying for each couple: C/N terrestrial/Delta T corresponding to mobile position.

The different couple of parameters are provided in the clause A.13.1.3 for each repeater configuration. To limit the number of cases, terrestrial, C/N is comprised between -10 dB and +10 dB (5 values). The detailed configurations are described in the clause A.13.1.3, but the correspondence between C/N and Delta T are summarized in table 7.13.

Table 7.13: Summary of Terrestrial C/N values and timing differences

Repeater at 30 m		Repeater at 45 m		Repeater at 60 m	
C/N (dB) terrestrial	Delta T (µs)	C/N (dB) terrestrial	Delta T (µs)	C/N (dB) terrestrial	Delta T (µs)
Sub Urban Case					
-10	57,07	-10,2	73,60	-7,6	73,60
-5,1	40,82	-4,7	48,24	-2,2	48,24
0	28,35	0,6	34,49	3	34,49
4,3	19,79	4,6	25,38	6,8	25,38
10,4	12,3	9,7	17,23	10,8	17,23
Rural Channel					
-9,6	107	-9,8	140,57	-6,9	140,57
-5,2	80,26	-4,8	100,30	-2	100,30
0	57,07	-0,2	73,60	2,4	73,60
4,9	40,82	5,3	48,24	7,8	48,24
10	6,99	9,4	25,38	-6,9	140,57

Two classes of hybrid channels can be defined:

- The Sub Urban channel, which is the combination of SU LMS channel and Sub Urban Cost Hata channel.
- The Rural Channel, which is the combination of ITS LMS channel and Sub Urban Cost Hata channel.

7.4.4.2 Simulation results

The simulations shown here have been made in SHA/SFN configuration with class 2 terminal only. In the simulator, with channel estimator, the reference C/N for ESR5 for 90 % fulfilment are the following:

- In ITS 50 km/h, C/N = 8,6 dB.
- In SU 50 km/h, C/N = 6,0 dB.

In the laboratory tests:

- In ITS 50 km/h, C/N = 9,5 dB.
- In SU 50 km/h, C/N = 6,0 dB.

Table 7.14 gives some simulations results and a comparison with the equivalent laboratory tests results.

Table 7.14: Examples of simulations results in hybrid channel

Modulation/ coding	Channel/ site height	C/N values (dB) for the simulations	Delta T (µs)	Simulated ESR5 Fulfilment %	Measured Laboratory ESR5 fulfilment % With C/N_{sat} = 9,5 dB
GI = 1/8					
QPSK 1/3	Sub-Urban 45 m	- 9,7 dB terr. - 6 dB sat	17	100	100
QPSK 1/3	Sub urban 45 m	- -10,2 terr. - 6 dB sat	73	85	80
QPSK 1/3	ITS 45 m	- -9,8 dB terr. - 8,6 dB sat	140	78	72
QPSK 1/3	ITS 30 m	- -9,6 terr - 9 dB (+0,4 dB)	107	82	89 [C/N_{sat} = 10,0 dB]
GI = 1/4					
QPSK 1/3	ITS 45 m	- -9,8 dB terr. - 9 dB sat	140	91	91 [C/N_{sat} = 9,7 dB]

Table 7.14 shows some degradation when Δ T is larger than the GT (here 44 µs with GI = 1/8). The different simulations results are quite in line with laboratory tests summarized in the following clause.

7.4.4.3 Laboratory results

As detailed results are provided in the clause A.13.1.3, a synthesis of the different tests results is provided, showing the impacts of hybrid channel.

First an example of results is provided in figure 7.42: it represents the different results in sub-urban channels with 16QAM 1/3 at 40 km/h with class 2 terminal, and repeater at 45 m height.

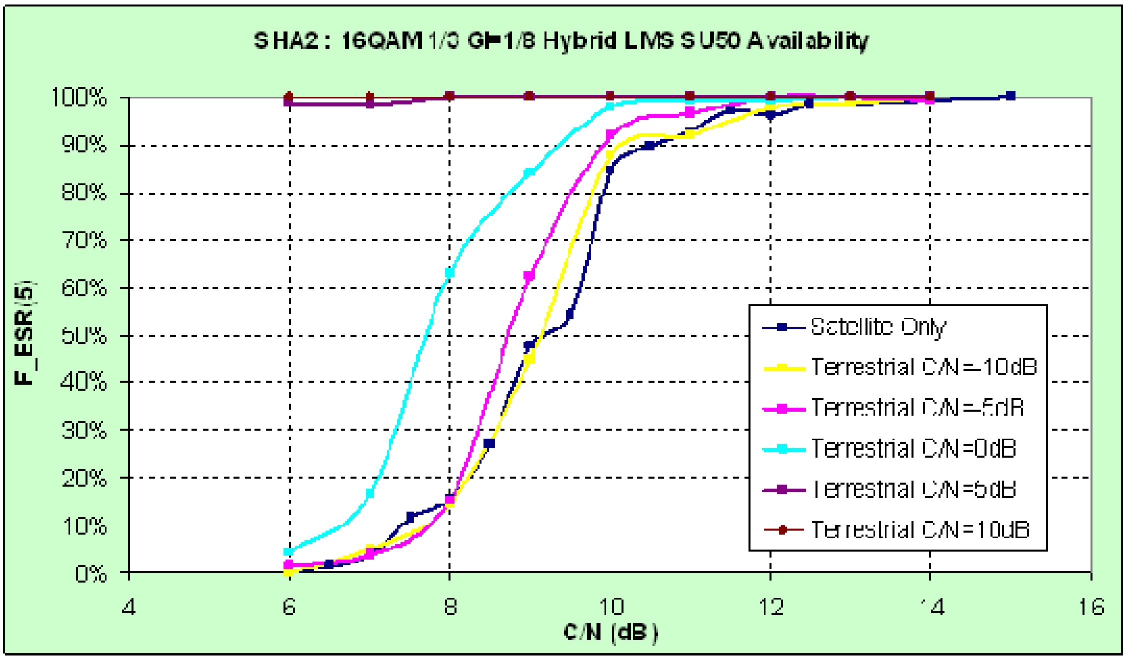


Figure 7.42: Examples of performances in Sub Urban area at 45 m with 16QAM 1/3

From this example we can derive the following conclusions:

- Most of the hybrid reception cases provide improvement compared to the satellite only reception as the required C/N is decreasing even when the differential delay is out of the GI (see table 7.15).
- When the terrestrial signal is under the C/N threshold (for instance C/N = 0 dB in QPSK while 2 dB are required), the resulting quality of service with satellite on (and even with very low satellite *C/N* values) is above 99 % ESR5;
- In QPSK, only the yellow curve with terrestrial received C/N equal to -10 dB, and with differential delay equal to 164 % of the GI_ shows some slight degradation in the satellite reception (< 0,5 dB).

Figure 7.43 represents the required C/N in satellite channel when in presence of terrestrial channel to get 90 % ESR5 fulfilment rate. All possible repeater heights have been considered. The first case is the QPSK 1/3 case, in two variants: GI = 1/8 and GI = 1/4. Abscise represents the terrestrial C/N, and different curves are representing for different repeater heights.

The blue horizontal line in figure 7.43 represents the C/N required in LMS channel only. The following results have been found:

- In SU channel the required C/N is 6 dB for ESR5 fulfilment of 90 %.
- In ITS channel, requited C/N is 9,5 dB for ESR5 fulfilment of 90 %.

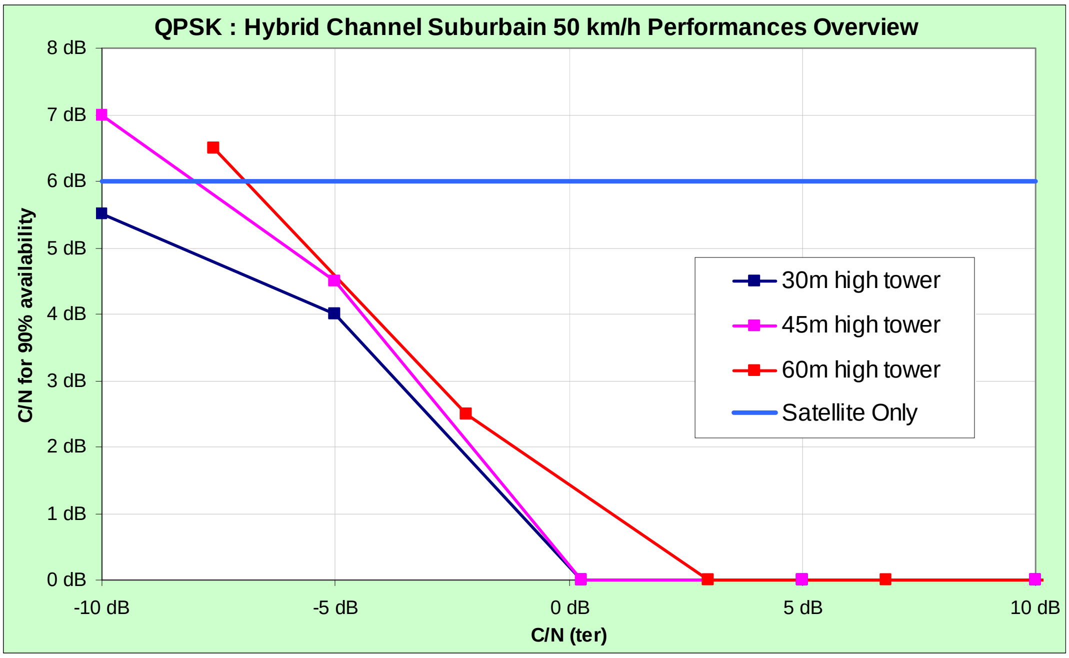


Figure 7.43: QPSK Synthesis in Sub Urban channel with GI = 1/8

In fact the points below the blue horizontal line correspond to the hybrid mode improvement region while the points above correspond to the degradation region, In the case of the 30 m height and terrestrial C/N = -5 dB, the required availability is obtained with satellite channel C/N of only 4 dB. It is remarked that the maximum loss of 1 dB in SU channel for GI = 1/8 occurs when the terrestrial C/N = -10 dB. At this terrestrial C/N the differential satellite/terrestrial delay is outside the guard interval. as it corresponds to Delta T of 140 µs, which represents 314 % of GI = 1/8, and 157 % at GI = 1/4.

For the case of GI = 1/4 reported in figure 7.44, the critical differential cases have disappeared, and there is improvement for the full terrestrial C/N range.

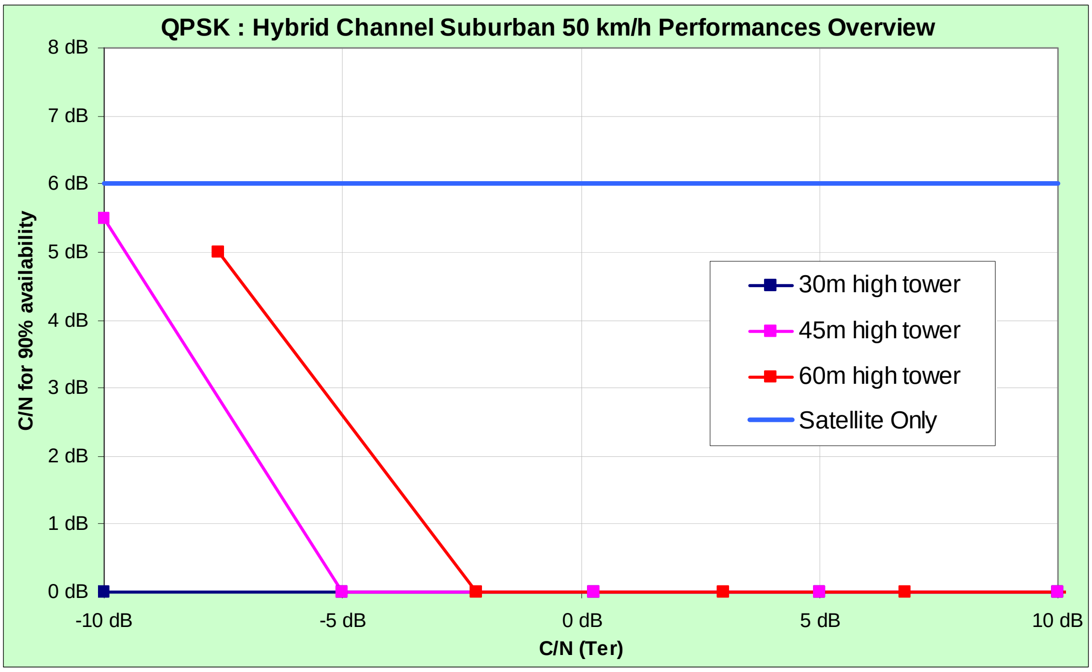


Figure 7.44: QPSK Synthesis in Sub Urban channel with GI = 1/4

For the rural channel case, the experimental findings are reported in figure 7.45 for GI = 1/8 and in figure 7.46 for GI = 1/4.

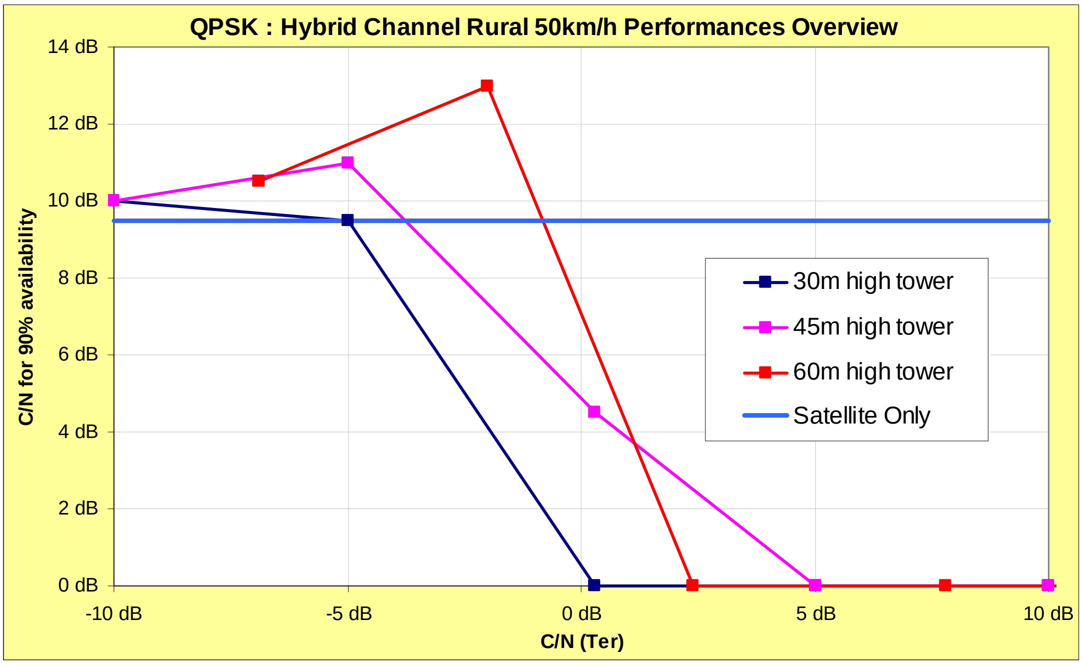


Figure 7.45: QPSK Synthesis in Rural Channel with GI = 1/8

Figure 7.45 results with GI = 1/8 shows some degradation (up to 3,3 dB) for 45 m and 60 m repeater height. Looking at figure 7.46 it is noted that the larger GI makes losses negligible i.e. less than 0,5 dB.

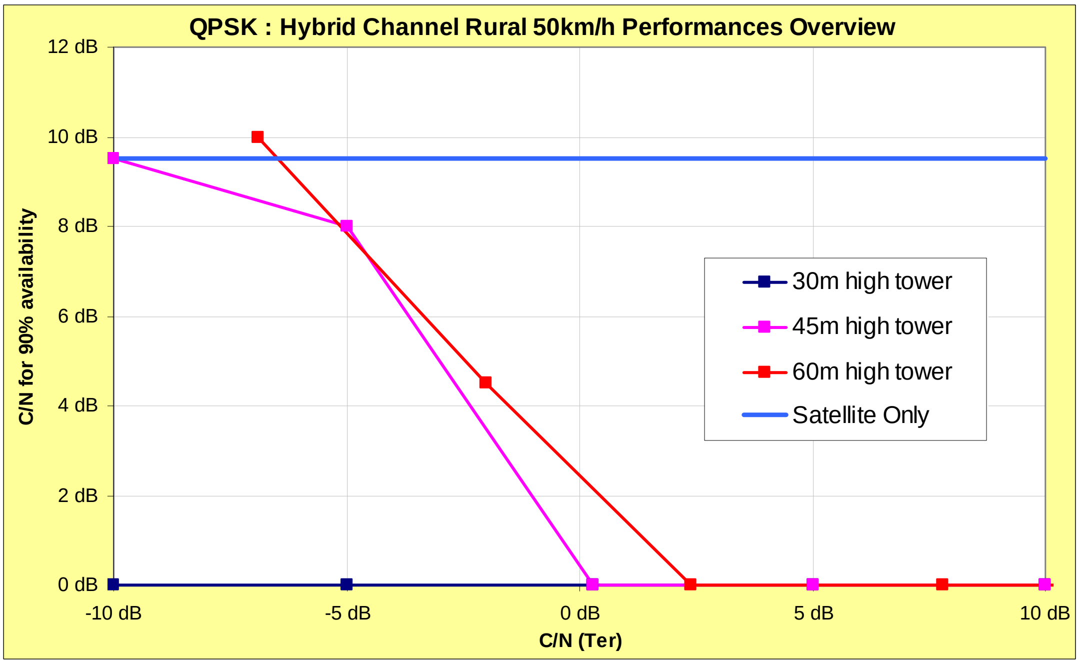


Figure 7.46: QPSK Synthesis in Rural Channel with GI = 1/4

Figure 7.47 provides the results for 16QAM modulation with GI = 1/8. No results are available for 16QAM modulation with GI = 1/4. It is expected that GI = 1/4 will give the same improvements as for QPSK 1/3.

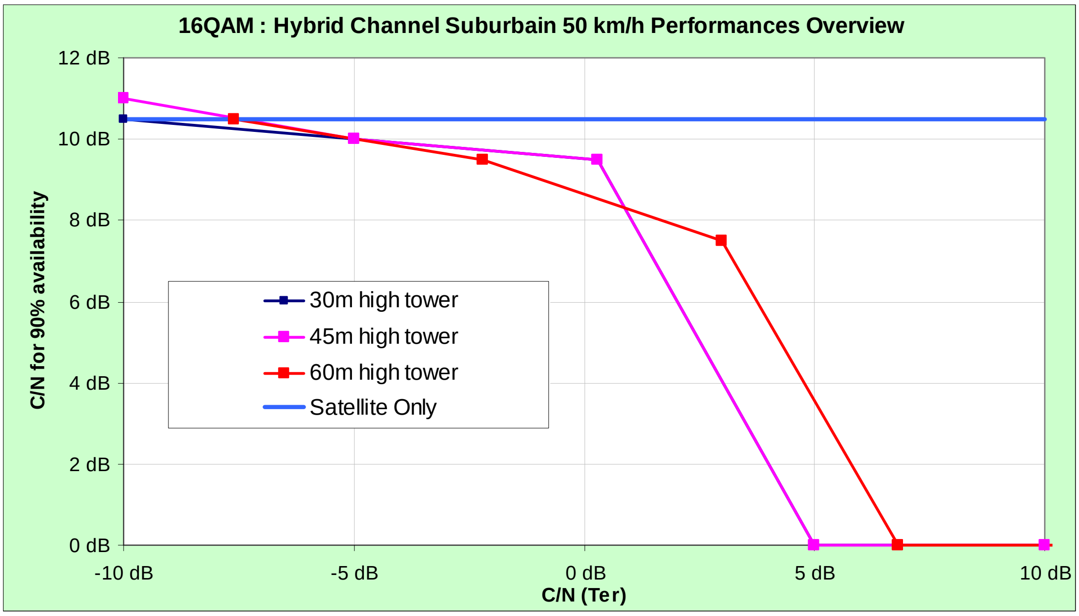


Figure 7.47: 16QAM Synthesis in Sub Urban channel with GI = 1/8

In all cases, there is an improvement compared with Satellite Channel Only or terrestrial channel only, except for 45 m at C/N = -10 dB, with 0,5 dB degradation.

Figure 7.48 provides results for the case of 16QAM modulation with rural channel and GI = 1/8. Figure 7.48 shows some degradation in particular for a tower at 60 m height but improvement have been measured for terrestrial C/N above 0 dB. As said above, the case of 1/4 with 16QAM 1/3 was not performed, but as degradation with 1/8 was lower for 16QAM than for QPSK, with GI = 1/4, it is expected that degradation will be lower.

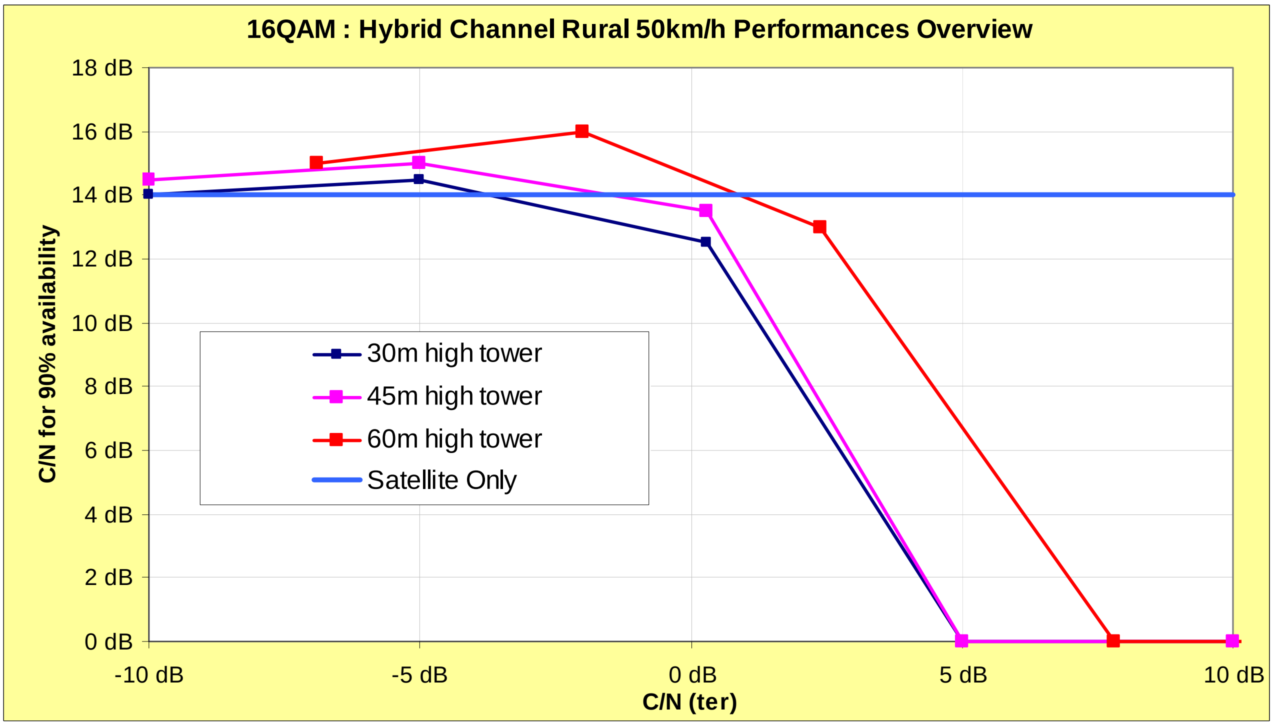


Figure 7.48: 16QAM Synthesis in Rural with GI = 1/8

For the case of MPE-IFEC, not all the cases have been measured and only GI = 1/8 has been considered. The most representative cases are summarized in table 7.15, which represents the ESR5 fulfilment ratio (three bottom rows) for various terrestrial C/N values and Delta T/GI ratio. The satellite C/N is set to 12 dB which provides 90 % ESR 5 fulfilment for SU LMS channel at 50 km/h.

Table 7.15: Summary of results for MPE-IFEC

MPE-IFEC	Delta T/GI ratio				
	0 %	50 %	100 %	150 %	200 %
C/N terrestrial	ESR 5 fulfilment ratio				
-6 dB	90 %	90 %	87 %	68 %	62 %
0 dB	92 %	92 %	91 %	51 %	8 %
6 dB	100 %	100 %	100 %	99 %	94 %

From table 7.15 we can derive the following conclusions:

- When the terrestrial signal is strong (C/N = 6 dB for instance) and whatever the differential delay, the improvement is always excellent: 100 % ESR5 when differential delay is up to 100 % of the GI. When differential delay exceeds 100 % of the GI, some improvement is also noticeable.
- When terrestrial signal C/N is around 0 dB and with a differential delay below 100 % of the GI, the results are good. The performance is degraded when the differential delay increases, but we observe ESR5 fulfilment reduction to 8 % corresponding to only 2 dB C/N degradation due to the sharp roll-off of the ESR5 curve.
- When the terrestrial signal is weak, (C/N around - 6 dB), there is only a slight degradation when differential delay is equal to 100 % of the GI (around 0,1 dB), but some degradation for larger delays, quite low in terms of C/N, even if the ESR (5) value varies quite a lot.

7.4.4.4 Conclusions on laboratory measurements

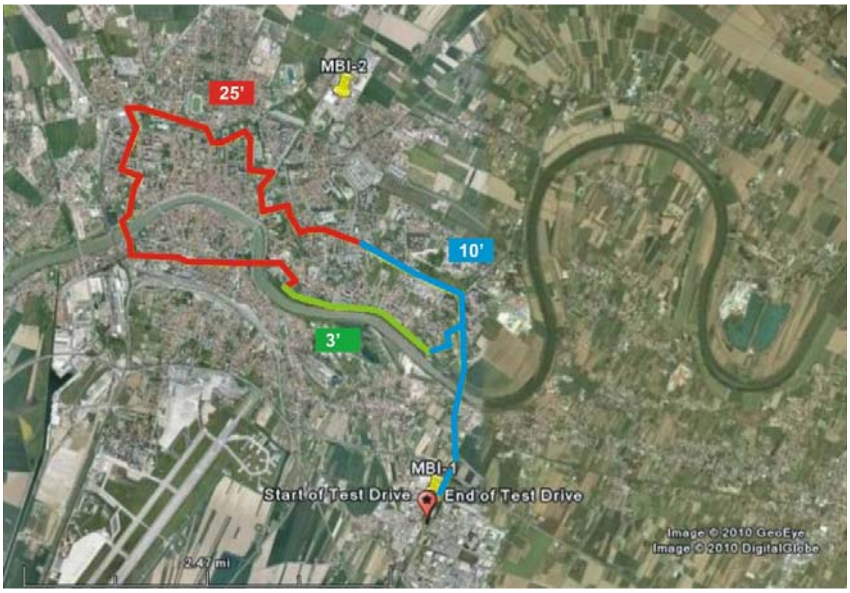
These previous clauses provided an extensive analysis of the hybrid SH-A-SFN architecture. The following conclusions can be derived:

- The hybrid channel architecture gives an evident gain of quality of services, as it increases in most of the cases the ESR5 fulfilment ratio.
- In the northern direction the differential delay is below 100 % of the GI even at long distance from the repeater. In this case even for GI = 1/8, and there is no risk of terrestrial interference to the satellite component, as the terrestrial signal becomes very weak at the edges.
- On the southern direction (in sight of the satellite), the hybrid combining gain is noticeable, even at more that 100 % of the GI. When the delay is more than 150 % of the GI, there can be some interference at the far edges, but they can be mitigated by using a large GI or by careful network planning, focusing power towards the northern part corresponding to sub urban areas.

7.4.4.5 Field trial results

7.4.4.5.1 Results from Field Trials - Italy, January 2010

Within the J-ORTIGIA Project [i.35], field trials have been carried out in the nearby of Pisa (Italy) during January 2010. In the field test area, a complementary ground component (CGC) with two terrestrial repeaters has been deployed. The position of the two CGC repeaters (named MBI-1 and MBI-2) with respect to the test area is indicated in figure 7.49. The two repeaters, both located at a rough height of 15 m, with the aim to cover the city centre of Pisa, equipped with a 50 W amplifier over a unique sector, providing an EIRP of ~47dBm+18 dB (antenna gain) - 3 dB (losses) = 62 dBm. Both the repeaters were directly pointing the center of Pisa. The CGC intends to complement the satellite coverage provided by the W2A satellite.



NOTE: The route used for the measurements is tracked (red-urban, blue-suburban, green-tree shadowing, below). Pisa, Italy. These pictures have been obtained via GoogleMaps (maps.google.com).

Figure 7.49: Position of the CGC repeaters over the test area (above)

The route used for the tests is depicted in figure 7.49. The route was selected to assess the performance in urban and sub-urban scenarios. Most of the route is located in the city of Pisa where urban environment is the predominant one. The first and the last part of the route are characterized by an open sub-urban environment over a fast route on the north-south axis.

Two SH-B configurations have been tested on the field:

- A class-1 configuration, with satellite component configured with modulation QPSK and coding rate 1/2, while the CGC part is configured with 16QAM and code rate 1/3. The scheme is complemented by an MPE-IFEC with B=17 and S=9, with nominal coding rate 2/3. The MPE-IFEC duration is 26 seconds ((B+S)*T_0, with B=17, S=9 and T_0=1 s (burst periodicity)). The actual measured code rate for MPE-IFEC is actually lower, due to padding in the ADT tables, and it is close to 1/2. Hence, the overall efficiency of the class-1 configuration for the TDM part is approximately 2*(1/2)*(1/2)=0,5 b/s/Hz.

- A class-2 configuration, with an interleaver of 5,4 s. The satellite component is protected by a rate 1/3 code, with 8-PSK modulation, while the CGC is still based on a 16QAM signalling and on coding rate 1/3. Hence, the overall efficiency of the class-2 configuration for the TDM part is approximately 3*(1/3)=1 b/s/Hz.

For the tests, two receivers provided by two distinct manufacturers have been simultaneously used. The measurement setup has been organized in a way that the two receivers had access to the same antenna signal. The receivers were operated with a G/T of -21,5 ÷ -22 dB / ºK. The LOS C/N measured on the satellite component is around 9 dB.

Tests have been carried out with 3 different network topologies: satellite-only (with CGC switched on) trials, terrestrial-only trials, and code-combining trials. A summary of test results in terms of service availability (ESR5(20) criterion) is provided in table 7.16. The class-1 configuration is clearly advantaged by its lower spectral efficiency with respect to the class-2 configuration. However, in both cases, the code combining feature has ensured a seamless hand over between satellite and terrestrial signals that is visible in the good service availability percentages achieved in hybrid reception. For the class-1 configuration, a service availability close to 95 % is achieved in hybrid reception, while a terrestrial-only reception would provide roughly 73 %. Similarly, for the class-2 configuration, a service availability of 90 % can be achieved thanks to code combining, while in the terrestrial-only case it would not exceed 70 %. In table 7.16, the performance of the two receivers in different sessions is depicted as well, confirming the stability of the obtained results.

Table 7.16: Summary of the performance achieved during the field trials Pisa, Italy, January 2010

Average values	**Class-1 configuration service availability (ESR5(20) criterion)**		**Class-2 configuration service availability (ESR5(20) criterion)**	
Spectral Efficiency	~0,5 b/s/Hz		~1 b/s/Hz	
Terrestrial-only	71 %		66,5 %	
Satellite-only, CGC on	52 %		40 %	
Code combining	94 %		90 %	
Complete results	RX 1	RX 2	RX 1	RX 2
Terrestrial-only	72 %	70 %	68 %	65 %
Satellite-only, CGC on	55 %/48 %	54 %/52 %	39 %	40 %
Code combining	94 %/95 %/93 %	95 %	89 %/91 %/90 %	90 %

More insights on the performance of the system can be obtained by evaluating the ESR5(20) criterion over location. The results for the class-1 case are provided in figure 7.50, while those for the class-2 configuration are presented in figure 7.51.

Figure 7.50: Service availability vs. location, class 1 configuration Terrestrial only (left), Satellite only (mid), code combining (right) Red spots identify areas in which the ESR5(20) criterion is not fulfilled Receiver 1 (top) and 2 (bottom)

In both cases, the satellite component seems to suffer mostly in the parts of the route which are developed along the East-West axis. Interestingly, in areas where neither the satellite component nor the CGC alone can provide coverage, their combination is successful. This is especially evident in figure 7.51, on the western and in the northern parts of the route.

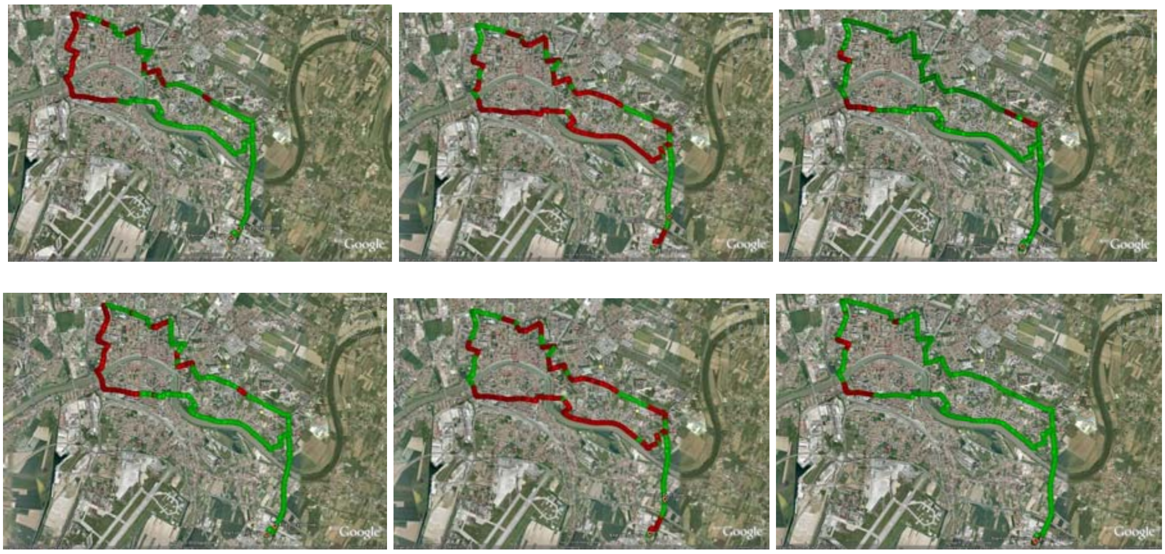


Figure 7.51: Service availability vs. location, class 2 configuration Terrestrial only (left), Satellite only (mid), code combining (right) Red spots identify areas in which the ESR5(20) criterion is not fulfilled Receiver 1 (top) and 2 (bottom)

Conclusions

These trials confirm the advantage brought by the hybrid coverage of DVB-SH systems. Indeed, satellite and terrestrial coverage not only prove to be quite complementary, but moreover the hybrid code combining mechanism allows receiving the signal in areas where neither the satellite nor the terrestrial signal was correctly received.

7.5 Synchronization

This clause explains how synchronization is achieved in a DVB-SH network. Distinction is made on the different hybrid radio frequency configurations (SFN and non SFN).

The synchronization aspects which are not interleaver-related are also given in this clause for all types of DVB-SH networks. The following items are addressed:

- distribution network delays and the use of the SHIP packet for synchronization;
- compensation of the satellite round-trip-delay using the SHIP mechanisms;
- SFN synchronization for OFDM transmitters;
- interleaver delay alignment for SH-A-MFN and SH-B networks.

7.5.0 Introduction to time alignment of satellite / terrestrial signals under consideration of SHIP signalling

This clause is related to the alignment of the satellite and terrestrial signals on air under consideration of the interleaver transit delays and the signalling via SHIP. For an SH-B network it is the target to ensure the time requirements given in clause 7.5.2 allowing for code combining in the receiver.

As an OFDM transmitter in an SH-A-SFN network from the beginning does not know about the network condition the first processing steps are also applicable it.

Whether time alignment under consideration of the interleaver delay is applicable to an SH-A-MFN network is not answered in the present document, as no guidance on code-combining for SH-A-MFN is given in [1]. In particular the different SH framing imposes special treatment. Thus the present document concentrates on SH-A-SFN and SH-B networks only.

This clause explains how the alignment of the TDM and OFDM SH frames can be performed at the transmission side using the SHIP signalling, i.e. in an automatic way without requiring any manual configuration of the transmitters.

Both TDM and OFDM transmitters have to:

- Check if the tdm_function() is present in the SHIP packet [1].
- If tdm_function() is present: Calculate the difference of latency (in time) between both time interleavers.
- If tdm_function() is present: Adjust the On-Air start of SH frame. This adjustment consists in the compensation of the difference of latency (in time) between the two time interleavers on the path with the smaller latency.
- Transmit the signal on-air under further consideration of the *synchronization_time_stamp* in the SHIP.

The complete procedure starting from parsing the SHIP packet is shown in figure 7.52.

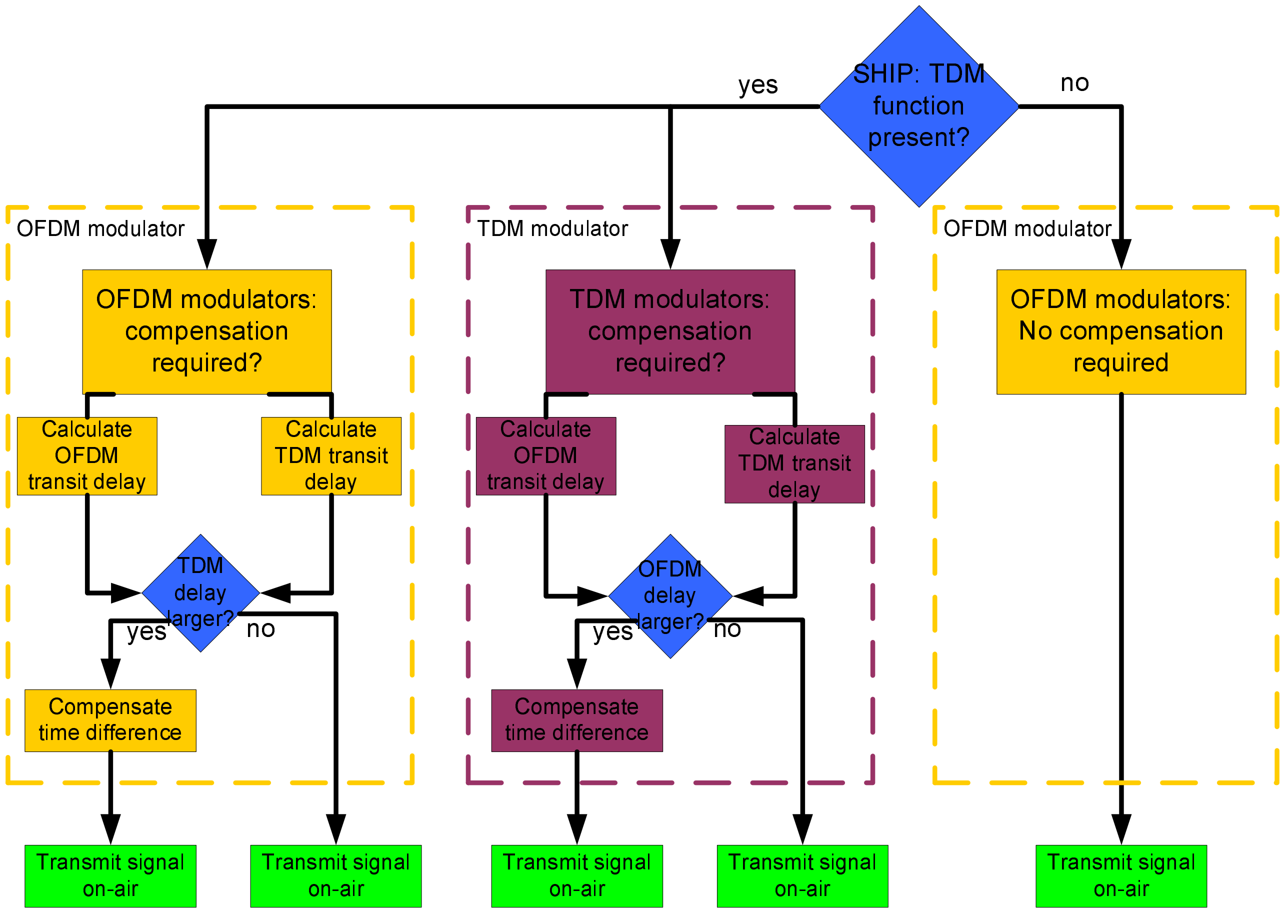


Figure 7.52: Rule to compensate the time interleaver latency based on the SHIP

Interleaver delay compensation is only relevant for transmitters in an SH-B network. In case no TDM component is available in the network then the OFDM transmitters do not need to compensate any interleaver delay, this is covered in clause 7.5.1.

In an SH-B network, if the transmitter has the smallest transmit delay, it needs to update the transit delay with the largest value so that on-air start of SH-frame is common to all transmitters and given by the following formula:

- $on\_air\_SH\_frame\_start_{comp} = on\_air\_SH\_frame\_start + largest\_interleaver\_transit\_delay$;

- where largest_transit_delay = max (OFDM_interleaver_transit_delay; TDM_interleaver_transit_delay); and
- on_air_SH_frame_start is defined in clause 7.5.1 for the OFDM transmitters and in clause 7.5.2 for the TDM transmitters.

where OFDM_interleaver_transit_delay and TDM_interleaver_transit_delay have to be calculated according to (1) and (2) in clause 7.2.3.3.4.2.

In practice, this implies a form of *time-compensation* on the transmitter having the smallest transit time. However the time-compensation technique is only one possible implementation and cannot be mandated. The standard specifies only a time requirement, which is compatible with different implementations, at MPEG2, interleaver, time, etc., and is easily understood by existing terrestrial modulators that are used to such time synchronization in traditional SFN networks. In clause 7.5.2 more details are provided.

In this clause the description is only dealing with the regular latency DVB-SH content. The specialties related to the low-latency extension are handled in Annex D.

7.5.1 Transmitter configuration in satellite-terrestrial SFN

7.5.1.1 Parameter selection

In the case of satellite-terrestrial SFN operation (SH-A and SFN), full synchronization between the satellite component and the terrestrial network is necessary. The terrestrial network includes all terrestrial repeaters belonging to one SFN cell. Throughout the whole DVB-SH network, different non-overlapping SFN cells may exist which have to be locally synchronized to the satellite but not between themselves.

Fine synchronization of terrestrial repeaters can be achieved, even when inclined GEO satellites are used as space component. This is achieved via:

- coarse time and frequency synchronization performed on the satellite signal inside the hub; with this coarse synchronization, the satellite signal is perfectly synchronized with the hub location;
- fine time and frequency synchronization performed at the terrestrial repeaters to compensate for variations within the satellite coverage, so between hub and SFN terrestrial re-transmitters positions. This synchronization can be performed on a per-repeater granularity.

With these two steps of synchronizations, it is regarded feasible that the satellite signal can be synchronized with a remaining time error in the range between -5 % and +5 % of the guard interval length at each repeater cell centre. The difference of propagation time between the centre and the border of the repeater cell depends mainly on the configuration of the terrestrial transmitters, but also on the size of the terrestrial cell.

Frequency synchronization inside the network shall guarantee for a small degradation with respect to the Doppler influence. For this reason the frequency accuracy shall be within ±5 % of the OFDM subcarrier spacing.

The information given below shows the order of magnitude for worst-case conditions (given a certain cell size). The principle behind the calculation of the delay called δd is given in figure 7.53. The following formula can be applied:

$$\delta d = \text{distance(satellite, repeater)} + \text{radius(repeater)} - \text{distance(satellite, receiver)}.$$

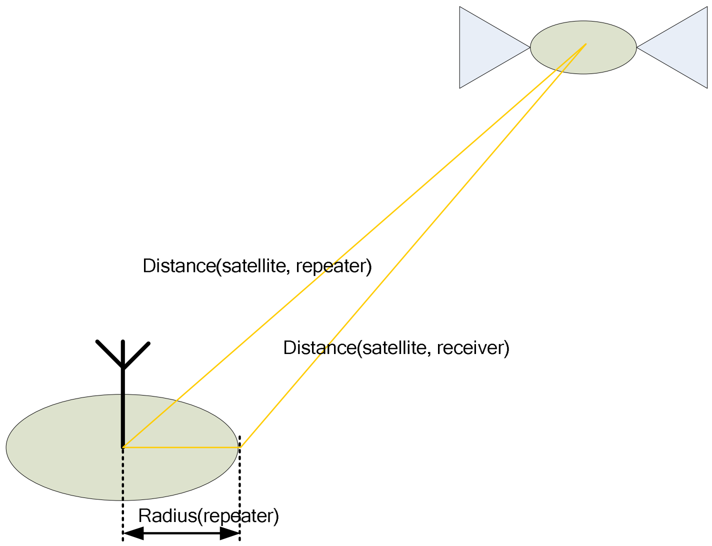


Figure 7.53: Delay calculation principle for SFN operation with 1 repeater

The distance is a function of longitudinal difference between repeater centre and satellite, latitude of the repeater centre and radius of the repeater. A parametric analysis shows the impact of the longitudinal difference (see figure 7.54).

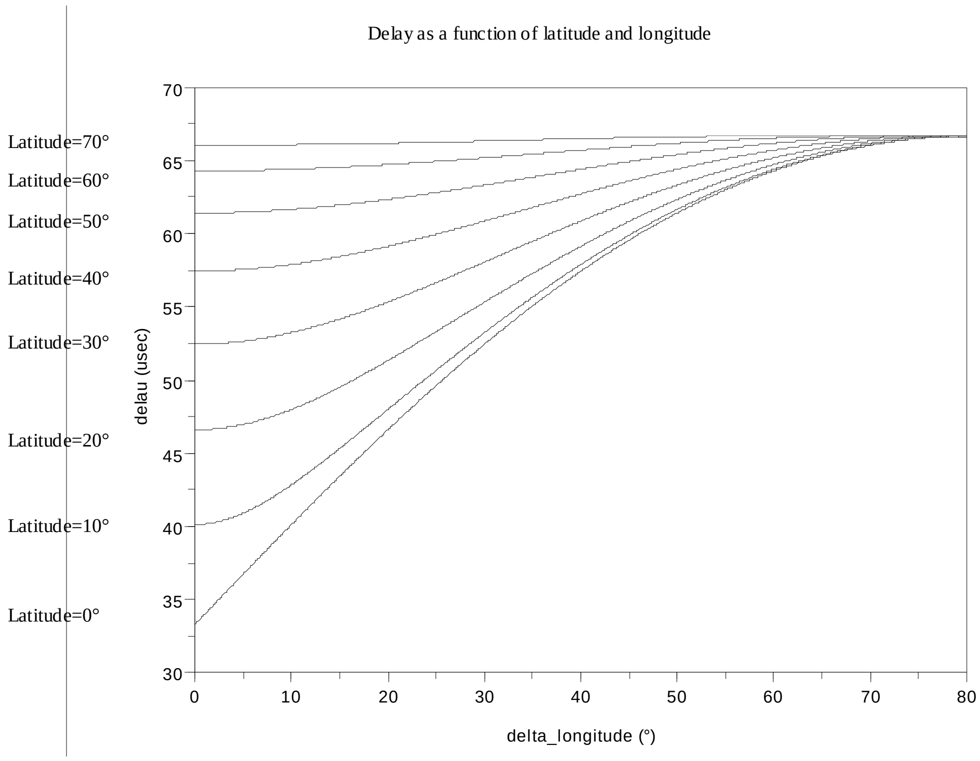


Figure 7.54: Delay as a function of latitude and longitude difference between satellite and terrestrial repeater cell, for a cell radius of 10 km

Taking the asymptotic values given with a delta_longitude at around 70°, the influence of the repeater radius can be displayed.

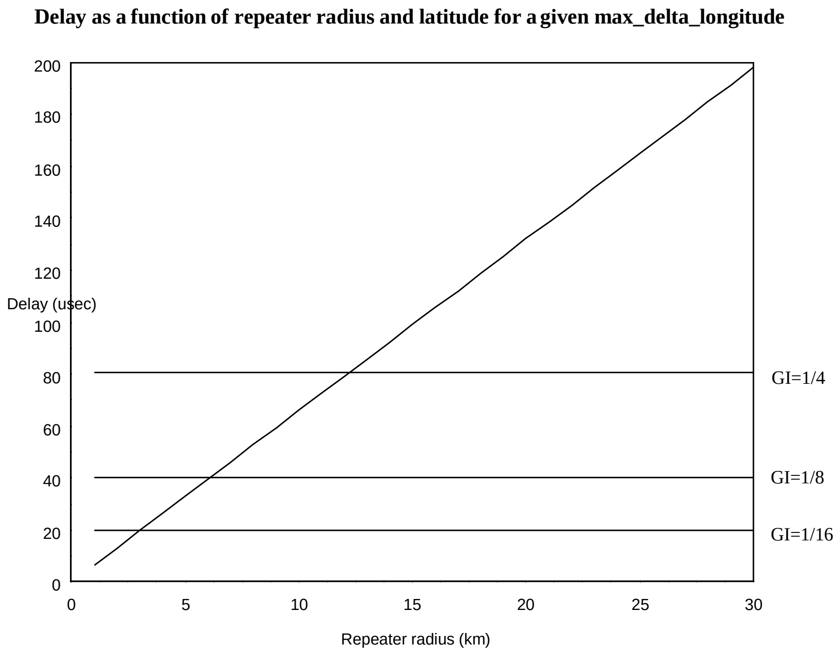


Figure 7.55: Delay as a function of the repeater radius for 70° delta_longitude

It must be recalled that these are minimum distances in the worst direction. In the opposite direction, the distance would be larger so that the "iso-δ zone" will not be a circle. The way this delay must be actually taken into account depends on network planning options:

- guard interval margins: reduction of the GI actual time due to lower constraints on repeaters synchronization; a typical value is 10 % requiring terrestrial transmitter precision of ±5 % of the guard interval value (e.g. for 2k mode 5 MHz, a precision of 1,12 µs is required for the shortest guard interval 1/32) leading to a decrease of 10 % of the GI actual value. Room for delay spread on the terrestrial transmission component must also be considered: this value is function of the propagation environment;
- propagation losses: the effect of the delay reduces when the relative power of the components increases; so, propagations losses have to be considered.

Practical values for OFDM 2k mode, 5 MHz bandwidth are given in table 7.17.

Table 7.17: Max cell radius to ensure SFN between the satellite and terrestrial network at edge of one repeater

Max Cell radius	Max delay in µs	GI = 1/4	GI = 1/8	GI = 1/16	GI = 1/32
12 km	79,8	80,64	40,32	20,16	10,08
6 km	39,9	80,64	40,32	20,16	10,08
3 km	19,65	80,64	40,32	20,16	10,08
1 km	6,55	80,64	40,32	20,16	10,08

For networks having several repeaters, the same principles apply, with additional contributions from the other receivers. See figure 7.56 for details.

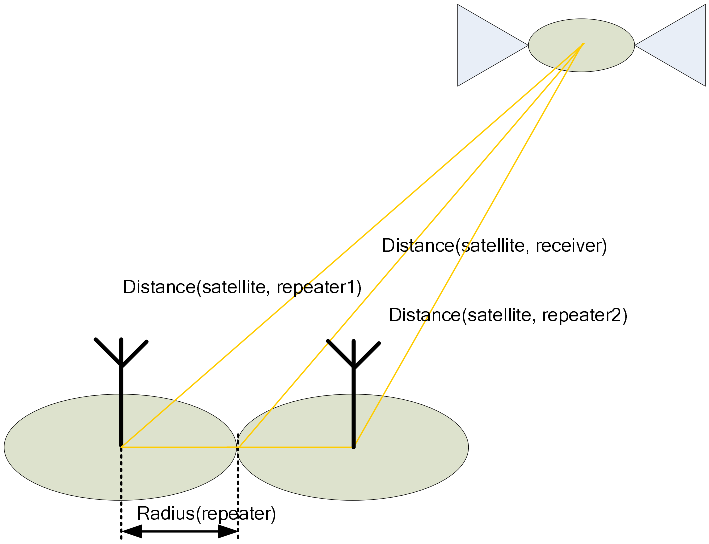


Figure 7.56: Delay calculation principle for SFN operation with 1 repeater

For the worst case assumption, the difference in distance is now:

$$\delta d = 2 * \text{radius(repeater)}$$

Provided the repeaters are synchronized with the satellite, they can be placed with distance equal to twice the maximum cell radius. Distances between repeaters could of course be made greater but then their reception would not be continuous, leading to a "spotted" map.

Depending on the type of network, the typical distances between repeaters would be in 2k FFT mode, 5 MHz:

- for 3G-based networks: 2 km allowing guard intervals of 1/32 or higher;
- for broadcast-centric networks: 12 km allowing guard intervals of 1/8 or higher.

7.5.1.2 Recommendation on network equipment

7.5.1.2.1 Principles of SFN architecture

In order to ensure exact synchronization between all transmitters, the transmitters are time synchronized by SHIP, providing functionality similar to MIP.

- In traditional DVB-H deployment, the MIP is inserted by dedicated equipment called MIP inserter located downstream of the IP encapsulator.
- In the context of DVB-SH, the SHIP insertion is highly recommended to be placed inside the IP encapsulator as the SHIP is also in charge of conveying SH services signalling that may be related to the DVB-H layers.

The SHIP enables SFN synchronization in a similar way as the MIP for DVB-T and most of the recommendations found in [DVB-T-IG] also apply as presented in figure 7.57. The SHIP includes two key parameters to enable synchronization:

synchronization_time_stamp: The *synchronization_time_stamp* of SHIP contains the time difference, expressed as a number of 100 ns steps, between the latest pulse of the "one pulse per second" reference (derived from GPS) that precedes the start of the SH frame M+1 and the actual start (i.e. beginning of first bit of first packet) of this SH frame M+1 at the output of the SHIP inserter. The granularity is 100 ns, the maximum value is 0x98 967F, equivalent to 1 s.

maximum_delay: The *maximum_delay* contains the time difference between the time of emission of the start of SH frame M+1 of the DVB SH signal from the transmitting antenna and the start of SH frame M+1 at the SFN adapter, as expressed by the value of its *synchronization_time_stamp* in the SHIP. The value of *maximum_delay* must be larger than the sum of the longest delay in the primary distribution network and the delays in modulators, power transmitters and antenna feeds. The granularity is 100 ns, the maximum value is 0x98 967F, equivalent to 1 s.

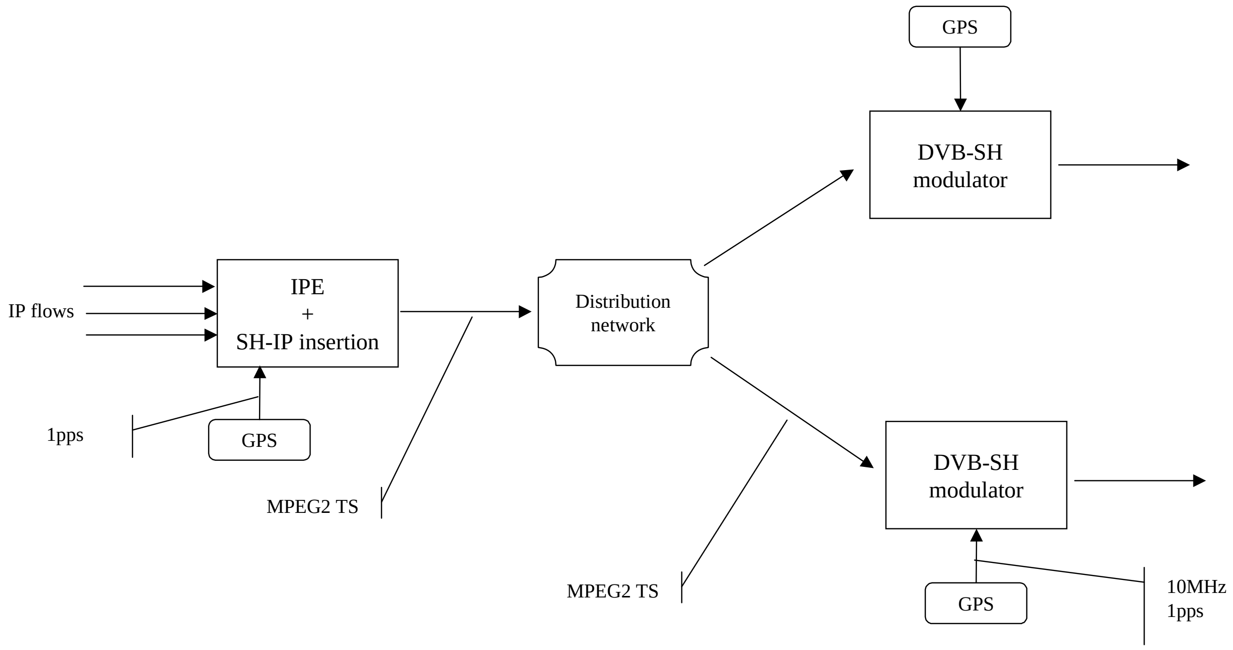


Figure 7.57: Principle of SFN synchronization in a DVB-SH network

There exist particularities that are explained below.

7.5.1.2.2 DVB-SH particularities

Repetition period: the first particularity concerns the signalling period of the SHIP. There is no more any mega-frame since this has been replaced by the SH-frame. SH-frame has a repetition interval that does not depend on the mode (1k, 2k, 4k or 8k) but only on the selection of guard interval and modulation.

This repetition interval is always smaller than 1 s. In general, the repetition frequency of the SH frame (and the SHIP) is larger than for the MIP.

As more than 1 SH frame per OFDM super-frame may occur (QPSK 8k, 16QAM 4k and 8k), it is recommended to check the frame-number (bit s23 and s24 of SHIP) to detect the first SH frame (number 0) of super frame, by the way, all exciters will transmit the same super-frame at the same time.

Interleaver control: the second particularity is the impact of the physical layer time interleaver. It spreads the data in time, with the full spread typically larger than the GPS 1pps impulse. This "latency" must not be confused with the delays introduced by the content delivery and the transmitter play-out, which are covered by the mechanisms above.

The following rules apply for SFN operation:

- modulators synchronize to the SFN network by evaluating the *synchronization_time_stamp* + *max_delay* and the GPS 1 pps and 10 MHz signalling, identical to DVB-H; this will take into account all transit delays brought by the digital and analogue processes except the interleaver transit delay;

- compensation is made for the interleaver latency. This is done on the transmitter side taking care of the start of an SH-frame on the OFDM signal according to clause 7.2.4.1. No compensation is made on the receiver side, as the structure of the time interleaver is inherently self-synchronizing. The shortest tap L[0] (as seen by the receiver) of the time interleaver is (by nature) always set to 0, forcing the very first OFDM symbol of an SH frame to carry already some useful information;

- "The instant for emitting the associated following SH-frame" shall be understood as the instant of the first sample of the guard interval of the OFDM symbol that carries the first IU of the SH-frame, the first IU being considered after the rate adaptation and the padding, but before the convolutional interleaver according to [1], figure 5.11.

  This first OFDM symbol is built when this same first IU of the first CW of the first SH-frame exits from the interleaver after the convolutional interleaver, but before the mapping on CU and SH framing according to [1], figure 5.11.

  This definition is directly related to the definition of the "start of the SH-frame" as given in clause 7.2.4.1.

  It is up to each modulator to take into account the transit delays brought by the digital and analogue processes, including processing time but not the delays for the time interleaver.

  The time chronogram is the following:

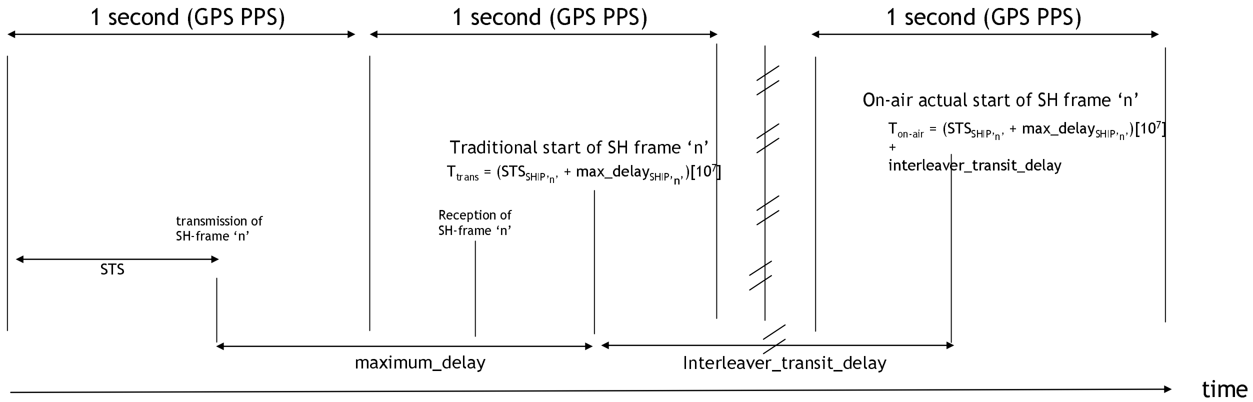


Figure 7.58: On-air start of SH-frame in OFDM SFN case

Where the interleaver_transit_delay has to be calculated according to (1) in clause 7.2.3.3.4.2.

The on-air start of SH-frame is therefore equal to the time-stamp signalled by SHIP STS within current second signalled by GPS PPS plus the interleaver_transit_delay given by above formula. This time is assumed to be computed with sufficient precision to ensure SFN. The SFN behaviour is guaranteed because all transmitters will start transmitting the first bit of the SH frame at the same time.

$$\mathrm{T\_trans} = \left(\mathrm{STS}_{\mathrm{SHIP}_n} + \mathrm{max\_delay}_{\mathrm{SHIP}_n}\right)\left[10^7\right] + \mathrm{interleaver\_transit\_delay}\,.$$

7.5.1.2.3 Possible implementations

- To sign the emission date of the following SH-frame M + 1, the SHIP inserter will copy into the 3-byte valid *synchronization_time_stamp* field of the SHIP packet contained in the SH-frame of index M the value that will be reached by its own counter when it will emit the first bit of the first packet of the SH-frame M + 1.
- An offset value, entered by the human operator of the distribution network, is then inserted by the SHIP inserter in the 3-byte field *maximum_delay*. This "offset" field must be greater than the maximum delay spread introduced by the network and is expressed as a certain number of 100 ns long (1/10 MHz) periods.
- Due to its a priori knowledge of the PID of the SHIP, each channel modulator will extract the SHIP from the SH-frame M of the incoming MPEG-TS and add both time field *synchronization_time_stamp* and *maximum_delay*. Then it waits for its local counter to reach the resulting value before inserting the associated following SH-frame M + 1 into the time interleaver.

7.5.2 Transmitter configuration in non-SFN hybrid networks

7.5.2.1 Overview

For configurations without SFN between satellite and terrestrial component (e.g. SH-A non-SFN or SH-B), the receiver has the possibility to combine the signals from different reception paths. Although not on-frequency, this can be considered as a kind of SFN, as the "same" content is transmitted.

The same rules for the use of the (terrestrial) SFN configuration apply as in SFN operation; however the satellite and terrestrial ground component have to meet less stringent timing and frequency accuracy requirements.

To allow combining between satellite and terrestrial signals, the alignment between the two components has to be ensured on the base of an SH frame. Therefore, the following requirements apply:

- at the terrestrial transmitter location, the time difference between the satellite and terrestrial component must be in the range between -5 ms and +5 ms (see note);
- no specific requirements are made for the frequency synchronization between the two components;
- the latencies of the different interleavers on satellite and terrestrial transmission have to be compensated to allow for proper alignment with minimum buffer requirements in the receiver.

NOTE: The ±5 ms requirement is below ±5 % of an SH frame duration for most of the possible DVB-SH waveform configurations, therefore the ±5 % have been discarded.

7.5.2.2 DVB-SH particularities

The requirements to compensate the latencies of the different interleavers on satellite and terrestrial transmission is a function of the time interleaver, the topic was introduced in clause 7.5.0.

Like the SH-A-SFN case it is important to clearly define the on-air timing for the satellite and terrestrial components (OFDM and TDM or OFDM):

- For OFDM the instant for "emitting the associated following SH-frame" shall be understood as the instant of the first sample of the guard interval of the OFDM symbol that carries the first IU of the SH-frame before the interleaver.

  This definition is directly related to the definition of the "start of the SH-frame" as given in clause 7.2.4.1.

  It is up to each modulator to take into account the transit delays brought by the digital and analogue processes, including processing time but not the delays for the time interleaver (same definition as in SH-A-SFN).

- For TDM the instant for "emitting the associated following SH-frame" shall be understood as the instant of the first sample of the PL slot that carries the SF right in front the first data IU of the SH-frame, the first data IU being considered after the rate adaptation, but before the convolutional interleaver according to [1], figure 5.12.

  This first PL slot is built with the SF when this same first data IU of the first CW of the first SH-frame exits from the interleaver after the convolutional interleaver, but before the mapping on CU and SH framing according to [1], figure 5.12.

  This definition is directly related to the definition of the "start of the SH-frame" as given in clause 7.2.4.2.

  It is up to each modulator to take into account the transit delays brought by the digital and analogue processes, including processing time but not the delays for the time interleaver.

  For TDM also the duration of the SF has to be considered as the SH-frame start is corresponding to the SF and not to a data IU. Additionally the pilot field in front of the according PL slot has to be considered.

Even though the formulas for interleaver delay compensation have an accuracy of 100 ns it has to be kept in mind that clause 7.5.2.1 imposes a less stringent requirement for OFDM/OFDM in SH-A-MFN or OFDM/TDM alignment in SH-B, thus the accuracy of the calculation can be reduced.

The duration of the TDM SF can be calculated according to the following formula:

$$\text{TDM\_SF\_duration} = \frac{6\,528}{(\text{TDM\_Modulation} * \text{TDM\_SymbolRate})}.$$

Accordingly the duration of the pilot field in front of a PL slot is:

$$\text{TDM\_pilot\_duration} = \frac{80}{\text{TDM\_SymbolRate}}.$$

7.5.2.3 Possible Implementations of Interleaver Delay Compensation

There are multiple solutions to achieve interleaver delay compensation inside the transmitter, which one is used is left open to implementers.

NOTE: It is important to keep in mind that even the descriptions below are given for the case where the satellite interleaver transit delay exceeds the one for the terrestrial path this cannot be assumed to be always true. The same rule applies to the satellite modulator if the terrestrial interleaver transit delay is the longer one.

At least the following implementations are possible and listed below for information.

7.5.2.3.1 Solution 1

The solution consists in using the existing capability of terrestrial modulators to synchronize at 100 ns in order to comply with any time reference objective. The principle is then to know which time value is sought at the output of the modulator. This solution makes usage of a large physical buffer after the interleaver output. The estimation of the buffering time is derived from difference in time between satellite and terrestrial interleaver transit times.

It can be seen that the setting of the modulator will resort to adding a delay of:

$$\text{largest\_interleaver\_transit\_delay} - \text{terr\_interleaver\_transit\_delay}.$$

But how this delay is fixed is completely left for implementation to the transmitter manufacturer.

This solution is depicted below in figure 7.59.

The same solution (by exchanging satellite with terrestrial modulators) applies to the satellite modulator if the terrestrial interleaver transit delay is the longer one.

7.5.2.3.2 Solution 2

This solution is similar to the one before, but puts the additional delay into the interleaver memory. Thus, this solution corresponds to the conceptual view as given in figure 7.21.

Regarding the depiction in figure 7.59 the only difference of this solution is to replace the sum of the two arrows 'terr_interleaver_transit_delay' and 'Additional buffer after interleaver' simply be the new 'terr_interleaver_transit_delay', which in fact then is the same length.

The same solution (with exchanging satellite by terrestrial modulators) applies to the satellite modulator if the terrestrial interleaver transit delay is the longer one.

7.5.2.3.3 Solution 3

The solution consists in using the existing capability of terrestrial modulators to synchronize with the GPS PPS. The principle is that if we need to synchronize an additional delay of x seconds (x>1), we can simply delay at MPEG2 level the required number of seconds and use an (internal) updated value of the SHIP time stamp and max_delay. The main advantage is that the buffer is now mainly at MPEG2 level instead of physical (coded) level, inducing large gains of memory. The figure below tries to capture the principle.

Followed steps by MPEG2 pre-processor:

1) After reception of the SH frame 'n', store the MPEG2 stream for a duration equal to:

sat_interleaver_transit_delay - terr_interleaver_transit_delay+maximum_delay-MPEG2_processing_max_delay;

- where MPEG2_processing_max_delay is the maximum delay to process the MPEG2 SH frame inside the transmitter.

2) Computes an updated STS' and max_delay' as follows:

$$\text{max\_delay}' = \text{MPEG2\_processing\_max\_delay}$$

$$\text{STS}' = \begin{pmatrix} (\text{STS}_{\text{SHIPn}} + \text{max\_delay}_{\text{SHIPn}}) \\ + \text{sat\_interleaver\_transit\_delay} - \text{terr\_interleaver\_transit\_delay} \\ - \text{MPEG2\_processing\_max\_delay} \end{pmatrix} [10^7]$$

- This {STS'; max_delay'} will be used in replacement of existing {STS; max_delay}, but the STS and max_delay value will not be modified within the SHIP to keep code combining possible.

3) Then the MPEG2 pre-processor injects the stream at maximum_delay' before expiration of STS'.

This solution is depicted below in figure 7.60.

The same solution (by exchanging satellite with terrestrial modulators) applies to the satellite modulator if the terrestrial interleaver transit delay is the longer one.

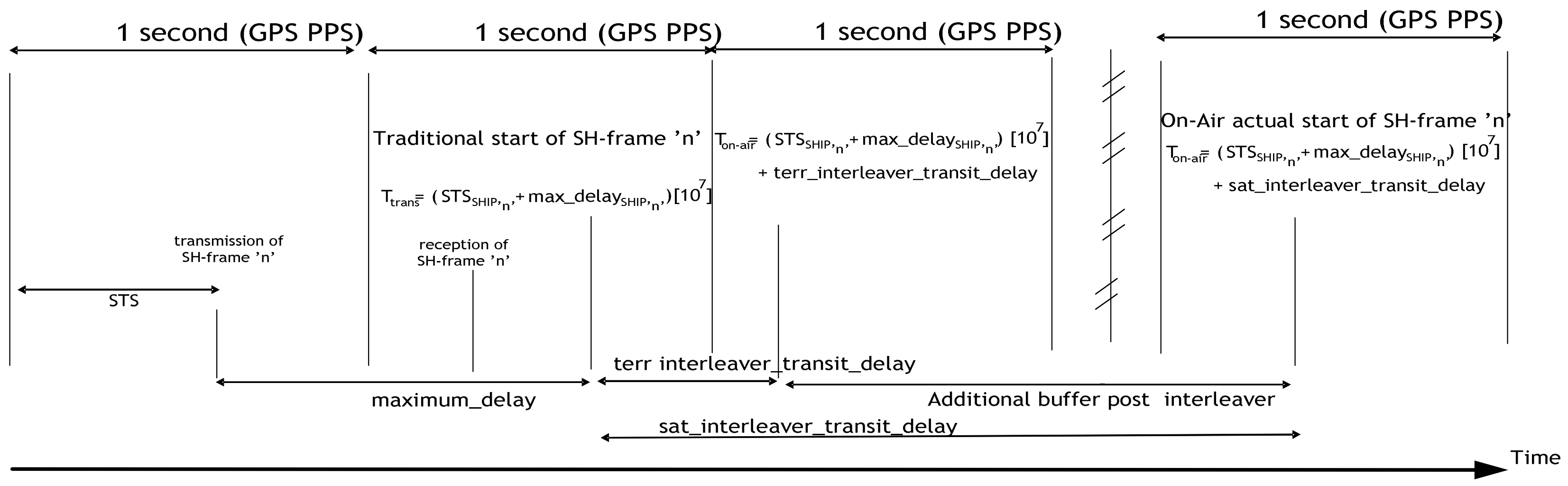


Figure 7.59: Description of Solution 1 (e.g. satellite path with longer interleaver transit delay)

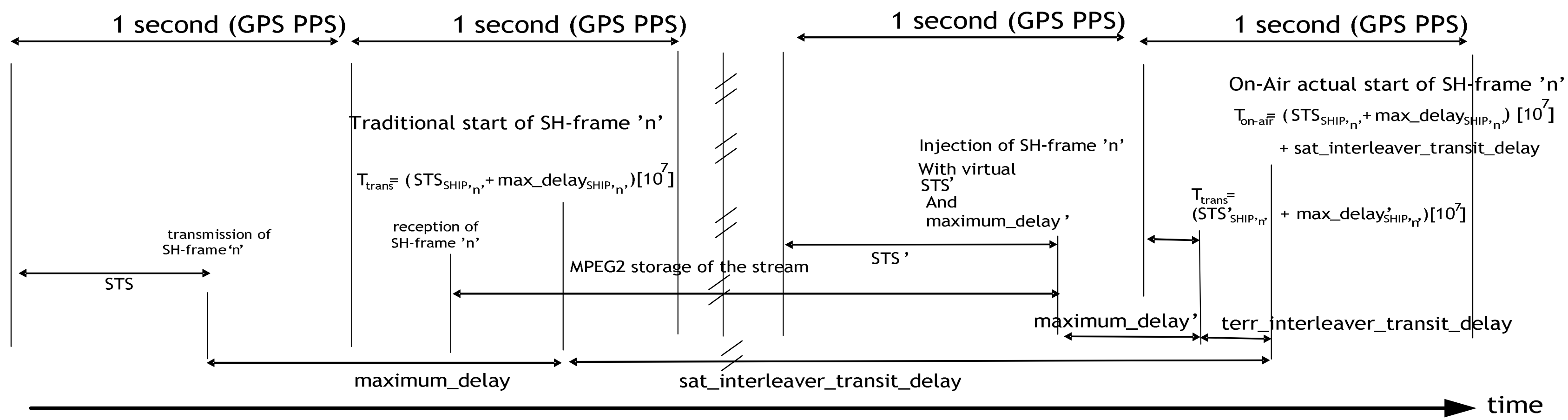


Figure 7.60: Description of Solution 3 (e.g. satellite path with longer interleaver transit delay)

7.5.3 Receiver synchronization and re-synchronization

This clause is for information.

To successfully decode one FEC packet, various modules in the receiver have to be synchronized to the block transmission structure of the turbo code. This clause tries to summarize the different steps for a successful decoding and presents different possible strategies.

The use of time slicing introduces some constraints on the synchronization, as receivers will have to either completely re-synchronize at the beginning of each burst, or use prediction from internal oscillators.

This clause describes various strategies to synchronize both the OFDM and the TDM part with the SH frame. The first two methods do not reuse any existing knowledge from previous time-sliced bursts; the third one reuses information derived from previous time-sliced bursts.

7.5.3.1 SH frame synchronization strategy 1 (without a priori information)

If no prediction of the SH frame structure is provided by the receiver, the synchronization to the SH framing has to be recovered at each switch-on.

For OFDM, the TPS bits are used to recover the framing. By reading the TPS of the OFDM super frame, the receiver discovers the following information:

- s_{23} and s_{24}: OFDM frame number;
- s_{35}: OFDM super frame number in SH frame.

However, the bit s_{35} acquisition is only required in modes where the number of super frames per SH frame is larger than 1, i.e. in QPSK 1k, QPSK 2k, and 16QAM 1k modes. For all other modes, the bit s_{35} is not used.

After a maximum of 2 OFDM super frames (for the combinations as mentioned above) and 1 OFDM super frame for all other modes, the receiver is able to determine the SH framing and, thanks to the coding parameters already acquired, derive exact boundaries of the code words.

Actually, the maximum duration can be 2 (respectively 1) OFDM super frame(s) if the receiver uses this strategy.

For TDM, the SOF preamble is used to derive the boundaries of the SH frame. The maximum duration is identical to OFDM as the SH frame duration has been chosen identically for OFDM and TDM.

7.5.3.2 SH frame synchronization strategy 2 (without a priori information, for OFDM only)

The second strategy is only applicable to OFDM and is tied to the absence of TDM. It is based on the alignment of interleaver and capacity units with the OFDM symbols and founds on the pattern of the scattered pilots. Without knowing the SH framing, the de-interleaver can immediately start the de-interleaving process, as soon as the first group of 4 OFDM symbols has been acquired.

This is possible due to the self-synchronizing structure of the convolutional interleaver. Each 48 interleaver units (48*126 bits = 6 048 bits) can be handled as a group and fed into the de-interleaver. For any 4 OFDM symbols, the number of bits to be processed is an integer multiple of 6 048 bits:

- 1k, QPSK, 4 symbols equivalent to 756*4*2 bits = 6 048 bits;
- 8k, 16QAM, 4 symbols equivalent to 6 048*4*4 bits = 96 768 bits = 16*6 048 bits.

After a sufficient de-interleaver duration (to receive enough parity data for decoding), it is possible to perform a limited number of "blind decoding trials".

7.5.3.2.1 Option 1

The number of tests is limited because the code words are aligned with the pattern of scattered pilots and, as a consequence, are aligned with the de-interleaver cycles. In practice, the number of cycles is equal to:

$$\frac{N_{BIL}}{6\,048}$$

(With N_{BIL} being the turbo block size after interleaver and puncturing), and is completely independent of the modulation.

The procedure is then the following:

- wait for 2 code words during $2*\frac{N_{BIL}}{bit\_rate}$;
- test for successful decoding;
- if the test is not successful, shift by one cycle by waiting for 6 048 bits during a duration of $\frac{6\,048}{bit\_rate}$ and retry the decoding after a waiting time of $2*\frac{N_{BIL}}{bit\_rate}$.

The maximum wait duration is $\frac{N_{BIL}}{6\,048}*\left(\frac{2*N_{BIL}+6\,048}{bit\_rate}\right)$, the average is half of this value.

7.5.3.2.2 Option 2

As soon as the receiver has done the scattered pilots acquisition, it is able to number the symbols in a row of 4 and to position the start of a code word with regard to the output of the smallest duration tap L[0]. The maximum number of tests depends on the modulation, FFT size and denominator of the fractional code rate. They are given in table 7.18 for code rate k/n where denominator n value is below or equal to 5.

The duration of the tests is implementation dependent and depends on the turbo-decoding strategy. Typical implementations will require reception of an equivalent of 2 code words for each try, implying reception duration of 2 code words multiplied by the number of tries.

The procedure is then the following:

- wait for 2 code words;
- test for successful decoding;
- if the test is not successful, move to next position without waiting and restart the test.

In case of successful test decoding, the receiver knows the code word boundaries and also the CBCOUNTER_FB which gives the position index of the code word inside the current SH frame. So the receiver knows exactly the SH-framing.

These two strategies (frame duration and decoding tries) can be applied in parallel and the first successful one will be used to minimize the acquisition time. As can be seen in table 7.18, code word acquisition is generally faster than OFDM frame or OFDM super frame acquisition except for the 1k mode.

Table 7.18: Typical synchronization times for strategy 1 and strategy 2

MOD	carriers	1/code_rate	Maximum number_of_tries	Option 1 Nof_tries*(2 CW + 6 048) (ms)	Option 2 Nof_tries * CW (ms)	SH frame acquisition time (ms)
16QAM	8k	2	1	16	3,58	243,712
16QAM	4k	2	1		3,58	121,856
16QAM	2k	2	1		3,58	60,928
16QAM	1k	2	2		7,17	45,696
QPSK	8k	2	1	32	7,17	243,712
QPSK	4k	2	1		7,17	121,856
QPSK	2k	2	2		14,34	91,392
QPSK	1k	2	4		28,67	45,696
16QAM	8k	3	8	35	43,01	243,712
16QAM	4k	3	4		21,50	121,856
16QAM	2k	3	3		16,13	60,928
16QAM	1k	3	3		16,13	45,696
QPSK	8k	3	4	70	43,01	243,712
QPSK	4k	3	3		32,26	121,856
QPSK	2k	3	3		32,26	91,392
QPSK	1k	3	6		64,51	45,696
16QAM	8k	4	1	61	7,17	243,712
16QAM	4k	4	1		7,17	121,856
16QAM	2k	4	2		14,34	60,928
16QAM	1k	4	4		28,67	45,696
QPSK	8k	4	1	122	14,34	243,712
QPSK	4k	4	2		28,67	121,856
QPSK	2k	4	4		57,34	91,392
QPSK	1k	4	8		114,69	45,696
16QAM	8k	5	8	94	71,68	243,712
16QAM	4k	5	7		62,72	121,856
16QAM	2k	5	5		44,80	60,928
16QAM	1k	5	5		44,80	45,696
QPSK	8k	5	7	188	125,44	243,712
QPSK	4k	5	5		89,60	121,856
QPSK	2k	5	5		89,60	91,392
QPSK	1k	5	10		179,20	45,696

7.5.3.3 SH frame synchronization strategy 3 (with prediction)

This strategy is applicable to both OFDM and TDM. The receiver may provide frame prediction according to the following schemes. The prediction times are considered to be feasible with consumer product oscillators, assuming a frequency accuracy of (±) 50 ppm.

For OFDM, the use of 2k mode, 5 MHz bandwidth with QPSK modulation and guard interval 1/8 has been selected for this example:

- prediction of the OFDM symbol number is possible over **one** second within an interval of (±) 12,4 % (50 µs at a total symbol length of 403,2 µs per OFDM symbol);
- prediction of the OFDM regular pilot structure (4 OFDM symbols) is possible over **one** second within an interval of (±) 3,1 % (50 µs at a total length of 4 OFDM symbols of 1 613 µs);
- prediction of the OFDM frame structure (68 OFDM symbols) is possible over **ten** second within an interval of (±) 1,8 % (500 µs at a total length of 68 OFDM symbols of 27,4 ms).

The length of the SH frame in OFDM frames depends on the selection of the modulation and guard interval and the selection of the FFT mode. The following extreme cases are possible:

- "very short SH frame": For 8k FFT and 16QAM modulation, one SH frame maps to 1 OFDM frame;
- "very long SH frame": For 1k FFT and QPSK modulation, one SH frame maps to 16 OFDM frames.

Combined with the highest (9,14 MHz) symbol rate available, this maps to the following prediction accuracies:

- "very short SH frame": Prediction of the SH frame structure is possible over **ten** seconds within an interval of (±) 0,73 % (500 µs at a total length of 68 OFDM symbols of 68,54 ms);
- "very long SH frame": Prediction of the SH frame structure is possible over **ten** seconds within an interval of (±) 0,36 % (500 µs at a total length of 16*68 OFDM symbols of 137,09 ms).

It can be concluded that, with low requirements on the receiver oscillators used, the receiver is capable of predicting the SH frame alignment for both OFDM and TDM over a time span of more than 10 s and even with oscillators worse than those used for the calculation exercise. Using this proposed scheme, re-acquisition time is only dependent on the re-acquisition time of the demodulators, but no longer on the SH framing, allowing a good prediction of code word boundaries.

7.5.4 HP/LP Synchronization in Hierarchical mode

In an SFN with hierarchy, the relative alignment of HP and LP streams must be the same at all modulators, in order to make them emit identical signals. HP and LP streams may originate at different locations, and so it is likely that HP and LP SFN adapters may not emit suitably synchronized streams.

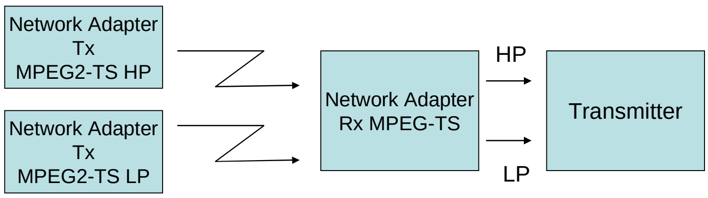


Figure 7.61: Distribution of 2 MPEG-2 transport streams in a hierarchical modulation context

A deterministic method of aligning the two streams is therefore required. As explained in [i.22], it is undesirable to align the SH-frame starts of the HP and LP streams, as this may involve large delays and result in a significant offset to the time_of_emission of one of the streams. The minimum requirement for alignment is that a sync byte of one stream (the slave stream) is coincident with the sync byte at the SH-frame start of the other stream (the master stream). Subsequently, only SH-frame information from the master stream may be used. This will ensure proper operation of various synchronized processes within the modulator. The requirement alignment is illustrated in figure 7.62.

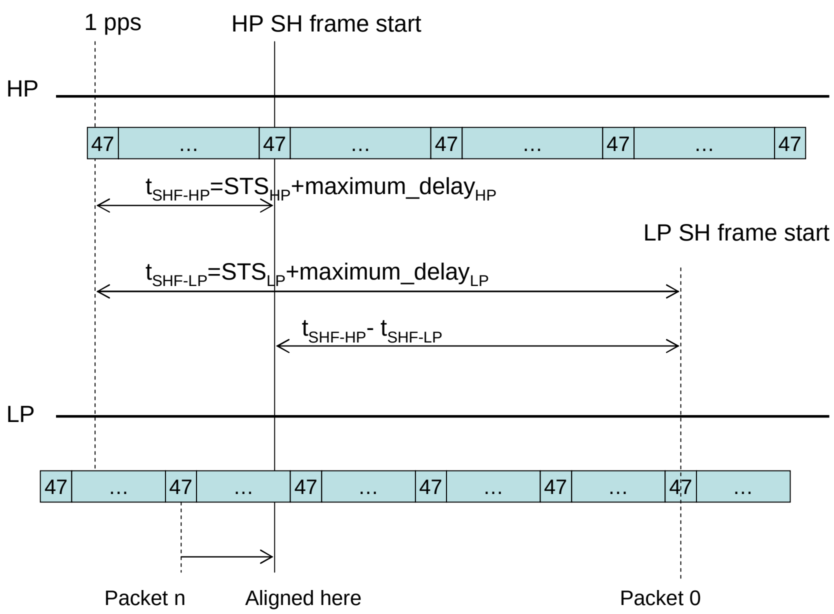


Figure 7.62: Principle of HP and LP synchronization

The HP stream is designated as the master and the LP as the slave. The alignment reference point is the SH-frame start of the HP stream. The LP stream is delayed with respect to the HP, to ensure an MPEG-TS packet start is aligned to the reference point. The amount of this delay is less than one complete MPEG-TS packet as follows: the packet n (counting from the LP mega-frame start) is aligned to the HP mega-frame start, where n is given by:

$$n = [(t_{SHF\text{-}HP} - t_{SHF\text{-}LP}) * P_{LP} / d_{MF}]$$

where:

- P_{LP} is the number of bytes in one LP SH frame and d_{MF} is the duration of one SH frame;
- n must be rounded down to the nearest integer, i.e. 2,7 would become 2; -2,3 would become -3.

This expression caters for the slave LP SH frame start either lagging (n is positive) or leading (n is negative) the master HP SH frame. The time is in 100 ns units.

7.6 System throughput calculations

DVB-SH capacity at MPEG TS interface level may be calculated from the parameters defined in the waveform definition [1]. This is performed in calculating first the number of MPEG TS packets in an SH Frame, and then the duration of the SH frame. The ratio of these two numbers give the bit rate capacity of the system.

Please note that the throughput calculation is independent of the selection of FFT mode in OFDM. To support seamless hand-over between OFDM and TDM, the SH frame length of the TDM part has been aligned to the SH frame length in OFDM. Therefore, the TDM parameters cannot be calculated independently but imply a selection of OFDM parameters which is represented in the following tables.

The Calculation results are for information.

7.6.1 Calculation of the number of MPEG TS packets per SH Frame

Each turbo code word contains exactly 8 MPEG TS Packets.

For OFDM, table 5.11 of the waveform definition [1] gives the number of turbo code words in an SH frame as a function of the FEC code rate.

For TDM, table 5.12 of the waveform definition [1] gives the number of Capacity Units (CU) per SH Frame. A CU is a set of 2 016 bits. From table 5.8 of the waveform definition [1], the number of CU per turbo code word may be calculated for each code rate. The total number of turbo code words in an SH frame is then the integer part of the division of the number of Capacity Units (CU) per SH Frame by the number of CU per turbo code word.

7.6.2 Calculation of the SH Frame duration

The SH frame length is defined as a function of the length of OFDM frames. Therefore, the frame length of both an SH-frame in OFDM and TDM mode are identical. The number of OFDM Frames per SH frame is defined in the table 5.10 of the waveform definition [1]. Each OFDM Frame is composed of 68 OFDM Symbols. The duration of an OFDM symbol is given in tables 5.22, 5.24 and 5.26 of the Waveform Definition document [1].

7.6.3 Typical MPEG-TS bit rates for OFDM

The following tables give the MPEG TS bit rates in Mbps of the OFDM waveform for various DVB-SH configurations.

7.6.3.1 Channel Bandwidth 5 MHz, OFDM

Table 7.19

	GI	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
QPSK	1/4	1,333	1,481	1,679	1,876	2,222	2,666	3,357	4,443
	1/8	1,481	1,646	1,865	2,085	2,468	2,962	3,730	4,937
	1/16	1,568	1,742	1,975	2,207	2,614	3,136	3,950	5,227
	1/32	1,616	1,795	2,035	2,274	2,693	3,231	4,069	5,386
16QAM	1/4	2,666	2,962	3,357	3,752	4,443	5,332	6,714	8,887
	1/8	2,962	3,291	3,730	4,169	4,937	5,924	7,460	9,874
	1/16	3,136	3,485	3,950	4,414	5,227	6,273	7,899	10,455
	1/32	3,231	3,591	4,069	4,548	5,386	6,463	8,139	10,772

7.6.3.2 Channel Bandwidth 1,7 MHz, OFDM

Table 7.20

	GI	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
QPSK	1/4	0,427	0,474	0,537	0,600	0,711	0,853	1,074	1,422
	1/8	0,474	0,527	0,597	0,667	0,790	0,948	1,194	1,580
	1/16	0,502	0,558	0,632	0,706	0,836	1,004	1,264	1,673
	1/32	0,517	0,574	0,651	0,728	0,862	1,034	1,302	1,723
16QAM	1/4	0,853	0,948	1,074	1,201	1,422	1,706	2,149	2,844
	1/8	0,948	1,053	1,194	1,334	1,580	1,896	2,387	3,160
	1/16	1,004	1,115	1,264	1,413	1,673	2,007	2,528	3,346
	1/32	1,034	1,149	1,302	1,455	1,723	2,068	2,604	3,447

7.6.4 Typical MPEG-TS bit rates for TDM

The following tables give the TDM MPEG TS bit rates in Mbps of the TDM waveform for various DVB-SH configurations.

7.6.4.1 Channel Bandwidth 5 MHz, TDM, with 15 % roll-off and QPSK in OFDM

Table 7.21

		TDM	Modulation	QPSK	Roll Off	15 %	OFDM modulation		QPSK
OFDM GI		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		1,530	1,728	1,925	2,222	2,567	3,110	3,900	5,184
1/8		1,536	1,755	1,975	2,249	2,633	3,127	3,950	5,266
1/16		1,510	1,684	1,917	2,207	2,556	3,078	3,891	5,169
1/32		1,556	1,676	1,915	2,214	2,573	3,112	3,890	5,146

Table 7.22

		TDM	Modulation	8PSK	Roll Off	15 %	OFDM modulation		QPSK
OFDM GI		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		2,320	2,567	2,913	3,308	3,900	4,690	5,826	7,800
1/8		2,359	2,633	2,962	3,346	3,950	4,718	5,924	7,899
1/16		2,323	2,614	2,904	3,311	3,891	4,705	5,866	7,841
1/32		2,334	2,573	2,872	3,291	3,890	4,668	5,805	7,779

Table 7.23

		TDM	Modulation	16APSK	Roll Off	15 %	OFDM modulation		QPSK
OFDM GI		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		3,110	3,456	3,900	4,443	5,184	6,221	7,800	10,417
1/8		3,127	3,511	3,950	4,498	5,266	6,308	7,899	10,532
1/16		3,136	3,485	3,891	4,472	5,227	6,273	7,841	10,455
1/32		3,112	3,411	3,890	4,428	5,146	6,224	7,779	10,353

7.6.4.2 Channel Bandwidth 5 MHz, TDM, with 15 % roll-off and 16QAM in OFDM

Table 7.24

		TDM	Modulation	QPSK	Roll Off	15 %	OFDM modulation		16QAM
		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		1,481	1,679	1,876	2,172	2,567	3,061	3,851	5,134
1/8		1,536	1,646	1,865	2,194	2,523	3,072	3,840	5,156
1/16		1,510	1,626	1,859	2,207	2,556	3,020	3,833	5,111
1/32		1,436	1,676	1,915	2,154	2,513	2,992	3,830	5,146

7.6.4.3 Channel Bandwidth 5 MHz, TDM, with 25 % roll-off and QPSK in OFDM

Table 7.25

		TDM	Modulation	QPSK	Roll Off	25 %	OFDM modulation		QPSK
OFDM GI		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		1,432	1,629	1,827	2,074	2,419	2,913	3,653	4,888
1/8		1,426	1,536	1,755	2,030	2,359	2,852	3,566	4,718
1/16		1,452	1,568	1,801	2,033	2,381	2,904	3,601	4,821
1/32		1,436	1,556	1,795	2,035	2,394	2,872	3,591	4,787

7.6.4.4 Channel Bandwidth 1,7 MHz, TDM, with 15 % roll-off and QPSK in OFDM

Table 7.26

		TDM	Modulation	QPSK	Roll Off	15 %	OFDM modulation		QPSK
OFDM GI		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		0,490	0,553	0,616	0,711	0,822	0,995	1,248	1,659
1/8		0,492	0,562	0,632	0,720	0,843	1,001	1,264	1,685
1/16		0,483	0,539	0,613	0,706	0,818	0,985	1,245	1,654
1/32		0,498	0,536	0,613	0,709	0,823	0,996	1,245	1,647

Table 7.27

		TDM	Modulation	8PSK	Roll Off	15 %	OFDM modulation		QPSK
OFDM GI		1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
1/4		0,743	0,822	0,932	1,058	1,248	1,501	1,864	2,496
1/8		0,755	0,843	0,948	1,071	1,264	1,510	1,896	2,528
1/16		0,743	0,836	0,929	1,059	1,245	1,505	1,877	2,509
1/32		0,747	0,823	0,919	1,053	1,245	1,494	1,857	2,489

7.7 Coexistence with other Physical layer elements

This clause is for information.

7.7.1 Introduction

The deployment of DVB-SH CGC raises the problem of coexistence of the DVB-SH system with other systems operating in adjacent S band, like 3G, WiFI, Bluetooth®, etc. Of particular interest is the case of coexistence with 3G networks as they are operating in the 2 110 MHz to 2 170 MHz on the down link and 1 920 MHz to 1 980 MHz on the uplink.

The coexistence problem is in fact a two fold problem:

- The first aspect is the possible impact of co-location, especially in the case of coexistence with 3G or other Mobile systems. Something that we could call short distance "interference" of course this concerns the reciprocal interference scenarios from DVB-SH to 3G and from 3G to DVB-SH.
- The second aspect is the long distance interference that reflects the mutual impact of the two systems (DVB-SH and the other one), mainly at the receiver level. This second aspect is also to be studied in both directions: from DVB-SH to 3G and vice-versa.

Due to the lack of measurements up to now, only the 3G case is covered, and partially.

No measurements have been done in WiFi or Bluetooth® environment. Most of the simulations and measurements were made on the impact of DVB-SH on 3G systems.

The 3G case

Concerning the 3G coexistence four aspects are covered in this clause.

- The first one is specific to some deployment strategies that include co-location of 3G and DVB-SH sites with cable and/or antenna sharing; this can have an impact on the link budget.
- The second one concerns the impact of the DVB-SH signals on the 3G service, based on real field tests.
- The third one is the result of simulations on the impact of a high power CGC on a UMTS network operating in the adjacent band, which is of course the worst case.
- The fourth one is the results of simulations concerning the impact of UMTS signals on DVB-SH signals, especially on satellite signals when there is no CGC.

The two first item results are based mainly on the outcomes of the SFR®/Vodafone™/Alcatel-Lucent™ (with the help of Thales Alenia Space™) experiment in Pau and also on some early Toulouse trials.

7.7.2 Site sharing

The general picture of the site co-sharing is the following.

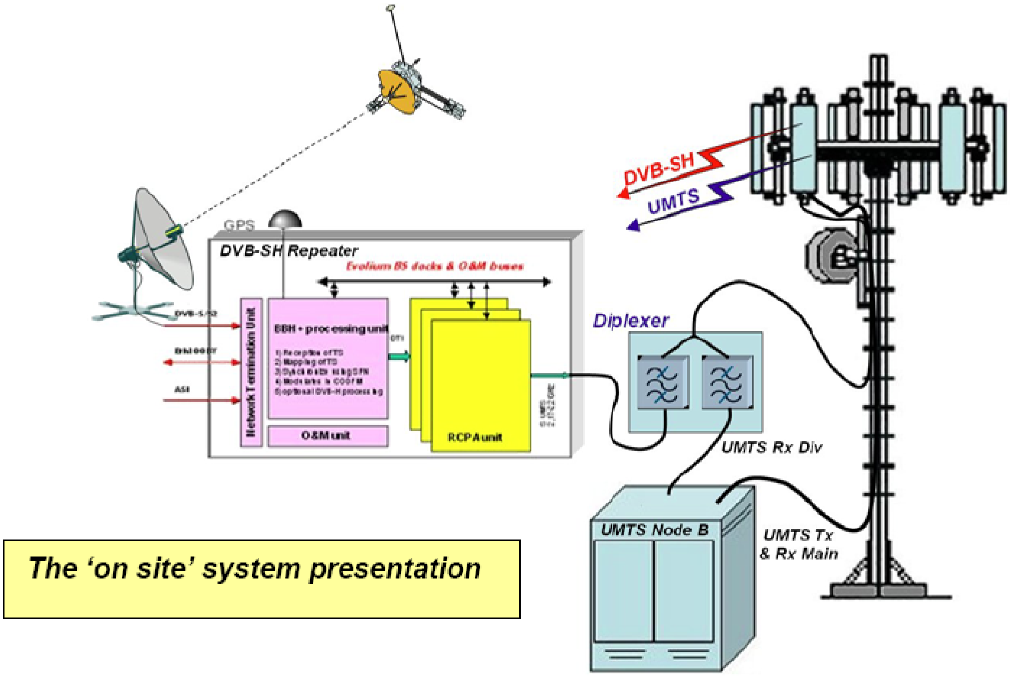


Figure 7.63: UMTS Base Station and DVB-SH transmitter general co location architecture

The impact on the 3G downlink link budget are the following

- Considering the previous antenna sharing architecture, the impact on the 3G downlink budget is 0,7 dB, due to diplexer and jumper.
- O the 3G uplink , there is the same attenuation of 0,7 dB that is compensated by the TMA (Tower Mounted Amplifier).
- The ACLR specification for DVB-SH in S-UMTS bands are the same as for 3G: 45 dB for ACLR 1 and 50 dB for ACLR 2. During the field trials in Toulouse, no interference case has been noticed.

On the MobileTV link budget, the influence of the site sharing is the following

- The impact on the link budgets (downlink only) is in the range 0,6 dB to 0,8 dB.
- The 3G antennas are generally compliant with MTV signal transmission; their transmission pattern is similar to the UMTS one as the frequency band is adjacent.

Impact of the DVB-SH on 3G Quality of Service

During the deployment of the Pau field trial, SFR® has performed different tests on their Key Indicator parameters (KIP). 3G Quality of Service was measured before, during and after installation of DVB-SH transmitters, based on the following KIP:

- Call set-up success rate (CSSR).
- Call completion success rate (CCSR).
- Interrupted call for various reasons (refer to label "Interrupt Rate" in the Graph").

As shown in figure 7.64, all quality indicators remained unchanged after installation of the DVB-SH transmitters sharing 3G sites and antennas.

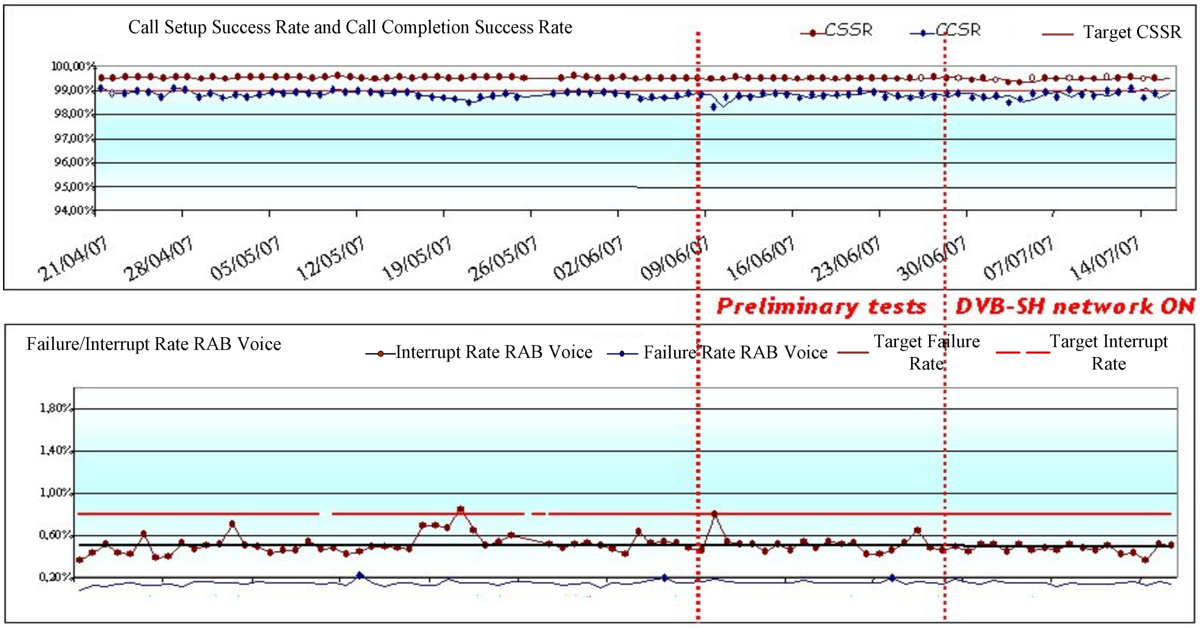


Figure 7.64: Results of SFR® tests on QoS

7.7.3 Results of simulations

Different simulations have been performed concerning the impact of CGC and specifically high power CGC on the reception of UMTS inside the same area. As the case of collocated CGC and 3G Base Station is not critical and there is no interference, the simulations have been concentrating on the case of non co-located sites. The simulations have taken the city of Torino as deployment site. Thus, we consider a UMTS network with a classical cellular topology, and a High Power CGC with two possible powers: 100 W and 300 W. The BS configuration parameters are the following, using a three-sector antenna and 20 W power amplifier and with 17 dBi antenna gain and 3 dB feeder losses. The configurations and parameters of the different sites are gathered in table 7.28.

Table 7.28: BS configuration

	Site name	Antenna height AGL	ASL	Number of sector	Azimuth spacing	Max EIRP/sector
Configuration 1	All Sites	30 m	196 m	3	120°	57 dBm

The UMTS traffic parameters are the following:

- Channel overhead: 10 %.
- Cell load: 100 %.
- Number of 12,2 kbps services: 40.
- Number of 64 kbps services: 15.
- Number of 128 kbps services: 10.
- Number of 384 kbps services: 4.

The UM "Voice service" UE parameters are listed in table 7.29.

Table 7.29: Voice service UE parameters

Central frequency	Power	Antenna	Antenna height	Power control	Ec/Io	Eb/No forward	Eb/ No reverse	Noise
2 167,5 MHz	0,125 W	Omni	1,5 m	60 dB	-14 dB	5 dB	6 dB	-102 dBm

The "Data service" UE parameters are listed in table 7.30 where E_c/I_0 is the energy per chip over noise plus interference power spectral density required for the CPICH channel and E_b/N_0 is required energy per bit over noise power spectral density for the traffic channel.

Table 7.30: Data services UE parameters

Central frequency	Power	Antenna	Antenna height	Power control	Ec/Io	Eb/ No forward	Eb/ No reverse	Noise
2 167,5 MHz	0,125 W	Omni	1,5 m	60 dB	-14 dB	2 dB	5 dB	-102 dBm

The TS terminals parameters are listed in table 7.32.

Table 7.31: CGC configurations

	Site name	Antenna height AGL	ASL	Number of sector	Azimuth spacing	EIRP/sector	ACLR 1st Channel
Configuration 1	Cernaia	82 m	248 m	3	120°	69 dBm	49 dB
Configuration 2	Cernaia	82 m	248 m	3	120°	64 dBm	49 dB

The summary of the different results are contained in tables 7.33 and 7.34.

Table 7.32: Influence of the CGC on the UMTS covered area

		UMTS only (REF)	UMTS + config1	UMTS + config2
CPICH	Surface (km²) covered having E_c/I_0 > -14 dB	764,3336	763,0632	763,6
	% covered having E_c/I_0 > -14 dB	100	99,83	99,90
	Difference/REF (km²)		-1,2704	-0,7336
Voice	Surface (km²) covered having E_b/N_0 > 5 dB	866,9536	866,9536	866,9536
	% covered having E_b/N_0 > 5 dB	100	100,00	100,00
	Difference/REF (km²)		0	0
Datas	Surface (km²) covered having E_b/N_0 > 2dB	865,8448	865,2012	865,666
	% covered having E_b/N_0 > 2 dB	100	99,93	99,98
	Difference/REF (km²)		-0,6436	-0,1788

Table 7.33: Influence of the CGC on the number of lost users

	REF	UMTS + config3	UMTS + config4
Eb/No min (dB)	5,19	2,76	4,54
Eb/ No max (dB)	24,43	24,43	24,43
Eb/ No ave (dB)	22,02	21,96	21,99
Nb of connected sub	**17 066**	17 066	17 066
% of connected sub	**100**	100	100
Nb of sub lost		**0**	**0**
Ec/Io min (dB)	-14	-24,93	-20,57
Ec/ Io max (dB)	-10,01	-10,01	-10,01
Ec/ Io ave (dB)	-11,55	-11,6	-11,57
Nb of connected sub	**17 066**	16 911	16 966
% of connected sub	**100**	99,09	99,41
Nb of lost subscribers		**155**	**100**

Conclusions

- We see that a high power CGC transmitter introduces a very low degradation of the UMTS services.
- In the worst case (EIRP=69 dBm and high antenna height) we have a loss of only 0,4 % of the covered area for the data service and 0,05 % for the voice.
- If the same CGC is located at a lower altitude, the interferences become negligible since there is no degradation of the voice service and only 0,07 % of the area is lost for the data.
- The number of lost subscribers is less than 1 % in the worst case without taking into account the possible reconnection to an other Node B.
- The simulations show that a high power CGC transmitter located in the same geographical area than a UMTS network and using an adjacent frequency degrades very few the UMTS service (less than 0,5 % of the covered area and less than 1 % of the subscribers).
- The degradation zones are clearly located around the high power CGC. The rest of the covered area is not affected by the CGC.
- Using interference reciprocity, one can deduce that DVB-SH will be as much impacted by UMTS Base Station operation in adjacent frequency band.

7.7.4 Theoretical impact of UMTS over DVB-SH

With the increasing number of DVB-SH systems, the coexistence of existing UMTS networks and of S-Band components is becoming an important issue to be addressed. For this reason, an investigation into the reciprocal interference between the two systems is needed.

The aim of this clause is to make an assessment of the interference between the Italian DVB-SH network and the UMTS network. The assessment has been carried out following a variety of procedures and a specific methodology that were defined in accordance with the constraints and targets related to this issue.

During the first two phases of the J-Ortigia project, the forward link DVB-SH transmissions were located in the first 5 MHz of the 2 170 MHz to 2 200 MHz band and were therefore adjacent to the band of an Italian UMTS service provider (Vodafone™).

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- A clear and complete citation must be supplied referencing the author, the full title and the version of clause 7.7.4.

7.7.4.1 Assessment of the Interference

The aim of this clause is to provide an assessment of the impact of UMTS interference on the pilot Italian DVB-SH network out in place around Pisa. This is done by quantifying the SNIR degradation and associating it to specific areas around the base stations. An example of how this can affect the quality of service is provided by addressing a typical use case and identifying the relative "exclusion zone". Finally, a set of on-field measurement extracted from a measurement campaign is presented in order to provide an example of realistic values.

7.7.4.1.1 Methodology

Due to the impossibility of controlling the UMTS network and to the low level of the interference signal, a theoretical assessment based on standard specification is carried out first. For the evaluation of the reciprocal interference of DVB-SH and UMTS networks, a symmetrical study can be performed. The analysis considers several useful values and figures defined in the harmonized standards, which can be adopted as reference system characteristics. There follows a brief description of these parameters.

BS/CGC Maximum Output Power, Pmax, is the mean power level per carrier measured at the antenna connector in specified reference condition.

Emission Mask specifies the maximum output power that a CGC or a BS shall not exceed. This is an absolute value, specified for various output power ranges and given in function of the frequency offset from the carrier frequency of the relevant system.

Adjacent Channel Leakage power Ratio (ACLR) is the ratio of the filtered mean power centered on the assigned channel frequency to the filtered mean power centered on the adjacent channel frequency. The ACLR is therefore a relative value.

Minimum receiver sensitivity is specified as the minimum input level required by the receiver; this value is taken as reference in order to characterize the interference with respect to the thermal noise power.

Table 7.34 summarizes the main useful information which characterizes both the UMTS and the DVB-SH network.

Table 7.34: RF Characteristics

	ACLR	Emission Mask Integration	Minimum receiver sensitivity	Maximum output power	
Network type/Unit	**(dB)**	**(dBm)**	**(dBm)**	**(dBm)**	
UMTS	-45 [15]	-0,75 [15]	$-106+(C/N)_{MIN}$ [i.37]	Wide Area BS [15]	No upper limit
				Medium Range BS [15]	≤ +38
DVB-SH	-45 [i.36]	0,77 [i.36]	$-105+(C/N)_{MIN}$ according to clause 10.4.4	HPA	53
				LPA	47

The following analysis will focus on the interference caused by the UMTS network to the DVB-SH: due to the similarity of the two systems, similar results can be expected for the interference caused by DVB-SH, which will be addressed later on. The analysis will consider *medium range BS* for urban environment characterization and *wide area* BS for rural areas.

The area around the base station (BS) is characterized based on the impact of interference on the overall SNIR. The following analysis provides therefore a quantification of out-of-band emissions on adjacent systems, but does not give specific information on the quality of service, which depends on the level of the received signal power. For example, in case of DVB-SH Complementary Ground Component (CGC) being co-located with a UMTS BS, the reception could still be possible even though significant SNIR degradation is present.

As derived from table 7.34, it is assumed there is a user terminal with a noise figure NF=2 dB and with a consequent noise power of -105 dBm/5 MHz. The scope of this assessment is to evaluate the impact of the interference level on the final SNIR and in order to this:

- The previous value of noise power is fixed and taken as reference.
- The interference power is expressed as a percentage of this noise power.
- The corresponding impact on the SNIR in terms of degradation is calculated.
- The areas where the various amount of degradation can be expected are identified and classified.

The results can be applied to any reception scenario by simply considering the level of received signal power in order to calculate the specific SNIR.

Following this methodology it is possible to identify three areas around the base station:

1) One area where the reception is limited by the interference level; this will be called Interference limited area. Taking into account the definition of minimum sensitivity for a DVB-SH receiver, it is possible to consider a value of -105 dBm/5 MHz as a threshold for the identification of the area. This corresponds to a receiver with a noise figure NF=2 dB. Here, the impact of the UMTS out-of-band power on the SH user terminal C/N is higher than 3 dB (noise temperature increase higher than 100 %) over the whole area and it is exactly 3 dB on the edge.

2) An intermediate region where the impact of the interference on the C/N goes from a maximum of 3 dB to a definitely negligible degradation of 0,04 dB (which corresponds to an increase in noise temperature due to UMTS interference less than 1 %, i.e. -125 dBm/5 MHz).

3) The interference free area, where the impact of the UMTS out-of-band signal is definitely negligible (less than 1 %). This is the area where the $\frac{C}{N+I} = \frac{C}{N} - \varepsilon$ and therefore no degradation due to UMTS can be expected.

This analysis is carried out both for an urban UMTS BS with a medium range output power and for a rural wide range BS. In the first case, the ACLR specified in [15] shall be used in order to evaluate the power of the out-of-band UMTS signal based on the in-band one. On the other hand, as no upper limit to the output power is specified, in the second case it is recommended to consider the integration of Emission Mask as an upper limit for the interference power. Both values are included in table 7.34. These considerations will be used in the RF coverage prediction tool.

Interference Limited Area

It identifies the area where the interference is predominant, i.e $I_{UMTS\_MAX} > N$ for a receiver with a NF=2 dB. In this area, the interference limits the C/(N+I). However, reception could still be possible in the case where a higher SH signal power is received (as long as the Csh/(N+I)>(C/N)min).

The radius of the interference limited zone is therefore given by the distance from the source of the interference where the following condition is fulfilled:

$$\begin{aligned} & R_{IL} = r \,|: \\ & I_{UMTS\_MAX}(r) = -105 \quad \text{dBm/5 MHz} \\ & \qquad\qquad\qquad \approx -172 \quad \text{dBm/Hz} \end{aligned}$$

In this area, the impact of the interference power on the overall C/(N+I) is higher than 3 dB and is exactly 3 dB on the edge of the area. A similar condition can be also applied when considering the DVB-SH as the interference source and the UMTS as the victim system.

For a medium range BS, an ACLR of 45 dB, it is possible to obtain the interference RF prediction by subtracting the ACLR from the in-band power:

$$I_{UMTS\_MAX} = C_{UMTS} - ACLR$$

By applying this result to the predictions of an urban BS in Pisa broadcasting with 38 dBm and a typical UMTS sector antenna of 18 dBi it is possible to evaluate theoretically the related interference limited zone.

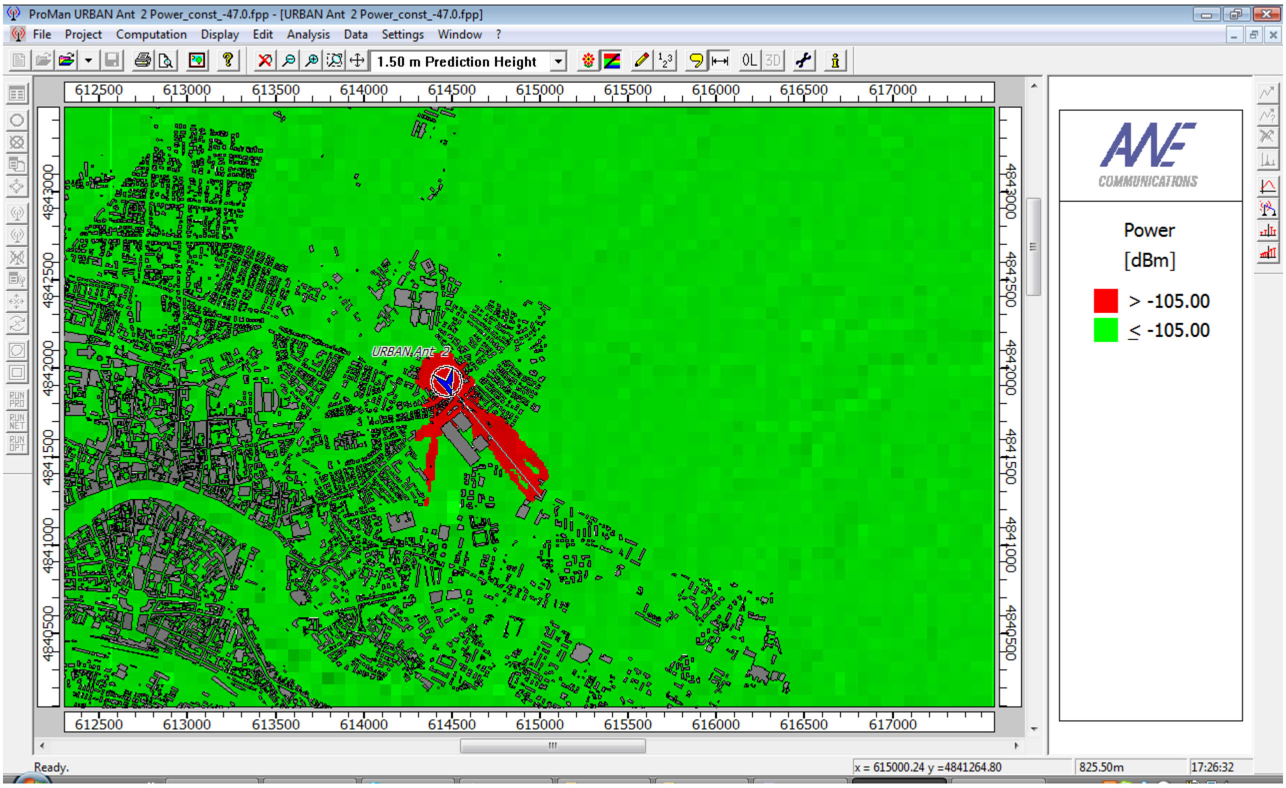


Figure 7.65: Interference limited area (in red) for an urban UMTS BS in Pisa. Max radius 825 m

With regard to the study of a rural BS, a wide range output power is considered, in order to deal with the worst case reception condition for rural cells. As for this type of BS there is not specified limit to the output power, the only limitation expressed in the standards is the UMTS Emission Mask absolute power, which can be calculated by an integral of the emission mask.

- This value indicated in table 7.35 is equal to -0,75 dBm for the UMTS.
- The relative coverage prediction, with the -105 dBm threshold, is shown in figure 7.66.

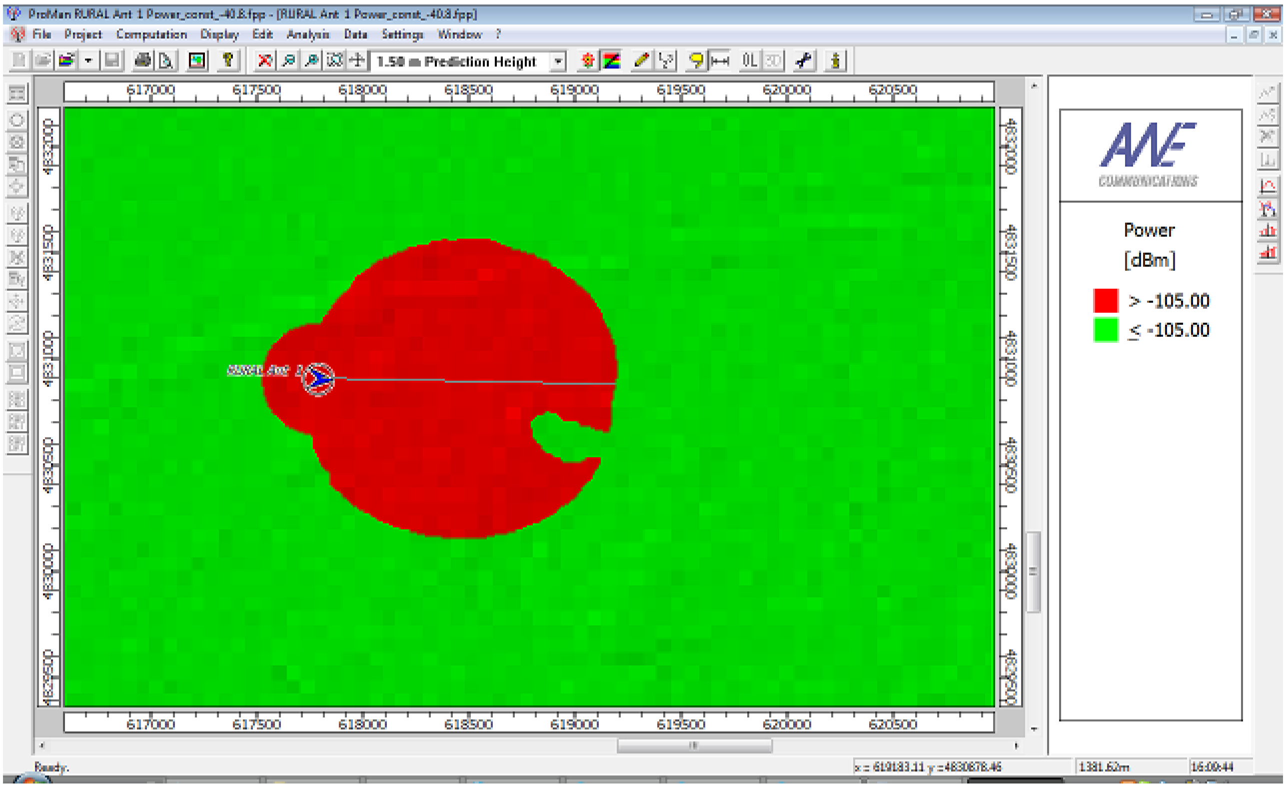


Figure 7.66: Interference limited area (in red) for a rural UMTS BS in Pisa. Max radius 1 380 m

From the previous figures, an interference limited zone of 825 m and 1 380 m is expected for the urban and rural base station respectively. This result indicates that in those areas the interference power could significantly affect the reception quality, whenever a weak S-band signal is received. However, as already underlined, the limits imposed by the standards represent only a boundary, as the real filters can perform better. The final quality of service is then determined by C_{SH} level.

Intermediate region

The distance where the impact of the interference power is exactly 3 dB identify the beginning of the intermediate region. In this region the increase in noise temperature due to UMTS interference goes from a maximum of 100 % (on the first edge), to a minimum of 1 % (on the second edge):

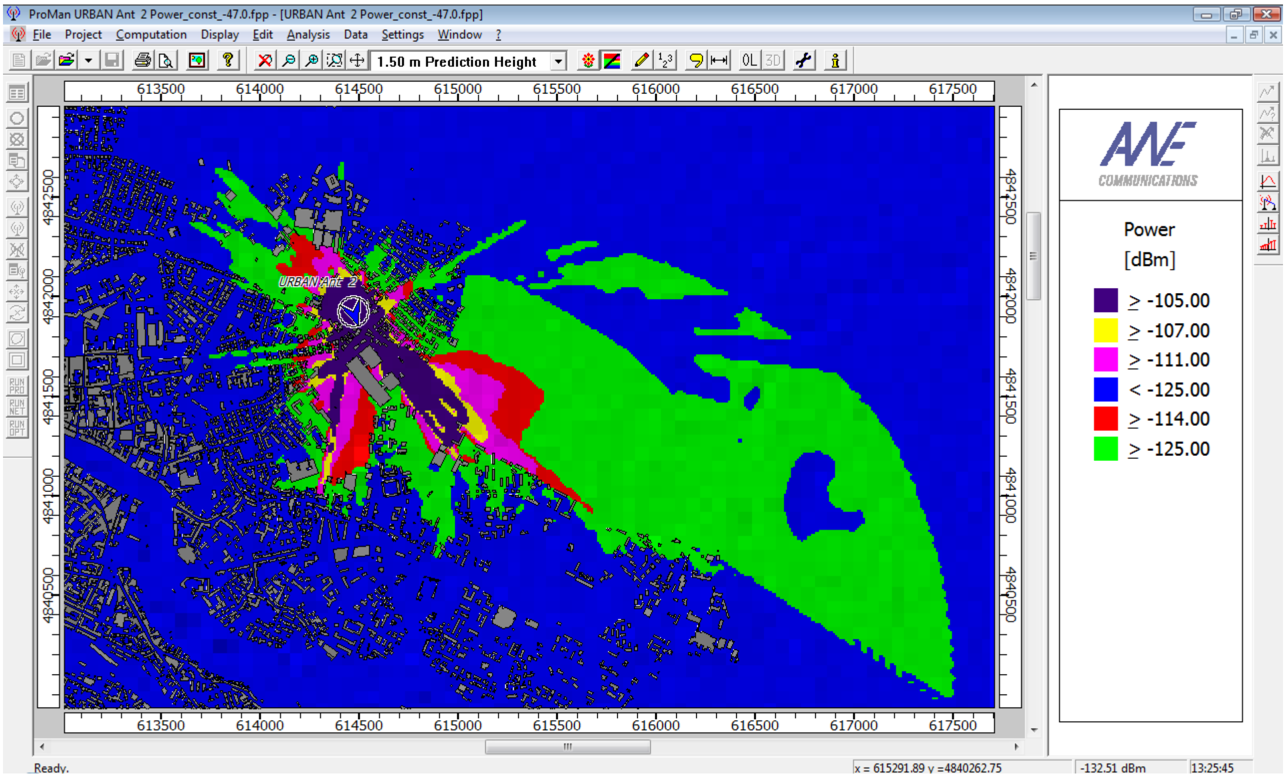
$$R_{IR} = r \mid :$$
$$-125 < I_{UMTS\_MAX}(r) < -105 \quad \text{dBm/5 MHz}$$

In this region, the interference does not dominate the noise power and this is why its impact on the final C/(N+I) is less than 3 dB. However, it is possible to identify other intermediate thresholds between the two mentioned values. In particular, table 7.35 summarizes the considered thresholds.

Table 7.35: Overall SNIR degradation, interference percentage and relative thresholds

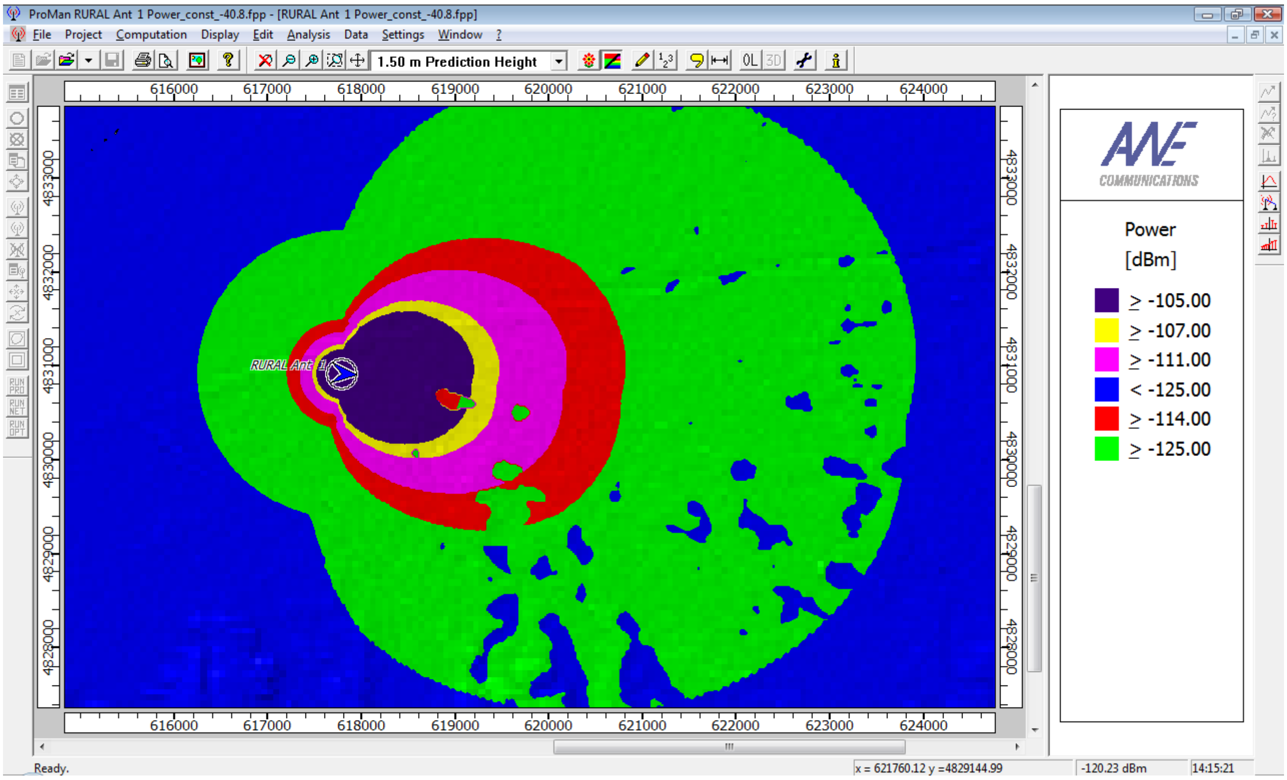
$\left(\frac{C}{N+I}\right)$ Degradation (D) [dB]	$\left(\frac{I}{N}\right)$%	I_umts Threshold
-3	100 %	-105
-2	58 %	-107
-1	25 %	-111
-0,5	12 %	-114
-0,04	1 %	-125

Based on this table, the evaluation of the extension of each zone can be obtained through RF coverage predictions. In particular, the ACLR value is still used to derive the interference power for an urban BS and the Emission mask for a rural one.



NOTE: Quasi Interference free area (in green).

Figure 7.67:Various Interference thresholds for an urban UMTS BS in Pisa



NOTE: Quasi Interference free area (in green).

Figure 7.68: Various Interference thresholds for a rural UMTS BS in Pisa

Interference Free Area

The interference created by the interfering system is considered negligible when the increase of noise temperature created by this interference is less than 1 %, with a consequent C/(N+I) degradation of 0,04 dB. This gives a threshold of:

$$-125 \quad \text{dBm/5 MHz}.$$

It is therefore possible to conclude that the interference from the UMTS system to DVB-SH is negligible if the UMTS out-of-band power is lower than this value. The radius at which this condition is fulfilled represents the BEGINNING of the interference free area:

$$R_{IF} = r \,|: \\ I_{UMTS\_MAX}(r) < -125 \quad \text{dBm/5 MHz}$$

NOTE: On the basis of a UE with NF=2 dB.

In this area, the interference does not limit the C/(N+I). For a urban BS, the interference free area begins at 3 830 m distance (see figure7.67) while, as shown in figure 7.68, for the rural one it begins after about 6 km. Table 7.36 summarizes the radius from the UMTS BS which characterizes each area, from the interference limited zone to the interference free area, with the intermediate region in-between.

Table 7.36: Maximum Radius of each area

$\left(\frac{C}{N+I}\right)$ **Degradation (D) [dB]**	$\left(\frac{I}{N}\right)$%	**Urban BS Max Radius [m]**	**Rural BS Max Radius [m]**
≥ -3	$\left(\frac{I}{N}\right)$ ≥ 100 %	825	1 340
-2 ≤ d < -3	58 % ≤ $\left(\frac{I}{N}\right)$ < 100 %	902	1 580
-1 ≤ d < -2	25 % ≤ $\left(\frac{I}{N}\right)$ < 58 %	1 200	2 340
-1 ≤ d < -0,5	12 % ≤ $\left(\frac{I}{N}\right)$ < 25 %	1 560	2 940
-0,5 ≤ d < -0,04	1 % ≤ $\left(\frac{I}{N}\right)$ < 12 %	3 580	6 045
d ≤ -0,04	$\left(\frac{I}{N}\right)$ ≤ 1 %	> 3 580	> 6 045

7.7.4.2 Exclusion zone with Satellite only reception

The term exclusion zone indicates the area where, in the absence of UMTS BS, DVB-SH reception is possible (i.e. the C/N is above the minimum required), however, when switching on the BS, the interference does not allow correct reception anymore. In other words, it is the area where the following two conditions are fulfilled:

$$\frac{C}{N} \geq \left(\frac{C}{N}\right)_{\min} \quad \& \quad \frac{C}{N+I} < \left(\frac{C}{N}\right)_{\min}$$

In order to simplify the evaluation of the extension of the exclusion zone, a satellite only reception scenario is considered, where the signal power is therefore fixed and nearly constant over the whole area considered. The user equipment is assumed to have a noise figure of NF=2dB. The reference satellite signal power in the city of Pisa, based on a passive antenna with NF=0 and a receiver with NF=2, is around -99,84 dBm. It is then assumed to have a QPSK signal with coding rate 1/2. For this configuration, in a LOS AWGN channel, the minimum required C/N for a TDM signal is equal to 1,6 dB (including 0,5 dB of implementation losses). In this context, the radius of the exclusion zone can be defined as:

$$R_{EZ} = r \,|: \left[\left(\frac{C_{SAT}}{I_{UMTS}(r)}\right)^{-1} + \left(\frac{C_{SAT}}{N}\right)^{-1}\right]^{-1} = \left(\frac{C}{N}\right)_{\min}$$

Consequently, all the points (x,y) where:

$$\left[\left(\frac{C_{SAT}}{I_{UMTS}(x,y)}\right)^{-1} + \left(\frac{C_{SAT}}{N}\right)^{-1}\right]^{-1} < \left(\frac{C}{N}\right)_{\min}$$

belong to the exclusion zone and no correct reception can be expected. This corresponds to the following condition on the interference power:

$$I_{UMTS}(x,y) > C_{SAT} \cdot \left[\left(\frac{C}{N}\right)_{\min}^{-1} - \left(\frac{C_{SAT}}{N}\right)^{-1}\right]$$

Considering that over the 5 MHz C_{SAT} = -99,84 dBm, $N = -105$ dBm and $\left(\frac{C}{N}\right)_{\min} = 1{,}6\,dB$, the exclusion zone is then characterized by a UMTS out-of-band power of:

$$I_{UMTS}(x, y) > \text{-}104 \quad \text{dBm/5 MHz}$$

Considering a rural BS, broadcasting with very high power, a satellite only reception would be a very critical case. With the above assumption, the resulting radius of the exclusion zone is 1 280 m, as shown in figure 7.69.

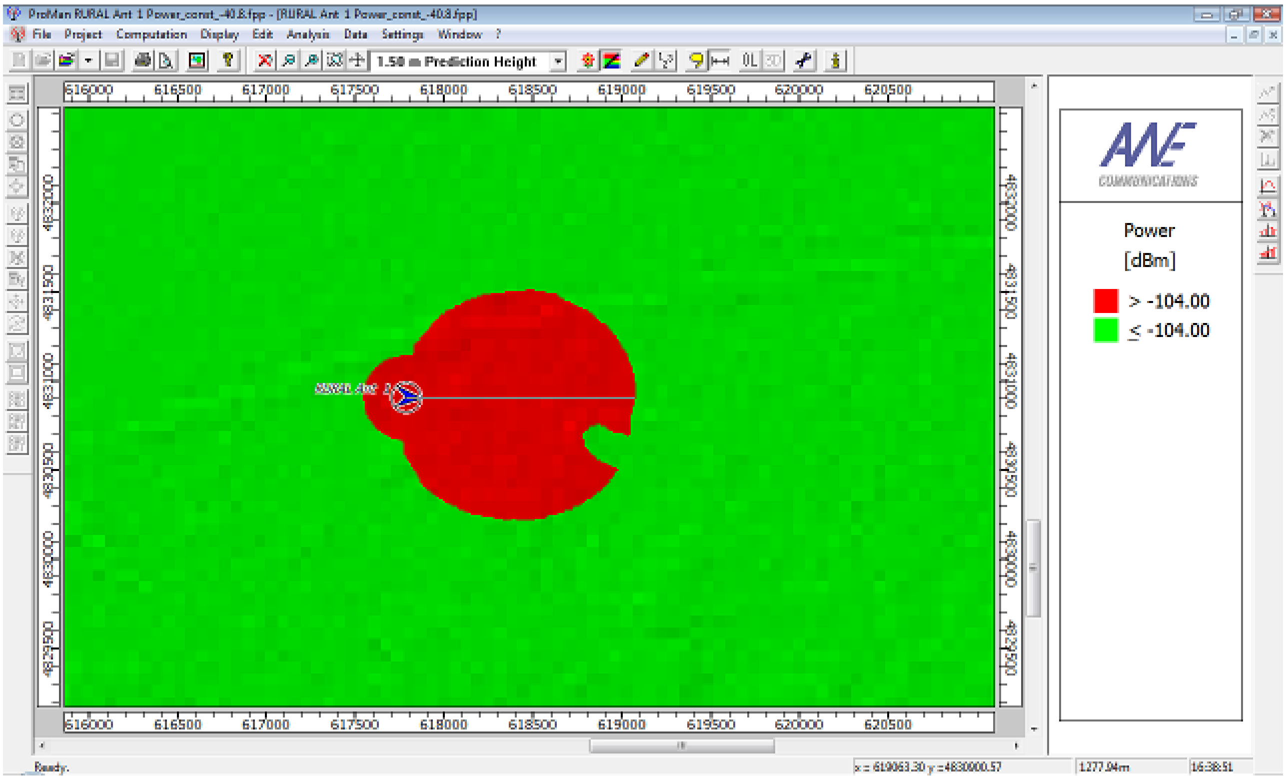


Figure 7.69: Rural wide area BS exclusion zone

7.7.5 Measurement results from On-Field Campaign

The UMTS BS output power is highly time variable. Due to this and to the low range of powers which characterizes the out-of-band signals, many of the on-field tests carried out cannot give evidence of the expected behaviour of interference. Some urban measurements are presented in this clause in order to provide an example of the assessment procedure and of some realistic values. The main outcome of the on-field measurement campaign is that a different and higher ACLR value has been measured in relation to an urban UMTS cell, which confirms that filters can perform also better than what specified in regulations. In this framework the determination of the ACLR is the main purpose of the assessment.

7.7.5.1 Measurements setup

The measurement setup is described in figure 7.70. The test setup is composed by a passive antenna +cascaded by an external low-noise amplifier and a spectrum analyser (Agilent MXA N9020A SA).

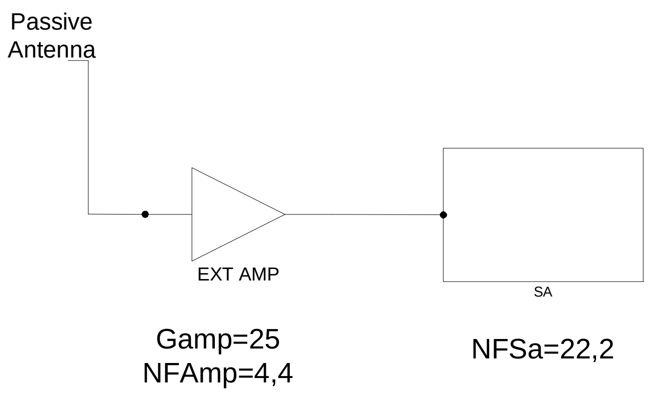


Figure 7.70: Measurement Setup

Table 7.37: Measurement Setup

Spectrum analyzer	**Agilent MXA N9020A**	
	DANL	-151,8 dBm/Hz
	Noise Figure	22,2 dB
External Amplifier	**Spin Electronics WBP 18 - 007**	
	Gain	25 dB
	NF @ 290 K	4,4 dB
Passive Antenna	**Procom**	
	Polarization	Vertical
	Gain	0 dB
	NF	0 dB

A specific procedure described in the Agilent Tutorial [i.39] defines how it is possible to calculate the sensitivity and the new DANL* of this configuration of the measurement setup. The resulting noise figure is:

$$NF_{sys} \approx 5{,}6\,dB$$

Where the total noise figure NF_{sys} is defined based on the specification of a calculation procedure, described in [i.39]. With the above configuration it is possible to have the following sensitivity and DANL*:

$$Sensitivity \approx -174 - 5{,}6 = -168{,}4\,dBm/Hz = -101{,}4dBm/5MHz$$

$$DANL^* \approx Sensitivity + G_{pre} = -143{,}4\,dBm/Hz = -76{,}4dBm/5MHz$$

All the following results were obtained by measuring the signal power in three adjacent bands: the first channel 2 165 MHz to 2 170 MHz gives the UMTS in-band power while the eventual interference power must be measured in the second channel 2 170 MHz to 2 175 MHz. However, since the interference level is often near the DANL* of the measurement chain, a third measurement in the upper adjacent channel 2 175 MHz to 2 180 MHz is needed in order to validate it. This procedure is needed because of the impossibility of switching off the UMTS network. The DVB-SH network is in the meanwhile switched off. The ΔP upper and lower values simply give the difference in dBc (dB to Carrier) of the right and left channels in comparison to the central channel (2 170 MHz to 2 175 MHz).

7.7.5.2 On-field Measurement Results

Table 7.38 lists some measurement results related to an urban UMTS and a suburban base station in Pisa, taken at different distances from the BS. The noise power related to the measurement chain is equal to the sensitivity previously calculated:

$$N = -101{,}4\,dBm/5MHz$$

Table 7.38: UMTS field strength measurements

Meas. Code	Distance from the BS [m]	Ch1 2 165 MHz to 2 170 MHz $\left[C_{UMTS}+N\right]_{dBm}+G_{pre}$ [dBm]	ΔP_{lower} [dBc]	Ch 2 2 170 MHz to 2 175 MHz $\left[I_{UMTS}+N\right]_{dBm}+G_{pre}$ [dBm]	ΔP_{upper} [dBc]	Ch 3 2 175 MHz to 2 180 MHz $N+G_{pre}$ [dBm]
Urb_168	168	-10,65	60,89	-71,54	-4,5	-76,05
Urb_263	263	-11,52	61,42	-72,94	-3,2	-76,17
Urb_377	377	-17,01	57,31	-74,32	-2,1	-76,47
Sub_40	40	-9,25	62,15	-71,4	-4,7	-76,04

The test locations are showed in figure 7.71.

Figure 7.71: Test locations for the urban UMTS BS

The measurements reveal a very high BS output power. This can be due either to the location of the BS or to the number of users connected to that station, which is proportional to the traffic and therefore to the output power. In order to validate the measurement of very low signals, like interfering signals, it is recommendable to:

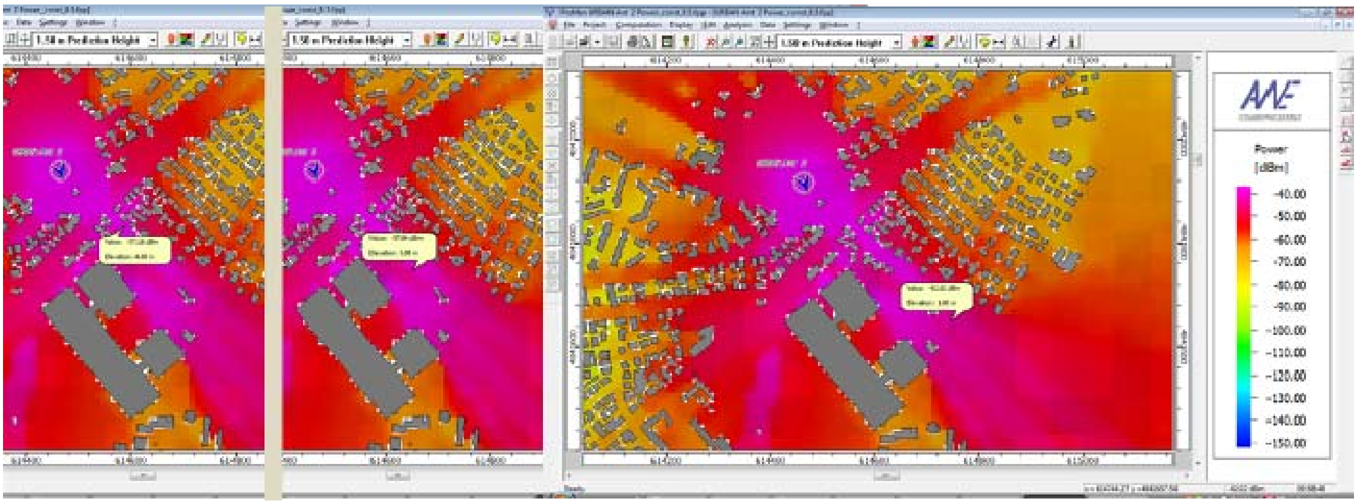
- Check that the obtained values are above the DANL.
- Check that the values obtained by cleaning measurements from the gain are above the sensitivity.
- Compare the obtained interference trend with the predicted results of the network planning tool (for the urban BS only).
- Compare the obtained values with the signal power of an adjacent channel where no signal is expected (Channel 3). This is because, being the interference level in that channel negligible, the displayed value can be taken as a reference lower bound (in fact it often corresponds to the DANL*). The value ΔP_{upper} in table 7.38 represents this difference.

The measures here selected confirm the previous checks. Evidence of the last two points is given below.

Table 7.39 summarizes the results predicted by the network planning tool relatively to the urban BS obtained by fixing the farthest measured value (377 m).

Table 7.39: Prediction and measurement on the interference

Distance from BS	Predicted C_{UMTS} (dBm)	Delta (dBm)	Measured C_{UMTS} (dBm)
168 m	-37,18	1,53	-35,65
263 m	-37,84	1,34	-36,5
377 m	-42,02	0,02	-42



distance = 168 m distance = 263 m distance = 377 m

Figure 7.72: Prediction on the interference trend

Looking at figure 7.72 the values predicted by the tool for the 168 m and 263 m points are very close to the measured ones. It is therefore possible to conclude that **the out of band measurements denote an interference trend coherent with what expected.**

The ΔP_{upper} value can be affected by measurement artefacts and it is therefore advisable to consider mainly the measurements in which such ΔP_{upper} is higher than 2 dB. This is once more the case of the quoted measurements: in this context, not only is the power significantly high, but also the ΔP_{upper} denotes clearly that the power measured in channel 2 represents the interference power.

Summarizing all the elements, it is possible to assert that:

- the measured value denote an interference trend coherent with what predicted;
- all the values obtained were above the available sensitivity/DANL;
- the urban BS measurements present always $\Delta P_{upper} > 2 \text{ dBc}$ which decreases for increasing distance.

Then, the real value of the interference I_{UMTS} can be calculated by linearly subtracting N obtaining the table 7.43 results.

Table 7.40: UMTS interference calculations

Meas. Code	Distance from the BS [m]	C_{UMTS} [dBm]	I_{UMTS} [dBm]	$ACLR$ [dB]
Urb_168	168	-35,65	-98,25	-35,65-98,25= **-62,6**
Urb_263	263	-36,52	-100,54	-36,52-100,54= **-64**
Urb_377	377	-42,01	-103,5	-42,01-103,5= **-61,5**
Sub_40	40	-34,25	-98,05	-38,25-98,05= **-63,8**

Note that the measurement results can be affected by a certain bias, linked to the stability of the used instruments. Furthermore, being interested in very low signals which are close to the thermal noise level accentuates this uncertainty, thus leading to consider few dB of approximation.

Based on these observations, it can be concluded that: **The urban UMTS Vodafone™ base station used an ACLR of approximately 60 dB.** With such values some degradation of the C/(N+I) can be expected, as they are above the -105 dBm threshold. This would mean that the measurement locations belong to the interference limited area and that the impact of the interference on the SNIR is higher than 3 dB degradation. However, the maximum distance where the measurement still give relevant results is 377 m, which is quite lower than the radius identified in figure 7.65 (825 m).

The results of this on field measurement session denote that better conditions can be found (i.e ACLR of 60 dB) and that the output power of a BS can be significantly high due to the number of users and amount of traffic.

7.7.6 Conclusions and Recommendations

The different trials, simulations and computations provided in the previous sections can be summarized as follows:

- In the case of terrestrial deployment, the impact of DVB-SH repeaters on UMTS reception is very low, quite undetectable.
- Even if no measurements have been performed, the reciprocal is also valid. In the different trails, no specific perturbations have been observed on DVB-SH reception.
- The reciprocal impact of UMTS and DVB-SH networks is in the same order of magnitude that the reciprocal impact of two UMTS networks.
- When DVB-SH reception is only performed through satellite (no CGC deployed), the possible presence of UMTS Base Stations can cause interference on the DVB-SH reception. In this case, the zone where the interference degrades the received satellite signal by a factor of 3 dB is called exclusion zone.
- During measurement campaigns, the measured radius of the exclusion zone is around 400 m in urban areas, far lower than the predicted value of 800 m, and the probable presence of a CGC will reduce this zone drastically.
- Though no measurements were reported on rural Base Station impact, one can extrapolate that the measured value will be also around half the predicted value, that is to say around 700 m instead of 1 300 m.
- Some results coming from J Ortigia confirm that the presence of S band signals in adjacent band degrades slightly only the DVB-SH signal received from the satellite, decreasing the ESR (5) fulfilment ratio by less than 10 %, which represents less than 0,5 dB degradation.
- We recommend therefore performing careful Radio Network Planning when deploying the system, in order to minimize possible, but still limited, reciprocal interference between UMTS Networks and CGC Networks.

8 Services

8.1 General considerations

From the service point of view, the following similarities between DVB-SH and its DVB precursor can be noted:

- DVB-SH signal is intended to be received by a variety of mobile and portable devices.
- DVB-SH offers high data rates, even in moving conditions and reuses technical elements of the DVB-H specification so that harsh conditions of terrestrial, urban mobile propagations can be addressed in the same manner.
- DVB-SH is a broadcast-centric delivery system: the same content is delivered to an unlimited audience without the risk of network saturation.
- DVB-SH has the flexibility of narrow-casting thanks to the support of multicast protocols.
- DVB-SH is an IP-based system using MPEG2-TS as the baseline transport layer (and consequently, DVB MPE as the default encapsulation protocol). Existing DVB signalling such as PSI/SI, SFN, Time-Slicing, MPE-FEC are mostly preserved.

- DVB-SH can carry DVB-IPDC in essentially the same way as DVB-H does. In this framework, it can support two-way interactive services when coupled with an appropriate interactive channel.

There are however key differences from DVB-H that should be kept in mind, due to the hybrid-network nature of DVB-SH:

- DVB-SH operates in frequency bands internationally allocated to Mobile Satellite Services.
- DVB-SH does not require that the same MPEG2 multiplex be shared with DVB-T or DVB-S/S2 services, nor does DVB-SH requires the use of DVB scrambling at MPEG2-TS for Access Control.
- Although the first deployments of DVB-SH will be based on MPE, the DVB-SH framing specifications has the provisions to allow migration to the new GSE protocol.
- When the 2 GHz S-band band is used, synergy with 3G telephony infrastructure would be exploited, especially in areas where such infrastructure already exists. Network planning for DVB-SH in urban areas could be similar to the 3G planning with the benefit that indoors coverage could be made essentially the same.
- A DVB-SH coverage is always composed of a satellite coverage complemented by a terrestrial coverage. The services offered in these two coverages are strongly linked but not necessarily the same.
- As a consequence of the above, DVB-SH services are a mix of Common services and Local services. Common services are usually those with very large audience while Local services have more fragmented audiences, possibly with geographical dependencies. A Local service package for one city/town may differ from the package for another city/town.
- There are challenges for DVB-SH due to higher mobility, satellite specific propagation channels, and, in some cases, higher frequency bands. Some of these constraints are addressed somewhat differently by SH-A and SH-B architectures.
- Although the Common services are available in both the SC and the CGC, the service attributes may differ depending on the user location, more precisely between different reception modes: satellite-only, terrestrial-only and combined satellite-terrestrial receptions. For example, specific physical parameters may be selected so that higher user speed is possible with satellite-only reception.
- DVB-SH interactive services may rely on a satellite return channel (e.g. in the 2 GHz MSS band) or on a terrestrial return channel (which could be independent from its CGC). The direct return path via satellite could be invaluable in areas without sufficient terrestrial coverage, or in case of catastrophic events leading to unavailability of terrestrial infrastructures.

8.2 Service classification

DVB-SH services could be considered as just a supplement to traditional TV broadcasting. In such a narrow view, DVB-SH just extends the service consumption from the homes to other places. Taking a more forward-looking view, it is expected that new services will naturally develop, exploiting close cooperation between the broadcast infrastructure, the mobile infrastructure and the content provisioning infrastructure. The characteristics of the whole spectrum of services that can be offered with DVB-SH with the above assumption and their implications on the network, the terminal and the user behaviour are discussed below.

8.2.1 Service categories

The services can be categorized using several criteria related to the way it is intended to be consumed. The following questions may be helpful in defining a particular focus for a particular DVB-SH implementation (see note).

- Is the service Common or Local?
- On which terminal category is the service most likely to be received? (Refer to clause 10 for terminal categories definitions and descriptions.)

- What are the likely use-cases? An important factor to take into account is the degree of "cooperation" that can be expected from the user. User cooperation means that the user can adapt his speed, can position/configure the terminal for maximum reception quality or accepts to refrain from certain actions when he is notified that the actual receive condition is not compatible with such actions. An example is a system that can warn the user that he is receiving under frequent signal blockages so that although the FEC is recovering the packet losses of the current service, the zapping time will be excessive due to the necessity to refill the FEC receive buffer for the new service.

- Another important fact to take into account is the expected speed of the terminals under use. A vehicle-mounted application will obviously be travelling at high speeds. However, hand-held devices such as mobile phones or portable media players will commonly be used by passengers in vehicles. Also, car manufacturers are increasingly supporting streaming audio from mobile handsets by Bluetooth® into the car stereo system; again this must work "at speed" and the extension of this to video should also be considered.

- Does the service always require a display and what is the maximum size of such display? A radio service does not require the display to be switched on (or the display can be used for another application, e.g. games). If the display is switched off, the user may wish to put the terminal in his pocket (in the same way he is used to with his MP3 player). The video bit rate should be sufficient in order to avoid visible video artefacts.

- Does the service require "multi-tasking"? For example, is it required to be able to watch a program while answering an incoming phone call with the same device? Another example is the ability to download a program while watching another one.

- Is the service time-critical? Can the content be delayed by several seconds compared to the actual time it is occurring or shown in traditional broadcasting? An example of time-critical service is a live TV program with participation of the selected audience by phone (mobile or fixed phone calls). Such services would not tolerate a latency of more than 600 ms.

- Are feedbacks of terminal information available? Terminal information could be: terminal Class, terminal location (at least coverage location, e.g. SC coverage or CGC cell-id), max speed, on-time periods, receive quality parameters, etc. These feedbacks could be used by the service provider or the network provider. These feedbacks could be in real-time during interactive sessions of an interactive service but they can also be periodically scheduled.

NOTE: The guidance for the selection of key DVB-SH technologies and architectures according to the main service requirements is for further study.

8.3 Service attributes

8.3.1 Definition

Service attributes describe the properties attached to a service as perceived by the end user in various reception conditions. This covers classical parameters found in DVB-H but also new parameters required by the DVB-SH specificities.

DVB-H classical attributes:

- bit rate profile (CBR, VBR, etc.);

- quality of service (Bit Error Rate, Frame Error Rate, ESR5(20), etc.).

DVB-SH specific attributes:

- coverage (satellite, terrestrial);

- zapping time;

- end-to-end delay;

- ease of use in various mobility situation (moving speed, static reception, various degrees of user cooperation, etc.).

8.3.2 Application

The service attributes can be defined at several interfaces.

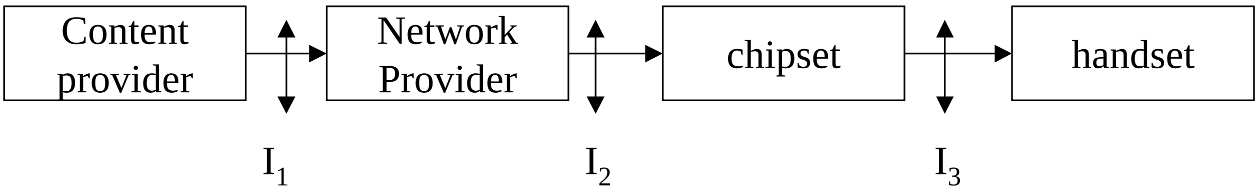


Figure 8.1: Application of service attributes concept

- **interface between content and network provider (I_1):** a service level agreement (SLA) is contracted between the content provider and the network operator so that the content operator requests specific quality, the level of channel and content protection, the coverage, the type of target terminals;
- **interface between network operator and chipset (interface I_2):** network operator transmits the "semantic" attached to the content using over-the-air ESG metadata in a non real time manner. The ESG from DVB-H needs to be updated with the added semantics required in DVB-SH;
- **interface the chipset and the handset (interface I_3):** only the chipset is able to know the real-time parameters such as if and when the user passes from one area covered by the satellite only to another covered by the CGC. This real time information is very "terminal-dependent" and is not be subject of the standardization process but left for implementation (the chipset may get this information by different means). By crossing the real time information with the semantics conveyed by the ESG, the handset is able to process the content and display it to the end user in an agreed manner. Different policies can be found:
  - the ESG display may differentiate Common from Local content. He could adopt his consumption behaviour based on this information (for a commuter between a rural and urban area, he would know where the rural content is aware, then he selects it);
  - for those contents that are not available everywhere, some hysteresis behaviour is possible at the boundary. It may be useful for the terminal to show the content inside the ESG as "selectable" only when its reception is stable enough;
  - zapping time may differ between those contents that are available via satellite (and which could necessitate a longer zapping time) and those that are available via terrestrial and should have a "terrestrial" zapping time. For the satellite contents, decision could be made depending on actual reception conditions to display immediately (typically under CGC) or to delay display (in case conditions may be less favourable, for instance under satellite-only coverage). But another policy could be to display immediately, whatever the actual quality of the stream is, in order to favour immediate perception of the user. So zapping time could be made variable in terms of content attributes and terminal conditions.

Table 8.1: Service attributes at interfaces

Service attributes	I_1	I_2	I_3
Bit rate profile	X		
Quality of service	X		
coverage (satellite, terrestrial)	X	X	X
zapping time requirements	X	X	X
end-to-end delay requirements	X	X	X
Target terminal (moving, fixed, cooperative, etc.)	X	X	X

8.4 Consequences of hybrid architecture on implementation of a DVB-SH service offering

The hybrid nature of DVB-SH systems has consequences on the several service aspects: handover implementation, service discovery and access, Electonic Service Guide (ESG).

8.4.1 Handover issue

For the Local services which are available only terrestrially, handover can be treated in the same way as in DVB-H. It should be noted that in the 2 GHz S-band, the CGC transmitters are likely to implement an SFN over large areas, at least at the level of a city coverage.

In a SH-A configuration, it is most likely that SFN between SC and CGC is implemented. There is no handover in this case for the Common services.

In a SH-B configuration, handover is required since the Common services must be transmitted on two different frequencies for the SC and the CGC, using (by definition) two different modulations and possibly different FEC and interleaver parameters (the same is true for SH-A *whenever SFN is not used*).

It should be noted that there are three differences between DVB-H handover and the DVB-SH-B handover:

- firstly, DVB-H handover assumes that there is only one front-end and one demodulator in the receiver. In SH-B, the TDM/OFDM configuration implies two demodulators working in parallel, one for the satellite signal and the other for the terrestrial signal (the RF front-end may be shared);
- secondly, the satellite TS and the terrestrial TS are not necessarily signalled by DVB-SI as two different TS. When the soft combining property is configured, the two signals coming from the satellite and terrestrial origins are merged before the Turbo decoder and appear at this decoder output as a single TS (see clause 7.2.2.3.3 Maximal ratio combining and complementary code), despite the fact that the actual content of this TS will vary as a function of the presence or absence of the CGC signal [21]. When soft combining is not used, handover is merely switching between two TS outputs. It is recommended that even in this case the PSI/SI be kept identical between the two TS;
- in DVB-H handover between terrestrial MFN cells, the receiver is usually able to estimate the power of the signal coming from different cells in order to optimize the moment when the handover is executed. In the DVB-SH-B hybrid environment, the satellite average signal power is almost constant, with occasional and almost unpredictable shadowing or fading due to terminal movement. Thus no handover prediction is possible.

8.4.2 Network and service discovery mechanisms

Network discovery in DVB-SH is based on DVB PSI/SI with the introduction of new descriptors (to be detailed in a new revision of EN 301 192 [9]).

Due to the possible use of SFN between the SC and its CGC, all the transport streams, whether transmitted via satellite or via terrestrial repeaters, belong the same network in the DVB sense.

Independently of the use of SFN, a DVB-SH network may be divided into "regions", within each a different terrestrial frequency plan is used. A simple example of the partitioning of an SH-network into regions is given in figure 8.2.

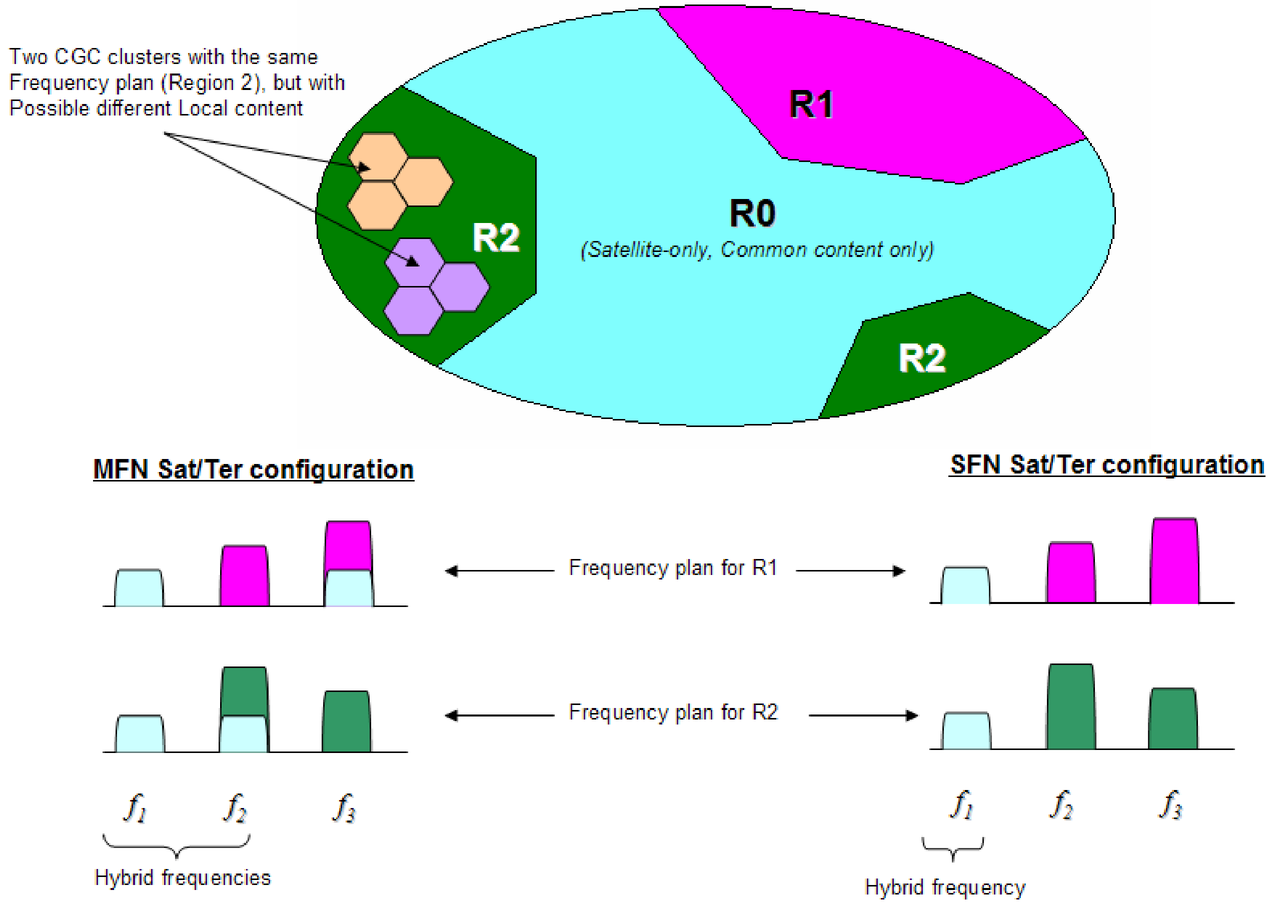


Figure 8.2: Illustration of the concept of partitioning an SH-network into regions

As illustrated by figure 8.2, a Region should **not** be interpreted as a contiguous area, or as a cluster of CGC coverages in SFN mode. For example, in Region R2, the figure shows two disjoint clusters of CGC coverage in SFN mode. The Local content in these two clusters may be different while their frequency plan is identical, by definition of the term "Region". When the SC and the CGC operate in MFN for the Common content, the frequency of the terrestrial retransmission (the "hybrid frequency") may be different between Regions, as depicted in the left part of the examples. When the SC and CGC operate in SFN for the Common content, the physical parameters chosen for the two "terrestrial-only" frequencies may differ between Regions, as shown in the right part of the examples. This may be imposed by interference constraints at the border of the satellite beam (see discussion on "exclusion zones" in clause 11). This is illustrated in the figure by indicating that frequency f3 has more terrestrial capacity than frequency f2 in Region R1 (and vice versa for Region R2).

The information on the validity of a frequency plan when the receiver is in a given region is signalled through a mechanism similar to the one used to signal the Cell_ID in DVB-T/H.

Usually, the Common services of a particular network are managed by a single IP Platform, possibly separated from the Local services which can be managed by several different IP Platforms. The user may have to select several IP Platforms.

It is desirable that, in normal operation, the need to monitor and parse SI information is limited to a minimum and that the discovery and selection of services are based on metadata signalled at the IP level.

8.5 Other considerations

DVB-SH terminal is expected to enjoy a regime of free circulation and use all over Europe as it is the case with mobile phones and receivers of broadcast services.

Terminals with transmit capability on satellite S-band (1 980 MHz to 2 010 MHz) may require compliance to the EN 301 442 [i.32].

It is possible that, through roaming agreements, a Common service in one satellite coverage could be made available also as a Local service in another satellite coverage. It is highly desirable that such feature be offered in the way that looks transparent to the user. In cross-border situations it is desirable that such feature be treated as a special handover.

9 Network configurations

9.1 Considerations on network configurations

Two main types of networks can be considered. On one side we have network that are designed to target portable devices (handsets, handheld devices) and on the other side, network that are designed to target vehicular reception.

9.1.1 Mobile TV network targeting portable devices

The Mobile TV service to handsets are/will be mostly deployed, managed and commercialized with the support of mobile operators. The typical handset will be a cellular handsets with a 2,2" to 4" screen that will allow to display mobile TV on top of regular features such as voice telephony, SMS or Internet access. Depending on countries, these handsets will be subsidized by the mobile operator that will get revenues from the mobile TV (flat fee). Interactive services, possibly VoD and other service will transit using the existing 3G network and will provide additional revenues to the mobile operators.

Another Mobile TV offer can also target non connected devices like Portable Multimedia Players. In this case the service, likely to be controlled by broadcasters, is an extension of broadcast digital TV paradigm to portable receivers essentially based on live channels reception combined with PVR functionalities. Both free to air or pay TV model would apply.

In both cases, services can be provided directly from a satellite in rural areas, but a terrestrial repeater network should be deployed to provide indoor coverage in urban areas where the satellite signal is insufficient. The terrestrial repeaters being designed for a smooth integration in existing 2G or 3G cellular sites. Depending on the requested indoor coverage quality, the number of TV channels, the number of cellular sites to be equipped with a low power DVB-SH repeaters will range from 1/4 up to 1/1. The terrestrial repeaters can be fed either via satellite signal using a DVB-S2 radio interface (typically in Ku or Ka band) or via terrestrial IP network.

In rural areas, satellite ensures nationwide direct reception. To improve indoor coverage, domestic gap-fillers can be used in rural areas to improve satellite reception. Repeaters in urban areas, mostly for indoor coverage, support satellite coverage. These repeaters re-transmit the nationwide channels and will therefore offer indoor coverage identical to a co-located UMTS system. To increase the system capacity in urban areas, adjacent carrier transmitters complement the satellite signal, allowing additional Local content. In each urban area, these complementary channels can be either national, regional or local ones. Finally, for handset based service, a cellular network, or a combination of broadcast type sites (high elevation towers) and cellular type repeaters, can be used to stream in unicast TV channel with limited audience TV and Video on Demand (VoD) as well as for interactivity return channel.

9.1.2 Mobile TV network targeting vehicle

The Mobile TV service to vehicle are mostly deployed, managed and commercialized directly by the broadcasters or the satellite operators, possibly without the support of mobile operators. The typical terminal will be installed in a car with typically a 7" screen, hence higher data rate per channel is expected. This screen can be coupled with other services such as GPS, Video player, MP3 player, etc. Possibly, limited interactive services will be possible using a return channel directly to the satellite.

The objective of the infrastructure is to optimize outdoor coverage across a territory. Services will be provided directly from a satellite in most areas but a terrestrial repeater network should be deployed to provide coverage in dense urban areas where the satellite signal could be insufficient (street canyon, underpass, etc.). As there is no need in this configuration for indoor coverage (except in tunnels), the repeater network will mainly use medium power DVB-SH repeaters that in most case will not be collocated with existing cellular infrastructure. Typically, only around 10 repeaters will be required to cover a typical urban metropolitan area. The terrestrial repeaters can be fed either via satellite signal using a DVB-S2 radio interface (typically in Ku or Ka band) or via terrestrial IP network.

Satellite ensures nationwide direct reception. Suitable spatial technologies (large antennas, high-power platforms, etc.) are used to provide the required net capacity. Repeaters in dense urban areas support satellite coverage. These repeaters re-transmit at the frequency of the satellite carrier. According to frequency scheme selection, a similar complement of urban channels can be implemented. Finally, a return link from the vehicular terminal up to the satellite can be used to provide interactive service using for example the ETSI standardized GMR 1 and 2 waveform already operational on Thuraya and AceS system.

9.2 Synchronization of Satellite and CGC for Common content

9.2.1 Introduction

Basic mode of operation of DVB-SH networks rely on hybrid architecture when satellite broadcast content is retransmitted terrestrially in areas when satellite link availability is reduced. When those terrestrial repeaters are deployed, SFN operation is of great importance and is required in the following conditions:

- when operating in SH-A mode, there is a need for dual SFN operations : between the different terrestrial repeaters, like in DVB-T or DVB-H networks, and between the satellite and the terrestrial repeaters, in a way called hybrid SFN operation;
- when operating in SH-B mode, there is a need of synchronicity between the different terrestrial repeaters and the satellite but the requirements are not as stringent as for SH-A mode.

In this clause the description is only dealing with the regular latency DVB-SH content. The specialties related to the low-latency extension are handled in Annex D.

9.2.2 Terrestrial SFN

The terrestrial SFN operation is described in TR 101 190 [i.22] "Implementation guidelines for DVB terrestrial services". Only basic principles are recalled here, and some differences between the two systems are described in more details.

9.2.2.1 Principle

In a SFN, all transmitters are synchronously modulated with the same signal and radiate on the same frequency. Due to the multi-path capability of the multi-carrier transmission system (COFDM) signals from several transmitters arriving at a receiving antenna may contribute constructively to the total wanted signal.

However, the limiting effect of the SFN technique is the so-called self-interference of the network. If signals from far distant transmitters are delayed more than allowed by the guard interval they behave as noise-like interfering signals rather than as wanted signals. The strength of such signals depends on the propagation conditions, which will vary with time. The self-interference of an SFN for a given transmitter spacing is reduced by selecting a large guard interval. It should be noted that the impact of delayed signals outside the guard interval may depend on receiver design. As an empirical rule, to successfully reduce self-interference to an acceptable value the guard interval time should allow a radio signal to propagate over the distance between two transmitters of the network. In order to keep the redundancy due to the guard interval down to a reasonably low value (25 %), the useful symbol length has also to be large given the transmitter spacing. On the other hand a smaller guard interval would lead to a higher number of transmitters. The SFN operation is spectral and power efficient as it uses the same frequency channel overall, and increases the coverage or the signal over noise ratio. The price to pay is an accurate time and frequency synchronization all over the coverage area.

9.2.2.2 SFN operation

The same content is received from distribution network and transmitted at the same time by all the transmitters of the SFN, with the same waveform strictly and exactly. Due to the nature of the signals, SFN operation have some constraints recalled below:

- *frequency synchronization*: if f_k denotes the ideal RF position of the k^{th} carrier, then each transmitter should broadcast this k^{th} carrier at $f_k \pm (\Delta f / 1\,000)$;

- *time synchronization*: the Guard Time has been introduced to cope with the different echoes in terrestrial channels, and also for SFN purpose. Thus, to avoid Guard Time wasting, a strong synchronization constraints must be put on the transmitters : a few µs difference should be the maximum (less than 10 % of the Guard Time).

The different constraints impose an absolute time reference like the GPS for all transmitters, and the transmission of start signal ("top") to allow all transmission at the same time. This is performed through the Mega Frame in DVB-T/H and through the SH Frame in DVB-SH.

9.2.2.3 SHIP solution

The general architecture of the SFN synchronization scheme is provided here in figure 9.1.

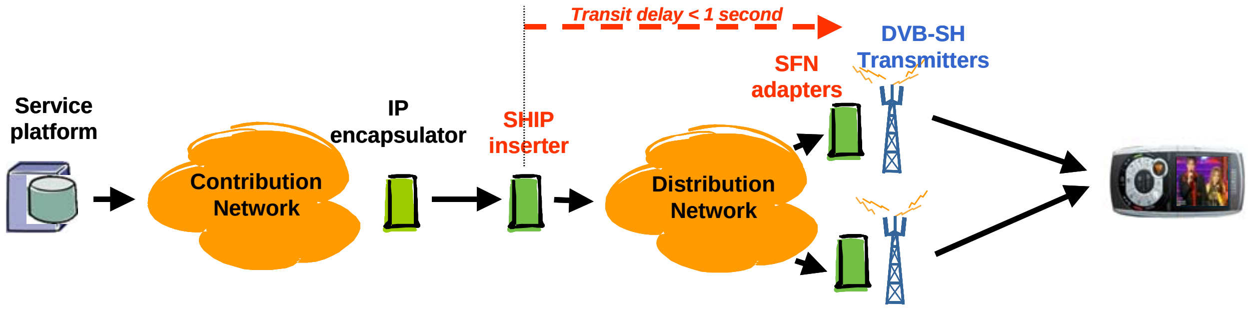


Figure 9.1: SFN synchronization architecture

Distribution network characteristics:

- heterogeneous network: satellite, terrestrial;
- differential transit delay performance from one path to another.

Synchronization principles:

- SH Frame Information Packet inserter: insertion of a GPS based timestamp (±0,1 µs accuracy) in the SH-FRAME indicating the transmission time of the beginning of the next SH Frame;
- SFN adapters in the transmitters (repeaters): buffering of incoming MPEG-TS packets and transmission of SH Frame aligned with GPS relative time stamp.

The definition and specification of the SHIP is provided in EN 302 583 [1].

The complete mechanism is described in clause 7.6.1.2.

9.2.3 Synchronization of Satellite and Terrestrial repeaters for the Common Content

9.2.3.1 SH-B and SH-A/MFN cases

In this configuration, content is broadcast on two separate carriers, each having dedicated modulation schemes. Two demodulators are required in the receiver; at this stage, no specific time or frequency synchronization is required. Each demodulator outputs a time de-interleaved SH Frame, one for the satellite path and one for the terrestrial path. Typical combination scheme consists in combining content prior to FEC decoding (see clause 7.3.2.3). It benefits from the FEC code combining performance and provides straightforward seamless reception. This requires time synchronization of both SH Frames, and controlled difference between symbol rates.

DVB-SH framing provides TDM and OFDM SH Frames of identical period; this eases the synchronization process in the receiver, as it provides simple repetitive and predictable processing. At network level, only relative delays have to be handled. Typically, the satellite component has the longest delay: it is the combination of the transmission path (about 250 ms), plus the length of the Physical Layer interleaver (potentially up to several seconds). Thus, at terrestrial repeater level, these satellite delays must be compensated to minimize the level of buffering memory with the receiver. Residual variation delay is less than a few milliseconds, which much less that the size of the memory required for physical layer time de-interleavers. Synchronization between transmit signals (satellite and terrestrial repeaters) is performed using the SHIP information, in relation with configurable geographical-dependant information in each repeater site. In the receiver, time synchronization between both components is ensured by the SH-Frame (Signalling Field for the TDM component, and OFDM frame for the OFDM component).

9.2.3.2 SH-A case

9.2.3.2.1 Principle

The global satellite coverage and the geographically distributed overlap of hybrid satellite/terrestrial zones require specific dispositions to ensure deriving maximum benefit of SFN capabilities to mobile broadcast services networks; this involves time phase and frequency shift control over the several possible paths used to transport broadcast signals to the User Equipment.

a) Satellite-originating broadcast signals are everywhere pregnant and hence constitute a de facto reference to which the terrestrial segment must be adjusted to ensure the signals transmitted over the different paths arriving at the receiver should be coherent in time, and frequency.

b) DVB-SH is the reference ETSI standard applicable to hybrid satellite/terrestrial coverage architectures; its waveform definition [1] provides through the SH frame a "container" structure for SFN synchronization data (SHIP), adapted from the DVBT/H standards.

c) Each hybrid satellite/terrestrial network requires the elaboration of customized SFN synchronization data, that are dependent on the actual specifics of the satellite orbital behaviour, which must distinguish the following cases:

   - geostationary orbit (GEO);

   - near-Geostationary orbit.

d) Relative Time phase and frequency shift variations between the satellite direct path and the indirect repeated last mile path affect the architecture performances and engineering optimization (cell size, etc.) of the ground network, and must be tuned carefully.

Hybrid SFN engineering considerations lead to ensure that:

- Time phase performances synchronization between direct and indirect path of some fraction (typically a few percents) of the Guard Interval duration.

- Frequency synchronization performances between direct and indirect paths of some fraction (typically a few thousandths) of subcarrier spacing (the which e.g. is 1 709 Hz for a 7 MHz band and a 4 K FFT size).

This requires that all equipments be synchronized onto a unique and stable clock reference and/or a unique and stable frequency reference. This is ensured by the synchronization of all terrestrial equipment (IPE/ Satellite gateway modulation stages/Repeater including frequency Up and down converters) onto the GPS 1 pps clock and/or 10 MHz frequency references.

However, the limiting effect of the Hybrid SFN technique is the so-called self-interference of the hybrid network. If the relative delay between the signal arriving from the satellite and the terrestrial repeater are delayed more than the amount allowed by the guard interval, SFN self interference occurs. The strength of such signals depends on the propagation conditions, which will vary with time. The self-interference of a Hybrid SFN for a given transmitter is reduced by selecting a large enough guard interval, which would reduce the useful multiplex througput. It should be noted that the performance impact of signals characterized by a differential delay exceeding the guard interval may depend on receiver design and on relative signals powers. In a hybrid network, with a GEO or near GEO satellite, the hybrid SFN zone around a given transmitter is an ellipse, of which one of the focuses is located at the transmitter; This ellipse offers a limited Hybrid SFN constructive zone in the south direction (in the north hemisphere) towards satellite position. When necessary, the size of the ellipse can be increased by advancing transmitter time of emission of certain amount of time. For convenience, this amount of time can be an integer number (N) of Guard Intervals. Typically one Guard Interval advance is selected.

In that case, the first bullet point of previous paragraph must be reformulated as:

a) time phase performances synchronization between direct and indirect path of some fraction (typically a few percents) of the Guard Interval duration (Modulo N GI).

9.2.3.2.2 Application with near GEO

With active house keeping, GEO satellite orbit should be kept within less than ±0,1° in both directions. The near GEO satellites may be inclined by as much as some few percents (typically up to (±)5°/6°. Though, correction procedures can be simplified for GEO satellite, we propose the analysis for the worst case, that is to say near GEO orbits.

9.2.3.2.2.1 Time delay

The direct path through this satellite experiences changes in the absolute time delay involving four location parameters:

a) Satellite gateway location.

b) Cell location in the hybrid satellite/terrestrial coverage zone.

c) Ground repeater location within the terrestrial cell.

d) UE location in the cell.

Maximum cell range resulting from CGC engineering results from ground network planning architecture and dimensioning choices. The following schematic displays the several path components contributing to relative delays between the direct satellite broadcast path and the indirect repeated last mile path. The distribution network for repeater content can be of different nature as for DVB-T/H networks. In the shown case, distribution network is using satellite S2.

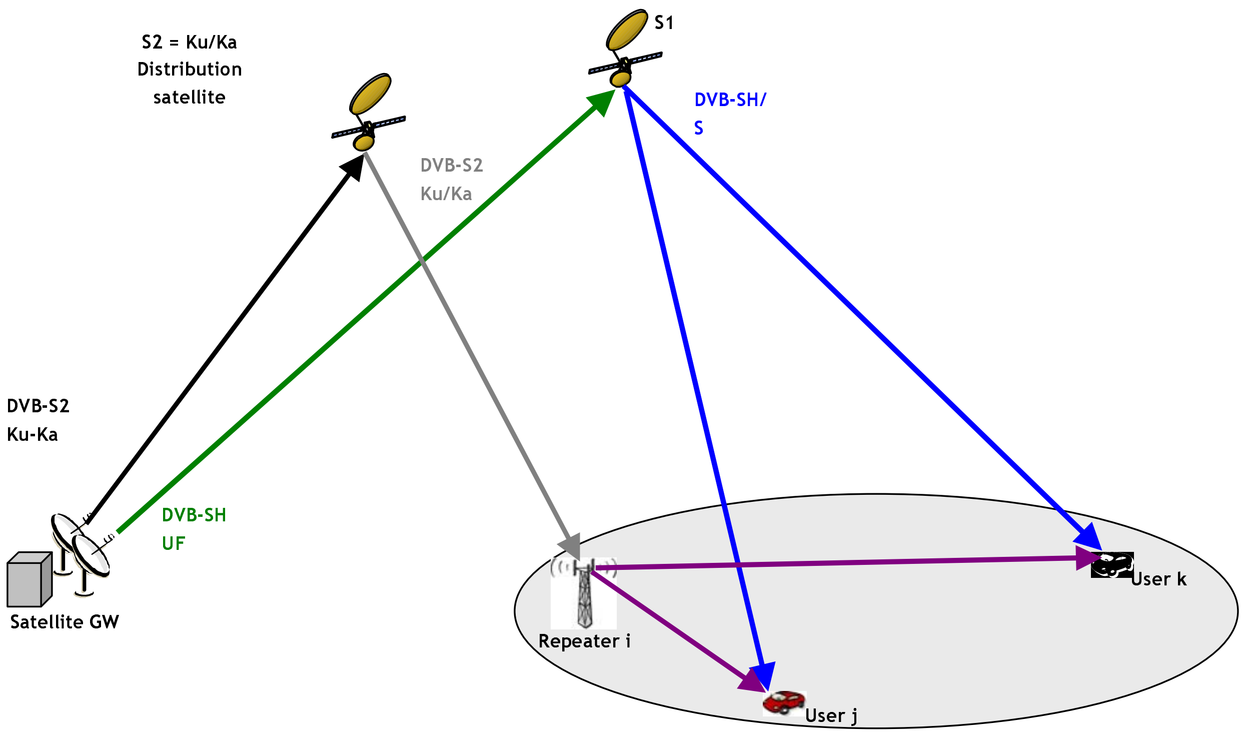


Figure 9.2: Overall system

The satellite apparent motion cause propagation time variation on the gateway-satellite path and on the satellite-repeater path (and on satellite to any location in the cell path). Depending on the satellite orbital characteristics and coverage considered in the hypothesis, the time delay variations may range to about ±4 ms in the whole coverage. A pre-compensation of the time delay variation must be done at the Gateway location. It is ensured by comparing the locally received signal with a local reference delayed by twice the gateway to satellite reference position path duration. This pre-compensation is generally sufficient for GEO satellites. For near GEO satellites, this pre-compensation reduces significantly the time delay variations on the coverage, however a differential delay of up to about ±400 µs remains between the direct and indirect paths. These differential delay variations must be compensated for at the level of each repeater using satellite ephemeris position to reach 10 % of the guard time (around 11 µs in QPSK 5 MHz 2k GI=1/4). The technique used to convey this information is out of the scope of the present document, however data could be sent by in-band techniques like SHIP private section.

In a near GEO case with 6° inclination, the following absolute delay variations can be observed, as shown in following figure, during 24 hours. Two repeaters at different locations are used.

- DT1: delay variation: satellite to repeater1 minus nominal value.
- DT2: delay variation: satellite to repeater 2 minus nominal value.
- DTG: delay variation: satellite to Gateway minus nominal value.

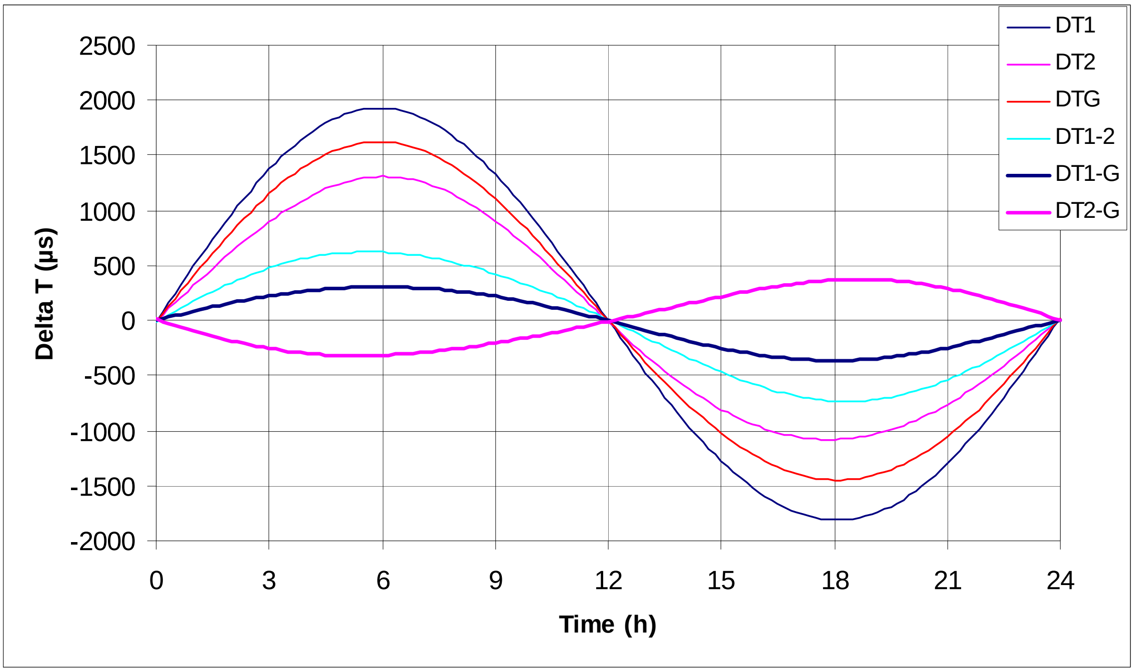


Figure 9.3: Examples of absolute delay variations (one way)

The differential following delay variations are shown in figure 9.4.

- DT1-G = DT1-DTG.
- DT2-G = DT2-DTG.
- DT1-2 = DT1-DT2.

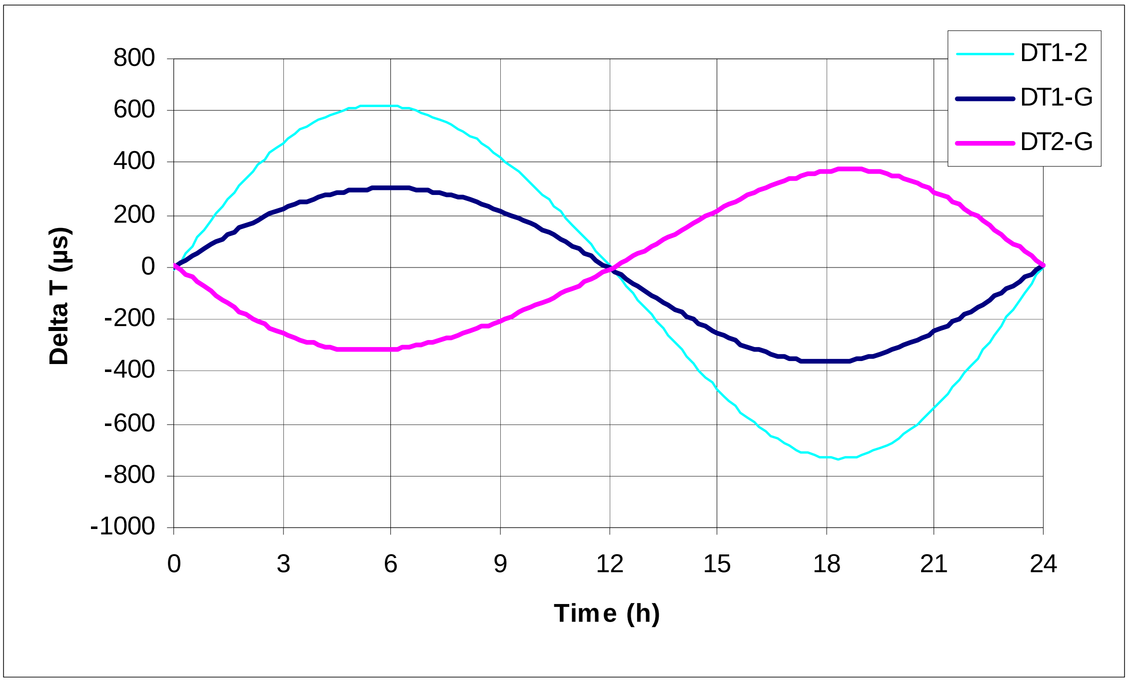


Figure 9.4: Examples of differential delay variations

9.2.3.2.2.2 Frequency shift

In parallel to the time variations, the satellite apparent motion results in Doppler shifts of the frequency by about a few hundred Hz (typical value circa ±300 Hz) which also require compensation. The absolute Doppler variations are shown here below.

- DF1: Delta F satellite/repeater 1.
- DF2: Delta F satellite/repeater 2.
- DFG: Delta F satellite/Gateway.

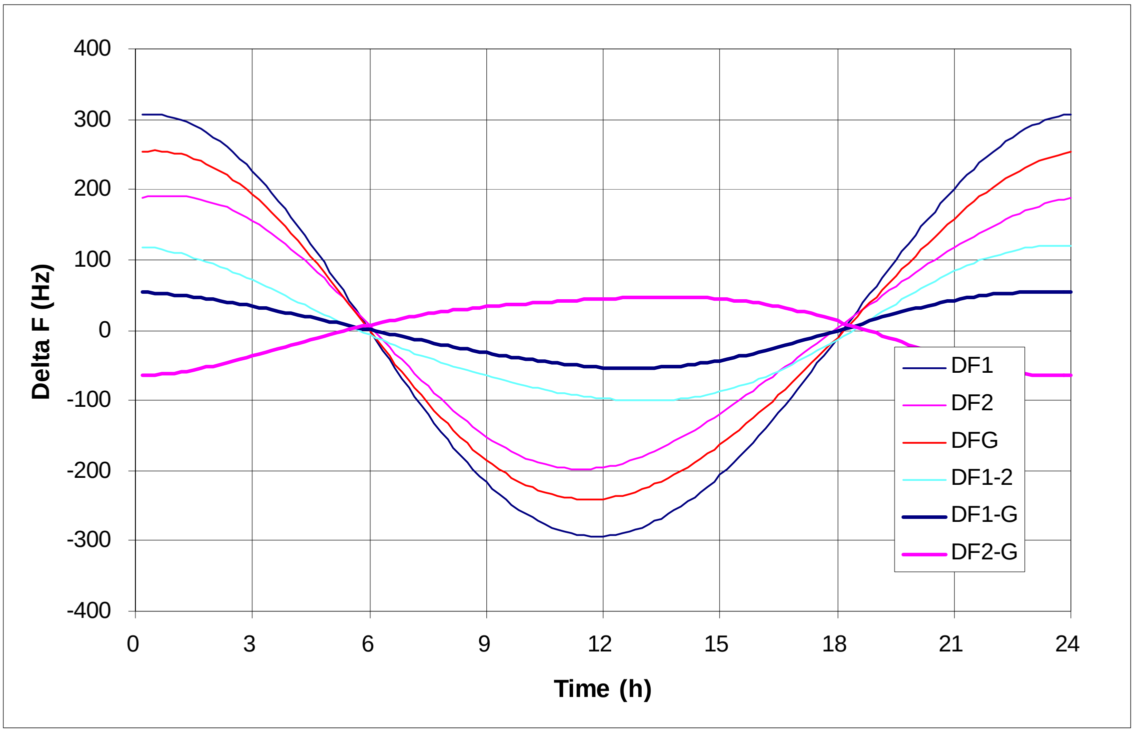


Figure 9.5: Examples of absolute Doppler variations (2,2 GHz on downlink)

A pre-compensation of the Doppler shifts must be done at the Gateway location. It is ensured by comparing the locally received signal with a local reference. This pre-compensation is generally sufficient for GEO satellites. For near GEO satellites, this pre-compensation reduces significantly the Doppler shifts on the whole coverage, however it remains a differential delay up to about ±100 Hz. These differential delay variations must be compensated for at the level of each repeater using satellite ephemeris velocity data the same way the time compensation is achieved.

9.2.3.2.3 Architecture

The satellite gateway provides signal processing for the satellite direct and indirect paths, and elaborates all SFN-related data required by the CGC elements (repeaters). The main services offered by the gateway are:

- transmission of data received from the IP Encapsulator towards the User Equipment via the satellite (S-Band), and towards the terrestrial repeaters through a commercial distribution satellite (Ku/Ka-Band);
- global SFN control and Time and frequency shifts corrections computation in a synchronized way between the satellite path and the repeater path.

The same MPEG TS are broadcast by S-band satellite (S1) and by a commercial satellite (S2). The MPEG-TS transmitted by the gateway to S1 will be DVB-SH modulated. This DVB-SH modulated signal, after transposition by S1 to a S-band carrier frequency f_o, is directly received by the User Terminals (direct path) under S1 coverage. The same MPEG-TS transmitted by the gateway to S2 will be DVB-S2 modulated in the Ku/Ka-band. S2 retransmits this signal in the Ku/Ka-band. The repeaters under the S2 coverage then demodulate the received DVB-S2 signal, and retransmit the extracted MPEG-TS using the same DVB-SH modulation scheme at the same previous S-band f_o frequency: the terrestrial DVB-SH modulated signal are then be also received by the User Terminal.

The SFN time and frequency synchronization processing in the gateway as illustrated by figure 9.6 consists in:

- comparing actual received S band signal with an internal reference;
- compensating time and frequency in the DVB-SH modulator sending to S1 ensuring perfect time and frequency compensation for the gateway location; and
- delivering to the repeaters through adequate means the relevant ephemeris information (one means could be to introduce such ephemeris into SHIP private functions that is sent over the global MPEG2 TS) for correcting remaining differential time variation and frequency shift, if required, at the repeater level.

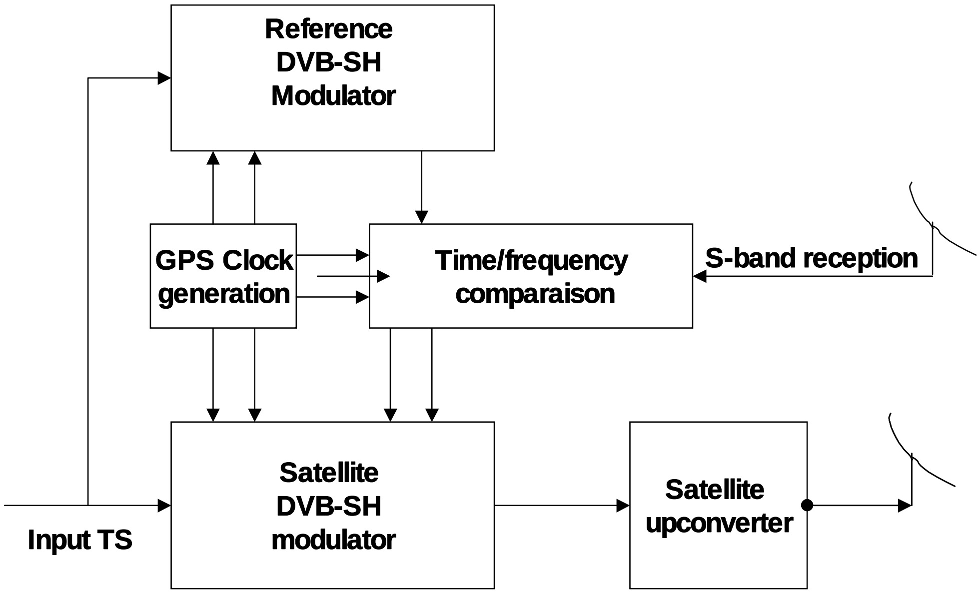


Figure 9.6: Hybrid SFN satellite gateway modulator

9.3 Signalling

As presented in clause 4, different signalling information are used in a DVB-SH system:

- physical layer: SHIP (all modulations), TPS (OFDM) and Signalling Field (TDM) are used;
- section: PSI signalling;
- link: MPE and MPE-IFEC;
- higher: ESG, files, etc.

9.3.1 SHIP, TPS and SF

Relevance of SHIP, TPS and SF signalling are detailed in clause 7. Procedures for signalling SHIP, in particular in complex situations, is detailed in [21].

9.3.2 MPE and MPE-IFEC

Relevance of MPE and MPE-IFEC signalling are detailed in clause 6.

9.3.3 PSI/SI case

Precise specification of the PSI/SI and SHIP can be found in [21]. Precise specification of the ESG can be found in [37].

The Service Information (SI), is not protected by the MPE-FEC so SI reception quality will heavily depend on burst and error rates witnessed before MPE-IFEC decoding.

The main PSI/SI needed by an IPDC/DVB-H terminal are:

- **NIT** Network Information Table;
- **INT** IP/MAC Notification Table;
- **PAT** Program Association Table;
- **PMT** Program Map Table; and
- **SDT** Service Description Table.

Table 9.1: PSI/SI tables reception performance

Table	Average size (bytes)	Size (TP)	Repetition period (ms)	Reason for acceptable performance quality	Remarks
NIT	3 000 to 6 000	20 to 37	10 000	Static over the network	Size depends on number of TS and cells
INT	500 to 13 000	4 to 72	30 000	Static over TS (HF and NHF)	Size depends on neighbour cell signalling policy and scenario
PAT	24 to 64	1	100	Static over the TS Repetition period	Size depends on scenario (SH-A/B)
PMT	140 to 1 400	1 to 8	100	May be variable depending on the location (HF and NHF) Repetition period	Size depends on scenario (SH-A/B) and local content policy
SDT	50 to 450	3	2 000	Static over ST (HF and NHF)	Size depends on scenario (SH-A/B) and local content policy (service_availability)

Concerning PAT/PMT, the size of the respective tables and the repetition rate (10 times per second) are such that robustness is not a problem, even in ITS channels where TS PER of 10 % are typical. The PAT is not variable but the PMT may vary depending on the location (this is a DVB-SH particularity). In order for the terminal to capture local variations, especially when local content is inserted, a specific procedure is described within the [21]: the terminal has to buffer for SH-frame durations before deciding on the actual value of the PSI with specific rules for erasing the incorrect values. Case of partial losses that can induce incorrect conclusions are managed by integrating over several SH-frames.

SDT, NIT and INT tables are made quasi-static and constant over the TS so the receiver does not have to actually receive them each time it is switched on or each time it performs handover, provided the tables are stored in the receiver. So *SI tables do not depend on geographical location within a country:*

- This case is naturally applicable for the hybrid frequency and is similar to the DVB-H case. In this case, NIT, INT and SDT are defined to cover a full spot beam so they could be stored in the receiver and updated only when they have changed. Any change of content would then be signalled in the PMT so a receiver would immediately be aware of this and could start downloading the updated tables. Robustness will not be a problem in this case. In this case, a receiver in SI acquisition mode would likely be tune on during the acquisition time in order to optimize the reception probability. In typical ITS environment, the average TP error rate is around 10 % to 15 % but these errors are distributed over 20 % of bursts (or 80 % of burst are error free). By listening continuously to burst, the receiver will quickly acquire the table since the repetition interval is in the order of a few seconds.

- This case is also applicable for non hybrid frequencies, a situation described by [21] as "content regionalization". If the SI were variables, it would have been necessary to actually receive these tables more frequently. However, due to the design choices made in [21], unique SI are maintained throughout the TS, wherever the terminal location. This will ease a lot the reception quality of INT, NIT and SDT.

In conclusion, INT and SDT are unique within the TS whereas the NIT is unique throughout the network and will not change (except when roaming between network, case not described here).

If one can assume that the receiver accesses the SI each time it is switched on and in connection with each handover, then this SI has to be receivable in a robust way without too much delay (preferably before the next burst). Due to the fact that the SI is tailor-made for the specific area, the size of the SI tables can be made highly limited in size. It is then possible to repeat the tables much more frequently than the required minimum (every 2 s form the SDT, every 10 s for NIT, every 30 s for INT). If the table sizes are small, they could be repeated, e.g. every second, and the probability of correct reception would increase dramatically thanks to the redundancy provided by the repetitions. This optimization procedure is also detailed in EN 302 304 [3], clause 9.3 and PSI/SI tailoring are detailed in the [21].

The PSI/SI to be used for DVB-SH will most probably be quasi-static over the hybrid frequency, since all content related information is sent over IP. This information can, therefore, in principle, be stored in the receiver, which will make access much less time critical. By the redundancy of the repetition of the PS/SI tables, correct reception is guaranteed, sooner or later in all reception conditions. In cases where fast access of SI, not previously stored, is required, this can be accomplished over non-hybrid frequency by increasing the repetition rate of the NIT, INT and SDT. With a repetition rate of, for example, one second, correct reception of SI tables can be obtained within a few seconds, also in very bad channel conditions.

9.3.4 ESG case

ESG are protected by MPE-IFEC, files are protected by IPDC content delivery mechanism. Specific mechanisms described in [ESGoverSHIG] manage regionalizaton of ESG and their protection by means of repetition (carrousel).

9.4 Considerations on the use of repeaters and their feeder links

This clause contains guidelines and recommendations for the use of repeaters (Complementary Ground Components) in DVB-SH networks.

9.4.1 TR(a) On-channel regenerative repeaters

The TR(a) CGC on-channel regenerative repeaters are broadcast infrastructure transmitters which complement reception in areas where satellite reception is difficult especially in urban areas. They are basically designed as any broadcast transmitters. Local content insertion is possible.

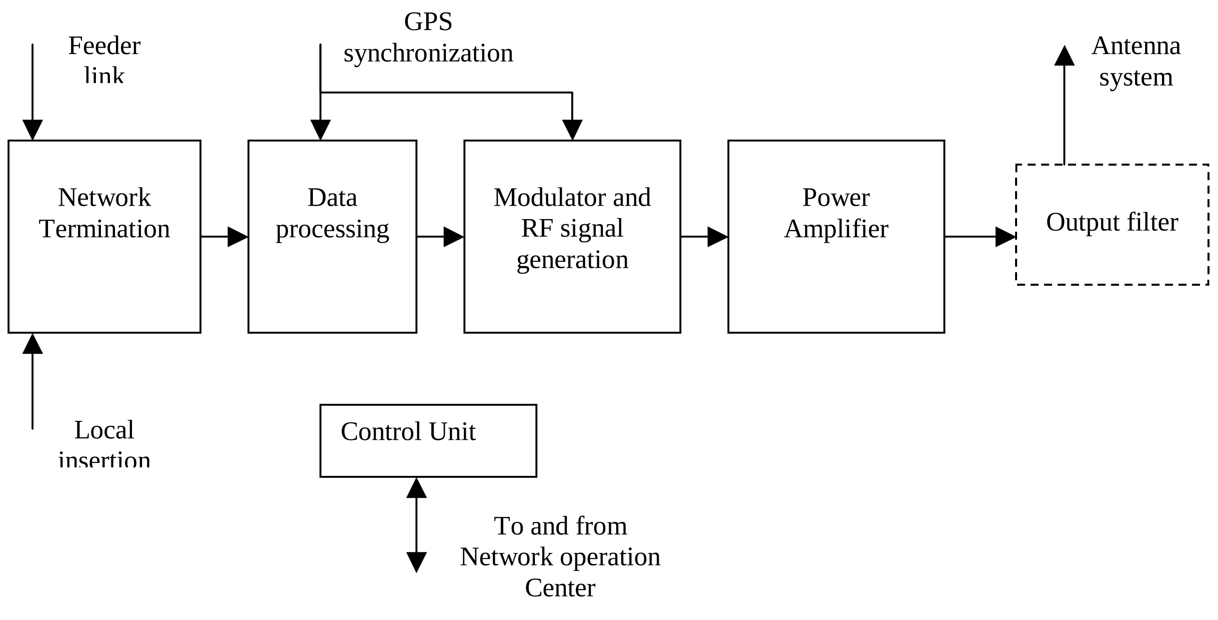


Figure 9.7: TR(a) functional block diagram

The TR(a) repeaters are composed of the main following functional blocks:

- A **network termination** function that interfaces with at least one feeder link. This feeder link ensures the backhauling function and brings the MPEG TS flows to the repeater in such a way that the SFN synchronization remains possible. The backhauling can be of any appropriate kind providing that the Quality of Service is compatible with the overall target (typically ten times better than the repeater to subscriber QoS). This aspect of feeder link QoS is not part of these guidelines. One of the most appropriate backhauling means is via a Ku or Ka-band geo-stationary satellite transponder (DVB-S or DVB-S2) thus offering the advantage of ensuring a wide area coverage of any number of repeaters. Other backhauling means are: Optical fibre network, microwave link, high speed DSL, etc. A mix of any compatible means is possible provided that the overall latency remains compatible with the SFN alignment. The network termination function ensures the interfacing with the backhauling network at the PHY and MAC levels. For adequate network management the network termination also ensures the monitoring of the quality of the incoming signal and provides the information to the control unit. A repeater can include several types of interfaces for either ensuring redundancy between feeder links or for the insertion of local contents or both.

- A **data processing function** that ensures the extraction of the selected bouquet(s) of MPEG TS from the backhauling flow(s). This data processing function might also be able to re-multiplex the MPEG TS in order to optimize the use of the feeder link. In particular local content can be extracted from the feeder link to generate the local multiplex. In that condition, only the envelope of all local contents is sent rather than the sum of all local multiplexes. This data processing function also achieves the SFN synchronization after extraction of the mandatory parameters of the SHIP packets. The time reference for the SFN synchronization is typically a 1 pps GPS pulse (or Galileo in the future). In case of loss of the time reference signal the TR(a) should stop transmitting (no radiated signal) or hold the transmission as long as the time shift can be guaranteed to be lower than 25 % of the guard interval and then stop transmitting. The extraction of the optional SHIP parameters is advisable. Note that the SFN synchronization as derived from the DVB-T/H assumes that the S band satellite is geo-stationary and that its residual relative movement can be compensated by the broadcast head end.

- A **control unit** function or **Operation and Maintenance** function that monitors and controls the repeater and is able to be linked to a centralized network management. If not linked to such a network operation centre the control function should be able to record the key parameters in such a way that the history can be later recovered for analysis. If no link to the network operation centre is possible then the use of optional SHIP parameters could be envisaged.

- A **modulator** function that generates the COFDM waveform followed by an up-converter or directly at RF. The RF frequency must also be synchronized by means of a GPS (Galileo) signal to ensure the correct SFN function.

- A **power amplifier** or set of power amplifiers to amplify the carrier signal up to the desired level. The output power limit is determined by the applicable safety regulation on one hand and to the radio planning target on the other hand.

- An optional **output filter** in order to comply with the output spectrum limits.

NOTE 1: For example, EN 302 574-1 [i.36] in the case of the S-UMTS band.

- The **antenna system** that radiates the RF signal.

Typical TR(a) parameters

- Feeder interfaces:
  - L-band interface from Ku or Ka-band LNB (950 MHz to 2 150 MHz).
  - ASI.
  - Ethernet (10/100 baseT or Gigabit Ethernet, optical or electrical).
- Synchronization:
  - GPS receiver.

- Output power (S-UMTS case):
  - More than 45 dBm per carrier per cell site.
- Other characteristics:
  - MER: > 23 dB.

NOTE 2: The MER (Modulation Error Ratio) value includes all the signal impairments as measured at the antenna level. The indicated value assumes that there is no non-regenerative means between the repeater and the subscriber terminal or that a non-regenerative devices inserted has no significant effect on the MER value seen by the terminal.

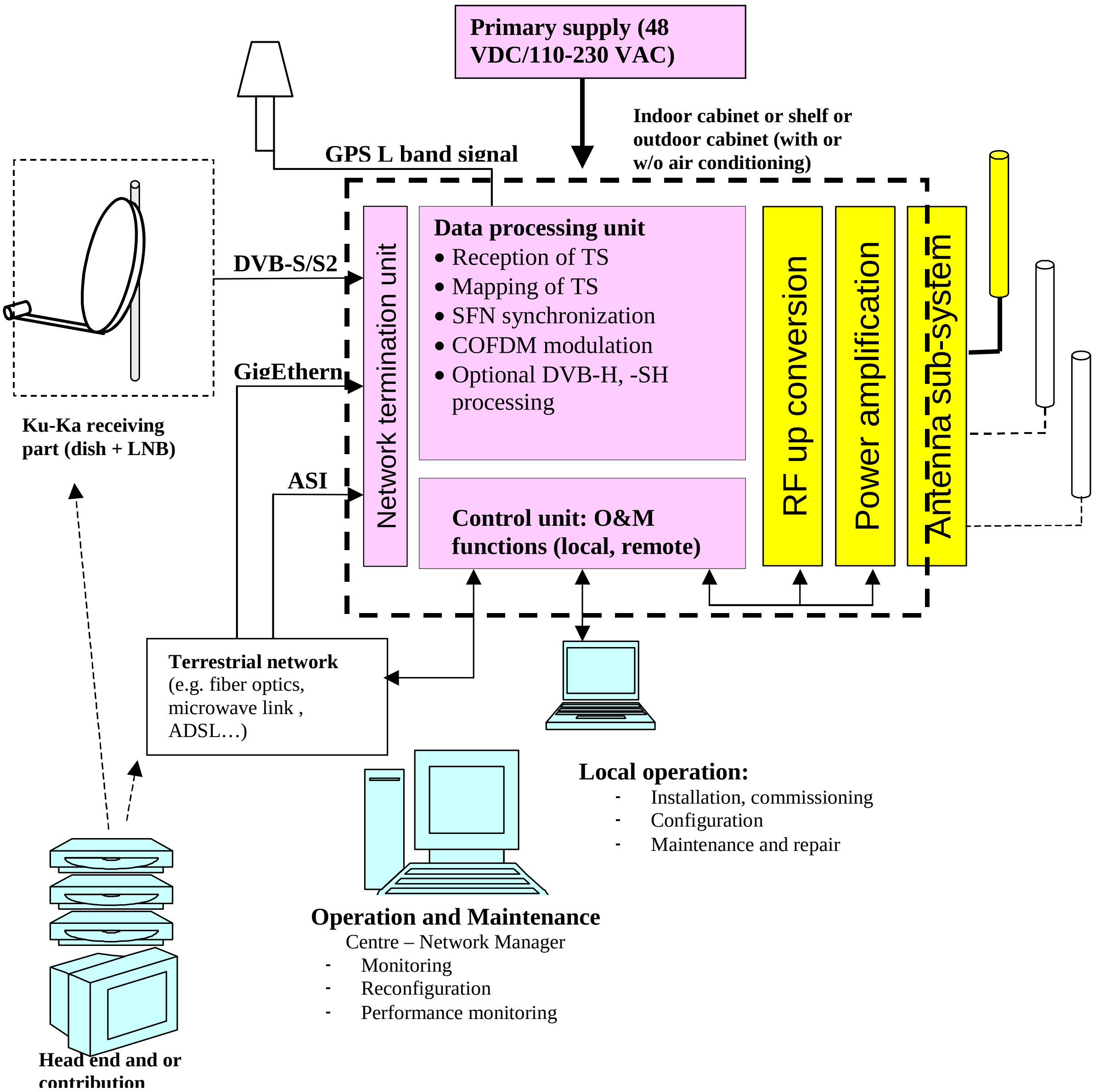


Figure 9.8: Example of implementation diagram of TR(a)

9.4.2 TR(b) Non-regenerative gap-fillers

9.4.2.1 Generalities

The non-regenerative gap-fillers (or simply gap-fillers) are personal or public devices of limited coverage providing local on-frequency re-transmission and/or frequency conversion; typical application is indoor enhancement under satellite coverage; no local content insertion is possible. The main benefits of the gap-fillers, when compared to regenerative repeaters, are easier deployment and lower cost. The delay induced by the whole process of reception, amplification and transmission must be substantially shorter than the guard interval of the used DVB-SH mode (a typical delay is 5 µs), so that a receiver receiving both signal from a gap filler and signal from a satellite or a regenerative repeater does not have to deal with interference but with a constructive addition of signals. Two difficulties arise when using a non-regenerative gap filer is the **filtering** and the difficulty to ensure the **remote management** of the gap-fillers when the deployment and operation have to be kept simple and economical.

9.4.2.2 Filtering issues in non-regenerative gap-fillers

The main obstacle in the deployment of on-channel gap-fillers is a problem inherent to its logic. The transmitted signal may be fed back to the input of the gap-fillers, thus creating a feedback-loop which generates two kinds of problems: ripple in the transfer function of the device, and, at worst, instability of the device.

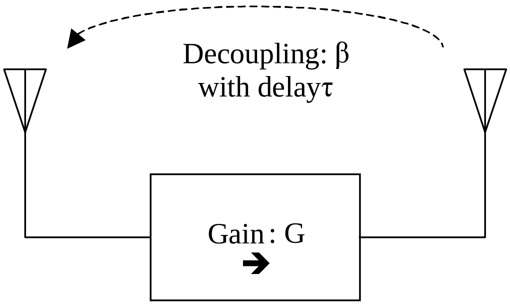


Figure 9.9: Illustration of an on-channel gap-filler coupling effect

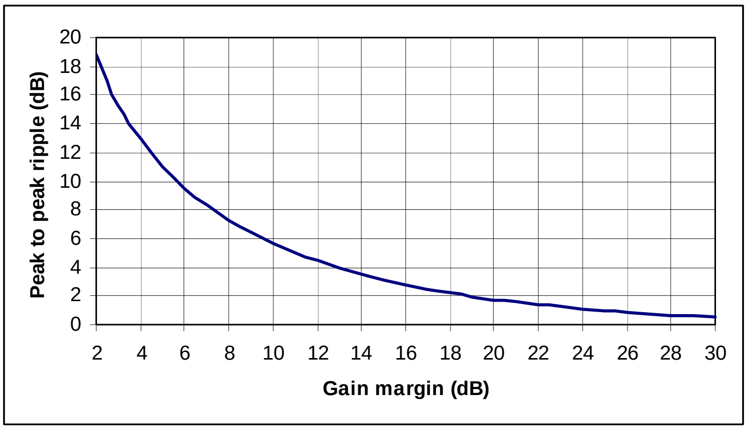


Figure 9.10: Peak to peak ripple as a function of gain margin

The amplitude ripple value in dB is given by:

$$R = 20\log\left(\frac{1+10^{-\frac{M}{20}}}{1-10^{-\frac{M}{20}}}\right)$$, M: margin in dB = -20 logGβ (with βG < 1)

There is also a group delay distortion and ripple but of negligible impact in OFDM: if the relative delay between input signal and re-injected loop-back signal is τ then the time delay variation in function of the frequency *f* is given by:

$$\theta = -\frac{d\varphi}{d\omega} = \tau.G\beta\frac{G\beta - \cos 2\pi f\tau}{1 + G^2\beta^2 - 2G\beta\cos 2\pi f\tau}\ .$$

Common values are 20logG around 70 dB and -20logβ greater than 80 dB.

Some on-channel gap-fillers include internal echo-cancellers in their devices. This element adds an internal decoupling to the external decoupling, thus allowing for a higher total effective decoupling. Improvements of more than 15 dB have been reported. The use of echo-cancellers in on-channel non-regenerative repeaters thus brings two benefits:

- the deployment of repeaters in sites where it would otherwise reveal itself unfeasible; and
- a reduction of in-band distortion thus improving quality of the received signal in the area covered by the repeater.

If no filtering is included in the gap-filler transfer function then other channels and also close or adjacent channels belonging to other services will be re-transmitted thus impacting the coverage of these other services. Proper site spectrum survey is thus necessary. In certain cases however this on the contrary can be advantageously used in the sharing of gap filling infrastructures between different services or applications. Typical examples are subways, tunnels and large indoor public areas (commercial malls, airports).

Regarding the gap-filler itself it is also important to note that protection filtering might be required in order to protect it against strong nearby sources that could drive it into non-linear behaviour and thus non-linear distortion and inter-modulation generation.

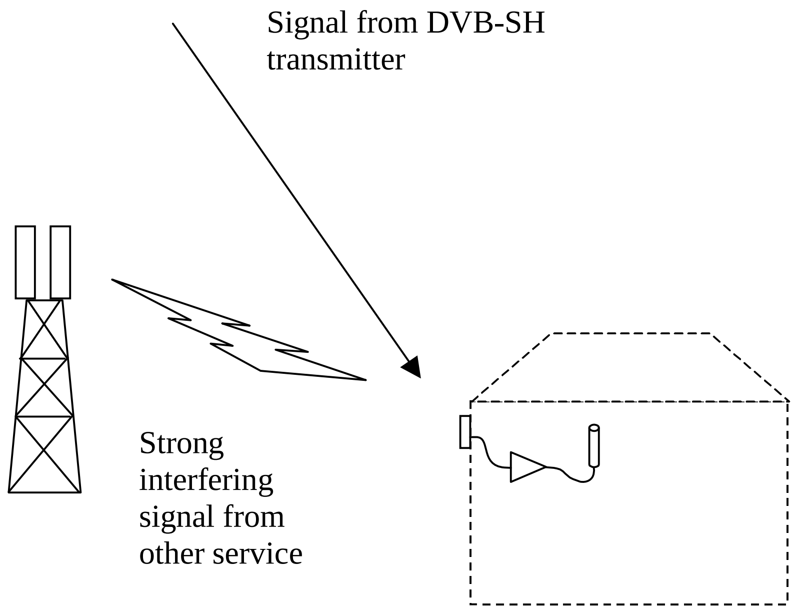


Figure 9.11: Filtering issues in non-regenerative gap-fillers

Recent gap-filler designs include efficient filtering in order to properly select the channel(s) to be retransmitted but this does not always overcome the issue of input overloading by strong out of band signal.

9.4.2.3 Frequency synchronized transposing non-regenerative gap-fillers

Such gap-fillers receive a signal on frequency F1 and re-transmit it on frequency F2 without any SFN resynchronization apart from the F2 carrier frequency. Demodulation at F1 then modulation at F2 of the received signal is not recommended so as to maintain the through time latency as low as possible. The carrier frequency F2 has to be synchronized to ensure the SFN between all the transposing gap-fillers this can be efficiently done by using a GPS locked frequency reference.

9.4.3 TR(c) Mobile transmitters

These transmitters are used to build "moving complementary infrastructures" on-board public transportation such as trains. Depending on waveform configuration and radio frequency planning, local insertion may be possible. Depending on the application both types of repeaters/gap-fillers can be employed for mobile coverage: regenerative and non-regenerative transmitters. In all cases it must be kept in mind that temporary interference with fixed repeaters might occur.

9.4.3.1 Regenerative TR (c)

Obviously if local insertion is required then regenerative TR(a) type is needed. A typical service that could be offered e.g. on-board trains could be pre-recorded programs locally inserted. That kind of service would be similar to what is offered now on-board planes. The backhauling of the main content is the most critical issue since it requires a relatively high data rate link between the vehicle and the fixed network. However depending on the desired capacity this will be easily achievable and actually achieved for other data services (WiMAX). Because there is no need for SFN with fixed sources it is also possible in that particular application and in the case of a hybrid network to recover the signal from the S-band satellite preferably by means of a relatively high gain antenna (6 dBi to 8 dBi) on the roof of the mobile and to demodulate this signal before re-modulation on the same frequency. The re-generated signal would then be redistributed through a radiating distributed local installation at a much higher level compared to the residual signal coming direct from the satellite.

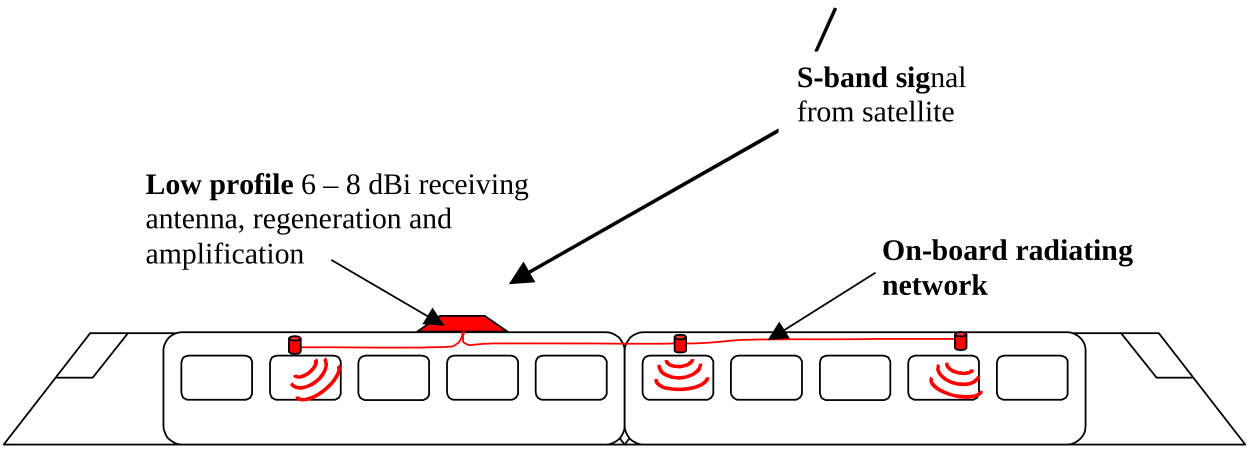


Figure 9.12: Example of possible TR(c) implementation on-board train

9.4.3.2 Non-regenerative TR(c)

If no local insertion is required then a non-regenerative solution could be envisaged. However attention has to be paid in the link budget and final C/N value depending on the main receiving antenna G/T. All the other drawbacks of non regenerative on-channel solutions (see TR(b)) apply but indeed in that case a shared installation with 3G networks could be contemplated. A transposing non-regenerating solution would help in solving the isolation constraint of the on-channel gap-fillers but the frequency availability or local usage might not be possible.

10 Reference Terminals

10.1 Top-level design considerations

10.1.1 Terminal categories

To address a wide range of market sectors, DVB-SH allows a large freedom in terminal implementations. Three main categories can be identified and are considered hereafter :

- category 1: car-mounted terminals (also called "vehicular");

- category 2: portable TV devices with 2 sub-categories:
  - 2a: large screen (≥ 10") portable devices, battery or mains powered;
  - 2b: pocketable (handheld) TV devices, mainly battery powered;
- category 3: handheld terminals with an embedded cellular telecom modem (or "convergence" terminal).

Car-mounted terminals can especially benefit from the nation-wide coverage and allow designers to include many of the advanced features of DVB-SH (Seamless complementary satellite/CGC coverage, SH-B configuration, high-order modulations, long time interleaver, etc.).

Portable TV devices with large screen are mainly stationary during reception. They could have an attached antenna, but also detachable antenna accessories. This latter case allows for high reception performance thanks to optimization of the antenna position by the user (i.e. find a LOS reception from satellite or optimized position for good reception from CGC).

Handheld terminals can especially benefit from the outdoor and indoor coverage in built up areas (similar to 3G coverage), but are more challenging due to their small form factor, the large number of functions to be integrated, limited battery power and coexistence with other active radio functions.

Category 2b has common characteristics with handset. Most often, pocketable terminals embed a large number of multimedia features without necessitating coexistence with radio modems.

Other features which are at the discretion of the manufacturers/markets include:

- antenna dedicated and/or optimized to satellite link;
- number of antennas and branches for diversity gain;
- power of processor embedded in terminal;
- embedded memory for physical layer processing;
- host memory for PVR functions.

10.1.2 Mobility aspects

10.1.2.1 Mobile channels

DVB-SH receiver mobility environments are defined in clause 11.

10.1.2.2 Antenna and diversity considerations

Due to the relative narrow bandwidth of the frequency bands considered (see clause 4.1.4.1 on frequency allocations), antenna performances could be better than that of their wide-band UHF counter part. Of course, this statement is not valid in the case of multi-band antenna (antenna shared between different missions on different bands).

For handheld with integrated linearly polarized antenna, -3 dBi gain can be achieved with omni-directional pattern (for example at 2,2 GHz/30 MHz bandwidth). If the antenna is external, antenna gain can be as high as 0 dBi for linear (0 dBic for circular antennas and omni-directional for both). The polarization choice depends on targeted terminal (optimization for satellite link or not).

For vehicular or external dedicated antenna, 4 dBic gain can be reached at elevation optimized to target satellite orbit and position (for example, antenna for SDARS system). Further details are given in clause 10.4.5.

Antenna diversity reduces the effect of the fast fading. As shown in figure 10.1 in a receiver exploiting antenna diversity, output signals obtained from several antennas are linearly combined using adjustable complex-value weighting factors before being decoded. Implementations may differ:

- by the antenna system's characteristics: number of antennas, relative positions, orientation and characteristics of each antenna (polarization, radiation pattern, etc.);
- by the algorithm used to compute and eventually iteratively adapt the weighting factors.

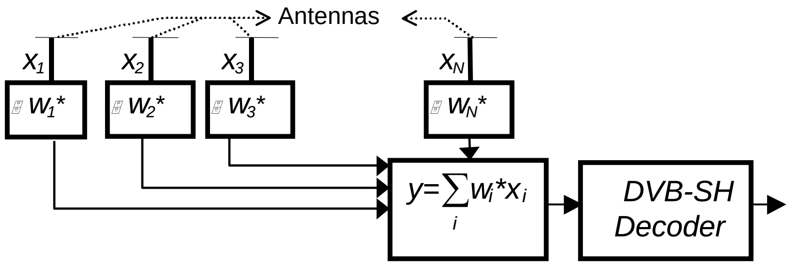


Figure 10.1: Antenna diversity receiver

In mobile reception conditions (especially for Terminal Category 1), antenna diversity gain is expected to permit a reduction of terrestrial repeaters transmit power by up to 6 dB for the same coverage. It should also allow increasing the mobile's maximum speed for correct reception. This result is given considering a multipath Rayleigh channel and no long time interleaver benefit. Tests results do not evidence contradictory effect between diversity and interleaving gains.

For satellite reception, at least 3 dB diversity gain can be achieved in LMS channels. The antenna diversity gain will increase with the mobile speed reduction.

Of course, diversity with higher order (3 or 4) could be envisaged on Terminal Category 1.

In indoor reception (especially for Terminals Categories 2a, 2b and 3) the mobile channel dynamic is typically slower than for outdoor mobile reception. However, considering the fading Rayleigh environment, diversity gain could be as high as 6 dB. Experimental results confirm this level of performance advantages (see clause A.13). It is to be remarked that the measured diversity gain may be impaired by issues like gain unbalance between antennas, antenna fading correlation dependency on terminal antennas respective positions and considered environment.

10.1.3 Service aspects

As for DVB-H, a trade-off has to be made between burst length, time-slicing off time period, power saving effectiveness and access time.

Receivers should minimize "wake up" time (time between RF switch-on and start of demodulation) by implementing various techniques such as keeping and using formerly acquired demodulation parameters, using fine evaluation of the off time period, etc.

As for DVB-H, time slicing allows seamless handover by scanning adjacent cells signals during idle time periods. Another aspect in DVB-SH is the receiver class (memory, see also clause 10.2) and its associated time interleaver depth capability.

Issues to be considered are:

- backward compatibility of receiver class 1 with long interleaver used by the transmitter;
- time slicing burst length/off time period;
- access (or zapping) time.

Coexistence (especially for Category 3 receivers)

For handheld receivers with an embedded cellular telecom modem, DVB-SH reception should not prevent or hinder cellular operations (initial synchronization, roaming, handover, communication). Reciprocally, reception of DVB-SH service should be possible in parallel with cellular operations. However, DVB-SH quality may be degraded by cellular uplink signal (see annex B).

Two receiver states may be considered:

- cellular function in standby, i.e. intermittent uplink transmissions (cell search, handover in mobile conditions): DVB-SH reception if active should provide nominal picture quality; and

- cellular function in active communication, i.e. uplink traffic transmission: DVB-SH reception if active may or may not be displayed, depending on user selection. When DVB-SH service display is switched off during call, demodulation should continue in the background and display should be automatically resumed at the end of the cellular call.

Coexistence conditions described in annex B consider a worst case

- DVB-SH full service while cellular handheld operation: standby and communication;
- low level DVB-SH downlink signal;
- maximum power cellular uplink;
- no DVB-SH time interleaver protection.

10.2 Memory requirements for DVB-SH processing

As shown in figure 10.2 the DVB-SH receiver baseband processing shows a similar high-level architecture as that of the DVB-H counterpart, namely analog-to-digital converter, OFDM demodulator, de-interleaver, Forward Error Correction (FEC), demultiplexing, multi protocol de-capsulation, MPE-FEC, IP filter and terminal host interfacing. New functions are mainly the Turbo decoder replacing the convolutional one of DVB-H and the physical layer time de-interleaver.

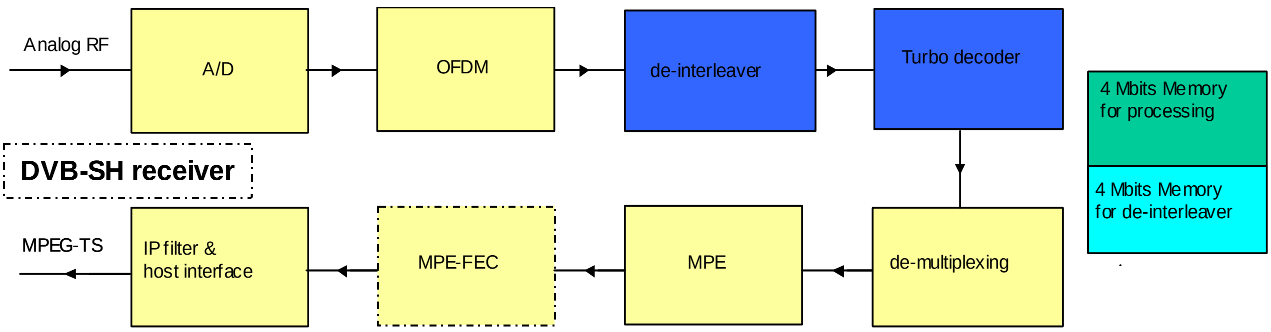


Figure 10.2

The Turbo decoder needs extra hardware which is available as an off-the-shelf building block from 3G technology development. The main challenge for implementing the Physical Layer Interleaver is its memory requirement which is dependent to the DVB-SH Receiver Class:

class 1

Receivers class 1 are required to handle an interleaver profile comprising one full SH-frame with 816 CU. Using the convolutional interleaver approach which typically halves the amount of memory needed, this transforms to:

- interleaver lengths of up to 240 ms (QPSK, uniform) and 120 ms (16QAM, uniform);
- capability to store and process up to 408 CU or 6 528 IU (can be handled with approx. 4 Mbits of memory).

Since support of DVB-H is likely for DVB-SH receivers, the memory dedicated to the DVB-H MPE-FEC could be allocated to the de-interleaver. In this case, the DVB-H MPE-FEC, if ever transmitted, is not processed. The choice between MPE-FEC processing or interleaving processing should be managed at system level. Therefore, DVB-SH class 1 receivers could be realized with the same amount of memory than previous DVB-H receivers.

class 2

Receivers class 2 are required to handle an interleaver profile comprising 64 full SH-frames with 52 224 CU. Using the convolutional interleaver approach which typically halves the amount of memory needed, this transforms to:

- interleaver lengths of up to 30 s (QPSK, non-uniform) and 15 s (16QAM, non-uniform);
- capability to store and process up to 26 112 CU or 417 792 IU (can be handled with 256 Mbits memory).

10.3 DVB-SH reference receiver model

10.3.1 Reference model

The receiver performance is defined according to the reference model shown in figure 10.3.

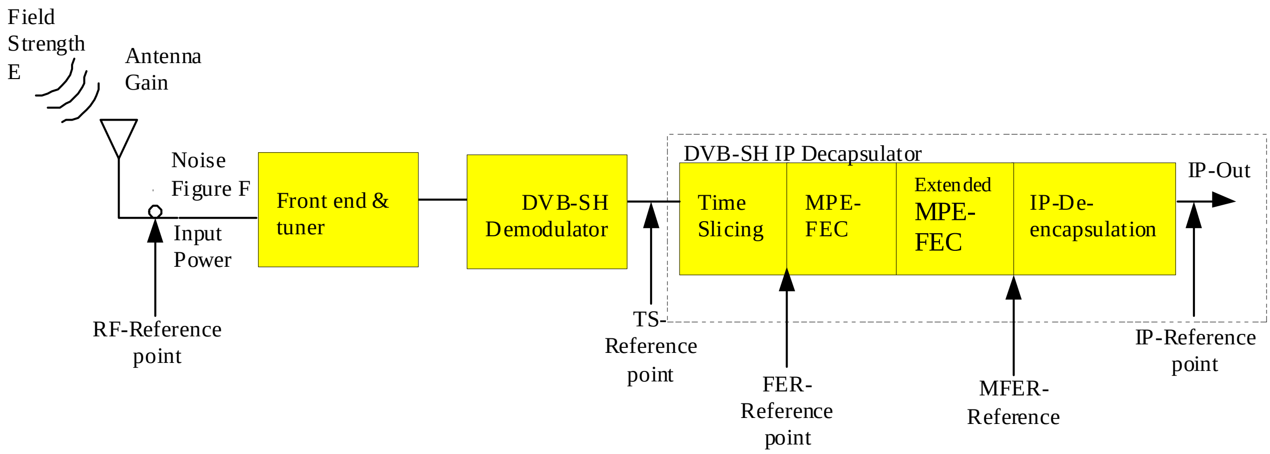


Figure 10.3: Reference model

Reference points are defined for:

- RF;
- transport stream;
- frame errors before MPE-FEC;
- frame errors after MPE-FEC and extended MPE-FEC;
- IP-stream.

All the RF receiver performance figures are specified at the RF-reference point, which is the input of the receiver.

10.3.2 Receiver for Vehicular terminals

Depending on terminal categories, various antenna and front-end solutions will be embedded in receiver. For terminal category 1 described in figure 10.4, according to vehicular reception constraints (clause 4.2.6.1), antenna with optimized diagram pattern and polarization will be used.

Additional Low Noise Amplifier will be connected directly to antenna to optimize noise figure and sensitivity (see details in clause 10.4).

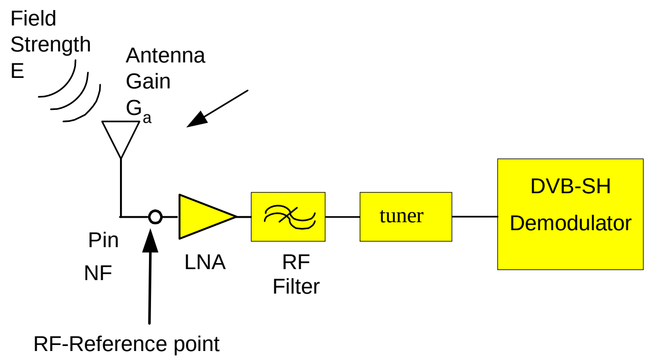


Figure 10.4: Radio receiver architecture for Terminal category 1

10.3.3 Receiver for Terminals with telecom modem

For category 3 terminals shown in figure 10.5, if the terminal includes telecom RF modems, a filter is mandatory between the antenna and the (optional) LNA and/or the tuner to protect the latter elements from telecom modem signal interference. However, this filter in front of the LNA degrades the DVB-SH receiver sensitivity.

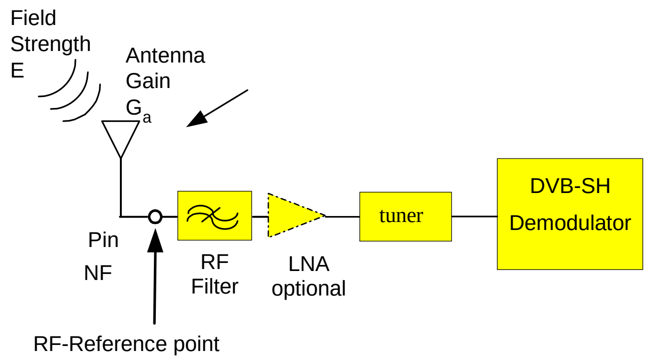


Figure 10.5: Radio receiver architecture for Terminal category 3

Basic receiver architectures for all terminal categories will be derived from both architectures presented above.

For SH-A system architecture, these synopsis will be used without any variants. For SH-B system architecture, the tuner will have two outputs each of them will correspond to TDM and OFDM signals respectively. These outputs are generated after down conversion into the tuner from two different RF frequencies, one is provided by satellite(s), the other by CGC segment. Demodulator for SH-B will have two inputs in accordance to figure 4.7 of clause 4.2.6.

For antenna diversity implementation, several receiver branches will be added to the single one. After diversity combining, transport stream will be provided to an unique input of DVB-SH IP decapsulator.

10.4 Minimum signal input levels for planning

10.4.1 Noise floor for vehicular receiver

The receiver should have a noise figure better than 2 dB at the reference point, at sensitivity level of each DVB-SH mode. This corresponds to the following noise floor power levels:

- Pn = -103,2 dBm [for 8 MHz OFDM channels, BW = 7,61 MHz];
- Pn = -103,7 dBm [for 7 MHz OFDM channels, BW = 6,66 MHz];
- Pn = -104,4 dBm [for 6 MHz OFDM channels, BW = 5,71 MHz];
- Pn = -105,2 dBm [for 5 MHz OFDM channels, BW = 4,76 MHz];
- Pn = -110,2 dBm [for 1,7 MHz OFDM channels, BW = 1,52 MHz].

10.4.2 Noise floor for handheld receiver

The receiver should have a noise figure better than 4,5 dB at the reference point, at sensitivity level of each DVB-SH mode This corresponds to the following noise floor power levels:

- Pn = -100,7 dBm [for 8 MHz OFDM channels, BW = 7,61 MHz];
- Pn = -101,2 dBm [for 7 MHz OFDM channels, BW = 6,66 MHz];
- Pn = -101,9 dBm [for 6 MHz OFDM channels, BW = 5,71 MHz];
- Pn = -102,7 dBm [for 5 MHz OFDM channels, BW = 4,76 MHz];
- Pn = -107,7 dBm [for 1,7 MHz channels, BW = 1,52 MHz].

10.4.3 Minimum C/N requirements

This clause provides in table 10.1 typical and possible DVB-SH receiver implementation losses. For AWGN minimum C/N requirement determination, it is recommended to add the implementation losses reported in table 10.1 to the C/N values given in clause 7.

Table 10.1: AWGN Implementation Losses

Modulation	Implementation losses (dB) Typical Receiver	Implementation losses (dB) Possible Receiver
TDM- QPSK	0,5	0,5
TDM- 8PSK	1,0	0,75
TDM- 16APSK	1,5	1,0
OFDM- QPSK	1,1	0,5
OFDM- 16QAM	1,5	1,0

The implementation losses are defined with respect to an ideal receiver with perfect channel estimator.

The implementation losses for typical receivers constitute an upper bound based on early receivers.

The possible receiver implements more sophisticated channel estimation algorithms, reducing implementation losses.

- Under TU6 channel conditions, the implementation losses for typical receivers are 1,6 dB for QPSK and 2 dB for 16QAM.
- Under TU6 channel conditions, the implementation losses for possible receivers are 1 dB for QPSK and 1 dB for 16QAM.
- Under Rice and Rayleigh terrestrial channel conditions, a provision of 2 dB for implementation loss is recommended.

Nevertheless, in all TU 6 computations of clause 11, measured C/N values are used.

10.4.4 Minimum input levels

10.4.4.1 Sensitivity for vehicular receiver

At RF-reference point, the receiver should observe reference criterion for a wanted signal level greater than Pn.

- Pn = -103,2 dBm + C/N [for OFDM 8 MHz];
- Pn = -103,7 dBm + C/N [for OFDM 7 MHz];
- Pn = -104,4 dBm + C/N [for OFDM 6 MHz];
- Pn = -105,2 dBm + C/N [for OFDM 5 MHz];
- Pn = -110,2 dBm + C/N [for OFDM 1,7 MHz].

Where C/N is specified in clause 7.2.2.6 and is depending on the channel conditions and DVB-SH modes.

10.4.4.2 Sensitivity for handheld receiver

At RF-reference point, the receiver should observe reference criterion for a wanted signal level greater than Pn.

- Pn = -100,7 dBm + C/N [for OFDM 8 MHz];
- Pn = -101,2 dBm + C/N [for OFDM 7 MHz];
- Pn = -101,9 dBm + C/N [for OFDM 6 MHz];
- Pn = -102,7 dBm + C/N [for OFDM 5 MHz];

- Pn = -107,7 dBm + C/N [for OFDM 1,7 MHz].

Where C/N is specified in clause 7.2.2.6 and is depending on the channel conditions and DVB-SH modes.

10.4.5 Antenna considerations

Receiver antennas must have good performance in various environments, receiving signal from satellite or/and CGCs. In this clause, intrinsic and integration characteristics are highlighted depending on the targeted terminal.

10.4.5.1 Antenna for terminal category 1

Antenna for vehicular terminals can be listed in two categories:

- antenna with circular polarization optimized for satellite reception; and
- antenna with linear polarization optimized for terrestrial reception.

Antennas with circular polarization for vehicular terminals have been developed for several systems like Satellite Digital Audio Radio System (SDARS) in US (XM Radio, Sirius). These antennas have relatively good circularly polarized gain at high elevation angle and acceptable linearly polarized gain at the horizon. They efficiently receive both satellite and terrestrial signals.

Antenna gain in Left or Right Hand Circular Polarization (LHCP/RHCP) can be +2 up to +8 dBic, depending on angle aperture and directivity required. Antenna gain in linear vertical polarization is less than -1 dBi at horizon. These antennas are designed in various technologies: small ground plane dependent patch etched on ceramics antennas, ground independent mast antennas such as quadrifilars, turnstile consisting in two 90° crossed dipoles able to generate circular polarization. Voltage Standing Wave Ratio (VSWR) can be less than 1,5 on 50 Ω load for narrow band operation, this performance maximize antenna efficiency. A Low Noise Amplifier (LNA) with a noise figure less than 1 dB can be connected directly at the antenna output to improve sensitivity. G/T of more than -21 dB/°K can be achieved.

For antennas with linear polarization, the practical standard for car reception is $\lambda/4$ monopole which use the metallic roof as the ground plane. Planar structures may be also used. Due to the use of the metallic roof, antenna pattern (and consequently the gain) is very dependent of the position of the antenna on the vehicle. Considering the 2 GHz MSS band, gain on the whole demi-sphere in linear vertical polarization is more than 0 dBi. Due to depolarization losses, the gain in RHCP is less than -3 dBic.

Other constraints have to be considered: Number of radio systems integrated in vehicle (Telecom modem car kit, GPS, etc.), larger antenna structures can be aesthetically displeasing when mounted on roof or glasses, etc.

Diversity features that increase the antenna number can lead to use a mix of technologies described above.

10.4.5.2 Antenna for terminal category 2a

This clause deals with antennas for Portable digital TV sets. These terminals are defined to be equipped with screen size larger than 25 cm (more than 10" diagonal), stationary during reception and antenna attached to the receiver. Integration of high performance antennas with circular polarization and high gain is now challenging due to the flat design of the terminal. In this case the manufacturers have to make compromise between design and performances. The performances of this terminal could be between the one of category 1 and category 2b, 3. If performances are favoured, by using for example a detachable antenna accessory, the terminal G/T could be as high as to -25 dB/°K (position adjusted by the user in line of sight of the source).

Diversity will be implemented advantageously with regard to mechanical size and potential distance between antennas.

10.4.5.3 Antenna for terminal category 2b and 3

This clause deals with antennas for pocketable digital TV receivers and handhelds with embedded Telecom modem(s), both being battery operated.

Two solutions are possible, one with an external antenna, the other with internal antenna. The majority of handset makers design terminals with internal antennas. A first consideration concerning antenna design is volume availability. Also, coexistence with a lot of components decreases expected gain due to dielectric and magnetic losses. This is why, low bulk antennas have less efficiency and more susceptibility to detuning. It is not conceivable to use multi band antenna that provide poor performances and coupling. Narrow band antenna will be considered as the basic antenna reference. For handheld 2b and 3 categories, better reception sensitivity will be reached with external antenna.

Antennas for handhelds can also be ordered in two categories:

- antennas with circular polarization optimized for satellite reception; and
- antennas with linear polarization optimized for terrestrial reception.

Circular polarization: the antenna pattern could be optimized for some pre-defined elevations, the main drawback is the position given by the user to the terminal. This leads to favour omni-directional pattern design rather than directional ones (at least in a hemisphere). The same assumptions have to be done with antenna with linear polarization.

Typical figures are given below for 2 GHz MSS band and in LOS measurement conditions:

- external RHCP antenna: gain 0 dBic (circular), gain in linear -3 dBi;
- internal RHCP antenna: gain -3 dBic (circular), gain in linear -6 dBi;
- external linear vertical polarized antenna: gain in linear 0 dBi, gain in circular -3 dBic;
- internal linear vertical polarized antenna: gain in linear -3 dBi, gain in circular -6 dBic;
- external RHCP antenna removable and adjustable by the user: gain +3 dBic, gain linear 0 dBi (antenna with directivity).

Antenna behaviour in multipath environment is more complex due to depolarization effect and integration of power in a sphere.

10.4.5.4 Antenna diversity considerations

Coming back on antenna diversity, different measurements on antenna diversity gain have been performed over the past years in various channels conditions and in S band (2 170 MHz to 2 200 MHz) ([i.33] and [i.48]). These references have been reporting the gain obtained using dual antenna.

It is thus recommended to apply the following values:

- 3,5 dB in reception conditions A, C and D;
- 5,5 dB in reception conditions B1 and B2 (I 33);
- 3 dB under satellite reception conditions and more generally $10 \cdot \log_{10}(N)$, where *N* is the number of antennas.

10.4.6 Maximum Input Power for Wanted and Unwanted Signals

The maximum allowed input level on the RF-reference point will depend on the antenna characteristics and RF front end architecture chosen (see clauses 10.3.2 and 10.3.3). Therefore, maximum value will differ with terminal categories.

For terminal category 3:

The maximum total average power from the wanted and unwanted signals should be less than +15 dBm. This value corresponds to a decoupling between DVB-SH antenna and modem antenna of more than 15 dB to 18 dB depending of the Telecom frequency band considered.

For terminal category 1 and 2a, 2b:

The maximum total average power from the wanted and unwanted signals should be less than -25 dBm. This value corresponds to mobile phone up link emissions close to the DVB-SH terminal at more than 1 meter from the source. Down link signal providing by base stations emissions will be below this value.

The assumptions considering interoperability with other radio systems, are given in annex B. In particular, real interferer environments of DVB-SH receiver will be highlighted including interfering signal patterns (with number of carriers, level and frequency band). In this clause, recommendations will be done for receiver linearity and selectivity.

10.4.7 G/T considerations

Table 10.2 gives typical figures of basic receivers (without diversity consideration) related to the terminal categories.

G/T calculations are based on various assumptions and the boundaries between classes of terminals are not so clear-cut.

Active antenna corresponds to an antenna with an attached LNA. Polarization losses are defined in the respective satellite and terrestrial budget links.

Assumptions:

- vehicular receiver (category 1): NF=2 dB (LNA 0,8 dB, filter 2,5 dB, tuner 3 dB);
- handheld receiver (category 2b and 3): NF=4,5 dB (filter 1,5 dB, tuner 3 dB).

Table 10.2: Receiver typical G/T versus terminal category

Usage	Handheld	Handheld	Portable	Vehicular
Classification	category 3	category 2b	category 2a	category 1
Antenna polarization	L or C	L or C	L or C	C
Tuner	DVB-SH	DVB-SH	DVB-SH	DVB-SH
Antenna gain (dBi) or (dBic)	-3	0	2	4
Total Gain (dB)	-3	0	2	4
Antenna Temperature (K)	290	290	200	150
Active antenna	No	No	Yes	Yes
Noise figure (dB) max	4,5	4,5	3	2
Total noise temperature (°K)	817	817	489	320
G/T (dB/K)	**-32,1**	**-29,1**	**-24,9**	**-21,0**

11 Network planning

DVB-SH system is a hybrid system with a satellite and a terrestrial components.

Terrestrial component network planning technique have been developed over the past 50 years for two distinct services:

- Classical TV broadcast in UHF, VHF, from for high towers, using high power transmitter.
- Cellular, mobile system from GSM/DCS to UMTS and more recently from WiMAX technology.

Although based on the same physics, practice differ in some extent to the very different markets they address:

- High power for the first one, low/medium power for the second.
- Roof top reception for the first one, indoor handset service for the second one.

It is not the objective of this clause to propose an integrated framework, but to expose the main rules of each world for the network planning exercise.

On the other hand the satellite world, also existing for 50 years, has obviously specificities: line of sight, shadowing, high path loss, and uses Land Mobile Satellite propagation models, for availability computation.

The present document is presenting an overview of different methods of radio network planning and using different dimensioning parameters which values are compiled from different sources and experiences. The proposed values cannot be seen as absolute references, especially on the cellular network approach, as the used parameters values can vary from a source to another, from a country to another. That is why a range of typical parameters values is proposed, and some examples provided for network planning.

11.1 Introduction to DVB-SH network planning

As DVB-SH system is a hybrid system with a satellite and a terrestrial component, the coverage considered in the DVB-SH network planning can be split in three areas:

- Terrestrial only coverage: defined as the area served only by the complementary ground component.
- Satellite only coverage: defined as the area served only by the satellite component.
- Hybrid coverage: defined as the area where both the satellite and the terrestrial signals can be received simultaneously and potentially combined.

Concerning terrestrial coverage, there are typically two possible methods to perform radio network planning:

- The first one is a typical broadcast approach based on the computation of the minimum field strength requirements, for different classes of terminals and different propagation channels. The broadcaster method has been developed for the use of high transmitting towers with very high power, and aiming at roof top coverage rather than street level receivers.
- The second one is based on cellular network radio planning tools, and is using the minimum received power level as planning criterion. Most of these tools uses only this method, and allow network planners to determine (given a certain transmitting power per site) the number of sites necessary to cover a city with a given quality of service. The cellular network method has been developed for lower tower and also lowers to medium power to cover dense cities at street level receivers, and targeting indoor coverage. This clause will give also in some examples the link budgets with the required transmitter power in some typical examples.

The link between the two methods is insured through the formula below with E representing the field strength and P_{imin} the minimum received power level, and Ga the antenna gain in dBi.

$$E\left[\text{dB}\mu\text{V/m}\right] = P_{s\min}\left[\text{dB(mW)}\right] - G_a\left[\text{dB}\right] + 77,2 + 20\log_{10} f\left[\text{MHz}\right]$$

One of the objectives of this clause is to compile long time practices in network planning domain applicable to DVB-SH network planning, that is to say review parameters and their possible value ranges enabling both type of network planning. These approaches can also be extended for hybrid coverage considering the satellite as an additional repeater that allows:

a) reduce the number or the power of terrestrial sites; or

b) increase the quality of the service in the hybrid coverage area.

For satellite coverage, the number of transmitters is dependent on the number of satellites serving the satellite coverage area. While in the terrestrial case we have a cluster of cells serving a geographically limited area, for the satellite the coverage can be a continent-size area served with several linguistic or regional spots. The objective of the satellite network planning is to assess the required satellite power given a certain coverage area and a certain quality of service for a given data rate. This exercise is described in the clause "Satellite Link Budgets", but it is done in a different way as in terrestrial network planning.

- in terrestrial links, given the coverage probability in some conditions, the required shadowing margins are computed and allow to dimension the transmitter;
- in satellite links, given the satellite EIRP in some conditions, the obtained link margins are used in simulations to compute the availability in the given conditions. Thus, by reverse engineering, given a an availability, required satellite EIRP can be estimated.

Therefore, after this introduction, the clause comprises the following clauses:

- definition of reception condition;
- coverage definition, following a parallel track for the three approaches;
- network planning factors.

Then, following also a parallel track:

- network planning methods and examples based on minimum filed strength;
- network planning methods and examples based on minimum received signal level;
- satellite network planning; and
- for next release: hybrid network planning.

11.2 DVB-SH reception conditions

11.2.1 Introduction

Reception conditions depend on the environment, the mobility conditions and the kind of terminal. The environments relevant for DVB-SH system are defined in clauses 4.2.1 to 4.2.3 and clause A.7 by propagation channel models. DVB-SH terminal categories are defined in clause 10. Furthermore, DVB-SH introduces two classes of receivers with different capability of processing time diversity elements in the transmitted waveform (see clauses 6.5, 7.3.3 and 10.2).

Concerning the mobility conditions, two main kinds of reception can be defined:

- Pedestrian speed reception where the channel conditions exhibit relatively low levels of fast fading (from movement of nearby objects such as trucks) and slow deep fades due to slow movements of the antenna and nearby blocking objects (≤ 3 kmph). The portable terminal with external or integrated antenna is used indoors or outdoors at a mean height of 1,5 m above ground.
- Mobile reception applies to the use of Car-mounted Terminals (Terminal Category 1) with speeds higher than 3 kmph. It is assumed that the receiving antenna is external and at a minimum height above ground level of 1,5 m. Terminal Category 2b and 3 used for direct reception within a car or train or any kind of vehicle could also be considered as a case of mobile reception. It can be called in vehicle reception.

11.2.2 DVB SH reception conditions

The five cases of reception conditions defined in DVB-H, called Class A, B1, B2, C and D (see EN 302 304 [3] and [19]), fit also the DVB SH terrestrial variety of contexts. An additional dimension can be introduced if we consider also the DVB-SH environments defined in clause 4: rural, urban, and suburban. These classifications can also be used for satellite or hybrid coverage, although not all reception cases are applicable in all environments due to the intrinsic power limitation of the satellite link. Conversely, some reception conditions in certain environments are only applicable to satellite coverage, due to the geographical limitation of the terrestrial coverage. An additional case can also be defined in DVB-SH; this case refers to the satellite indoor coverage achieved due to a domestic gap filler (defined as TR (b) in the system specifications) that receives and amplifies the satellite signal.

Note that this partition can apply equally to SH-A and SH-B system architectures.

In all scenarios, and in the rest of the document, receiver is placed at 1,5 m above ground.

Table 11.1: Usage scenario

Reception Condition	Situation	Characteristics	Environment	Coverage	Channel Characteristics	Typical channel parameter Relevant DVB-SH parameters
Reception condition A	Outdoor pedestrian	Up to 3 kmph	Rural	Satellite	Stationary : Low delay/low spread	LOS :AWGN/Rice K>10: Additional margin to cope with fading For shadowed, Rice below 7 dB Time interleaving to mitigate effects
					Low speed: large signal variation	LMS channel model at low speed. Time interleaving
			Urban	Terrestrial	Stationary : Rayleigh/Very low Doppler	TU6 channel? /low code rate improves. Antenna diversity also improves
					Low speed / Rayleigh /low Doppler	Higher margins to cope with slow fading effects
			Suburban	Terrestrial, Hybrid	For terrestrial same as above	For terrestrial same as above
					No hybrid channel model available	No hybrid channel model available
Reception condition B1	Light-indoor	Up to 3 kmph, lightly shielded building	Rural	See note (1)		
			Urban	Terrestrial	Channel is the same as Reception A with high penetration margins	TU6 channel? /low code rate improves. Antenna diversity also greatly improves
			Suburban	Terrestrial, Hybrid	Same as above for terrestrial No hybrid channel model available	
Reception condition B2	Deep-indoor	Up to 3 kmph, highly shielded building	Rural	See note (1)		
			Urban	Terrestrial	Channel is the same as Reception A with higher penetration margins as in B1	TU6 channel? /low code rate improves. Antenna diversity also greatly improves
			Suburban	Terrestrial	Same as above (lower margins)	Same as above
Reception condition C	Mobile (vehicle) with roof-top antenna	Up to 200 kmph	Rural	satellite	Large signal strength variation depending on environment	LMS channel model at medium/high speeds for different environments
			Urban	Terrestrial	Multiple Rayleigh fading paths Delay spread depends mainly on network characteristics	Channel models like TU6 cover this scenario at least for low or medium power repeaters. Critical SFN scenarios require channel models with higher delay spread
			Suburban	Terrestrial, Hybrid	For terrestrial same as above No hybrid channel model available	For terrestrial same as above No hybrid channel model available
Reception condition D	Mobile (portable) in-car	Up to 130 kmph	Rural	See note (2)		
			Urban	Terrestrial	Multiple Rayleigh fading paths Delay spread depends mainly on network characteristics	Channel models like TU6 cover this scenario at least for low or medium power repeaters. Critical SFN scenarios require channel models with higher delay spread
			Suburban	Terrestrial	Same as above	Same as above

Reception Condition	Situation	Characteristics	Environment	Coverage	Channel Characteristics	Typical channel parameter Relevant DVB-SH parameters
NOTE: According to clause 10 classification, terminals used for the different reception conditions are the following: reception conditions A, B1, B2 and D are associated to terminal categories 2a, 2b and 3 (handset, handheld and portable): (1) for Reception conditions B1 and B2 in the rural environment, under satellite coverage, it is assumed that the satellite signal is assisted by terrestrial repeater (TR(b)). The link budget applies to these TR(b), not to the end-user terminal; reception condition C is associated to terminal category 1 (vehicular); (2) for Reception condition D in the rural environment, under satellite coverage, it is assumed that the satellite signal is assisted by terrestrial repeater (TR(c)). The link budget applies to these TR(c), not to the end-user terminal.						

11.3 Coverage definition

In general, coverage is defined by a reference area and a percentage of that area where the signal is received with a given required quality of service a given percentage of time. Consideration is also given to the edge of the signal area (whether satellite beam, or terrestrial transmit signal). The edge is taken as the contour beyond which the signal degrades (either from leaving the satellite beam footprint, or moving too far from the terrestrial transmitter) below the point required to deliver the given QoS level. DVB-SH coverage is made up of three different components: satellite only coverage, terrestrial only coverage, and hybrid coverage.

The coverage area definition is differing, first between satellite and terrestrial coverage, and second between the broadcaster and cellular network approaches.

11.3.1 Broadcaster approach

In defining the coverage area for each reception condition, a three level approach is taken (see EN 302 304 [3]).

Receiving location coverage: the smallest unit is a receiving location with dimensions of about 0,5 m. In the case of portable antenna reception, it is assumed that optimal receiving conditions will be found by moving the antenna or moving the handheld terminal within 0,5 m in any direction. Such a location is regarded as covered if the required carrier-to-noise and carrier-to-interference values are achieved for 99 % of the time.

Small area coverage: the second level is a "small area" (typically 100 m × 100 m).

Coverage area: the third level is the coverage area. The coverage area of a transmitter, or a group of transmitters, is made up of the sum of the individual small areas in which a given class of coverage is achieved.

11.3.2 Cellular approach

Cell coverage: rather than using the three level approach coverage area is defined as the aggregation of several circular (approximately) cells defined by their radius. The transmitter is in the centre of each cell and can use omnidirectional antenna or trisector or any other type of antennas.

Cell radius: when using cellular network methods, coverage are is defined as a cell of radius R, defined, as the extreme points where the link budgets is satisfied to meet the required level of signal.

Cell edge: is defined at distance R of the transmitter.

11.3.3 Satellite coverage

For satellite-only coverage: the basic covered area is the satellite antenna footprint, typically tailored to a satellite spot beam. However, not all points in the satellite coverage area are reached with the same quality. Depending on the service definition, service coverage will have to take into account environments such as shadowing, which can be caused by trees, tall buildings, or even low buildings in the case of low satellite elevation angle.

11.3.4 Quality of coverage

The coverage is characterized by a certain quality of service characterized by a quality criterion. For each propagation channel (Gaussian, Rice, TU6, etc.) and each modulation and coding state, this criterion is satisfied by a minimum C/N value, which is used in network planning (called network planning value).

The cell coverage probability is defined as the percentage of locations inside the cell where the criterion is satisfied or where C/N is above the minimum corresponding value.

Under terrestrial coverage (TU6), clause A.12 utilizes two quality criteria:

- FER 5 %, equivalent to MFER 5 %, which is used in DVB-H: 5 % of frames in error; and
- ESR5; defined in clause A.8: 1 s maximum in error every 20 s, which is more demanding and a fulfilment criterion defined by ESR5(20).

As shown in clause A.12, less than 1 dB in C/N separates the two criteria. In the computations associated to TU6, an ESR5(20) fulfilment criterion is chosen.

For satellite channel, ESR5(20) fulfilment criterion is chosen.

BMCO [19] defines the different qualities of service corresponding to different coverage probabilities, and depending on the different reception conditions. They are defined as "good" and "acceptable":

- the terrestrial coverage area is declared *"good"*:
    - at least 99 % of receiving locations within the area are covered for reception conditions C and D;
    - at least 95 % of receiving areas at the edge of the area are covered for reception conditions A and B;
- the terrestrial coverage area is declared *"acceptable"*:
    - at least 90 % of receiving locations within the area are covered for reception condition C and D;
    - at least 70 % of the receiving locations at the edge of the area are covered for reception conditions A and B.

Table 11.2: Coverage probability for terrestrial coverage (broadcast method)

	"Good"	**"Acceptable"**
Reception Condition A	95 % edge	70 % edge
Reception Condition B	95 % edge	70 % edge
Reception Condition C	99 % overall	90 % overall
Reception Condition D	99 % overall	90 % overall

In the cellular coverage, there is no specific qualification of coverage quality or required percentage of coverage (cell or edge of cell). With given percentage of edge or overall coverage. There are some coverage obligation under the regulation process.

Usually, coverage probability can be defined on the whole cell, or at the edge of cell coverage. There is no specific rule.

The most usual qualities of coverage are 90 % and 95 % overall coverage for all reception conditions. Hence, other coverage probability can be used upon request of operator.

Under satellite coverage: **under satellite coverage, there are no typical values of coverage probability.**

In the satellite clause, the obtained coverage probability are provided in the different examples.

11.4 Network planning factors

11.4.1 Introduction

The network planning factors constitute the ensemble of parameters that are needed in the different network planning computation and tools; As they are common to the different methods, they are described in this clause. One can distinguish three categories of criteria:

- channel dependant factor: relative to the shadowing and in building penetration margins for instance;
- technology dependant factor: bandwidth, noise factor, antenna gain, etc.; and
- network architecture factor or implementation optional parameter: SFN topology gain, antenna diversity gain.

11.4.2 Channel dependant factor

The first factor is the in building or in vehicle entry or penetration losses, applicable to reception conditions B and D.

Depending on the different terrestrial planning methods, the range of values is somehow different, and so the taken values are give in each corresponding clause.

The other factor is called location correction factor, or shadowing margins or fading margins in the satellite links.

Correction location factor is defined by the product: $C_l = \mu * \sigma$ in EN 302 304 [3] and [19] where: μ is called the distribution factor, and σ is the standard deviation of the distribution of the signal variation.

The possible values are provided in clause 11.5.3.1.

The term shadowing margins is used currently in the cellular network approach. The most commonly used propagation model is Log normal attenuation with the standard deviation σ.

Considering the distance to transmitter d and R the cell radius, probability that signal exceeds a defined threshold at distance d in an area dA (elementary area) is given by:

$$P_e = \frac{1}{2}\left\{1 - \operatorname{erf}\left[\frac{-C_l + 10n \log\left(\frac{d}{R}\right)}{\sqrt{2}\sigma}\right]\right\}$$

where: C_l is the shadowing margin associated with the coverage probability and σ is the shadowing standard deviation.

So, when being at edge of cell, d = R and the equation becomes:

$$Pe = \frac{1}{2}\left[1 - \operatorname{erf}(a)\right]$$

$$\text{with } a = -\frac{C_l}{\sigma\sqrt{2}}$$

It must be noticed that the ratio C_l / σ is the location correction factor μ.

The relationship between overall coverage probability and shadowing margins is given by the Jakes formula, obtained by integrating the previous equation all over the cell.

$$P_c = \frac{1}{2}\left[1 - \operatorname{erf}(a) + \exp\left(\frac{1-2ab}{b^2}\right)\left(1 - \operatorname{erf}\left(\frac{1-ab}{b}\right)\right)\right]$$

where: P_c is overall cell coverage probability, a the same variable as above; and

$$b = \frac{10n\log(e)}{\sigma\sqrt{2}}$$

n is the path loss exponent (usually 3,5 in cellular networks);

e the exponential constant.

This probability is calculated at each location with the appropriate values, taking into account standard deviation.

There are many ways to use these formulas, whether a cell edge coverage or a whole cell coverage is required.

As the requirements are given in terms of whole coverage, the margins will be computed accordingly, and the cell edge coverage indicated.

In the satellite link budgets, there are different ways to consider the link margins:

- for stationary reception, a Rice channel is assumed, leading to similar computation as above, providing availability versus link margins. Then link budgets give the possible link margins, and the corresponding availability;
- for mobile reception (above 3 kmph), the link margins are given by the link budgets, and simulations provide the availability depending on conditions.

11.4.3 Technology dependant factors

By this we mean the antenna gain, frequency band and the receiver noise factor.

The different possible frequency bands have been explored in clause 4.

In this context, the computations are done in S band (2 170 MHz to 2 200 MHz) all over the document, but L band is also a candidate band, and the results can be easily adapted.

The noise bandwidth can have all the values described in the DVB-SH standard. For the computed examples provided in this clause, a noise bandwidth of 4,76 MHz is assumed, corresponding to the OFDM 5 MHz case.

The terminal related factors are the Noise Factor, the antenna gain,, the antenna polarization, and the antenna temperature, resulting in the factor of merit G/T, used in the satellite link budgets.

The different terminal categories and their related characteristics are described in clause 10.

11.4.4 Network architecture or implementation optional factors

This clause concerns some factors depending on some specific system architecture, mainly SFN gain and on implementation antenna diversity gain, that may additionally be taken in account by the Network planning tool.

a) The SFN configuration

The SFN topology can be considered in two cases:

- terrestrial only coverage;
- hybrid satellite/terrestrial coverage.

In the terrestrial only coverage, it is an important feature of the DVB-SH. The SFN gain will depend on the network topology. When overlap is made possible by construction and the cell transmitters are properly synchronized, there is SFN gain with respect to a single-cell case. Similarly, the UMTS networks are also built as Single Frequency Network, probably more known under the name of networks with frequency reuse factor "1".

In UMTS networks, the handover procedure occurs when a mobile is at the edge of two or three cells, where signal received from different nodes B are of similar strength; Thanks to the Rake receiver, terminal can acquire and combine the different signals coming from the Nodes B, increasing signal strength by the amount of the so called Soft Handover Gain (SHG). This point is studied in the literature. Different references provide analytical and experimental studies of Handover gain, leading to 3 dB to 4,6 dB Soft Handover gain.

In DVB-SH/H SFN networks at the cell boundaries, the signals can be added coherently, thanks to the Guard Interval, leading to SFN gain.

This gain can be translated into the following advantages:

- increase of the coverage probability when dealing with equal transmit power as for a single cell;
- decrease of Transmit power while keeping the same coverage and cell radius (equivalent to shadowing margins reduction);
- increase of cell coverage.

The resulting so called SFN gains are in a range between 3 dB and 5 dB.

The network planner will take into consideration the requirements related to the SFN configuration. Namely, the deployment of the transmitters must be such that the signal coming from different stations can be added coherently (i.e. relative delay falls within the guard interval). When this is not respected the network will endure an additional level of self interference that may degrade the overall performance.

The hybrid SFN topology and possible associated gain is not studied in this release of the Implementation Guidelines.

b) Receiver antenna diversity

Antenna diversity has been studied and implemented for a long time in the telecommunication industry, and also by the car equipment manufacturers, as it can provide improvement of reception quality. The antenna diversity is not a standard feature. Implementation depends on wavelength and terminal size. Typically, the use of S band allows implementing dual antenna diversity on handsets, while it will be more adapted for PDA in L band for instance.

As reported, in different trial results, dual diversity gains using Maximum Ratio Combining are the following:

- 3,5 dB in conditions A, C and D;
- 5,5 dB in conditions B1 and B2;
- 3 dB in Satellite Reception conditions.

It is not a standardized feature, though mentioned in clause 10, but as its efficiency has been on field proven, it will be used as an option in the different computations.

11.5 Terrestrial Network Planning based on minimum equivalent field strength (broadcast approach)

11.5.1 Introduction

Traditional TV broadcast is based on roof service characteristics, typically 10 m height, from high towers, high powered sites, to high gain Yagi type receiving home antennas. Quality of service provided is measured through the computation of the minimum median equivalent field strength.

Broadcast Network planning tools are usually proprietary to each broadcaster, that allow them to determine the transmitters optimal topology, and accordingly defined delivered quality of coverage.

For DVB-H, these methods have been extensively discussed at the BMCO forum and have been enhanced to take into account the specifics of indoor service planning to individual terminals, such as in building penetration losses, shadowing margins (hereunder defined as location correction factor), thus introducing near ground coverage (1,5 m) in urban and rural areas, and also technology terms such as emission antenna gain, and reception antenna diversity gain.

Only few DVB-H networks are actually deployed on large scale, thus providing little experience feedback; likewise actual cellular topologies and network planning based on these tools have limited background for broadcast services, and mostly rely on experiences from mobile radio services planning (detailed in the cellular approach clause).

The following clause gives an illustration of applicable methods and typical parameterizations as recommended by the BMCO forum. Mobile services, derived methodologies are presented in the cellular approach In the following, different calculations are made for the different reception conditions identified earlier and are given for S band only, but they can be extended easily to L band or to any other usable frequency band.

11.5.2 Quality of coverage

The quality of coverage is provided in clause 11.2.4.

11.5.3 Network planning based on field strength computation methodology

To compute the median filed strength, the following reception cases are considered for terrestrial reception:

- reception condition A (outdoor portable);
- reception condition B1 and B2 (indoor portable);
- reception condition C Mobile (vehicle with roof top antenna);
- reception condition D (portable, in vehicle with no roof top antenna).

The C/N values will be the same as used in the previous clause on minimum signal level. To calculate the minimum median power flux density or equivalent field strength needed to ensure that the minimum values of signal level can be achieved at the required percentage of locations, the following formulas are used:

$$\Phi_{min} = P_{s\,min} - A_a$$

$$Aa = G + 10 \log 10 (\lambda^2/4\pi)$$

$$E = \sqrt{4\pi\eta \frac{P_{s\min}}{G_a}} \cdot \frac{f}{c}$$

or, in dB:

$$E\left[\text{dB}\mu\text{V/m}\right] = P_{s\min}\left[\text{dB(mW)}\right] - G_a\left[\text{dB}\right] + 77{,}2 + 20\log_{10} f\left[\text{MHz}\right]$$

This equation gives the link between the two possible methods,

where: $\eta = 120\pi\ \Omega$;

$E_{min} = \Phi_{min} + 120 + 10 \log (120\pi) = \Phi_{min} + 145{,}8$;

$\Phi_{med} = \Phi_{min} + P_{mmn} + C_l + L_o$ (Reception condition A or C);

$\Phi_{med} = \Phi_{min} + P_{mmn} + C_l + L_b + L_o$ (Reception condition B);

$\Phi_{med} = \Phi_{min} + P_{mmn} + C_l + L_v + L_o$ (Reception condition D).

In case of antenna diversity use, all value of minimum power flux density must be written differently:

$$\Phi_{med/div} = \Phi_{med} - G_{div}$$

$$Emed = \Phi med + 120 + 10 \log (120\pi) = \Phi med + 145{,}8$$

An equivalent formula is given here below:

$$E_{med} = NF + 10*Log(k_B T\ B) + C/N - G + 107{,}2 + 20*Log(f) + L_b \text{ (or } L_v) + \mu*\sigma + L_o \text{ (other losses)}$$

where: C/N : RF signal to noise ratio required by the system (dB) for the required performance and modulation/coding scheme;

Φ_{min} : minimum power flux density at receiving location (dBW/m2);

E_{min} : equivalent minimum field strength at receiving location (dB⌠V/m);

A_a : effective Antenna Aperture;

G : antenna gain (dBi);

L_b : building penetration loss (dB);

L_v : vehicle entry loss (dB);

L_o : all other losses (polarization mismatch, etc.);

P_{smin} : minimum receiver signal input power (dBW);

P_{mmn} : allowance for man made noise (dB);

C_l : location correction factor (dB);

Φ_{med} : minimum median power flux density, planning value (dBW/m²);

E_{med} : minimum median equivalent field strength, planning value (dB⌠V/m).

For calculating the location correction factor C_l a log-normal distribution of the received signal is assumed. The location correction factor can be calculated by the formula:

$$C_l = \mu * \sigma$$

where: μ is the distribution factor, and σ is the standard deviation of the distribution.

It must be noticed that this value of μ depends on the model assumed for the distribution (normally lognormal is considered for shadowing), as well as the standard deviation. The product is in fact the taken shadowing margins. Possible values are given in further clauses.

11.5.4 Network planning factors

11.5.4.1 Location Correction

11.5.4.1.1 Location correction for reception condition A

In reception condition A (portable outdoor), the location variation factor is given by:

$$C_l = \mu * \sigma$$

The common used value for σ is 5,5 dB (see ITU-Recommendation M.1225 [11]).

This gives the table for the different qualities of coverage.

Table 11.3: Location correction factor for reception condition A

Edge of area coverage (%)	Distribution factor (μ)	Location correction factor (dB)
70	0,52	3
95	1,64	9

11.5.4.1.2 Building penetration losses and location correction for reception conditions B1 and B2

In reception conditions B, signal are subject to in building penetration losses. The location correction factor at indoor locations is the combined result of the outdoor variation and the variation factor due to building attenuation. These distributions are expected to be uncorrelated. The standard deviation of the indoor field strength distribution can therefore be calculated by taking the root of the sum of the squares of the individual standard deviations. As a consequence, the location variation of the field strength is increased for indoor reception.

$$\sigma = (\sigma^2_o + \sigma^2_p)^{1/2}$$

Table 11.4 gives the building penetration losses values proposed.

Table 11.4: Building penetration losses

Band	S band
Penetration loss	Loss
Condition B1	14
Condition B2	18

As an example, table 11.5 gives the location correction factor in deep indoor (B2 reception) with overall:

$$\sigma = 10 \text{ dB}$$

Table 11.5: Location correction factor for reception condition B2 in S band

Edge of area coverage (%)	Distribution factor (µ)	Location correction factor (dB)
70	0,52	5,2
95	1,64	16,4

11.5.4.1.3 Location correction for reception conditions C and D

Mobile reception is defined as a reception with moving receiver or at location where large objects moving around the receiver.

Two cases are possible:

- mobile reception with handset terminal inside a vehicle, car or inside any moving object (Reception condition D);
- mobile reception with a vehicular terminal (Reception condition C).

Clause 7 includes the values that should be used for planning in mobile reception. It should be taken into account the great influence of Turbo code FEC on the C/N and maximum Doppler shift (and therefore maximum speed in a particular channel). Any way, link budgets are performed with speed below the limit.

a) Location percentage requirements for mobile reception (conditions C and D)

For reception categories C and D, some additional margins are added to cope with mobile environments (see EN 302 304 [3]).

Table 11.6: Location correction factor for reception conditions C and D

Overall area coverage (%)	Distribution factor (µ)	Location correction factor (dB)
90	1,28	7
99	2,33	12,8

b) Vehicle entry loss

For mobile reception inside cars or any other vehicle entry loss must be taken into account. For instance, an entry loss is 7 dB is taken for in car penetration losses. The location correction factors will be the same for in vehicle reception.

11.5.4.2 Network architecture and implementation optional factors

a) The SFN possibility

The SFN gains appear at the cell overlap points. The field strength computation is made all over the area and not at specific points. So SFN gain will not be considered in this clause.

b) Receiver antenna diversity

As recalled in clause 11.4.4 b) there will be two cases: single antenna reception and dual antenna reception. The proposed gains are the following:

- 3,5 dB in conditions A, C and D;
- 5,5 dB in conditions B1 and B2.

The filed strength reductions are equal to the gain provided above.

11.5.5 Field strength computation examples

The objective of the link budget is to provide the minimum field strength values at 1,5 m height. Normally, the exercise can be made for all possible values of C/N. We limit the examples to a few values corresponding to some situations in the clause A.12. Extrapolation from the proposed values is straightforward.

An example of each reception condition is provided in S band, only to limit the number of tables.

Antenna diversity is used for reception conditions A, B and D, though it can also be implemented for vehicle roof top reception.

Two modulation and coding schemes are studied:

- QPSK 1/3; and
- 16QAM 1/3.

The required C/N values are extracted from the clause 12. In the following clause, we give an example of table, and a synthesis for different examples. The C/N values are coming from laboratory measurements and thus include the implementation losses, and are given for FER 5 %.

It must be noticed that maximum speed in the tables (conditions C and D) is 50 kmph, below the CN + 3 dB Doppler limit.

11.5.5.1 Field strength computation for reception condition B2

The studied case is with Handheld Category 3 in reception condition B2 (3 kmph) and the link budgets are computed for 99 % of ESR5(20) fulfilment at different coverage qualities as indicated in the table 11.7.

Table 11.7: Median field strength in reception condition B2 with QPSK 1/3 and 16QAM 1/3 single antenna

Modulation and coding		QPSK 1/3	16QAM 1/3
Frequency band	(MHz)	2 182,5	
Equivalent Noise Bandwidth	B (MHz)	4,76	
Receiver Noise Figure	NF (dB)	4,5	
Minimum C/N required by system	(dB)	2,8	9,1
Min receiver signal input power	Psmin (dBW)	-129,90	-123,60
Min. equivalent receiver input voltage, 50 Ω	Us min (dBµV)	7,1	13,4
Antenna gain relative in dBi	G (dBi)	-3	
Effective Antenna aperture	Aa (dBm²)	-31,23	
Min. power flux density at receiving location	Φ min (dBW/m²)	-99	-92
Min. equivalent field strength at receiving location	Emin (dBµV/m)	47	53
In building or in car penetration losses	Lb (dB)	18	
Location correction factor for 70 % edge of coverage	Cl (dB)	5,2	
Minimum median power flux density at 1,5 m	Φmed (dBW/m²)	**-75**	**-69**
Minimum median equivalent field strength at 1,50 m	Emed (dBµV/m)	**70**	**77**
Location correction factor for 95 % edge of coverage	Cl (dB)	16,4	
Minimum median power flux density at 1,5 m	Φmed (dBW/m²)	-64	-58
Minimum median equivalent field strength at 1,5 m	Emed (dBµV/m)	**82**	**88**

Table 11.8: Median field strength in reception condition B2 with QPSK 1/3 and 16QAM 1/3 dual antenna

Modulation and coding		QPSK 1/3	16QAM 1/3
Frequency band	(MHz)	2 182,5	
Equivalent Noise Bandwidth	B (MHz)	4,76	
Receiver Noise Figure	NF (dB)	4,5	
Minimum C/N required by system	(dB)	2,8	9,1
Min receiver signal input power	Psmin (dBW)	-135,40	-129,10
Min. equivalent receiver input voltage, 50 Ω	Us min (dBµV)	1,6	7,9
Antenna gain relative in dBi	G (dBi)	-3	
Effective Antenna aperture	Aa (dBm²)	-31,23	
Min. power flux density at receiving location	Φ min (dBW/m²)	-104	-98
Min. equivalent field strength at receiving location	Emin (dBµV/m)	42	48
In building or in car penetration losses	Lb (dB)	18	
Location correction factor for 70 % edge of coverage	Cl (dB)	5,2	
Minimum median power flux density at 1,5 m	Φmed (dBW/m²)	-81	-75
Minimum median equivalent field strength at 1,50 m	Emed (dBµV/m)	64,83	71,13
Location correction factor for 95 % edge of coverage	Cl (dB)	16,4	
Minimum median power flux density at 1,5 m	Φmed (dBW/m²)	-70	-63
Minimum median equivalent field strength at 1,5 m	Emed (dBµV/m)	76	82

11.5.5.2 Synthesis

Table 11.9 gives a synthesis of the previous link budgets based on minimum filed strength.

Table 11.9: Overall synthesis (Field Strength in dBµV/m) single antenna reception

Reception conditions/terminal	Quality of Service cases	QPSK 1/3	16QAM 1/3
Condition A/ Handheld Cat 3	70 % edge	50	57
	95 % edge	56	63
Condition B2/ Handheld Cat 3	70 % edge	70	77
	95 % edge	82	88
Condition C/Vehicle Cat 1	90 % overall	47	52
	99 % overall	53	58
Condition D/ Handheld Cat 3	90 % overall	60	66
	99 % overall	66	72

Table 11.10: Overall synthesis (Field Strength in dBµV/m) dual antenna reception

Reception conditions/terminal	Quality of Service cases	QPSK 1/3	16QAM 1/3
Condition A/ Handheld Cat 3	70 % edge	47	54
	95 % edge	53	60
Condition B2/ Handheld Cat 3	70 % edge	65	72
	95 % edge	77	83
Condition C/Vehicle Cat 1	90 % overall	44	49
	99 % overall	50	55
Condition D/ Handheld Cat 3	90 % overall	57	63
	99 % overall	63	69

11.5.6 Use of field strength based Radio Network Planning Tool example for DVB-SH Network Design

For further study.

11.6 Terrestrial Network Planning based on minimum received signal level (cellular approach)

Network Planning tools allow to compute precise site distribution and cell ranges. They use propagation models to predict with high accuracy path loss, and different cell ranges. These models integrate field based calibration. Some Practical Network Planning tools, include a propagation model called Standard Propagation Model, based on Cost Hata model with different correction terms allowing range of application down to 200 m from the BTS.

Others suppliers provide tools based on ITU-R Recommendation P.526-11 [27] model.

The complete description, parameterization of the tool, and procedures of use is not in the range of the present document.

A network planning tool will compute the coverage using an intermediate parameter called "the minimum signal level for coverage". This parameter corresponds to the minimum signal power level that will ensure service reception. It takes into account propagation characteristics associated to the considered type of scattering environment. The network planning tool will plot coverage maps.

This clause gives an overview of the common practices of the cellular industry (cellular operators), and provides a first level approximation of DVB-SH networking, including the computation of cell range, given the transmitter EIRP.

After some considerations on network topology, we provide the basic equations used in the network planning tools.

11.6.1 Cellular network topology

Without going into complex topological details, a cellular network is composed of an aggregation of cells that can be considered in first approximation as circle of radius R. This radius defines the edge of cell and is considered in the link budget computation..

Figure 11.1 shows a regular cellular network with trisectorized cells and in SFN possible topology.

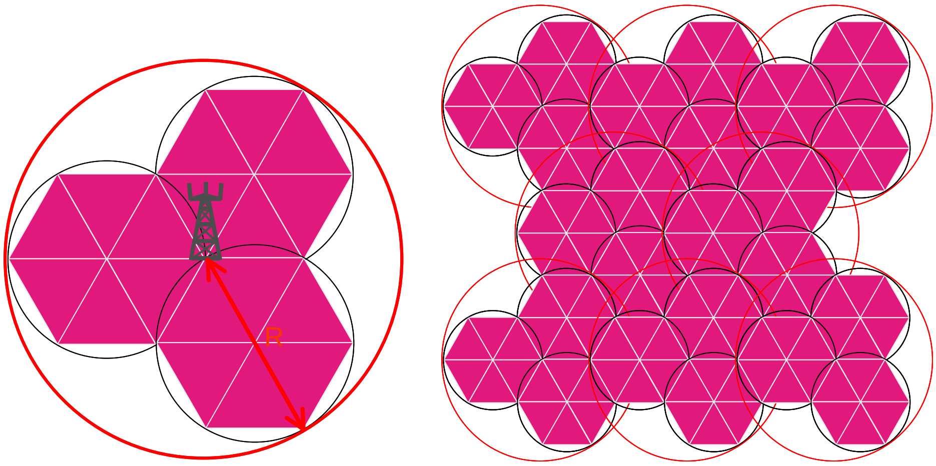


Figure 11.1: Trisectorized cell topology

For the trisectorized antenna on the left of the figure, the coverage area of the three sectors is approximated by a regular hexagon. This approximation makes it easy to show the overall coverage area in the figure on the right, as the hexagons completely cover the plane area without overlap.

The surface of a cell is the following S= 1,95 R^2.

Given the intersite distance D, relationship between D and R is the following: D = 3R/2.

This formula can be used to provide estimates of the different of required sites to cover an area.

11.6.2 Quality of coverage

Given a certain quality criterion, different whole cell coverage probability can be considered. The 90 % coverage probability means that 90 % (for instance) of the cell can receive at least the minimum signal level required for insuring the target C/N value.

Usually 90 % and 95 % overall cell coverage are considered. This corresponds respectively to 77 % and 88 % edge of cell coverage. But other values can be as well required.

Usually, network dimensioning is made for indoor service with handheld at 90 % or 95 % cell coverage, Such indoor coverage probabilities imply de facto in very high outdoor coverage probability, above 99 %.

11.6.3 Network planning tool

A typical numerical network planning tool uses the plotting of the Minimum Received Signal Level.

Starting by the following equation:

$$P_n = NF + 10 * \log (k.T0.B);$$

$$Ps\ min = P_n + C/N.$$

where: NF is the Noise Factor of the receiver;

K: Boltzmann constant;

T_0: receiver temperature;

B: noise bandwidth of the signal;

C/N is the required C/N for a given quality of service, depending on the modulation, coding and propagation channel.

So the minimum received signal input power at antenna input can be written:

$$P_{amin} = P_{s\,min} - G + L_p = C/N - G + L_p + NF + 10 * \log (k * T_0 * B).$$

It must be noticed that power unit is dBm.

Where: G is the antenna gain in dBi and L_p is possible polarization mismatch losses.

On the other hand the basic link budget equation can be written as follows, considering the received C/N:

$$C/N = EIRP_{Tx} - PL - Ms - L_b - L_v + G - NF - 10 * \log (k * T_0 * B) + G_{SFN}$$

Where: $EIRP_{Tx}$: transmitter EIRP;

PL: path loss;

Ms: shadowing margins, following a Log Normal law of standard deviation σ;

L_b: building penetration loss (dB) (if any);

L_v: vehicle entry loss (dB) (if any);

L_p: polarization losses (if any).

$$EIRP_{Tx} = Tx_{power} - (losses) + Tx\ (antenna\ gain)$$

Where: transmitted power is in dBm, losses in dB, and antenna gain in dBi.

Path loss is computed using different path loss models that are included in the network planning tool. There is a variety of PL models (Cost 231 Hata, Welfish Ikegami, Xia-Bertoni) that are usually on field calibrated. For the purpose of global link budgets for rough order of magnitude estimates, the Cost Hata model is used in the link budgets with some calibrated factors.

After some elementary algebra, equation (1) can be written:

$$C/N - G + L_p + NF + 10 * \log (k * T_0 * B) + Ms + L_b + L_v - (G_{SFN}) = EIRP_{Tx} - PL$$

The left part of the equation is the Minimum Received (Rx) Signal level, independent of transmitted power.

Other definitions can be introduced:

- System Gain (SG): SG = $EIRP_{Tx} - P_{amin}$;
- Maximum Allowable Path Loss (MAPL) = SG - Ms - L_b - L_v + G_{SFN}.

Leading to equivalent definition of the Minimum Received (Rx) Signal level.

11.6.4 Network planning factors

11.6.4.1 Shadowing margins

In cellular network radio planning it is usual to distinguish between different environments, while this classification does not exist in EN 302 304 [3].

- Dense Urban (DU): dense cities like Paris.
- Urban (U): cities like immediate Paris suburbs called also "Dense Individual, Mean collective".
- SubUrban (SU): residential areas, called also "Mean individual, mean collective" including as well industrial zones.
- Rural (RU): open areas : small towns and villages, highways, fields.

The last case is not applicable for DVB-SH, as rural areas will be under satellite coverage.

Shadowing margins follow a Lognormal law of standard deviation usually in the range of 6 dB to 10 dB, or even more for some authors [i.11], [i.10]. For network planning exercise, a common value of 8 dB is assumed for urban and suburban areas in [i.11], [i.12], [i.13], [i.14] and [i.15].

Concerning in building penetration losses, there is also a wide range of values. For instance, deep indoor penetration losses are in the range of 18 dB to 21 dB, and sometimes more.

In the following clauses, we give the possible ranges of values, and give an example for network planning purpose.

11.6.4.1.1 Shadowing margins for reception condition A

As proposed value for standard deviation is 8 dB, table 11.11 gives the shadowing margins for various conditions and different cell coverage.

Usual coverage probability values whole cell are 90 % and 95 %.

Table 11.11: Shadowing margins for reception condition A DU/U/SU

Cell Coverage (%)	Shadowing margins (dB)
90	5,5
95	8,7

11.6.4.1.2 In building penetration losses and shadowing margins for reception condition B

In building penetration margins values depend on the different materials and the environments. Table 11.12 presents the possible range of values, and the sued values in the network planning. But the taken values can vary from an operator to another.

Table 11.12: Indoor penetration margins in S band for reception condition B

Environment/B1 or B2	DU/B2	U/B2	SU/B1
Indoor penetration margins range (dB)	18 to 21	15 to 18	13 to 15
Proposed value for network planning	18	15	13
NOTE: For the shadowing margins, the same formula is applied when dealing with the combination of indoor and outdoor signal variations.			

The following standard deviation values are taken:

- outdoor standard deviation: 8 dB for DU, U and SU;
- indoor standard deviation: 6 dB for all cases.

The shadowing margins are provided in table 11.13.

Table 11.13: Shadowing margins in reception conditions B1 and B2

Environment	DU	U	SU
Standard deviation (dB)	10	10	10
Shadowing margins @ 90 % cell coverage	7,6	7,6	7,6
Shadowing margins @ 95 % cell coverage	11,6	11,6	11,6

11.6.4.1.3 Shadowing margins for reception conditions C and D

In reception condition C, the same shadowing margins will be used as for reception condition A, as we are in outdoor reception.

In reception conditions D, an in vehicle penetration loss of 7 dB is required, and the shadowing margins are the same as for condition C.

11.6.4.2 Network architecture optional parameters

This clause concerns some criteria depending on some specific system architecture, mainly SFN gain and antenna diversity gain.

a) The SFN configuration

The used topology is similar to the UMTS one. As the link budgets are computed at cell edge, SFN gain can be taken into account and 4,7 dB gain is used in the link budgets.

b) Receiver antenna diversity

As recalled in clause 11.4.4 b) there will be two cases: single antenna reception and dual antenna reception. The proposed gains are the following:

- 3,5 dB in conditions A, C and D;
- 5,5 dB in conditions B1 and B2.

11.6.5 Link budgets examples for DVB-SH network planning based on minimum received signal level

For cellular network, global estimation uses link budgets for budgetary purposes. As it gives only a rough order of magnitude of the coverage, Cost Hata model is also commonly used below 1 km.

In the link budgets, the power of each transmitter is provided, with different losses (cable losses, diplexer losses if any: between 3 dB and 4 dB overall), the antenna gain (18 dBi per sector in the following examples), resulting in the transmitted EIRP.

Diplexer losses are considered when the DVB-SH transmitter is sharing equipments with other BTS (3G for instance). When stand alone repeater, non diplexer loss is considered.

The results of the link budgets are the Minimum Signal Level for Network Planning (green line), and also the cell radius, and cell surface, given the reception condition, the terminal properties, and the required C/N.

Two modulation and coding schemes are studied:

- QPSK 1/3; and
- 16QAM 1/3.

The required C/N values are extracted from clause A.13 (table A.13.26). In the following clause, we give an example of table, and a synthesis for different examples. The C/N values are coming from laboratory measurements and thus include the implemnation losses, and are given for FER 5 %.

It must be noticed that maximum speed in the tables (conditions C and D) is 50 kmph, below the CN + 3 dB Doppler limit.

The following cases are provided:

- reception condition A: Outdoor pedestrian in dense urban area;
- reception condition B (B1 and B2): indoor pedestrian in dense urban area;
- reception condition C: vehicle at 50 kmph in suburban area;
- reception condition D: in car user at 50 kmph in suburban area.

As in the clause 11.5, we give an examples for reception condition B, and the synthesis for the other cases.

11.6.5.1 Link budgets for Reception condition B2

The studied case is with Handheld Category 3 and the link budgets are computed for 99 % of ESR5(20) fulfilment over 95 % overall coverage in dense urban.

Table 11.14: Link budget for reception condition B2 @ 95 % with Tx power of 12 W and QPSK 1/3

SHA Terrestrial Link Budgets			
Radio interface parameters	**Unit**	**Dense Urban value/No Diversity**	**Dense Urban value/ Diversity**
Channel bandwidth	MHz	5,00	5,00
Frequency	MHz	2 182,50	2 182,50
Mode		2 048,00	2 048,00
Radio interface mod code		QPSK1/3	QPSK1/3
Antenna type		TRI	TRI
1/Guard Interval		8	8
Guard interval		0,125	0,125
Total number of subcarriers		1 705,00	1 705,00
Number of data subcarriers		1 512,00	1 512,00
Tu duration	µs	358,40	0,18
GI duration	µs	44,80	358,40
Ts duration	µs	403,20	44,80
Subcarrier spacing	kHz	2,79	403,20
Useful bandwidth occupancy	MHz	4,76	2,79
Useful data rate at MPEG2-TS interface	Mbit/s	2,490	2,490
Transmitting end		**Tx**	**Tx**
Power amplifier per carrier and sector	W	12,0	12,0
Tx Power at antenna input	dBm	40,8	40,8
Cable loss	dB	3,0	3,0
Diplexer loss	dB	0,7	0,7
Tx antenna gain	dBi	18,0	18,0
EIRP	*dBm*	*55,1*	55,1
ERP	*W*	*323,0*	323,0
Receiving end		**Rx**	**Rx**
Rx antenna gain	dBi	-3,0	-3,0
Polarization mismatch	dB	0,0	0,0
Noise figure	dB	4,5	4,5
Antenna temperature	K	290,0	290,0
Ambient temperature	K	290,0	290,0
kT	dBm/Hz	-174,0	-174,0
Equivalent Rx band	dBm/Hz	66,8	66,8
Rx noise floor	dBm	-102,7	-102,7
Required C/N	dB	2,8	2,8
Rx sensitivity	dBm	-99,9	-99,9
Minimum Rx level at antenna	*dBm*	*-96,9*	-96,9
Minimum Signal Level for Network Planning	**dBm**	-72,01	-77,51
System Gain	**dB**	**152,0**	**152,0**
Margins			
Average building penetration loss	dB	18,0	18,0
Target level of signal penetration		B2	B2
Std shadowing outdoor	dB	8,00	8,00
Std shadowing indoor	dB	6,00	6,00
Std dev of Fading Margin	dB	10,00	10,00
Propagation constant		3,5	3,5
Shadow fading Margin - whole cell	dB	11,6	11,6
Coverage Probability - whole Cell		**95,0 %**	**95,0 %**
SFN network gain	dB	4,7	4,7
Rx gain (antenna diversity)	dB	0,0	5,5
MAPL	**dB**	**127,1**	**132,6**
Cell range computation			
Cost 231-HATA model			
Cell range	**km**	**0,593**	**0,850**
Surface	km^2	**0,69**	**1,41**
H-BTS	m	30,0	30,0
H-MS	m	1,5	1,5
K1	dB	138,08	138,08
K2	dB	35,22	35,22
Kc	dB	-3,0	-3,0
a(CPE)	dB	0,00	0,00

SHA Terrestrial Link Budgets
NOTE 1: Cell radius is 593 m in dense urban environment with single antenna reception.
NOTE 2: Cell radius is 850 m in dense urban environment with dual antenna reception.

11.6.5.2 Overall synthesis

As in the minimum field strength approach, the tables 11.15 and 11.16 provide a synthesis concerning different reception conditions.

Table 11.15: Overall synthesis @ 95 % coverage with single antenna reception

Reception Condition	A in dense urban		B2 in dense urban		C in suburban		D in suburban	
Modulation /coding	QPSK 1/3	16QAM 1/3	QPSK 1/3	16QAM 1/3	QPSK 1/3	16QAM 1/3	QPSK 1/3	16QAM 1/3
Power per sector (W)	7	27	12	32	7	25	7	25
Required C/N (dB)	2,8	9,1	2,8	9,1	2	7,2	2	7,2
Minimum Signal Level for Network planning (dBm)	-93	-87	-72,	-66	-100	-95	-87	-81
Cell range (km)	1,997	1,940	0,593	0,519	5,102	5,202	2,111	2,157

Table 11.16: Overall synthesis @ 95 % coverage with dual antenna reception

Reception Condition	A in dense urban		B2 in dense urban		C in suburban		D in suburban	
Modulation /coding	QPSK 1/3	16QAM 1/3	QPSK 1/3	16QAM 1/3	QPSK 1/3	16QAM 1/3	QPSK 1/3	16QAM 1/3
Power per sector (W)	7	27	12	32	7	25	7	25
Required C/N (dB)	2,8	9,1	2,8	9,1	2	7,2	2	7,2
Minimum Signal Level for Network planning (dBm)	-98	-90	-77	671	-104	-98	-90	-85
Cell range (km)	2,510	2,439	0,850	0,744	6,413	6,552	2,653	2,711

It must be noticed, that its is easy to extrapolate the Minimum Signal Level to other values of C/N.

11.6.6 Use of radio network planning tool example for DVB-SH network design

The Radio Network Planning Tool allows to compute and show coverage maps based on the minimum received power level required to ensure service reception (depends on scattering environments).

The design is based on the use of a Radio Network Planning Tool (RNPT) and a geographical database.

The inputs of the network design are:

- the DVB-SH link budget for a precise configuration of the RNPT;
- the clutter and terrain model databases;
- the minimum signal levels for coverage (design level); and
- eventually the characteristics of existing 2G/3G sites if available for an eventual reuse of the antennas and feeders. (This is not a requirement: it can exist stand alone repeaters.)

The outputs of the network design are:

- coverage maps with received power level or C/N+I level;
- number of sites;
- coverage statistics.

The goal of the network design process with the RNPT is to reach the minimum received power level defined in the link budget (i.e. coverage) for all scattering environments over the area considered in the design). A RNPT allows the designer performing the coverage of one area by positioning transmitting sites within the database.

The Minimum Rx signal level to be considered in the RNPT can be computed as follows:

Minimum Rx signal level = Tx EIRP - Maximum Allowable Path Loss (MAPL)

Using the Link budget formula provided above, it can be further computed from as follows:

Minimum Rx signal level = Min Rx input power + (Shadowing Margin + Building Penetration loss [see note 1]) - (SFN gain [see note 2] + Rx diversity gain)

NOTE 1: If Indoor coverage is targeted. Depending on the situation it can be also in vehicle penetration losses.

NOTE 2: SFN impact can also supported by the tool.

For a highly accurate network design especially in dense urbanized areas, ray-tracing simulations can be considered for a better reliability of the coverage of the network. For these simulations a deterministic propagation model (ray optical propagation model) is used, requiring a 3D building high-resolution database. This combination of ray-tracing and 3D vector database allows a higher accuracy and reliability of the coverage maps.

In some tools, SFN gain is added "manually", other advanced tools integrates already the SFN gain.

11.7 Satellite Network Planning

For the satellite network planning, the following reception categories are considered:

- Reception condition A (outdoor portable) for rural environment.
- Reception condition B1 with domestic gap filler.
- Reception condition C Mobile (roof top antenna).

For satellite network planning purposes, the following satellite to terminal propagation environments will be considered:

1) suburban or rural areas (villages);
2) open area or also rural (open field);
3) rural intermediate tree shadowing (road crossing a forest).

We have excluded from this analysis the urban and rural heavy tree shadowing environments as the satellite is not expected to deliver the DVB-SH service in these kind of environments.

11.7.1 Methodology for Satellite Coverage Calculation

The DVB-SH satellite coverage is a composite of the three environments mentioned earlier: suburban, open area and intermediate tree shadowing. Each environment has distinctive propagation characteristics and therefore DVB-SH physical and link layer performances are expected to be different. Moreover the required quality criteria in terms of ESR5(20) is also dependent on the propagation conditions. For terrestrial channels that can be characterized as Rayleigh channels the requirement is 99 % of ESR5(20) fulfillment; however, the satellite channel presents very different dynamics and therefore this requirement cannot be directly translated. At the present moment, we lack sufficient information to conclude on the required ESR5(20) fulfillment necessary to guarantee a good video quality via satellite. The calculations presented in this clause are based on the assumption that a 90 % ESR5(20) fulfillment is required, however the methodology is considered of general validity.

To get satellite coverage figure over a wide area it is necessary to:

- classify the satellite coverage region according to three environment categories listed above and associate a probability of occurrence $p_E(i) \leq 1,\ i = 1,2,3$ for each of them over the satellite coverage;
- select typical mobile terminal speed for each environment;

- select an ESR5(20) target fulfillment rate per environment $\left[F_{ESR5}(i)\right]_{\min}$, $i=1,2,3$.

As explained earlier, this target fulfillment rate depends on the environment considered. For the calculations on the present document we will consider that for the satellite environments 90 % of ESR5(20) fulfillment is required:

- define the system parameters required to compute the link budgets providing LOS *C/N* and *C/I* values as described in the following clause;
- define the DVB-SH waveform parameters required to perform the simulations as described in clause A.3;
- analyze the ESR5(20) fulfillment rate $F_{ESR5}(i),\ i=1,2,3$ achieved in each of the environments given the selected system and waveform parameters (see clause A.12 for typical results);
- compute the satisfaction index $\chi(i),\ i=1,2,3$ for each satellite coverage environment defined as:

$$\chi(i)=\begin{cases}1 \text{ if } F_{ESR5}(i)\geq\left[F_{ESR5}(i)\right]_{\min}\\ 0 \text{ if } F_{ESR5}(i)<\left[F_{ESR5}(i)\right]_{\min}\end{cases}\quad i=1,2,3$$

- compute the overall satellite coverage C_{SAT} (in % as):

$$C_{SAT}(\%)=100\sum_{i=1}^{3}p_E(i)\ \chi(i);$$

where: $p_E(i)$ is the probability of occurrence of each environment.

Note that the above described methodology assumes that the satellite elevation angle is not changing in an appreciable way over the coverage region. In case of large satellite beams, as it is the case of a global beam covering a large part of the continent, the above described procedure must be extended by splitting the satellite coverage over regions of similar elevation angle.

11.7.2 Basic formulas for Satellite link budgets calculation

As was done for the terrestrial case, the basic formulas for the calculation of the Signal to Noise ratio C/N in the satellite link are recalled hereafter. Satellite link budget is made of two components: uplink (from Gateway to Satellite) and a downlink (from Satellite to User Terminal). The formula for the uplink C/N calculation can be expressed as follows:

$$\left[\frac{C}{N}\right]_U=\frac{1}{kB}EIRP_{GW}\frac{1}{L_U}\left[\frac{G}{T}\right]_{SAT}$$

where: k: Boltzmann's constant;

B: Noise Bandwidth;

$EIRP_{GW}$: Gateway Equivalent Isotropic Radiated Power;

L_U: Uplink losses;

$\left[\frac{G}{T}\right]_{SAT}$: Satellite Antenna Gain over the Satellite Noise Temperature;

$$EIRP_{GW}=\frac{P_{GWTX}\,G_{GW}}{L_T\,L_{FTX}}$$

where: P_{GWTX} is the transmitted power of the Gateway;

G_{GW} is the gateway antenna gain;

L_T is the de-pointing losses; and

L_{FTX} is the feeder loss between the transmitter and the antenna;

$$L_U = L_{FS} L_A$$

where: L_{FS} is the free space losses (see formula below) and L_A is the atmospheric attenuation losses;

$L_{FS} = \left(\frac{4\pi R}{\lambda}\right)^2$ with R defined as the distance between the gateway and the satellite.

In a similar manner, for the downlink the following formula applies:

$$\left[\frac{C}{N}\right]_D = \frac{1}{kB} EIRP_{SAT} \frac{1}{L_D} \left[\frac{G}{T}\right]_{ES}$$

where: $EIRP_{SAT}$: satellite Equivalent Isotropic Radiated Power;

L_D: Downlink losses;

$\left[\frac{G}{T}\right]_{ES}$: Terminal Antenna Gain over the Terminal Noise Temperature;

$EIRP_{SAT} = \frac{P_{SAT} G_{SAT}}{L_T L_{FTX}}$;

where: P_{SAT} is the transmitted power of the Satellite;

G_{SAT} is the satellite antenna gain;

L_T is the de-pointing losses; and

L_{FTX} is the feeder loss between the transmitter and the antenna.

$L_D = L_{FS} L_A$;

where: L_{FS} is the free space losses (same formula as above considering R as the distance between the satellite and the terminal); and

L_A is the atmospheric attenuation losses in the downlink.

Additional to the uplink and downlink C/N components, the link budget needs to take into account the interference contribution. In general, interference is modelled as an additional source of noise:

$$\left[\frac{C}{N+I}\right]^{-1} = \left[\frac{C}{N}\right]^{-1} + \left[\frac{C}{I}\right]^{-1}$$

where: I is the interference component caused by multiple sources of interference: intermodulation (IM), co-channel interference (CC), and adjacent channel interference (ACI):

$$I = I_{IM} + I_{CC} + I_{ACI} .$$

In the following we will assume a transparent GEO satellite system (no modulation or coding on-board) which is limited by the downlink component (from satellite to user mobile terminal). This last assumption is consistent with the DVB-SH scenario where small hand-held or mobile terminals are targeted.

The basic link budget for transparent satellite systems is computed as follows:

$$\left[\frac{C}{N+I}\right]^{-1} = \left[\frac{C}{N+I}\right]_{uplink}^{-1} + \left[\frac{C}{N+I}\right]_{downlink}^{-1}.$$

When the downlink part is driving the link budget, the overall link budget can be approximated by the downlink component:

$$\left[\frac{C}{N+I}\right] \approx \left[\frac{C}{N+I}\right]_{downlink}.$$

11.7.2.1 Satellite link margin

In order to close the link, the available Signal to Noise plus Interference ratio needs to be greater than the required Signal to Noise ratio: for given waveform configuration (MODCOD):

$$\left[\frac{C}{N+I}\right] \geq \left[\frac{C}{N}\right]_{\text{REQUIRED MODCOD}}.$$

This required C/N guarantees the target BER and it depends on the Modulation and Coding and on the channel (in general considered as being AWGN).

Link margin is generally defined as the difference between the available Signal to Noise plus Interference ratio and the required signal to noise ratio:

$$M = \left[\frac{C}{N+I}\right](dB) - \left[\frac{C}{N}\right]_{\text{REQUIRED MODCOD}}(dB).$$

11.7.2.2 Satellite fade margin

When there is a fade in the downlink (cause by shadowing, partially blockage of the signal, etc.) we can expect the nominal signal power to be degraded by an attenuation α.

$$C_{fade} = C.\alpha$$

When the main source of interference comes from the intermodulation and from the co-frequency beams the fade affecting the carrier also affects the interference. Therefore the ratio C/I remains unchanged.

$$\left[\frac{C.\alpha}{I.\alpha}\right] = \left[\frac{C}{I}\right] \text{ if } I = I_{IM} + I_{CC}.$$

And therefore the overall C/(N+I) in the presence of fade affecting both the useful signal and the main sources of interference is as follows:

$$\left[\frac{C}{N+I}\right]_{Fadedownlink}^{-1} = \left[\frac{C\alpha}{N}\right]^{-1} + \left[\frac{C}{I}\right]^{-1}.$$

Now, by fade margin we refer to the value of α so that the link budget can be closed (i.e. is equal or greater than the required):

$$\left[\left[\frac{C\alpha}{N}\right]^{-1} + \left[\frac{C}{I}\right]^{-1}\right]^{-1} = \left[\frac{C}{N}\right]_{\text{REQUIRED MODCOD}}.$$

And therefore, the fade margin (in dB) can be calculated by:

$$M_{fade} = -\alpha(dB) = \left[\frac{C}{N}\right](dB) + 10\log\left[\left[\frac{C}{N}\right]^{-1}_{\text{REQUIRED MODCOD}} - \left[\frac{C}{I}\right]^{-1}\right].$$

When the link budget is not limited by the interference, i.e. when C/I>> required C/(N+I) and C/(N+I)≈C/N; or when the interference is dominated by the components not affected by the same fade as the signal (i.e. adjacent channel interference or co channel interference from other systems), then the fade margin is the same as the link margin:

$$M_{fade} \approx \left[\frac{C}{N}\right](dB) - \left[\frac{C}{N+I}\right]_{REQUIREDMODCOC} \approx M\ .$$

11.7.3 Required margins computation

The above stated formulas can be used to calculate the Signal to Noise ratio received by the terminal in a AWGN channel, but in order to close the link in the propagation channels applicable to DVB-SH, it is required to assess the additional fade endured by the satellite signal due to the mobile propagation channel. Contrary to the terrestrial case, the satellite propagation channel is considered non-frequency selective. Therefore, the fade associated with the satellite to mobile propagation channel will homogeneously affect the satellite signal along all its bandwidth.

This clause analyses the required fade margins for the satellite link. As stated before, these margins depend on the reception conditions and on the environment considered.

For reception condition A, the methodology proposed here considers that the terminal is in direct LOS and in a static situation (neither the terminal nor the surrounding geometry changes or changes are very slow compared to physical and upper layer countermeasures available in DVB-SH). In this static condition, time diversity mechanisms such as the physical layer interleaver or the link layer FEC do not provide any advantage. Fade due to multipath needs to be counteracted via a modulation and coding that allows for a sufficient link margin (i.e. $M_{fade} > Fade$) or via spatial diversity mechanisms such as antenna diversity (the latter not analyzed in the following).

Concerning reception conditions C and D, the mechanisms introduced in DVB-SH to counteract the signal fluctuations in the satellite to mobile channel impact the link margin required to guarantee the service quality (ESR5(20)). If in general the margin is calculated so that the link budget is closed under fade conditions for a certain modulation and coding (and a given BER), in the presence of physical layer FEC interleaver and/or link layer FEC, the quality criteria (ESR5(20)) can be met even if the fade temporarily exceeds the available margin. In fact, within the physical layer or link layer protection time, the fade can exceed the available margin for a period of time as long as this period does not go beyond the correction capabilities of the interleaver or link layer. The correction capabilities depend on the specific implementation of the physical or link layer protection. For any given link margin, the overall signal quality performance depends on this correction capability and on the dynamics of the channel (i.e. kind of environment and terminal speed). Therefore, it is not possible to derive a required fade or link margin value of general validity. The margins proposed in the clause dedicated to mobile reception are based on the available physical layer simulation results for different environments. A methodology will be also proposed on how this results can be used to calculate the availability over the coverage area.

11.7.3.1 Margins required reception condition A

In portable reception conditions addressed in this clause we consider that the terminal is in a static situation where the fade due to multipath can be counteracted only with sufficient link margin.

In the case of reception condition A (i.e. outdoor pedestrian), satellite reception is associated to a rural environment. Direct reception of satellite signal to handheld terminals (terminal categories 2a, 2b) is achieved only when the terminal is in direct Line-of-Sight. In these conditions, the satellite-to-mobile terminal channel can be considered as a Ricean channel with a Carrier to Multipath Ratio that depends on the geometry of the environment. In order to calculate the required margin associated to the Ricean channel, the following calculations apply:

Let us consider a signal *s* with a Signal to Noise Ratio in AWGN channel hereafter called $[C/N]_{AWGN}$ (calculated according to the formulae given in the previous clause). This signal experiences a fade that instantaneously degrades the received *C/N*. This instantaneous *C/N*, hereafter called γ, can be expressed as follows:

$$\gamma = \left[\frac{C}{N}\right]_{AWGN} r^2 \text{ where r is the fading process amplitude.}$$

In the case of a Ricean channel the distribution of the fading amplitude *r* is given by the Rice distribution (see also clause A.7):

$$p_R(r) = \frac{r}{\sigma^2} e^{-\left(r^2+s^2\right)/2\sigma^2} I_0\left(\tfrac{rs}{\sigma^2}\right),$$

where: σ_2 is the variance of either the real or the imaginary components of the multipath; and

s is the amplitude of the LOS signal component.

To find the distribution of the instantaneous *Signal to Noise ratio* γ from the Rice distribution, we use the identity:

$$p_\Gamma(\gamma) = p_R(r)\frac{dr}{d\gamma}.$$

We also define *Rice factor K* as in clause A.7 and we normalize to unity the LOS signal power:

$$K = \frac{s^2}{2\sigma^2};\ s^2 = 1.$$

Therefore, we obtain the following expression for the Probability Density Function (PDF) of the *instantaneous Signal to Noise ratio* γ:

$$p_\Gamma(\gamma) = \frac{K}{\left[\frac{C}{N}\right]_{AWGN}} e^{-K\left(\frac{\gamma}{\left[\frac{C}{N}\right]_{AWGN}}+1\right)} I_0\left(2K\sqrt{\frac{\gamma}{\left[\frac{C}{N}\right]_{AWGN}}}\right).$$

And the CDF of γ can be expressed by:

$$CDF_\Gamma(\gamma) = \int_{-\infty}^{\gamma} p_x(x)\,dx = 1 - Q_1\left(\sqrt{2K}, \sqrt{2K}\sqrt{\frac{\gamma}{\left[\frac{C}{N}\right]_{AWGN}}}\right).$$

where: Q_1 is the Marcum's Q function:

$$Q_1(a,b) = e^{-\left(a^2+b^2\right)/2} \sum_{t=o}^{\infty} \left(\tfrac{a}{b}\right)^t I_t(ab) \quad b > a > 0\,.$$

According to the above expressions, figure 11.2 represents the *cdf* of the degradation of instantaneous Signal to Noise Ratio with respect to the Signal to Noise Ratio in AWGN for different values of Carrier to Multipath ratio *K*.

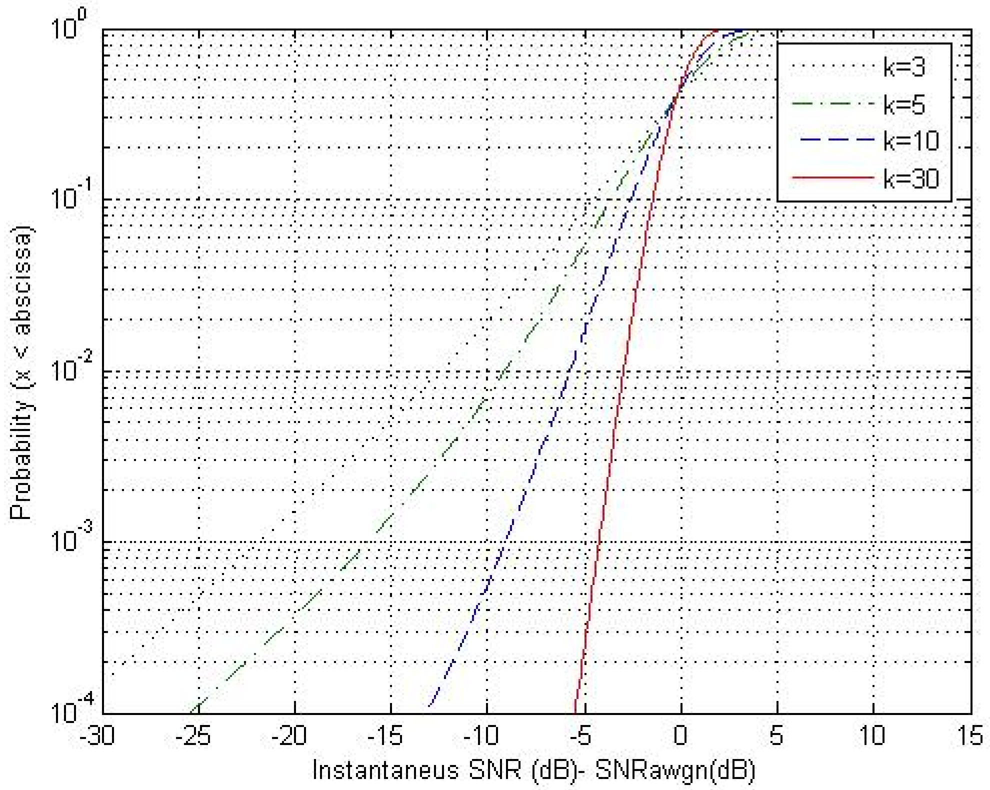


Figure 11.2: Ricean fade power loss CDF for different Rice factors K (in dB)

In a mobile channel, this instantaneous *Signal to Noise ratio* γ, is a time variant variable. When the terminal is in a static situation, as it is the case of category A reception, the Signal to Noise ratio becomes a position dependent variable. The time probability can be mapped into coverage probability and therefore the margins required to reach a certain percentage of the coverage can be calculated using the CDF above. This approach is homologous to the location correction factor used in the terrestrial coverage calculations seen earlier.

$$P_{out} = \Pr\left\{\gamma < \left[\frac{C}{N}\right]_{REQ}\right\} = \Pr\left\{\left[\frac{C}{N}\right]_{AWGN} - \gamma > \left[\frac{C}{N}\right]_{AWGN} - \left[\frac{C}{N}\right]_{REQ}\right\}$$

Where the quantity $\left[\frac{C}{N}\right]_{AWGN} - \left[\frac{C}{N}\right]_{REQ}$ represents the Fade Margin as defined earlier (in the case of no interference).

According to the previous calculations, figure 11.3 summarizes the fade margin required for different values of Rice factor *K* and different percentages of covered area.

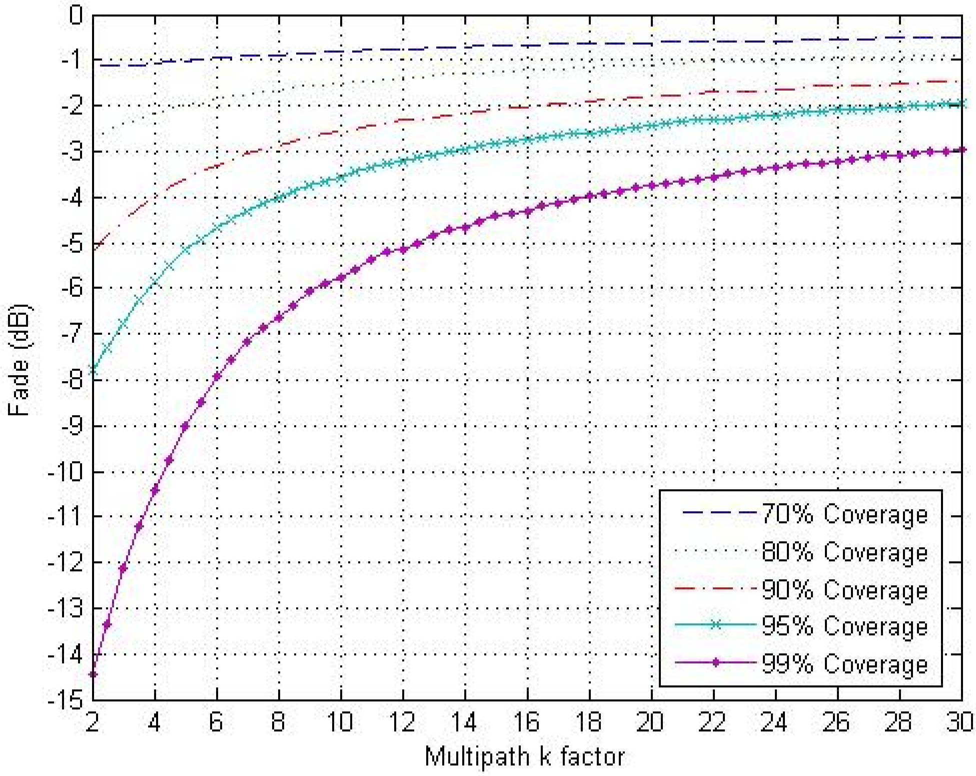


Figure 11.3: Fade margin required for different carrier to multipath ratios and coverage probability

Typical values for the Rice factor *K* range between 5 dB and 10 dB for hand-held terminal. Values can be higher in open areas.

It has to be noted that in the case of static reception conditions, the presence of a long physical layer interleaver or link layer FEC does not provide a real advantage in terms of additional coverage or availability since these mechanisms are effective to counteract the fluctuations of fade that occur only when the terminal is on the move. Instead the presence of mobile terminal antenna diversity may greatly help to reduce the required link margin for this case.

11.7.3.2 Margins required in reception condition C

Mobile reception conditions for satellite refer to reception condition C for the rural environment. Mobile speeds analyzed in this clause cover also low speeds (3 kmph).

The Empirical Roadside Model briefly described in clause 4 allows determining the shadowing endured by the satellite link for a certain coverage probability in an equivalent way as the location correction factor does for the terrestrial network planning. However, as discussed previously, DVB-SH implements a long physical layer interleaver or/and a Link Layer FEC that alters the amount of shadowing that can be endured by the satellite link (without causing outage) when the terminal is moving with a certain speed. Therefore, this model cannot be directly translated into a required shadowing margin for our network planning calculations.

In order to take into account the time diversity introduced by the DVB-SH physical and link layer, the network planning needs to rely on the simulations of the physical and link layer over a channel model with representative fade/interfade durations. In the case of DVB-SH, many simulations have been performed using the time series generated with the well-known Perez-Fontan LMS model [16]. Clause A.12 presents the results obtained for different environments, speeds and fade margins. These results give an indication of the achievable ESR5(20) criteria in different environments for a given LOS *C/N*. The required margin for satellite reception is heavily dependent on the satellite elevation angle, the location, the type of environment, the mobile terminal speed and on the receiver physical and link layer configuration (overall redundancy, length and configuration of the physical layer interleaver and/or link layer FEC, etc.). Therefore, the tables in clause A.12 present a set of representative DVB-SH cases but cannot be considered to have general or universal applicability.

Clause A.12 simulations are organized by environments that are considered representative of DVB-SH usage (Intermediate Tree Shadowing, Suburban, etc.). Network planners need to take into consideration the percentage of time where a DVB-SH user is under each of the analyzed environments in order to derive the final availability.

11.7.4 Antenna diversity Gain

The use of antenna diversity in the receiver can improve the link budget and allow to counteract multipath fading. In combination with Maximum Ratio Combining, the C/N gain of the antenna diversity with 2 antennas has been estimated between 3 dB to 5 dB both in laboratory tests and in on field trials.

11.7.5 Link budget examples

This clause presents several examples of satellite link budgets calculated following the formulae given above. It also presents the available fade margin for different physical layer configurations.

The selected link budget parameters are chosen to be representative of a DVB-SH GEO satellite system, although the link budget parameters (EIRP and interference contribution, etc.) always depend on the specific system configuration.

For these link budget calculations two satellite EIRP values have been selected: 63 dBW and 68 dBW which are representative of low and medium power satellites. It has also been considered an overall downlink C/I component of 14 dB, where the interference contribution comes from the inter-modulation noise and from the co-channel intra system interference. Both interference components are affected by fade in the same amount as the signal and therefore the distinction between fade and link margin defined in clause 11.7.2.2 Satellite fade margin applies.

In the following link budgets, there are four columns representative of the four different categories of terminals defined in clause 10. Main terminal characteristics (polarization, antenna gain, noise figure) are extracted from clause 10.

11.7.5.1 Link budget for SH-A

Table 11.17 represents the link budget for SH-A configuration in the case of a satellite with an effective EIRP towards the analyzed beam of 63 dBW and a 5 MHz channel. Table 11.18 shows the same link budget but considering antenna diversity of order 2 in the receiver.

Table 11.17: SH-A link budget for 5 MHz channel and 63 dBW EIRP

SH-A, 5 MHZ channel					
Physical Layer	**Unit**	**Handheld category 3**	**Portable category 2b (see note)**	**Portable category 2a**	**Vehicular category 1**
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Up-link C/(N+I)	dB	19,5	19,5	19,5	19,5
Satellite transmission					
Tx frequency	GHz	2,2	2,2	2,2	2,2
EIRP effective/beam	dBW	**63,0**	**63,0**	**63,0**	**63,0**
Sat to RX Terminal propagation					
Free Space Loss	dB	190,7	190,7	190,7	190,7
Atmospheric attenuation	dB	0,5	0,5	0,5	0,5
Total attenuation	dB	191,2	191,2	191,2	191,2
Terminal Rx reception					
Terminal G/T	dB/K	-32,1	-29,1	-24,9	-21,0
Polarization losses	dB	3,0	3,0	3,0	0,0
Downlink Results					
C/N down-link	dB	-1,8	1,2	5,5	12,3
C/I down-link total	dB	14,0	14,0	14,0	14,0
Down-link C/(N+I)	dB	-1,9	1,0	4,9	10,1
Total C/(N+I)	dB	-1,9	0,9	4,7	9,6
NOTE: The results of this link budget are also representative of a handheld terminals with circular polarization.					

Table 11.18: SH-A link budget for 5 MHz channel and 63 dBW EIRP and antenna diversity

DVB-SH-A, 5 MHZ channel					
	Unit	**Handheld Category 3**	**Handheld category 2b**	**Portable category 2a**	**Vehicular category 1**
OFDM noise bandwidth	MHz	4.76	4.76	4.76	4.76
Up-link C/(N+I)	dB	19.5	19.5	19.5	19.5
Satellite transmission					
Tx frequency	GHz	2.2	2.2	2.2	2.2
OBO	dB	2.0	2.0	2.0	2.0
EIRP effective/beam	dBW	63.0	63.0	63.0	63.0
Sat to RX Terminal propagation					
Free Space Loss	dB	190.7	190.7	190.7	190.7
Atmospheric attenuation	dB	0.5	0.5	0.5	0.5
Total attenuation	dB	191.2	191.2	191.2	191.2
Terminal Rx reception					
Terminal G/T	dB/K	-32.1	-29.1	-24.9	-21.0
Polarisation losses	dB	3	3	3	0
Antenna Diversity Gain	dB	3	3	3	3
Downlink Results					
C/N down-link	dB	1.2	4.2	8.4	15.3
C/I down-link total	dB	14.0	14.0	14.0	14.0
Down-link C/(N+I)	dB	1.0	3.8	7.4	11.6
Total C/(N+I)	dB	0.9	3.6	7.1	10.9

This link budget can be closed for different physical layer configurations, given different fade margins. In order to close the link, the required C/N and implementation margin values shown in clauses 7.2.2.6 and 10.4.3 respectively are chosen. Tables 11.19 to 11.21 show the link budget closure for three different physical layer configuration (Code rate 1/2, 1/3 and 1/5) for the single antenna case.

Table 11.19: Link budget closure for code rate 1/2 at physical layer and 63 dBW EIRP

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/2 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1 705	1 705	1 705	1 705
Number of useful carriers		1 512	1 512	1 512	1 512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/2	1/2	1/2	1/2
Useful bit rate at physical layer	Mb/s	3,38	3,38	3,38	3,38
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Required C/N at physical layer at BER 10^-5	dB	1,4	1,4	1,4	1,4
Implementation Loss in AWGN	dB	1,1	1,1	1,1	1,1
Received C/(N+I)	dB	-1,9	0,9	4,7	9,6
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**-4,4**	**-1,6**	**2,2**	**7,1**
Available Fade Margin at Physical Layer	**dB**	**-4,7**	**-1,7**	**2,5**	**9,4**

Table 11.20: Link budget closure for code rate 1/3 at physical layer

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/3 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1 705	1 705	1 705	1 705
Number of useful carriers		1 512	1 512	1 512	1 512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/3	1/3	1/3	1/3
Useful bit rate at physical layer	Mb/s	2,25	2,25	2,25	2,25
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Required C/N at physical layer at BER 10^-5	dB	-0,9	-0,9	-0,9	-0,9
Implementation Loss in AWGN	dB	1,1	1,1	1,1	1,1
Received C/(N+I)	dB	-1,9	0,9	4,7	9,6
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**-2,12**	**0,73**	**4,54**	**9,40**
Available Fade Margin at Physical Layer	**dB**	**-2,22**	**0,78**	**5,02**	**11,87**

Table 11.21: Link budget closure for code rate 1/5 at physical layer

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/5 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1 705	1 705	1 705	1 705
Number of useful carriers		1 512	1 512	1 512	1 512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/5	1/5	1/5	1/5
Useful bit rate at physical layer	Mb/s	1,35	1,35	1,35	1,35
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Required C/N at physical layer at BER 10^-5	dB	-3,6	-3,6	-3,6	-3,6
Implementation Loss in AWGN	dB	1,1	1,1	1,1	1,1
Received C/(N+I)	dB	-1,9	0,9	4,7	9,6
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**0,58**	**3,43**	**7,24**	**12,10**
Available Fade Margin at Physical Layer	**dB**	**0,60**	**3,59**	**7,83**	**14,68**

According to previous link budgets, only code rate 1/5 allows closing the link for the handheld terminals. With antenna diversity, the link budget can be closed for all terminals with code rate 1/3 as seen in table 11.22. In the most performing terminals, the antenna diversity provides also a clear advantage in terms of fading margin. It is worth to note that antenna also counteracts the multipath fade (if antennas are sufficiently separated) therefore the gain assumed in this link budgets is clearly conservative.

Table 11.22: Link budget closure for code rate 1/3 at physical layer

Physical Layer		Handheld Category 3	Handheld category 2b	Portable category 2a	Vehicular category 1
		CODE 1/3 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1705	1705	1705	1705
Number of useful carriers		1512	1512	1512	1512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/3	1/3	1/3	1/3
Useful bit rate at physical layer	Mb/s	2.25	2.25	2.25	2.25
OFDM noise bandwidth	MHz	4.76	4.76	4.76	4.76
Required C/N at physical layer at BER 10^-5	dB	-0.9	-0.9	-0.9	-0.9
Implementation Loss in AWGN	dB	1.1	1.1	1.1	1.1
Received C/N	dB	0.9	3.6	7.1	10.9
LOS Margin at Physical Layer wrt AWGN	**dB**	**0.7**	**3.4**	**6.9**	**10.7**
Available Fade Margin at Physical Layer	**dB**	**0.8**	**3.8**	**8.0**	**14.8**

In the same way, the tables 11.23 to 11.26 apply for the a medium power satellite radiating 68 dBW towards the beam.

Table 11.23: SH-A link budget for 5 MHz channel and 68 dBW EIRP

Physical Layer	**Unit**	**DVB-SH-A, 5 MHZ channel**			
		Handheld category 3	**Portable category 2b**	**Portable category 2a**	**Vehicular category 1**
FDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Up-link C/(N+I)	dB	19,5	19,5	19,5	19,5
Satellite transmission					
Tx frequency	GHz	2,2	2,2	2,2	2,2
EIRP effective/beam	dBW	68,0	68,0	68,0	68,0
Sat to RX Terminal propagation					
Free Space Loss	dB	190,7	190,7	190,7	190,7
Atmospheric attenuation	dB	0,5	0,5	0,5	0,5
Total attenuation	dB	191,2	191,2	191,2	191,2
Terminal Rx reception					
Terminal G/T	dB/K	-32,1	-29,1	-24,9	-21,0
Polarization losses	dB	3,0	3,0	3,0	0,0
Downlink Results					
C/N down-link	dB	3,2	6,2	10,4	17,3
C/I down-link total	dB	14,0	14,0	14,0	14,0
Down-lin k C/(N+I)	dB	2,8	5,5	8,8	12,3
Total C/(N+I)	dB	2,8	5,4	8,5	11,6

Table 11.24: Link budget closure for code rate 1/2 at physical layer

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/2 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1 705	1 705	1 705	1 705
Number of useful carriers		1 512	1 512	1 512	1 512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/2	1/2	1/2	1/2
Useful bit rate at physical layer	Mb/s	3,38	3,38	3,38	3,38
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Required C/N at physical layer at BER 10^-5	dB	1,4	1,4	1,4	1,4
Implementation Loss in AWGN	dB	1,1	1,1	1,1	1,1
Received C/N	dB	2,76	5,35	8,49	11,57
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**0,3**	**2,9**	**6,0**	**9,1**
Available Fade Margin at Physical Layer	**dB**	**0,3**	**3,3**	**7,5**	**14,4**

Table 11.25: Link budget closure for code rate 1/3 at physical layer

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/3 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1 705	1 705	1 705	1 705
Number of useful carriers		1 512	1 512	1 512	1 512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/3	1/3	1/3	1/3
Useful bit rate at physical layer	Mb/s	2,25	2,25	2,25	2,25
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Required C/N at physical layer at BER 10^-5	dB	-0,9	-0,9	-0,9	-0,9
Implementation Loss in AWGN	dB	1,1	1,1	1,1	1,1
Received C/N	dB	2,76	5,35	8,49	11,57
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**2,56**	**5,15**	**8,29**	**11,37**
Available Fade Margin at Physical Layer	**dB**	**2,76**	**5,75**	**9,99**	**16,84**

Table 11.26: Link budget closure for code rate 1/5 at physical layer

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/5 at PL			
DVB-SH mode		2K	2K	2K	2K
Number of carriers		1 705	1 705	1 705	1 705
Number of useful carriers		1 512	1 512	1 512	1 512
Useful symbol duration	msec	358	358	358	358
Guard Time fraction		1/4	1/4	1/4	1/4
Modulation Order		QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate		1/5	1/5	1/5	1/5
Useful bit rate at physical layer	Mb/s	1,35	1,35	1,35	1,35
OFDM noise bandwidth	MHz	4,76	4,76	4,76	4,76
Required C/N at physical layer at BER 10^-5	dB	-3,6	-3,6	-3,6	-3,6
Implementation Loss in AWGN	dB	1,1	1,1	1,1	1,1
Received C/N	dB	2,76	5,35	8,49	11,57
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**5,26**	**7,85**	**10,99**	**14,07**
Available Fade Margin at Physical Layer	**dB**	**5,57**	**8,56**	**12,80**	**19,65**

11.7.5.2 Link budget for SH-B

Table 11.27 represents the link budget for SH-B configuration in the case of a satellite with an effective EIRP towards the analyzed beam of 63 dBW and a 5 MHz channel.

Table 11.27: SH-B link budget for 5 MHz channel and 63 dBW EIRP

Physical Layer	Unit	DVB-SH-B, 5 MHZ channel			
		Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
TDM occupied bandwidth	MHz	4,89	4,89	4,89	4,89
Up-link C/(N+I)	dB	20,0	20,0	20,0	20,0
Satellite transmission					
Tx frequency	GHz	2,2	2,2	2,2	2,2
EIRP effective/beam	dBW	63,0	63,0	63,0	63,0
Sat to RX Terminal propagation					
Free Space Loss	dB	190,7	190,7	190,7	190,7
Atmospheric attenuation	dB	0,5	0,5	0,5	0,5
Total attenuation	dB	191,2	191,2	191,2	191,2
Terminal Rx reception					
Terminal G/T	dB/K	-32,1	-29,1	-24,9	-21,0
Polarization losses	dB	3,0	3,0	3,0	0,0
Downlink Results					
C/N down-link	dB	-1,3	1,7	5,9	12,8
C/I down-link total (adjacent, co-channel, NPR, etc.)	dB	14,0	14,0	14,0	14,0
Down-link C/(N+I)	dB	-1,4	1,5	5,3	10,3
Total C/(N+I)	**dB**	-1,4	**1,4**	**5,2**	**9,9**

As before, this link budget can be closed for different physical layer configurations. Tables 11.28 to 11.30 show the link budget closure for three different physical layer configuration (Code rate 1/2, 1/3 and 1/5) according to the thresholds and implementation margins shown in clause 7 for the TDM case.

Table 11.28: SH-B link budget closure for code rate 1/2 at physical layer and 63 dBW EIRP

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/2 at PL			
TDM Roll-off	-	0,15	0,15	0,15	0,15
TDM Symbol rate	-	4,25	4,25	4,25	4,25
Modulation	-	QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate	**-**	**1/2**	**1/2**	**1/2**	**1/2**
Useful bit rate at physical layer (minus 8 % of pilots)	Mb/s	3,91	3,91	3,91	3,91
TDM noise bandwidth	MHz	4,25	4,25	4,25	4,25
Required C/N at physical layer at BER 10^-5 in AWGN	dB	1,1	1,1	1,1	1,1
Implementation Loss in AWGN channel	dB	0,5	0,5	0,5	0,5
Received C/N	dB	**-1,4**	**1,4**	**5,2**	**9,9**
Link budget results					
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**-3,0**	**-0,2**	**3,6**	**8,3**
Available Fade Margin at Physical Layer	**dB**	**-3,2**	**-0,2**	**4,0**	**10,9**

Table 11.29: SH-B link budget closure for code rate 1/3 at physical layer and 63 dBW EIRP

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/3 at PL			
TDM Roll-off	-	0,15	0,15	0,15	0,15
TDM Symbol rate	-	4,25	4,25	4,25	4,25
Modulation	-	QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate	**-**	**1/3**	**1/3**	**1/3**	**1/3**
Useful bit rate at physical layer (minus 8 % of pilots)	Mb/s	2,61	2,61	2,61	2,61
TDM noise bandwidth	MHz	4,25	4,25	4,25	4,25
Required C/N at physical layer at BER 10^-5 in AWGN	dB	-1,2	-1,2	-1,2	-1,2
Implementation Loss in AWGN channel	dB	0,5	0,5	0,5	0,5
Received C/N	dB	**-1,4**	**1,4**	**5,2**	**9,9**
Link budget results					
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**-0,7**	**2,1**	**5,9**	**10,6**
Available Fade Margin at Physical Layer	**dB**	**-0,8**	**2,2**	**6,5**	**13,3**

Table 11.30: SH-B link budget closure for code rate 1/5 at physical layer and 63 dBW EIRP

Physical Layer	Unit	Handheld category 3	Portable category 2b	Portable category 2a	Vehicular category 1
		CODE 1/5 at PL			
TDM Roll-off	-	0,15	0,15	0,15	0,15
TDM Symbol rate	-	4,25	4,25	4,25	4,25
Modulation	-	QPSK	QPSK	QPSK	QPSK
Physical Layer Coding rate	**-**	**1/5**	**1/5**	**1/5**	**1/5**
Useful bit rate at physical layer (minus 8 % of pilots)	Mb/s	1,56	1,56	1,56	1,56
TDM noise bandwidth	MHz	4,25	4,25	4,25	4,25
Required C/N at physical layer at BER 10^-5 in AWGN	dB	-3,9	-3,9	-3,9	-3,9
Implementation Loss in AWGN channel	dB	0,5	0,5	0,5	0,5
Received C/N	dB	**-1,4**	**1,4**	**5,2**	**9,9**
Link budget results					
LOS Margin at Physical Layer w.r.t. AWGN	**dB**	**2,0**	**4,8**	**8,6**	**13,3**
Available Fade Margin at Physical Layer	**dB**	**2,0**	**5,0**	**9,2**	**16,1**

The same approach can also be repeated for a satellite radiating 68 dBW per beam, obtaining 5 dB more margin for the same physical layer considered previously.

As in SHA link budget example, same calculation regarding the antenna diversity gain can also be applied in this case.

11.7.5.3 Implementation losses and link budgets

In all previous link budgets tables, the used implementation losses are the typical values presented in table 10.1. The same table presents "Possible Receiver Implementation Losses" that are 0,25 dB to 0,6 dB below the typical values. For instance, in OFDM QPSK, implementation losses are 0,5 dB instead of 1,1 dB for typical receivers. The use of these implementation losses would increase consequently the obtained fade margins.

11.7.5.4 Example of availability calculation for reception condition A

The fade margin derived in the above link budget can be used to calculate the availability obtained for portable category of reception in a given area conditioned to the multipath factor over that area. Tables 11.31 and 11.32 show the availability obtained for the different physical layer configurations and for the different terminal types. The availability results are derived by matching the fade margin obtained in each of the link budgets with the fade margin required for a given multipath factor shown in figure 11.2.

Table 11.31: Availability for reception condition A and 63 dBW of EIRP

EIRP=63 dBW SH-A	QPSK 1/2		QPSK 1/3		QPSK 1/5	
	Fade margin available	**Availability**	**Fade margin available**	**Availability**	**Fade margin available**	**Availability**
-4,7 dB		Out of Coverage	-2,2 dB	Out of Coverage	0,6 dB	70 % (if *K*>15 dB)
Terminal Category 2b (see note)	-1,7 dB	Out of Coverage	0,8 dB	70 % (*K*>12 dB)	3,6 dB	80 % 90 %(if *K*>10 dB)
Terminal Category 2a	2,5 dB	80 % 90 % (if *K*>10 dB)	5,0 dB	90 % 95 % (if *K*>5 dB)	7,8 dB	95 % 99 % (if *K*>6 dB)
NOTE: Also representative of terminal category 3 (handheld) with circular polarized antenna.						

Table 11.32: Availability for reception condition A and 68 dBW of EIRP

EIRP=68 dBW SH-A	QPSK 1/2		QPSK 1/3		QPSK 1/5	
	Fade margin available	**Availability**	**Fade margin available**	**Availability**	**Fade margin available**	**Availability**
Terminal Category 3	0,3 dB	Coverage only in AWGN	2,8 dB	80 % 90 % (if K>10 dB)	5,6 dB	90 % 95 % (if K>6 dB)
Terminal Category 2b	3,3 dB	80 % 90 % (K>6 dB)	5,75 dB	90 % 95 %(K>5 dB)	8,6 dB	95 % 99 %(if K>5 dB)
Terminal Category 2a	7,5 dB	95 % 99 % (if K>7 dB)	10 dB	95 % 99 % (if K>4 dB)	12,8 dB	95 % 99 % (if K>3 dB)

11.7.5.5 Example of availability calculation for reception condition C

The link budgets shown above have been used in the physical layer simulations presented in Clause A. These simulations give quality in terms of ESR5(20) obtained in a set of scenarios representative of DVB-SH satellite to mobile channel. Tables 11.33 to 11.36 summarizes the percentage of ESR5(20) fulfilment for the Intermediate Tree Shadowing (ITS) and SubUrban (SU) environment. The terminal considered is a vehicular terminal (category 1) with a long physical layer interleaver.

Table 11.33: ESR5(20) fulfilment for SH-A for the category 1 (vehicular) terminal in reception condition C

SH-A Long physical layer interleaver 63 dBW EIRP Terminal category 1 (Vehicular)			**Physical Layer**			
			QPSK 1/3		16QAM 1/5	
			Available Fade Margin = 11,9 dB (After implementation losses)		Available Fade Margin = 9,3 dB (After implementation losses)	
			Uniform	Uniform-Late	Uniform	Uniform-Late
Mobile Channel	ITS	50 kmph	100 %	99 %	95,5 %	86 %
	SU	50 kmph	100 %	100 %	100 %	99 %

Table 11.34: ESR5(20) fulfilment for SH-A in reception condition A with terminal category 2b with circular polarized antenna

SH-A Long physical layer interleaver 68 dBW EIRP Terminal category 2b with linear polarized antenna and 3 with circular polarized antenna			Physical Layer			
			QPSK 1/3		16QAM 1/5	
			Available Fade Margin = 5,0 dB (After implementation losses)		Available Fade Margin = 2,2 dB (After implementation losses)	
			Uniform	Uniform-Late	Uniform	Uniform-Late
Mobile Channel	SU	3 kmph	84 %	75 %	51 %	42 %

Table 11.35: ESR5(20) fulfilment for SH-B for Category 1 (vehicular) terminal in reception condition C

SH-B Long physical layer interleaver 63 dBW EIRP Terminal category 1 (Vehicular)			Physical Layer			
			QPSK 1/3		8PSK 2/9	
			Available Fade Margin = 13,3 dB (After implementation losses)		Available Fade Margin = 11,2 dB (After implementation losses)	
			Uniform	Uniform-Late	Uniform	Uniform-Late
Mobile Channel	ITS	50 kmph	97 %	96 %	95,3 %	93,7 %
	SU	50 kmph	98,2 %	97,3 %	99 %	96,8 %

Table 11.36: ESR5(20) fulfilment for SH-B in reception condition A with terminal category 2b with circular polarized antenna

SH-A Long physical layer interleaver 68 dBW EIRP Terminal category 2b with linear polarized antenna and 3 with circular polarized antenna			Physical Layer			
			QPSK 1/3		8PSK 2/9	
			Available Fade Margin = 6,5 dB (After implementation losses)		Available Fade Margin = 4,4 dB (After implementation losses)	
			Uniform	Uniform-Late	Uniform	Uniform-Late
Mobile Channel	SU	3 kmph	77,9 %	72,6 %	75,2 %	73,5 %

11.7.6 Methodology for Satellite Coverage Calculation for reception condition C

The DVB-SH satellite coverage is a composite of several environments: urban, suburban, open area, heavy tress shadowing and intermediate tree shadowing. Each environment has distinctive propagation characteristics and therefore DVB-SH physical and link layer performances are expected to be different. Moreover the required quality in terms of ESR5(20) is also dependent on the propagation conditions. For terrestrial channels that can be characterized as Rayleigh channels the requirement is 99 % of ESR5(20) fulfilment; however, the satellite channel presents very different dynamics and therefore this requirement cannot be directly applied. At the present moment, we lack sufficient information to conclude on the required ESR5(20) fulfilment necessary to guarantee a good video quality via satellite. The calculations presented in this clause are based on the assumption that a 90 % ESR5(20) fulfilment is required, however the methodology is considered of general validity.

In the following, a methodology to calculate the satellite coverage area served with a given quality is presented. This methodology has been developed and used in the framework of the ESA contract "role of satellite in 4G mobile networks" [i.43]. To get satellite coverage figure over a wide area it is necessary to:

- Classify the satellite coverage region according to the environment categories listed above. In order to do so, the topographic data provided by the Global Land Cover Facility (GLCF, see [i.43]) gives the type of land cover of each location under the coverage. It enables to attribute a specific user environment to every location.
- Using the topographical data, associate a probability of occurrence $p_E(i) \leq 1,\ i = 1, 2, 3$ for each of the environments over the satellite coverage.
- Select typical mobile terminal speed for each environment.

- Select an ESR5(20) target fulfilment rate per environment $\left[F_{ESR5}(i)\right]_{\min}$, $i=1,2,3$.

As explained earlier, this target fulfilment rate depends on the environment considered. For the calculations on the present document we will consider that for the satellite environments 90 % of ESR5(20) fulfilment is required,

- define the system parameters required to compute the link budgets providing LOS *C/N* and *C/I* values as described in the following clause;
- define the DVB-SH waveform parameters required to perform the simulations as described in clause A.3;
- analyze the ESR5(20) fulfillment rate $F_{ESR5}(i),\ i=1,2,3$ achieved in each of the environments given the selected system and waveform parameters (see clause A.12 for typical results);
- compute the satisfaction index $\chi(i),\ i=1,2,3$ for each satellite coverage environment defined as:

$$\chi(i)=\begin{cases}1 \text{ if } F_{ESR5}(i)\geq\left[F_{ESR5}(i)\right]_{\min}\\ 0 \text{ if } F_{ESR5}(i)<\left[F_{ESR5}(i)\right]_{\min}\end{cases}\quad i=1,2,3$$

- compute the overall satellite coverage C_{SAT} (in % as):

$$C_{SAT}(\%)=100\sum_{i=1}^{3}p_E(i)\ \chi(i)\ ;$$

where $p_E(i)$ is the probability of occurrence of each environment.

Note that the above described methodology assumes that the satellite elevation angle is not changing in an appreciable way over the coverage region. In case of large satellite beams, as it is the case of a global beam covering a large part of the continent, the above described procedure must be extended by splitting the satellite coverage over regions of similar elevation angle.

11.7.7 Example of satellite coverage calculation

Previous clauses have explained the link budget calculations and the margins required for different reception conditions and different terminals. It has to be noted that physical layer simulations refer to one specific environment (Intermediate tree shadowing or suburban) and characterized the quality of DVB-SH physical layer (with a specific code rate and interleaver length) under that environment. In this context, the environments are those defined by the LMS model used for the physical layer simulations [16] and [17].

In this example, we consider a satellite coverage area composed of 6 linguistic beams covering 8 service zones: Belgium, France, Germany, Italy, Poland, Portugal, Spain and UK.

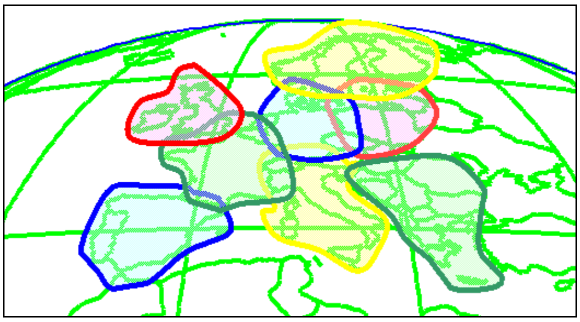


Figure 11.4: Example of Satellite Coverage area

A world land cover database based on the measurements of the Advanced Very High Resolution Radiometer (AVHRR) from the NOAA polar-orbiting satellite is used to determined the distribution of the different environments under the satellite coverage. Here, the smallest resolution is used: (0,5' x 0,5'), (around 1 km² at NADIR). A 0,5' x 0,5' cell defines one location and is characterized by a homogeneous land cover. These database provides 14 classes of land environments. Table 11.37 provides a conservative correspondence matrix between the land cover classes of the AVHRR database and the LMS propagation environments used for the physical layer assessment.

Table 11.37: land cover classes and LMS Perez-Fontan Classes

AVHRR land cover classes	***Pérez-Fontan classes***
Evergreen needle leaf forest	Heavy Tree Shadowing (HTS)
Evergreen broadleaf forest	HTS
Deciduous needle leaf forest	HTS
Deciduous broadleaf forest	HTS
Mixed forest	HTS
Woodland	Intermediate Tree Shadowing (ITS)
Wooded grassland	ITS
Closed shrub land	Open
Open shrub land	Open
Grassland	Open
Cropland	Open
Bare ground	Open
Urban and built-up	Urban/suburban

Figure 11.5 shows the percentage of coverage per environment considering the data from AVHRR and the classification done in table 11.37 in each of the 8 service zones.

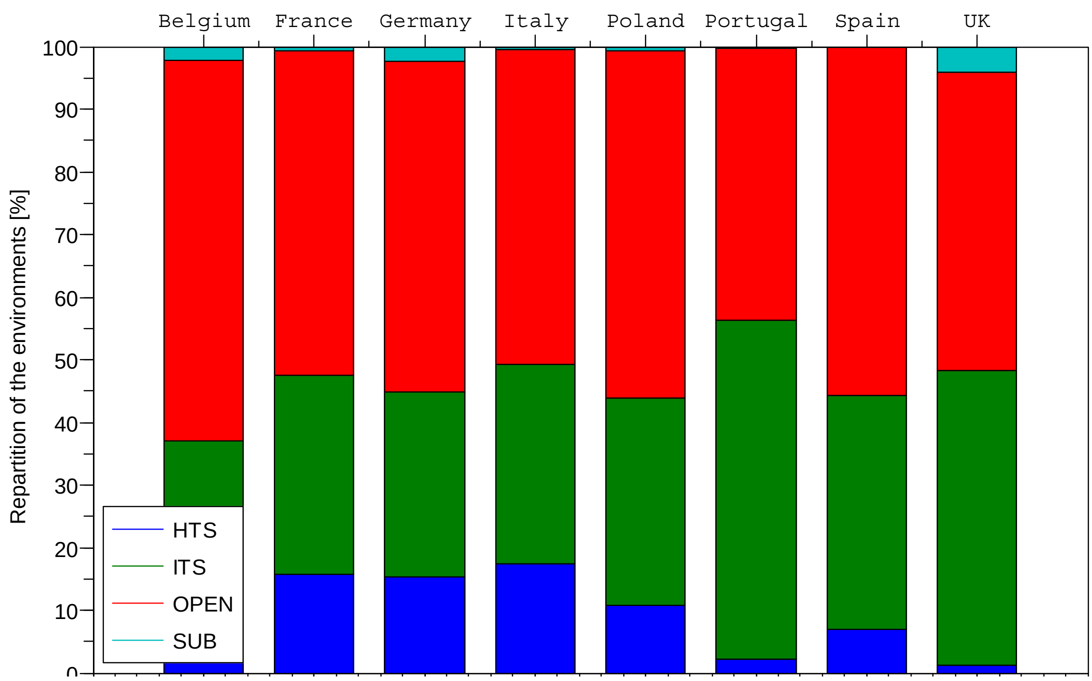


Figure 11.5: Distribution of environments in the satellite coverage area

According to figure 11.5, most of the coverage area can be characterised as open or Intermediate tree shadowing environment. Suburban and urban environments represent less than 4 % of the area. However, the concentration of population in those areas make the analysis of the suburban environment relevant for the satellite. Contrary to that, the Heavy Tree Shadowing environment can represent up to 15 % of the coverage in certain service zones, however, the population is likely to be rather scarce in those areas.

The satellite antenna diagram in this example with a resolution of 0,2° x 0,2°) has been used to compute both the antenna gain and the carrier-to-interference ratio (C/I) due to the frequency re-use (see figures 11.6 and 11.7).

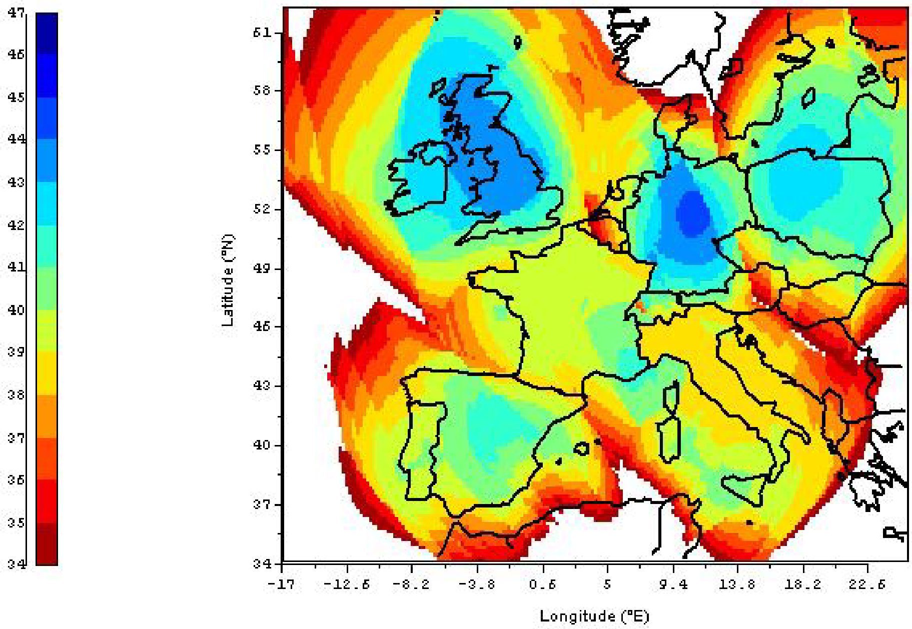


Figure 11.6: Example of Satellite Antenna Gain over the coverage area

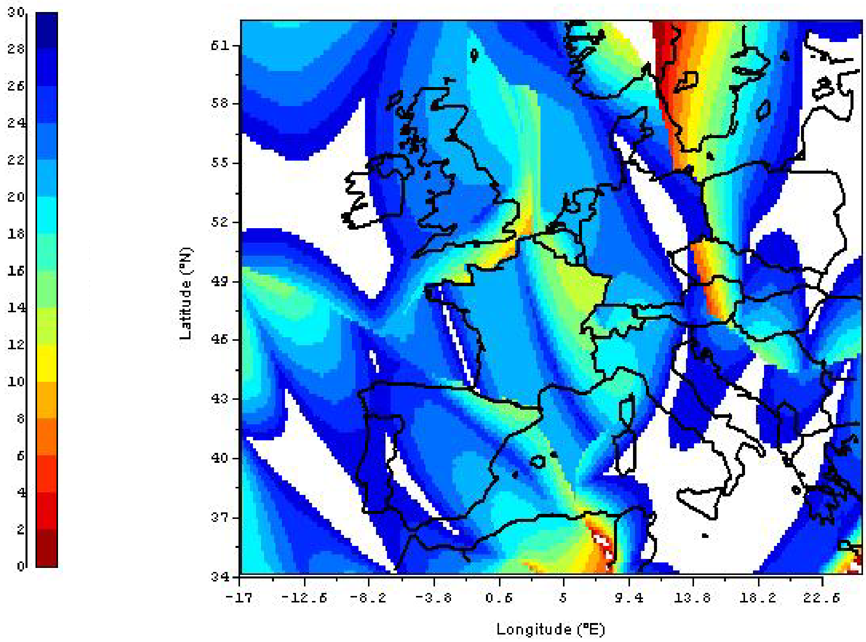


Figure 11.7: Example of Satellite Antenna C/I for a frequency reuse 1:2

Link budgets have been computed using the same resolution as the antenna diagram. The C/(N+I) values over the coverage obtained from the link budgets can be translated into service availability (ESR5) using the environmental data described earlier. The results of the link budgets are interpolated onto a given service zone with a 0,5' x 0,5' resolution and associated with the type of propagation environment. To each environment, type of modulation, interleaver length and code rate, there is a graph that links the C/N with the ESR5(20) quality (see clause A.12). This process is depicted in figure 11.8.

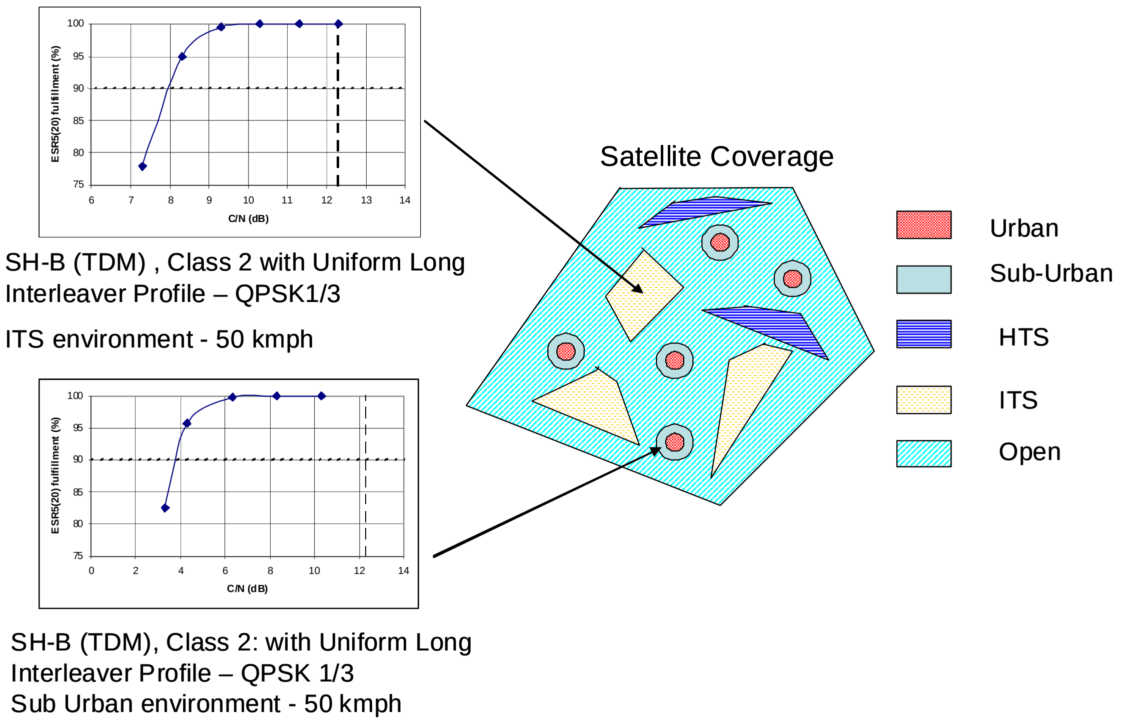


Figure 11.8: ESR5 versus C/N performance association for different environments over the coverage area

The result of this process is a map that provides the Service availability over the satellite coverage for a given interleaver profile, code rate, modulation type and mobility situation (see figure 11.9).

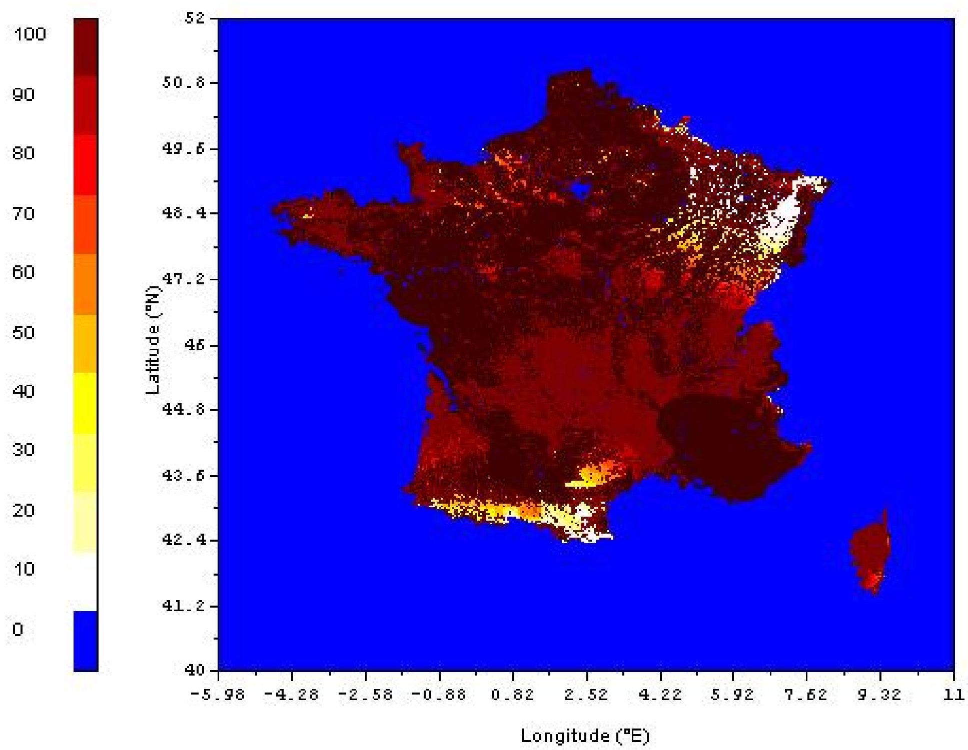


Figure 11.9: Example of service availability over France

The last stage requires the setting of a targeted service availability. In this example, the target service availability is given in terms of ESR5 and in the present document is set to be 90 % for all satellite environments.

11.8 Hybrid network planning

11.8.1 Introduction

While classical Fixed TV broadcast either satellite-based or terrestrial-based addresses roof top service to directional antennas, Mobile TV broadcast addresses terminals with very reduced RF performances with regard to roof top antennas, located at street level (for outdoor services) or at various levels of floors with insuperable penetration losses for a satellite signal. Hence Global Mobile Broadcast solutions that rely on satellite umbrellas also require Complementary Ground Components to distribute the service to users in the urban/dense urban environments.

Architectures will rely:

- Either on the SH-B version of the DVB-SH standard, where satellite and terrestrial components use different modulations (TDM versus OFDM) and different frequency channels or even bands; satellite/terrestrial overlap zone is just another case of service management across the spectrum.
- Either on the SH-A MFN version of the DVB-SH standard, where satellite and terrestrial components use the same modulation (OFDM) but different frequency channels or even bands; satellite/terrestrial overlap zone is likewise just another case of service management across the spectrum.
- Either on the SH-A SFN version of the DVB-SH standard, where satellite and terrestrial components use the same modulation (OFDM) and the same frequency channel (and thus band); satellite/terrestrial overlap zones require co-engineering on both segments in order to maintain SFN on the whole hybrid coverage and avoid destructive interferences; this entails constraints on parameters such as Guard Interval for the CGC.

It is not easy to derive the hybrid SFN gain as the satellite to terrestrial repeater power ration will rapidly evolve while the mobile user is moving from some rural or sparse suburban context at the theoretical terrestrial border to standard suburban or even some level of urban environment in more central zones.

In fact, not only the SFN gain can be observed, which is beneficial for the QoS, but also some destructive interference can appear, due to the fact that the terrestrial receiver coverage outside the terrestrial SFN area may create interference with the satellite signal. This is because the terrestrial gap fillers signals may be seen in LOS propagation for oriented towards open fields, especially in the direction of the satellite (south) and the differential delay of terrestrial signal versus the satellite one may exceed the OFDM guard interval.

An other consequence of the hybrid architecture and frequency reuse in the case of multi beam satellite is the possible existence of interference on terrestrial coverage of a given country coming from adjacent country beam at the same frequency, due to beam secondary lobes.

So the following subjects are covered in the different subsections.

- Hybrid network system model.
- Impact on coverage: improvements and interferences.
- Interference limited zones in multibeam hybrid networks.

11.8.2 Hybrid network system model

11.8.2.1 Overview

Figure 11.10 gives an overview of the system architecture that must be considered in the following clauses.

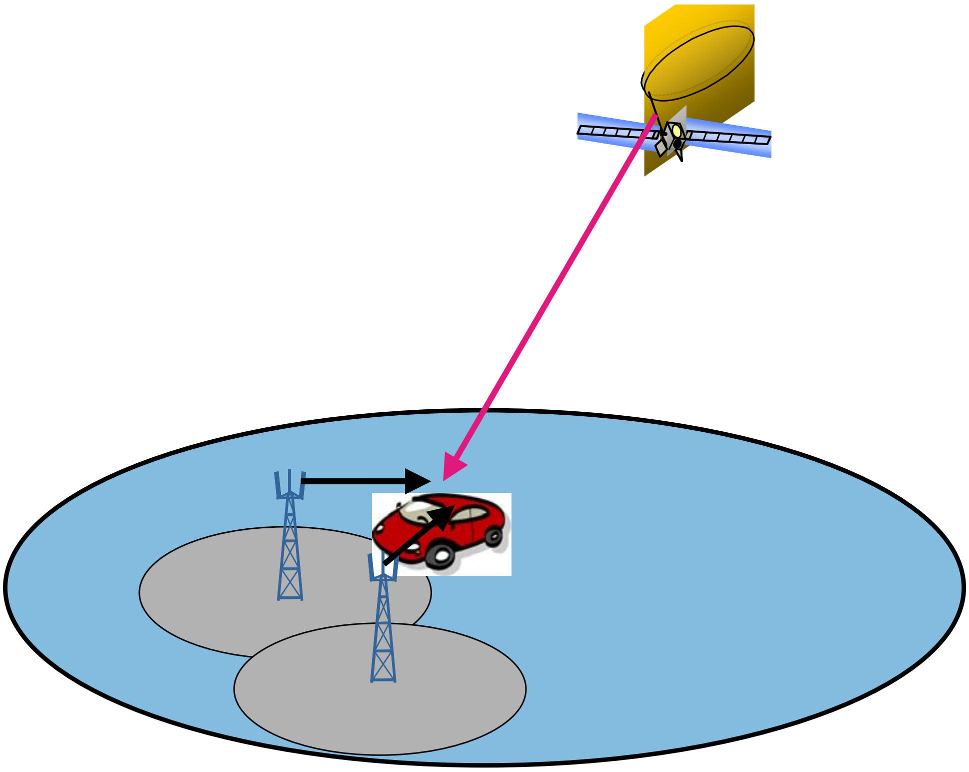


Figure 11.10: Overall System

In the most general situation, the system is composed of the following elements under the DVB-SH satellite coverage.

- One or more repeaters.
- A DVB-SH receiver, mainly vehicle mounted receiver.

Though some hybrid situation can occur in urban areas, the ideal locations for hybrid situation are suburban areas, where satellite signal can be received with a reasonable quality of service and which can be under one or many terrestrial repeaters. Of particular interest is the case where a repeater is covering the edge of the area, at the limit of rural area, open or ITS.

For the terrestrial channel, we consider TU6 channel at 50 km/h (3 km/h?) in suburban environment, using Cost Hata model.

For the satellite channel, three possible cases, based on Perez-Fontan LMS channel, can be used:

- Sub urban channel.
- Open channel: obviously when repeater is at the edge of an open area.
- Intermediate Tree Shadowing.

There are many differences when considering SFN (SHA) and MFN cases.

In MFN case, the two signals are received in different frequency blocks (inside the same frequency band or not), and simultaneous reception (though ad hoc terminals are not yet existing) requires two receiving chains (two antennas, two tuners two demodulator but probably a single decoder).

The two signals (OFDM/OFDM or TDM/OFDM) can be received without any stringent SFN conditions, and even outside the Hybrid SFN geometric zone, the signal from the repeater can be added to the satellite signal, provided it is received with a sufficient level to be detected and combined.

In the case of SFN (hybrid SHA), the two signals are received through the same chain (two chains, but with the same content in case of receiving diversity), they are added intrinsically as in terrestrial SFN and decoded.

11.8.2.2 Hybrid channel model and Geometrical Aspects

The hybrid channel can be novelised as the addition of the terrestrial signal following a typically TU6 channel model, and the satellite channel, following a LMS channel model. Figure 11.11, that represents the emulator block diagram, gives a schematic view of hybrid channel model.

The terminal will receive the signal from the two sources:

- The terrestrial repeater(s) following a TU6 propagation channel, or any repeaters.
- The satellite following a LMS propagation channel.

The signals are impaired differently, suffering from multipath, shadowing and blocking effects on the different path. Doppler effects will also be different.

Receiver will perform an estimation of the right window to remove the CP, and then will use a channel estimator in order to demodulate.

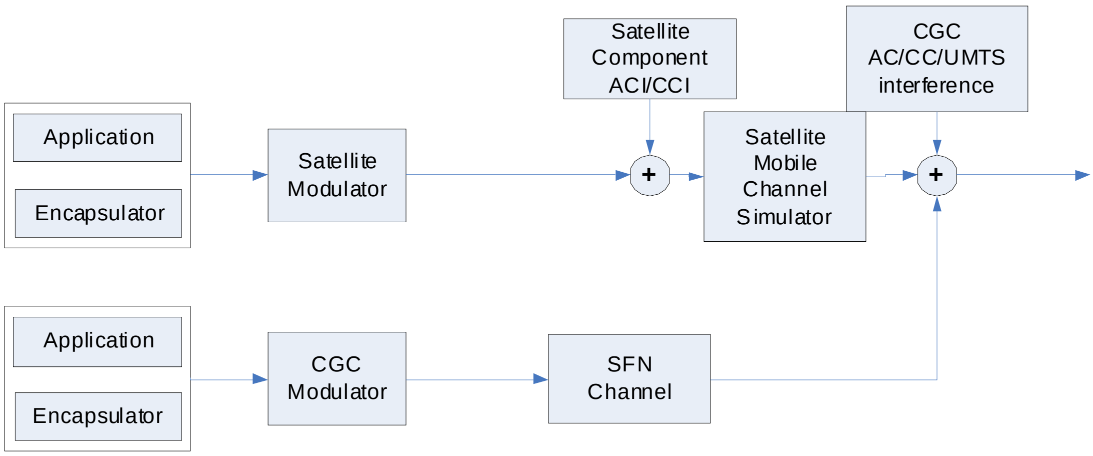


Figure 11.11: Possible lab configuration

The geometrical Analysis intends to determine in the case of a single repeater the Hybrid SFN zone using a very simplified model.

The schematics of the geometry is represented here below.

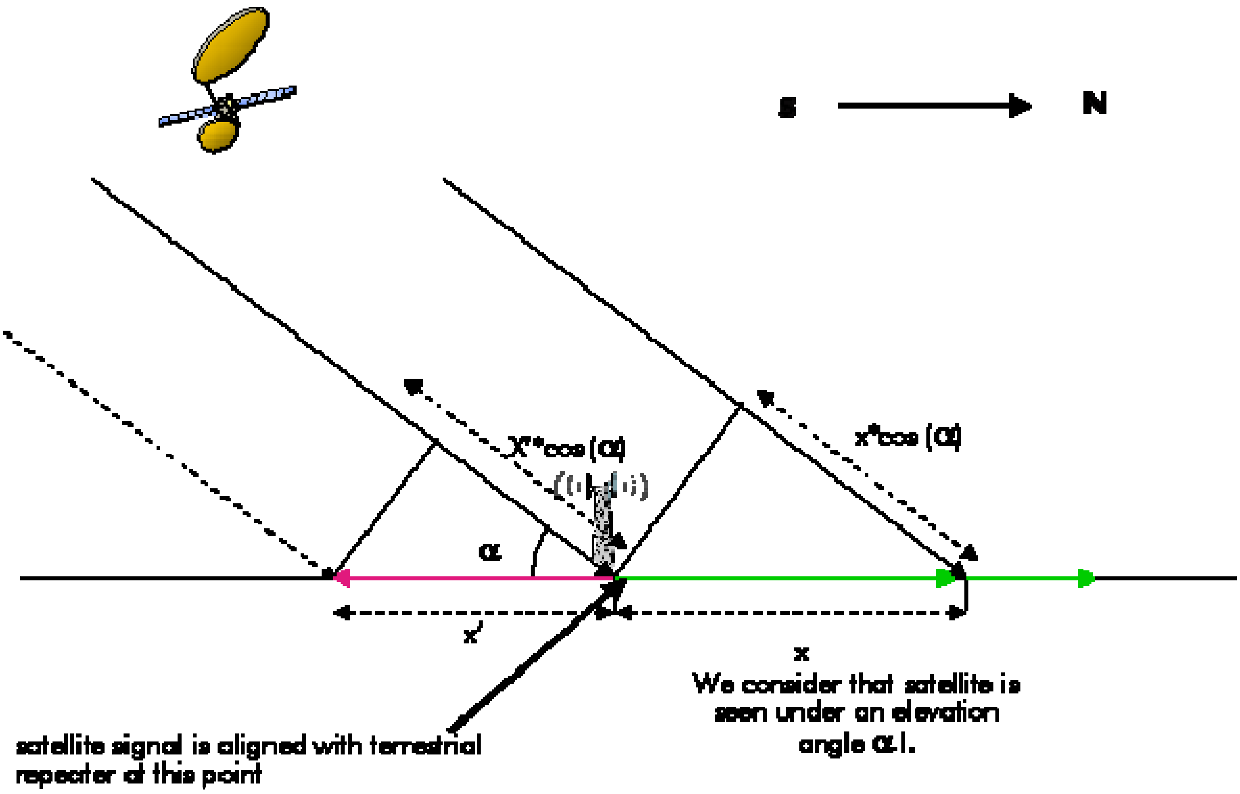


Figure 11.12: Schematics of the geometrical analysis

The repeater is called R1 and is considered as the time reference. The direction N/S is indicative, in fact, the considered direction is approximately SE/NW, if we are located around Las Vegas (115°W), and satellite ICO G1 is at 91°W, and also around Paris with W2A satellite located at 10°E. The positive axis is towards north direction. And we make a rough computation along this axis.

Satellite is seen under an elevation angle α. So if we consider a point M at distance x from the repeater, the satellite signal will arrive at the following relative time to the repeater: x cos(α)/c.

If x is positive (north of the repeater), satellite signal will arrive later that at the repeater. In case M is south of the repeater, satellite signal will arrive in advance compared to the repeater.

The origin of time is defined by the time of arrival of the satellite signal at the repeater, which is also the time of transmission of the repeater signal. This synchronisation can be realised thanks to the SHIP in the SH frame.

Now, at point M, with these conventions:

- The signal from the repeater arrives at time $\frac{|x|}{c}$ at point M.
- The signal from the satellite arrives at time $\frac{|x|\cos(\alpha)}{c}$.

So the general hybrid SFN condition can be written as:

$$\frac{|x|}{c} - \frac{|x|\cos(\alpha)}{c} \leq GT \ .$$

Where GT is the guard time. Introducing do =c*GT.

So the Hybrid SFN condition can be written as $|x| - |x|\cos(\alpha) \leq d_o$.

Two possible cases are therefore foreseen: x>0 and x<0, and the Hybrid SFN conditions can be written:

- $x \le \dfrac{d_o}{1-\cos(\alpha)}$ when M is located north of the repeater.
- $|x| \le \dfrac{d_o}{1+\cos(\alpha)}$ when M is located south of the repeater.

Some examples are necessary to illustrate the previous equations.

In all following cases, GT = 89,6 μs for GI =1/4 /2k/5 MHz.

Let us first consider the case of point located North of the repeater:

- α=33°: x max = 166 km (around Paris).
- α=45°: x max = 91 km.
- α=41°: x max = 109,59 km.

When M is located South, the SFN condition can be written as:

- α=33°: x max = 14,6 km.
- α=45°: x max = 15,75 km.
- α=41°: x max = 15,3 km.

These figures suggest that the hybrid SFN zone is an ellipse north oriented, as roughly described in figure 11.13.

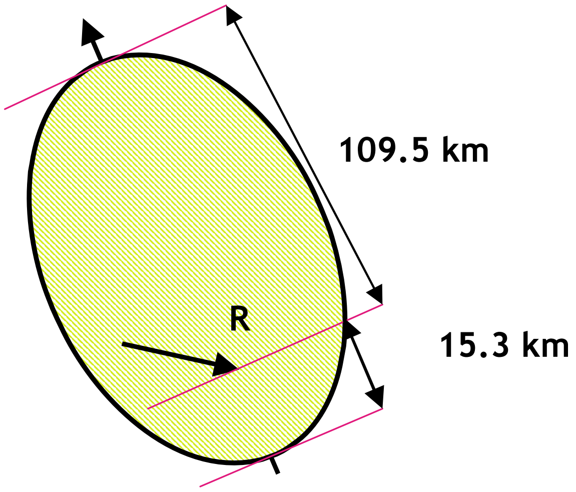


Figure 11.13: One repeater plus satellite SFN coverage @ 41° elevation

With GI 1/8 dimensions are roughly divided by two. The hybrid SFN zone si quite narrow in the southern part.

11.8.3 Impact on coverage

11.8.3.1 The Possible zones in the case of SHA Hybrid SFN

In the case of SHA Hybrid SFN, and when considering a hybrid coverage zone there are three main areas with special scenarios. Nevertheless while transiting from one area to another the transition is smooth:

- When approaching the transmitter, the terrestrial signal is much more powerful than the satellite one, and there is no impact on the reception quality.
- When satellite and terrestrial signals are of comparable strength, they can add until the mobile is out of the SFN zone defined by the Guard Interval. Leaving the overlap zone the terrestrial component fades out smooth. This situation is especially important when the system is working in DVB-SH-A-SFN, because the combining gain is maximum here.
- When the mobile is out of the Hybrid SFN zone, the terrestrial signal can act as an interferer whose strength depends on terrestrial repeater distance, EIRP, antenna diagram, propagation channel. Again this case mainly concerns the pure SH-A-SFN case, as no (in-band) interference can occur in the MFN case.
- When terrestrial signal C/N vanishes (below -10 dB), there is no more interference, and we are in a satellite only coverage zone.

It is in principle necessary to distinguish the SH-A-SFN case from the MFN case (SH-A or SH-B, which is quite the same). The difference between MFN and SFN is the type of interference, for SFN the interference is in-band (e.g. if the guard interval is violated), for MFN the interference is in another frequency band and can be mitigated by receiver filtering.

Figure 11.14 shows a qualitative representation of the different zones.

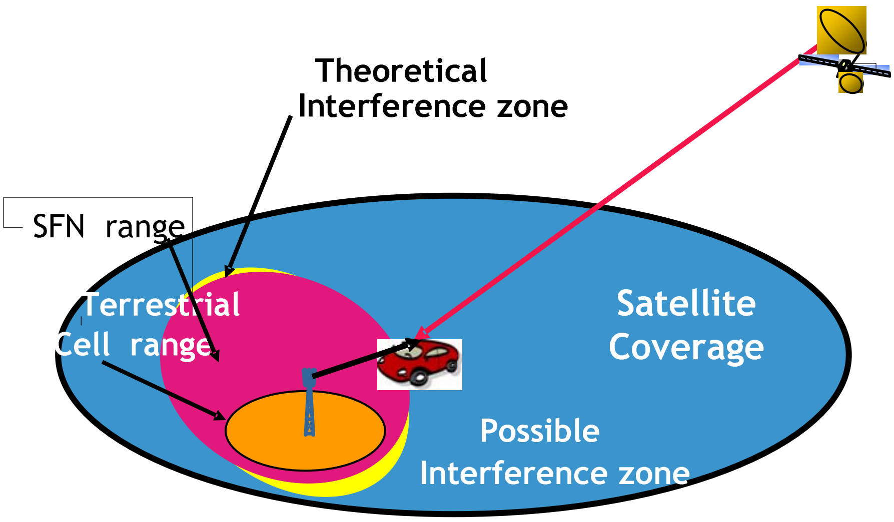


Figure 11.14: Hybrid System Overview

The different computations have shown that the dimensions of the ellipse are quite high in the north edge around 100 km with a repeater located in Paris, and a satellite at 10°E and GI of 1/8.

On the southern direction, distance is around 8 km; so except the southern direction, the geometrical SFN zone is quite huge, and the risk and level of interference are very low. This is why the different simulations and measurements have been concentrated on the critical cases. The different results are presented in clauses 7 and A.13.

The main parameters to consider are the terrestrial received C/N and the time difference between the satellite signal time of arrival and the repeater signal time of arrival at the receiver. The time reference is defined as the time of arrival of the satellite signal at the repeater. This time difference is called Δ T.

Summarizing some of different laboratory measurements, the following graph, illustrates the different zones.

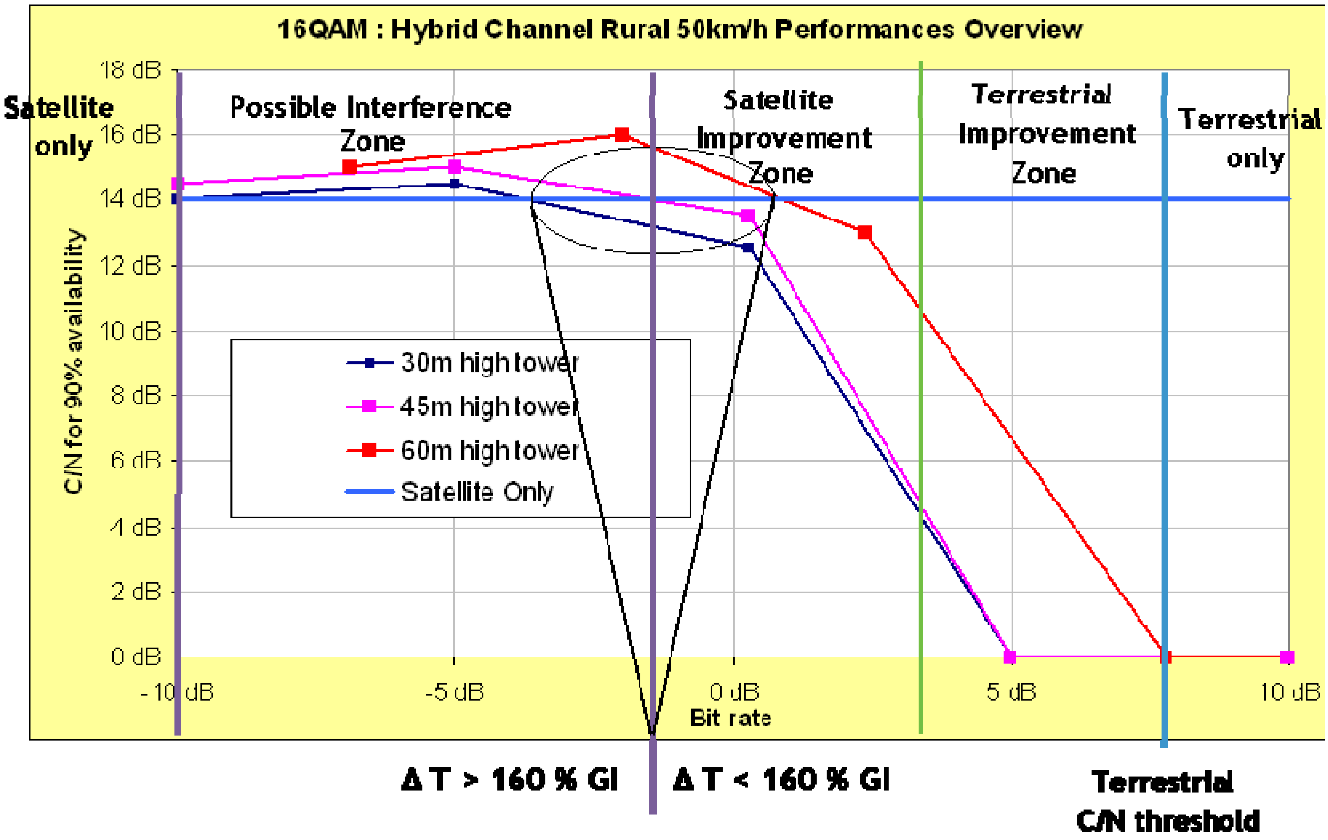


Figure 11.15: Illustration of the different zones in rural areas with 16QAM 1/3

It is possible to distinguish the different zones:

- When received terrestrial signal C/N is above required threshold, we can consider that we are under terrestrial coverage, and the satellite signals provides some improvement, but is not essential.
- The second zone (starting from right in the diagram) corresponds to a zone where satellite signal is improved by the terrestrial one. Even if the satellite signal is quite low (for instance if received by a handset, the combination of the two signals result in a good quality signals.
- The third zone corresponds to an improvement of the satellite signal with a "residual" terrestrial signal.
- The fourth zone corresponds to possible interference zone between the satellite and terrestrial signals. The border between the two previous zones corresponds to a differential delay of around 160 % of the GI.
- The last zone corresponds to satellite only signal as terrestrial signal is falling below -10 dB, but it can have some interference impact.

In the case of suburban area, the interference zone is reduced as shown in figure 11.16.

Figure 11.16: Illustration of the different zones in Sub Urban areas with 16QAM 1/3

11.8.3.2 Hybrid coverage improvement

The previous clauses have shown that the interference can occur in the southern direction. The way to mitigate the interference is to increase the SFN zone: of course the first way is to use higher Guard Time, but this will not solve all the situations. An important feature that can be implemented is the timing advance:

- Usually, the time of transmission of a repeater is aligned with the time of arrival of the satellite signal at the repeater.

- In the timing advance mechanism, time od transmission is advance by a integer multiple of the use Guard Time.
- By doing so, the dimensions of the ellipses are multiplied by the same integer number.

The following pictures provide plots of hybrid SFN zones using W2A satellite at 10°E, and one repeater in two different locations: Paris and Frascati (ESRIN) 5 MHz and GI 1/8 (44,6 µs) are used.

Figures 11.17 and 11.18 plot the hybrid SFN contours with no time advance, one GI advance and 8 GI advance. So the contours represent maximum time difference of one GI, two and nine GI.

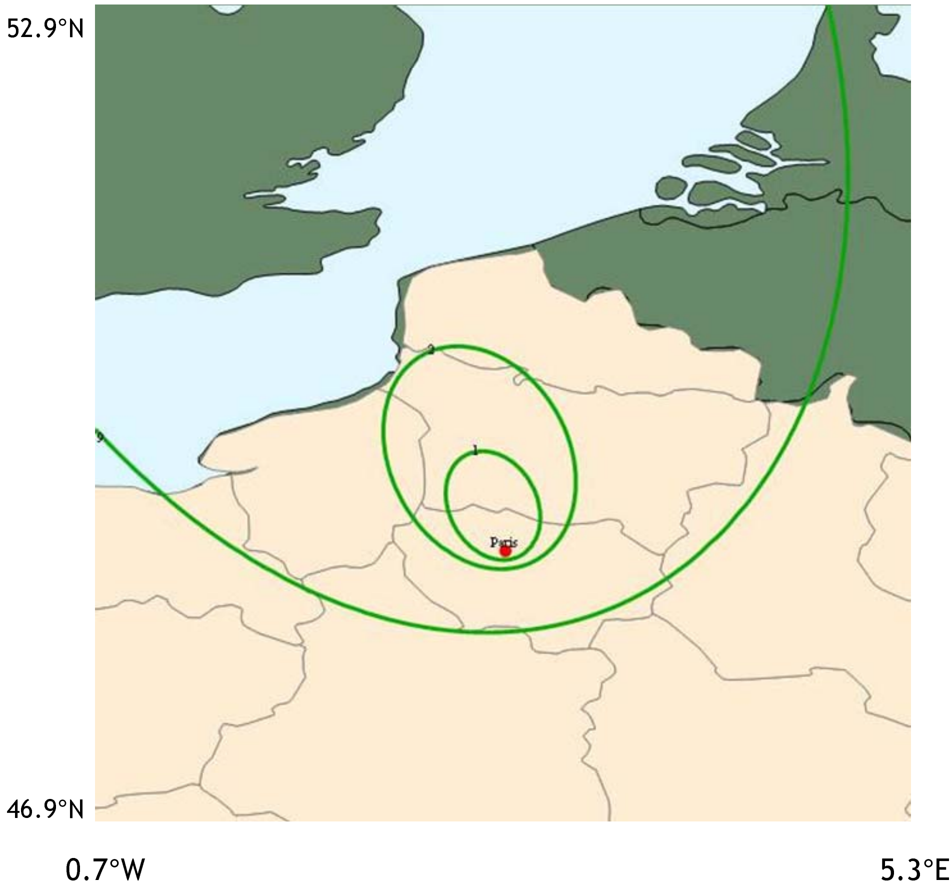


Figure 11.17: Hybrid SFN contour lines in Paris

And at ESRIN:

Figure 11.18: Hybrid SFN contour lines in Frascati

The different plots show clearly that the DFN zone is improved by using timing advance.

11.8.3.3 Combining gain

Some conclusions and rules can be derived from clause 11.8.3:

- The hybrid channel architecture gives an evident gain of quality of services, as it increases in most of the cases the ESR (5) ratio.
- In the northern direction, the differential delay is below 100 % of the GI on along distance, even for GI 1/8, and there is no risk of interference, as the terrestrial signal becomes very low at the edges.
- On the southern direction (in sight of the satellite), the gain is noticeable, even at more than 100 % of the GI.

- When the delay is more than 150 % of the GI approximately, there can be some interference at te edges of the contour, but they can be mitigated by different means: using a large GI when possible, implement timing advance at repeaters level, or reducing radiated power towards southern direction. The repeater could use two sector antenna at the edge of the zone and oriented towards northern directions.

Hence a hybrid network planning tool (whose design is of course out of the scope of the present document) should in fact proceed as follows for the great lines:

- First compute the geometrical hybrid SFN zones taking into account the possible timing advance: roughly the zones where differential delay is below or over 150 % (TBC) of the GI.
- Then compute terrestrial only coverage with a given C/N threshold, and consider it as terrestrial only coverage.
- Then compute inside the geometrical SFN zone the possible gain in LOS of the satellite signal, by using next formula. In this case the QoS could be determines by using maps like in clause 11.7.4, The result will give inputs that will give inputs for possible reduction of sites or power.

$$\left(\frac{C}{N}\right)_{eq} = 10 \times Log(10^{0,1\times\left(\frac{C}{N}\right)_{Sat}} + 10^{0,1\times\left(\frac{C}{N}\right)_{Ter}}) \text{ in dB.}$$

- Outside the geometrical zone, the tool can compute the impact on satellite C/N, and then the possible degradation I some areas, In that case the terrestrial C/N is seen as a C/I for the satellite signal, and the resulting C/N is now computed by:

$$\left(\frac{C}{N}\right)_{eq} = -10 \times Log(10^{-0,1\times\left(\frac{C}{N}\right)_{Sat}} + 10^{-0,1\times\left(\frac{C}{N}\right)_{Ter}}) \text{ in dB.}$$

11.8.4 Interference limited zones in multibeam hybrid networks

11.8.4.1 Introduction

In mutibeam satellites, there is the possibility of large overlaps between adjacent beams (see for example clause 4, figure 4.2). In satellite-only systems, a suitable intra-system C/I level is maintained by segmenting the overall bandwidth in colours and reusing each colour only in the beams where there is a good antenna isolation. However, in hybrid terrestrial/satellite systems, all the colours are used in each beam (either by the satellite or by the terrestrial component). Therefore, the terrestrial transmitter in a beam may be interfered by the satellite signal of the adjacent beam when it reuses the frequency of that adjacent beam.

In the following, we will define the interference limited zone as the coverage area that would be served if the interference component was not present. In this analysis, the source of the interference is the satellite signal from the adjacent beams. The situation is depicted schematically in figure 11.19.

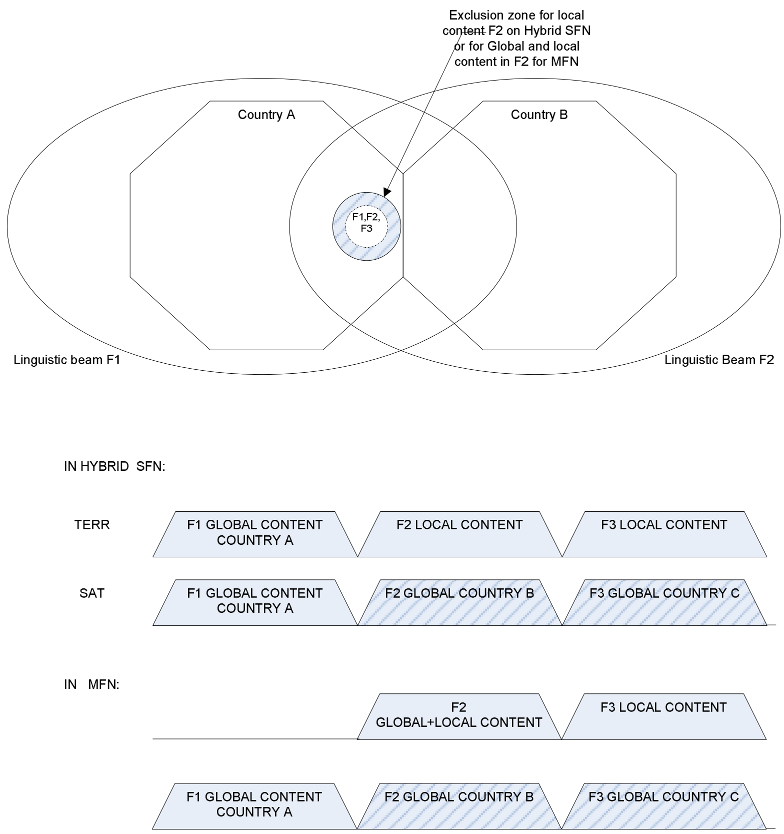


Figure 11.19: Interference limited zone in Country A caused by satellite beam over country B

This kind of interference limited zone impacts the local and the common content as follows:

- Case a): for the Local content only: the interference limited zone is the fringe coverage area of the transmitter carrying the Local content that would be covered if the interfering satellite signal were removed.
- Case b): for the Common content in an MFN (and only MFN): the fringe area of the transmitter carrying the Common content where the wanted satellite signal is not usable (e.g. blocked) and that would be covered if the interfering satellite signal were removed.

In both cases, the result is a reduction of the cell radius covered by the CGC. The new cell radius can be calculated considering an increase in the noise floor (or equivalently a degradation of the receiver G/T) caused by the interference signal arriving from the satellite. The following link budget exemplifies the approach for a satellite EIRP of 63 dBW.

11.8.4.2 Reception conditionss B1 and B2: Indoor reception

In the case of indoor reception, the satellite signal is heavily attenuated by the building penetration loss. Hereafter, table 11.38 shows the cell reduction caused by the satellite signal experienced in reception conditions B1 (light indoor) and B2 (deep indoor), for different environments.

Table 11.38: Link budgets results in reception conditions B1 and B2

DVB-SSP in S band		INDOOR NO INTERFERENCE				INDOOR WITH INTERFERENCE			
Terrestrial link budget		**Dense Urban**	**Urban**	**Sub-urban**	**Rural**	**Dense Urban**	**Urban**	**Sub-urban**	**Rural**
Radio Interface Parameters	Unit	**Value**	**Value**	**Value**	**Value**	**Value**	**Value**	**Value**	**Value**
Channel bandwidth	MHz	5.00	5.00	5.00	5.00	5.00	5.00	5.00	5.00
Frequency	MHz	2182.50	2183	2183	2183	2182.50	2183	2183	2183
Mode		2048.00	2048.00	2048.00	2048.00	2048.00	2048.00	2048.00	2048.00
Radio interface mod code		QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3
Antenna type		TRI	TRI	TRI	TRI	TRI	TRI	TRI	TRI
1/Guard Interval		8	8	8	8	8	8	8	8
Guard interval		0.125	0.125	0.125	0.125	0.125	0.125	0.125	0.125
Total number of sub-carriers		1705.00	1705.00	1705.00	1705.00	1705.00	1705.00	1705.00	1705.00
Nber of data sub-carriers		1512.00	1512.00	1512.00	1512.00	1512.00	1512.00	1512.00	1512.00
Elementary period	us	0.18	0.18	0.18	0.18	0.18	0.18	0.18	0.18
Tu duration	us	358.40	358.40	358.40	358.40	358.40	358.40	358.40	358.40
GI duration	us	44.80	44.80	44.80	44.80	44.80	44.80	44.80	44.80
Ts duration	us	403.20	403.20	403.20	403.20	403.20	403.20	403.20	403.20
Sub-carrier spacing	kHz	2.79	2.79	2.79	2.79	2.79	2.79	2.79	2.79
Symbol rate	ksymb/	2.480	2.480	2.480	2.480	2.480	2.480	2.480	2.480
Useful bandwidth occupancy	MHz	4.75	4.75	4.75	4.75	4.75	4.75	4.75	4.75
Useful data rate at MPEG2-TS interface	Mbit/s	2.50	2.50	2.50	2.50	0.00	0.00	0.00	0.00
Spectrum efficiency		0.50	0.50	0.50	0.50	0.00	0.00	0.00	0.00
Transmitting end		**Tx**	**Tx**	**Tx**	**Tx**	**Tx**	**Tx**	**Tx**	**Tx**
Power amplifier per carrier and sector	W	12.0	12.0	12.0	12.0	12.0	12.0	12.0	12.0
Tx Power at antenna input	dBm	40.8	40.8	40.8	40.8	40.8	40.8	40.8	40.8
cable loss	dB	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0
Diplexer loss	dB	0.7	0.7	0.7	0.7	0.7	0.7	0.7	0.7
Tx antenna gain	dBi	18.0	18.0	18.0	18.0	18.0	18.0	18.0	18.0
EIRP	*dBm*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*
EIRP	*W*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*
Intereference source									
Satellite received power (LOS)	dBm	-1000	-1000	-1000	-1000	-107	-107	-107	-107
Building Penetration loss		18.0	16.0	14.0	12.0	18.0	16.0	14.0	12.0
Interference From satellite		-1018.0	-1016.0	-1014.0	-1012.0	-125.0	-123.0	-121.0	-119.0
Receiving end		**Rx**	**Rx**	**Rx**	**Rx**	**Rx**	**Rx**	**Rx**	**Rx**
Rx antenna gain	dBi	-3.0	-3.0	-3.0	-3.0	-3.0	-3.0	-3.0	-3.0
Polarization mismatch	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Noise figure	dB	4.5	4.5	4.5	4.5	4.5	4.5	4.5	4.5
Antenna temperature	K	290.0	290.0	290.0	290.0	290.0	290.0	290.0	290.0
Ambiant temperature	K	290.0	290.0	290.0	290.0	290.0	290.0	290.0	290.0
kT	dBm/Hz	-174.0	-174.0	-174.0	-174.0	-174.0	-174.0	-174.0	-174.0
Equivalent Rx band	dBm.Hz	66.8	66.8	66.8	66.8	66.8	66.8	66.8	66.8
G/T	dB/K	-32.1	-32.1	-32.1	-32.1	-32.1	-32.1	-32.1	-32.1
Rx noise floor	dBm	-102.7	-102.7	-102.7	-102.7	-102.7	-102.7	-102.6	-102.6
Required C/N	dB	2.8	2.8	2.8	2.8	2.8	2.8	2.8	2.8
Rx sensitivity	dBm	-99.9	-99.9	-99.9	-99.9	-99.9	-99.9	-99.8	-99.8
Minimum Rx level at antenna	*dBm*	*-96.9*	*-96.9*	*-96.9*	*-96.9*	*-96.9*	*-96.9*	*-96.8*	*-96.8*
Minimum Signal Level for Network Planning	dBm	-72.01	-74.01	-71.31	-73.31	-71.98	-73.97	-71.24	-73.21
System Gain	dB	**152.0**	**152.0**	**152.0**	**152.0**	**152.0**	**152.0**	**151.9**	**151.9**
Margins									
Average Hand Loss	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Average building penetration loss	dB	18.0	16.0	14.0	12.0	18.0	16.0	14.0	12.0
Target level of signal penetration		B2	B2	B1	B1	B2	B2	B1	B1
Std shadowing outdoor	dB	8.00	8.00	8.00	8.00	8.00	8.00	8.00	8.00
Std shadowing indoor	dB	6.00	6.00	6.00	6.00	6.00	6.00	6.00	6.00
Std dev of Fading Margin	dB	10.00	10.00	10.00	10.00	10.00	10.00	10.00	10.00
Propagation constant		3.5	3.4	3.5	3.4	3.5	3.4	3.5	3.4
Shadow fading Margin - whole cell	dB	11.6	11.6	11.6	11.6	11.6	11.6	11.6	11.6
Coverage Probability - whole Cell		**95.0%**	**94.9%**	**95.0%**	**94.9%**	**95.0%**	**94.9%**	**95.0%**	**94.9%**
Coverage Probability - edge of cell	%	**87.7**	**87.7**	**87.7**	**87.7**	**87.7**	**87.7**	**87.7**	**87.7**
SFN network gain	dB	4.7	4.7	0.0	0.0	4.7	4.7	0.0	0.0
Rx gain (antenna diversity)	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
MAPL	dB	**127.1**	**129.1**	**126.4**	**128.4**	**127.1**	**129.1**	**126.3**	**128.3**
Cell range computation									
Cost231-HATA model									
Cell range	km	**0.593**	**1.107**	**0.899**	**2.376**	**0.592**	**1.104**	**0.895**	**2.359**
Surface	km2	**0.69**	**2.39**	**1.58**	**11.00**	**0.68**	**2.38**	**1.56**	**10.86**

As expected, the impact of the satellite interference is negligible in the case of indoor propagation.

11.8.4.3 Reception conditions A: Outdoor pedestrian

In the case of outdoor reception, the satellite signal is not impaired by the building penetration loss and the cell radius calculated considering the interference from the satellite signal homogenously present in the cell is around 8 % for the handsets as seen in table 11.39.

Table 11.39: Link budgets results in reception condition A

DVB-SSP in S band		OUTDOOR NO INTERFERENCE				OUTDOOR WITH INTERFERENCE			
Terrestrial link budget		**Dense Urban**	**Urban**	**Sub-urban**	**Rural**	**Dense Urban**	**Urban**	**Sub-urban**	**Rural**
Radio Interface Parameters	Unit	**Value**	**Value**	**Value**	**Value**	**Value**	**Value**	**Value**	**Value**
Channel bandwidth	MHz	5.00	5.00	5.00	5.00	5.00	5.00	5.00	5.00
Frequency	MHz	2182.50	2183	2183	2183	2182.50	2183	2183	2183
Mode		2048.00	2048.00	2048.00	2048.00	2048.00	2048.00	2048.00	2048.00
Radio interface mod code		QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3	QPSK1/3
Antenna type		TRI	TRI	TRI	TRI	TRI	TRI	TRI	TRI
1/Guard Interval		8	8	8	8	8	8	8	8
Guard interval		0.125	0.125	0.125	0.125	0.125	0.125	0.125	0.125
Total number of sub-carriers		1705.00	1705.00	1705.00	1705.00	1705.00	1705.00	1705.00	1705.00
Nber of data sub-carriers		1512.00	1512.00	1512.00	1512.00	1512.00	1512.00	1512.00	1512.00
Elementary period	us	0.18	0.18	0.18	0.18	0.18	0.18	0.18	0.18
Tu duration	us	358.40	358.40	358.40	358.40	358.40	358.40	358.40	358.40
GI duration	us	44.80	44.80	44.80	44.80	44.80	44.80	44.80	44.80
Ts duration	us	403.20	403.20	403.20	403.20	403.20	403.20	403.20	403.20
Sub-carrier spacing	kHz	2.79	2.79	2.79	2.79	2.79	2.79	2.79	2.79
Symbol rate	ksymb/	2.480	2.480	2.480	2.480	2.480	2.480	2.480	2.480
Useful bandwidth occupancy	MHz	4.75	4.75	4.75	4.75	4.75	4.75	4.75	4.75
Useful data rate at MPEG2-TS interface	Mbit/s	2.50	2.50	2.50	2.50	0.00	0.00	0.00	0.00
Spectrum efficiency		0.50	0.50	0.50	0.50	0.00	0.00	0.00	0.00
Transmitting end		**Tx**	**Tx**	**Tx**	**Tx**	**Tx**	**Tx**	**Tx**	**Tx**
Power amplifier per carrier and sector	W	12.0	12.0	12.0	12.0	12.0	12.0	12.0	12.0
Tx Power at antenna input	dBm	40.8	40.8	40.8	40.8	40.8	40.8	40.8	40.8
cable loss	dB	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0
Diplexer loss	dB	0.7	0.7	0.7	0.7	0.7	0.7	0.7	0.7
Tx antenna gain	dBi	18.0	18.0	18.0	18.0	18.0	18.0	18.0	18.0
EIRP	*dBm*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*	*55.1*
EIRP	*W*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*	*323.0*
Intereference source									
Satellite received power (LOS)	dBm	-1000	-1000	-1000	-1000	-107	-107	-107	-107
Building Penetration loss		0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Interference From satellite		-1000.0	-1000.0	-1000.0	-1000.0	-107.0	-107.0	-107.0	-107.0
Receiving end		**Rx**	**Rx**	**Rx**	**Rx**	**Rx**	**Rx**	**Rx**	**Rx**
Rx antenna gain	dBi	-3.0	-3.0	-3.0	-3.0	-3.0	-3.0	-3.0	-3.0
Polarization mismatch	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Noise figure	dB	4.5	4.5	4.5	4.5	4.5	4.5	4.5	4.5
Antenna temperature	K	290.0	290.0	290.0	290.0	290.0	290.0	290.0	290.0
Ambiant temperature	K	290.0	290.0	290.0	290.0	290.0	290.0	290.0	290.0
kT	dBm/Hz	-174.0	-174.0	-174.0	-174.0	-174.0	-174.0	-174.0	-174.0
Equivalent Rx band	dBm.Hz	66.8	66.8	66.8	66.8	66.8	66.8	66.8	66.8
G/T	dB/K	-32.1	-32.1	-32.1	-32.1	-32.1	-32.1	-32.1	-32.1
Rx noise floor	dBm	-102.7	-102.7	-102.7	-102.7	-101.3	-101.3	-101.3	-101.3
Required C/N	dB	2.8	2.8	2.8	2.8	2.8	2.8	2.8	2.8
Rx sensitivity	dBm	-99.9	-99.9	-99.9	-99.9	-98.5	-98.5	-98.5	-98.5
Minimum Rx level at antenna	*dBm*	*-96.9*	*-96.9*	*-96.9*	*-96.9*	*-95.5*	*-95.5*	*-95.5*	*-95.5*
Minimum Signal Level for Network Planning	dBm	-92.91	-92.91	-88.21	-88.21	-91.53	-91.53	-86.83	-86.83
System Gain	dB	**152.0**	**152.0**	**152.0**	**152.0**	**150.6**	**150.6**	**150.6**	**150.6**
Margins									
Average Hand Loss	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Average building penetration loss	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Target level of signal penetration		A	A	A	A	A	A	A	A
Std shadowing outdoor	dB	8.00	8.00	8.00	8.00	8.00	8.00	8.00	8.00
Std shadowing indoor	dB	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Std dev of Fading Margin	dB	8.00	8.00	8.00	8.00	8.00	8.00	8.00	8.00
Propagation constant		3.5	3.4	3.5	3.4	3.5	3.4	3.5	3.4
Shadow fading Margin - whole cell	dB	8.7	8.7	8.7	8.7	8.7	8.7	8.7	8.7
Coverage Probability - whole Cell		**95.0%**	**94.9%**	**95.0%**	**94.9%**	**95.0%**	**94.9%**	**95.0%**	**94.9%**
Coverage Probability - edge of cell	%	**86.2**	**86.2**	**86.2**	**86.2**	**86.2**	**86.2**	**86.2**	**86.2**
SFN network gain	dB	4.7	4.7	0.0	0.0	4.7	4.7	0.0	0.0
Rx gain (antenna diversity)	dB	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
MAPL	dB	**148.0**	**148.0**	**143.3**	**143.3**	**146.6**	**146.6**	**141.9**	**141.9**
Cell range computation									
Cost231-HATA model									
Cell range	km	**2.327**	**3.970**	**2.713**	**6.503**	**2.127**	**3.618**	**2.480**	**5.926**
Surface	km2	**10.56**	**30.74**	**14.36**	**82.45**	**8.82**	**25.53**	**12.00**	**68.48**

It is worth to notice that the above link budgets consider the satellite always in LOS. However, the probability of the satellite signal being shadowed or blocked depends on the environment. Therefore, for the same probability of coverage, the cell reduction caused by the satellite will be less significant in the dense urban and urban case than in the suburban and rural scenario. The link budgets above shall be considered as worst case calculations for all scenarios.

The impact of the satellite signal can also be evaluated by assessing the loss of coverage probability caused by the satellite interference at constant cell radius. For the cell radius considered above, probability (in the whole coverage) goes down from 95 % to 93,5 % due to the satellite interference.

11.8.4.4 Conclusions

When the terrestrial CGC is dimensioned for indoor reception, the link budgets show that the impact of the satellite signal in the cell coverage is negligible. If we call this indoor coverage cell radius Ro, for distances below Ro of the transmitter, the impact of satellite reception is also negligible for outdoor reception conditions (obtained cell radius is always larger than Ro even in the presence of interference).

When only outdoor reception in considered, the cell radius of the terrestrial CGC will be reduced by the satellite interference particularly in sub-urban and rural areas. The impact of this cell reduction is relatively small for handsets at 95 % of coverage probability. Cell radius could be kept constant at the expense of loosing 1,5 % of coverage probability in the whole cell.

More accurate characterization of the interference limited zones will require a description of the statistics of the satellite signal in the area of interest including the satellite antenna gain over that area.

Annex A (informative): Data and methodology for DVB-SH assessment

This clause presents the data and methodology used by the DVB TM-SSP group to assess the efficiency of the specified DVB-SH Physical Layer, separately, and together with the selected Upper Layer complementary FEC.
TM-SSP members have agreed on a common simulation and experimental performance collection framework for performance assessment of the different DVB-SH configurations.

A.1 Geostationary Satellite Payload characteristics (S-band)

A.1.1 Payload architectures

The satellite payload distortions are of two types: linear distortions caused by the filters amplitude and group delay and the nonlinear distortions (see note) caused by TWTA(s) operated near its saturation point. The general analysis is complicated by the fact that linear and nonlinear distorting elements are typically cascaded. Nevertheless, the linear distortions are of second order compared to the nonlinear distortions.

The target EIRP of a mobile TV satellite in S-band is in the order of 63 dBW to 72 dBW, with the 68 dBW being a possible limit between "medium" and "high" power satellites broadcasting to handheld devices. For comparison, the EIRP of satellites broadcasting to fixed reception in the Ku-band are in the region of 50 dBW to 55 dBW.

Antenna and power amplification architectures play an equally central role whose final objective is the achievement of the target EIRP and the desired coverage areas. Antenna gain is upper-bounded by the theoretical limit relevant to the solid angular view of the coverage area.

EIRP requirements and antenna performance limit lead in most cases to a demanding RF power that exceeds the generation capabilities of single amplifiers available for space applications and "power combining" should be adopted.

Two main payload classes can be distinguished depending on the power amplification and combining architecture: "Linear" and "Non-linear" classes.

For a single-beam satellite (or multibeam satellites that do not require flexible power-sharing between beams), the required EIRP is typically obtained by using parallel TWTAs through network or polarization combining. Each TWTA amplifies only one signal (but several TWTAs amplify the same signal). The TWTA can be driven close to saturation if needed. The satellite is then said to belong to the "non-linear" class, providing a power advantage for the TDM waveform, which does not require the same TWTA back-off as the OFDM waveform.

When there is a need to reallocate available power to the different beams after the satellite is launched, a different amplification architecture is adopted. In this case, the amplification is performed through a low-power BFN followed by a shared TWTA network exploiting a stack of multiport amplifiers (MPA) connected to the transmit antenna feeds. Each TWTA now amplifies a weighted sum of several signals. This architecture implements spatial power combining that provides the flexible power allocation to the various beams. This flexibility is achieved at the expense of operating the TWTAs in multicarrier mode (i.e. with sufficient back-off) and therefore the power advantage of TDM over OFDM is diminished.

NOTE: The multiplicative phase noise impact is discussed in clause A.11.1.2.

A.1.2 TWTA characteristics and Operating Point Optimization

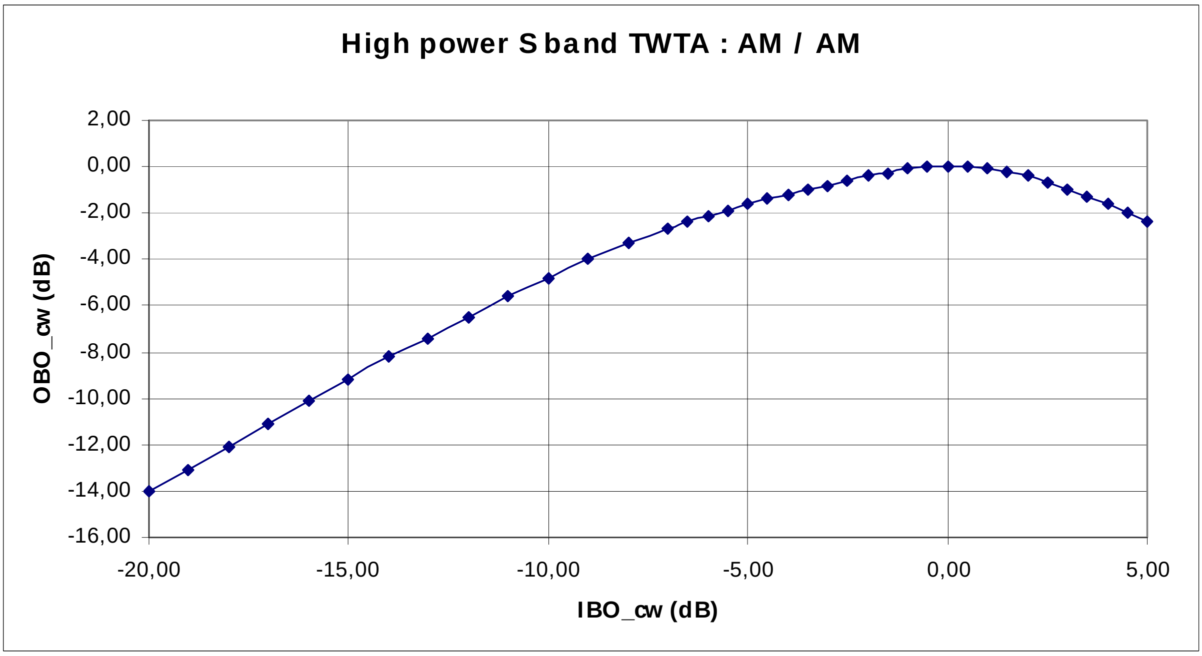
The satellite on-board platform DC power limitation calls for an TWTA(s) operation as close as possible to the maximum TWTA saturated power. At system level the TWTA is typically characterized by a maximum (saturated) power and three key performance characteristics: the AM/AM, AM/PM and DC to RF conversion efficiency curves.

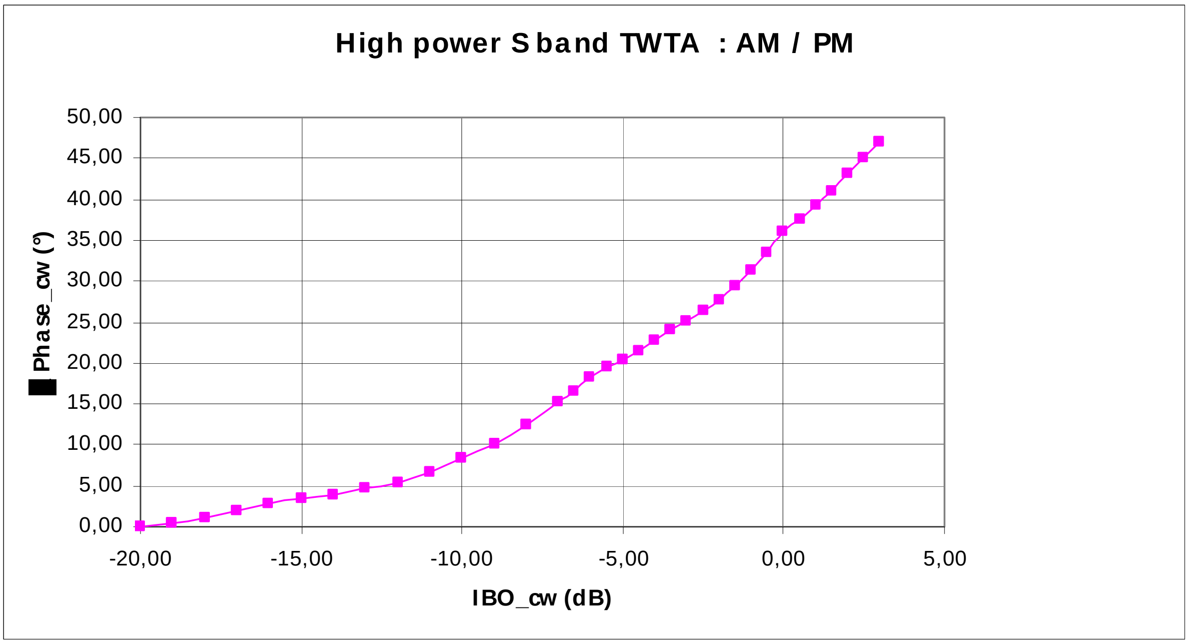
From a system perspective the quantity to minimize is the total payload degradation D_{TOT} in dB given by:

$$D_{TOT}(\text{IBO}) = \left[\frac{E_s}{N_0}\right]_{req}^{NL}(\text{IBO}) - \left[\frac{E_s}{N_0}\right]_{req}^{AWGN} + \text{OBO(IBO)-10log}\left[\overline{\eta}(\text{IBO})\right]$$

being $\left[\frac{E_s}{N_0}\right]_{req}^{NL}$ and $\left[\frac{E_s}{N_0}\right]_{req}^{AWGN}$ expressed in dB the average symbol energy over noise density required to achieve the target Frame Error Rate (FER) in the nonlinear and linear channel respectively, IBO, OBO represent the TWTA input and output backoff respectively in dB, while $\overline{\eta}$ represents the TWTA DC to RF conversion efficiency at the current drive level. It is important that the OBO and $\overline{\eta}$ are computed with the real DVB-SH signal at the selected drive level (IBO). It is apparent that an accurate payload loss assessment and TWTA operating optimisation involve quite complex physical layer simulations and accurate system modelling. Intuitively the IBO optimisation will result in the best trade-off between the loss of TWTA efficiency represented by the term OBO and $-10\log\left[\overline{\eta}(\text{IBO})\right]$ which are growing with IBO and the channel distortion losses represented by the term (in dB) $\left[\frac{E_s}{N_0}\right]_{req}^{NL}(\text{IBO}) - \left[\frac{E_s}{N_0}\right]_{req}^{AWGN}$ which will reduce with IBO.

For a DVB-SH type of system requiring large RF power (in the order of hundreds of Watts per beam) typically Travelling Wave Tubes (TWTA) are adopted. Figure A.1.1 shows typical AM/AM, and AM/PM characteristics for an S-band TWTA.





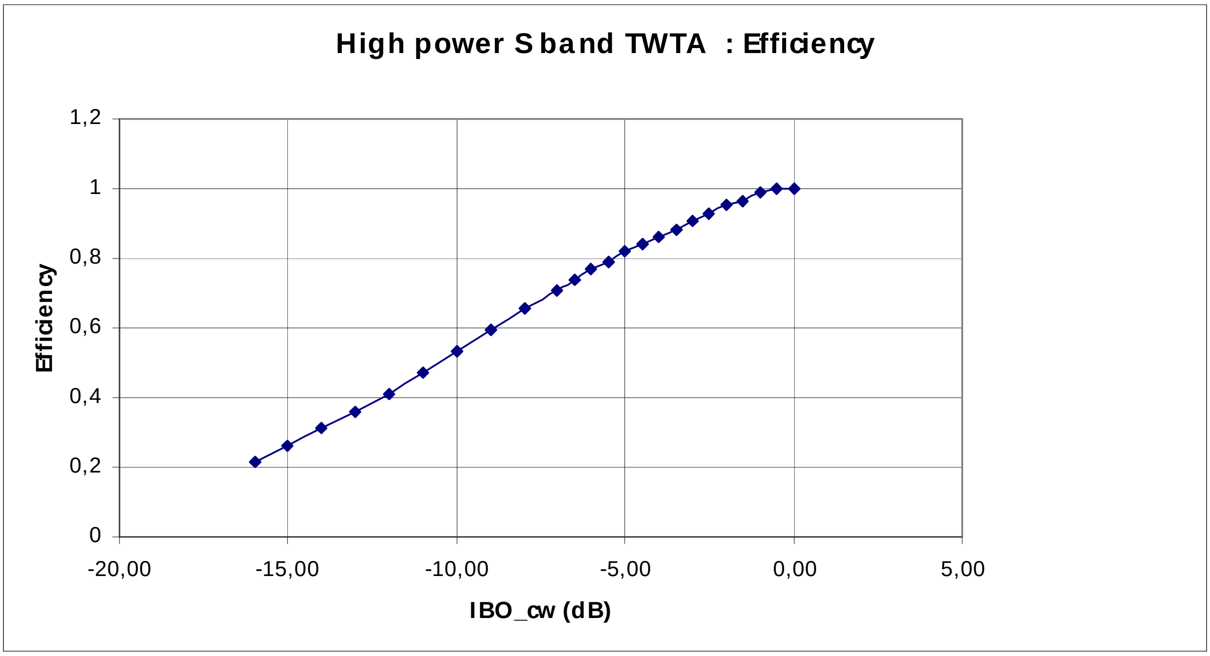


Figure A.1.1: S-band TWTA typical AM/AM, and AM/PM and efficiency characteristics

Over nonlinear channels the DVB-SH signal constellation, when observed at the demodulator symbol matched filter decimated output, is affected by two main kinds of impairment:

1) The constellation centroids (see note 1) warping due to the AM/AM and AM/PM TWTA nonlinear characteristic.

NOTE 1: By centroid we consider the compilation of received constellation cluster center of mass conditioned to each constellation point.

2) The clustering effect due to the Inter-Symbol Interference (ISI) experienced at the demodulator matched filter output.

The warping phenomenon is responsible for the reduction of the distance among 16QAM points (SH-A) or 16APSK rings (SH-B) (AM/AM compression) as well as a differential phase rotation among them (AM/PM differential phase).

The ISI causing the constellation clustering is related to the demodulator SRRC filter mismatch on the received signal due to the combination of the signal band limiting introducing memory in the channel, the IMUX filter linear distortion, the TWTA memory less nonlinearity, the OMUX linear filter distortions, resulting in a nonlinear channel with memory. These effects can be easily understood recalling the TWTA AM/AM and AM/PM characteristic of figure A.1.1. Clearly the warping effect has no impact on single ring constellations such as QPSK and 8PSK (see note 2). Degradations for 16QAM and 16APSK may be compensated in the receiver using pilots.

NOTE 2: The phase and amplitude PSK distortion will be compensated for by the demodulator AGC and phase recovery sub-systems.

A.1.3 Performance over nonlinear channels

A.1.3.1 Simulations

The reference demodulator as defined in clause A.11 has been used to derive via simulation the reported performances. In particular, the demodulator implements two-dimensional pilot-aided channel estimation, therefore, the impact of imperfect channel estimation is included. Tested physical layer configuration is QPSK-1/3 for both OFDM and TDM.

The optimum IBO/OBO is derived via simulation as *C/(N+I)* link margin maximisation, according to the definitions in clause A.1.2. However, the total payload degradation D_{TOT} was evaluated neglecting the TWTA DC to RF conversion efficiency. The formulae hereafter hold:

$$D_{TOT}(\mathrm{IBO}) = \left[\frac{E_s}{N_0}\right]_{req}^{NL}(\mathrm{IBO}) - \left[\frac{E_s}{N_0}\right]_{req}^{AWGN} + \mathrm{OBO}(\mathrm{IBO})$$

$$Losses_{IBO,\,ESR} = \left.\frac{C+I}{N}\right|_{IBO,\,ESR} - \left.\frac{C}{N}\right|_{ESR}$$

A.1.3.1.1 Single HPA per beam

Figure A.1.2 shows the total degradation D_{TOT} as function of IBO when an AWGN channel is considered.The optimum IBO (OBO) has been found to be 2,3 dB (1,8 dB) for the SH-A configuration and 0,3 dB (0,7 dB) for the SH-B configuration.

It is worth noticing that the D_{TOT} curve is almost flat around the IBO_{OPT}, resulting in a reduced degradation when a sub-optimum IBO is chosen. Also, the OBO variation is within 0,5 dB when the IBO is kept below 3 dB. Thus, IBO/OBO point could be adapted according to the design of the payload with little degradation in terms of end-to end performance.

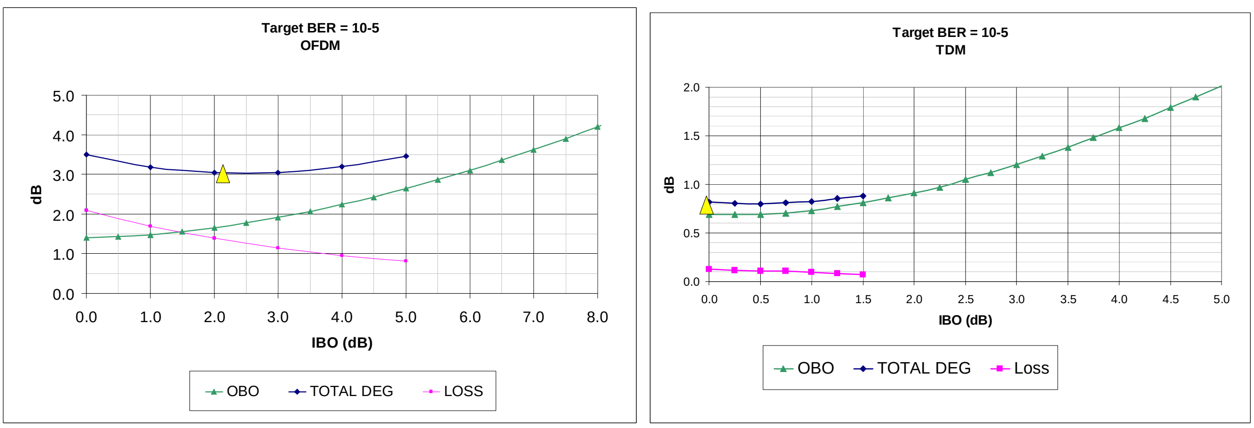
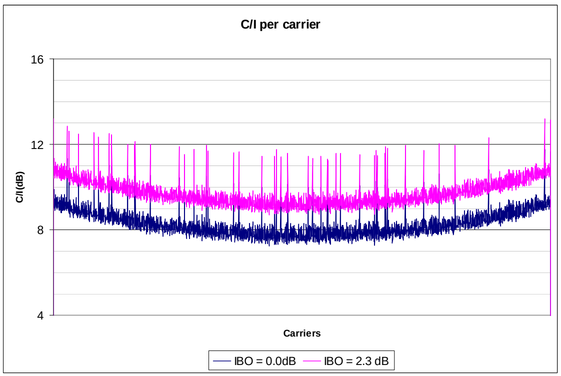


Figure A.1.2: IBO/OBO optimization for single HPA per beam, AWGN channel: a) SH-A (a) and b) SH-B

Figure A.1.3 shows the in-band and out-of-band characteristics when a SH-A waveform is considered. The out-of-band signal emission characteristic in figure A.1.3b) has been evaluated through Power Spectrum Density (PSD).The in-band C/I values shown in figure A.1.3a), have been derived with a per FFT-bin correlation method.

This method is applicable to any complex QAM signal and it is independent from the payload architecture. It requires alignment of transmitted and received symbol data sequences for a Data Aided Maximum Likelihood Estimation (DA-MLE).

Two cases are reported: worst C/I case when the HPA is driven at saturation and when driven at optimum IBO as per figure A.1.2. In the latter case, the minimum C/I value (FFT center) decreases from the 9 dB when IBO = 0 dB to roughly 7,5 dB; it is clear that external FFT-bins suffer for less interference, then they are characterized by improved C/I.



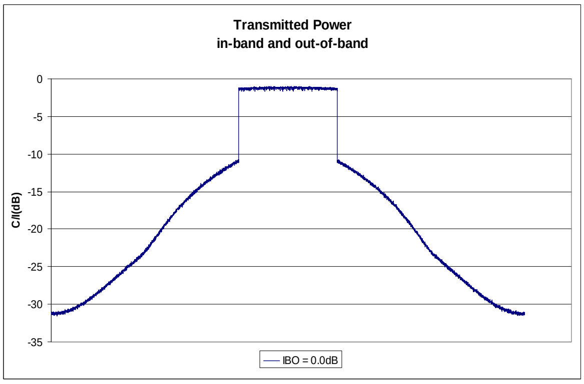


Figure A.1.3: SH-A In-band and out-of-band characteristics: a) C/I per FFT-bin and b) out-of-band emission

A.1.3.1.2 Multiport Amplifier

This clause refers to the multiport amplifier (MPA) payload configuration as described in clause A.1.1. In detail an 8 by 8 MPA (multiple multiplex per HPA, as per figure A.1.4) followed by high power BFN has been considered. No mismatch has been introduced and the HPAs have been modeled assuming they have the same AM/AM and AM/PM characteristics shown in figure A.1.1. Also, the same input power to all amplifiers after the input hybrids has been considered.

Performances have been verified for different level of MPA load, up to 3 active inputs over 8 available ports. The evaluation of the total payload degradation D_{TOT} always refer to a single DVB-SH signal input, defined as target signal.

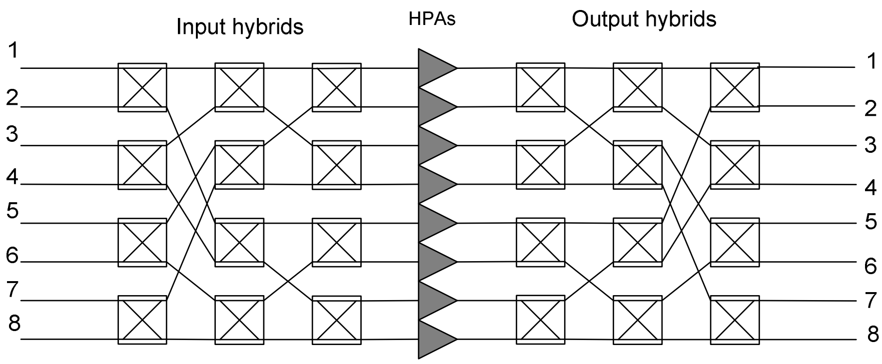


Figure A.1.4: Multi-Port Amplifier configuration

Figure A.1.5 shows the D_{TOT} as function of IBO when SH-A configuration in AWGN channel is considered. The optimum IBO (OBO) has been found to be 2,8 dB (1,9 dB) with 1 active input and 2,3 dB (1,8 dB) with 3 ports. It appears that SH-A OFDM MPA D_{TOT} performance are practically the same as for single HPA.

On the contrary, in case SH-B TDM configuration would be considered the D_{TOT} performance would be degraded in case of MPA exploitation being the HPAs composing the MPA operating with multicarrier input even if the individual input ports are driven by a quasi-constant envelope TDM signal. In case of MPA the D_{TOT} for TDM and OFDM is almost identical. It is worth noticing that the D_{TOT} curve is almost flat around the IBO_{OPT} point, resulting in a reduced degradation when a different IBO is chosen. Also, the OBO variation is within 0,5 dB when the IBO is kept below 3 dB.

Thus, IBO/OBO point could be adapted according to the design of the payload with little degradation in terms of end-to end performance.

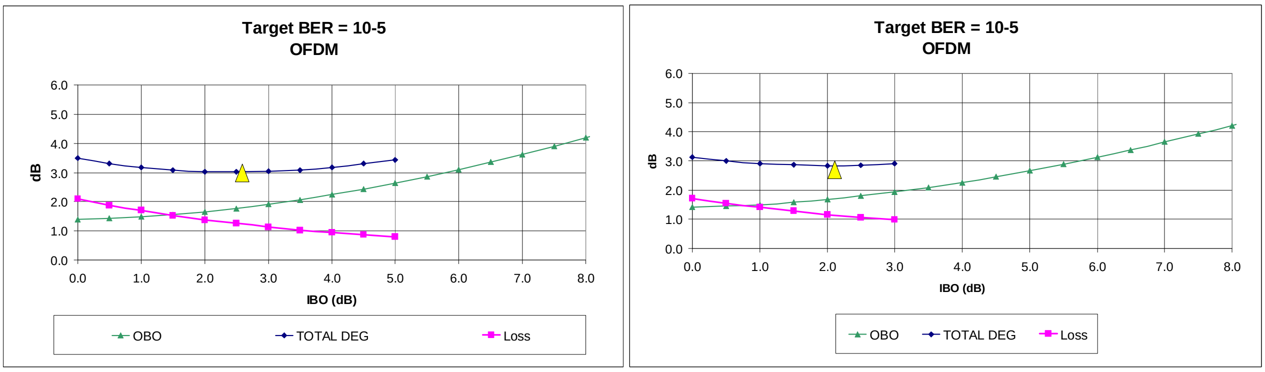


Figure A.1.5: IBO/OBO optimization for SH-A with different MPA loads: a) 1 SH-A carrier and b) 3 SH-A carriers

A.1.3.2 Single HPA per beam - Laboratory Measurements

The measurements reported in this clause have been obtained for a SH-A QPSK-1/3 configuration in AWGN channel with no co-channel interferer and for different IBO values. The target figure of merit is 95 % ESR5 fulfillment.

The values of $OBO|_{IBO}$ used for the calculation of the C_{sat}/N are obtained by simulation, whereas $(C+I)/N$ is measured with real equipment. C_{sat} is defined as the relative satellite power at amplifier's saturation point needed to close the link budget, thus:

$$\left.\frac{C_{sat}}{N}\right|_{IBO} = \left.\frac{C+I}{N}\right|_{IBO} + OBO|_{IBO}$$

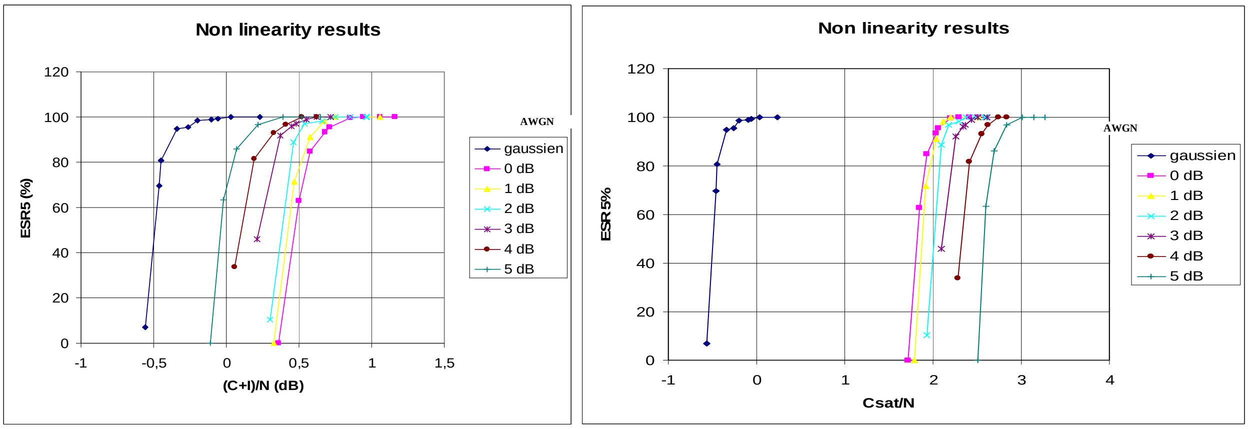


Figure A.1.6: ESR5 fulfillment performance for QPSK coding rate 1/3 in AWGN channel versus: a) *(C+I)/N* and b) *C_{sat}/N*

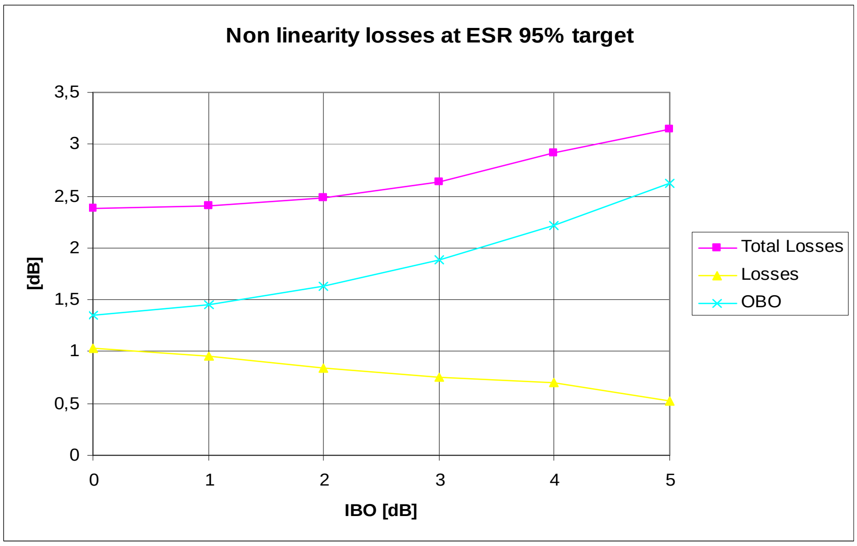


Figure A.1.7: Measured Total Degradation as a function of the IBO

Laboratory measurement results reported in figures A.1.6 and A.1.7 confirm the simulation findings that a large possible IBO working zone with a reduced degradation when the IBO is kept between 0 dB and 3 dB (corresponding to a working zone of 1,4 dB to 1,68 dB for OBO). Moreover, IBO/OBO point could be adapted according to the design of the payload.

A.2 Void

A.3 Simulation Conditions

A.3.1 Simulations with ideal channel estimator in demodulator

A.3.1.1 System parameters

The first focus is to optimize the interleaving scheme according to the channel impairments. The imperfect channel estimator in the demodulator is neglected at this stage.

To limit the number of cases, the system parameters are those considered relevant to hybrid S-Band systems that are the first to make use of DVB-SH. Waveform spectrum efficiencies resulting in bitrates at MPEG-TS level ranging between 2,2 Mbps and 2,8 Mbps, for a 5 MHz channelization, are considered. The number of services chosen ranges between 8 and 11.

A reference value of roughly 10 s has been considered for the end-to-end delay introduced by the DVB-SH processing. This allowance is to be shared between the physical layer and the LL-FEC, Turbo FEC delays being negligible (one Turbo FEC block is in the order of several msec).

Different configurations selected are shown in tables A.3.1 to A.3.4. Direct comparison of power efficiency is possible when the bit rate is strictly the same, otherwise the bit rate should be kept in mind when comparing different cases. The interest to increase the modulation order while keeping the net bit rate is to take advantage of lower Turbo code rates that can bring higher channel erasure resilience. Also different sharing between PL and LL-FEC for class 1 receivers for the same overall spectral efficiency is investigated.

System definitions for OFDM and TDM are reported in tables A.3.5 and A.3.6 respectively.

Finally, note that TDM and OFDM configurations are designed so that they can be synchronous in an SH-B system, so that code combining can be performed.

Table A.3.1: OFDM Reference configurations for class 1 receivers (5 MHz)

Parameter/Case name	16QAM_1/4_S	16QAM_2/7_S	QPSK1/2_S	QPSK2/3_S
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	QPSK	QPSK
PHY FEC rate	1/4	2/7	1/2	2/3
LL-FEC rate (recommended)	2/3	7/12	2/3	1/2
OFDM Symbols/coded FEC	8,00	7,00	8,00	6,00
Services	8	8	8	8
Bit rate/service (at TS level)	279,8 kbps	273,6 kbps	279,8 kbps	277,7 kbps
MPEG TS total bit rate	3,357 Mbps	3,752 Mbps	3,357 Mbps	4,443 Mbps
Early interleaver duration	0 ms	0 ms	0 ms	0 ms
Late interleaver duration	211 ms	211 ms	211 ms	211 ms

Table A.3.2: OFDM Reference configurations for class 2 receivers (5 MHz)

Parameter/Case name	QPSK1/3_U	QPSK_1/3_UL	16QAM_1/5_U	16QAM_1/5_UL
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	16QAM	16QAM
PHY FEC rate	1/3	1/3	1/5	1/5
LL-FEC rate (recommended)	1	1	1	1
OFDM Symbols/coded FEC	12	12	10	10
# Services	8	8	9	9
Bit rate/service (at TS level)	277,7 kbps	277,7 kbps	296,2 kbps	296,2 kbps
MPEG TS Total bit rate	2,222 Mbps	2,222 Mbps	2,666 Mbps	2,666 Mbps
Early interleaver duration	11 000 ms	10 000 ms	11 000 ms	10 000 ms
Late interleaver duration	0 ms	215 ms	0 ms	215 ms

Table A.3.3: TDM Reference configurations for class 1 receivers, compatible with OFDM QPSK (GI 1/4)

Parameter/Case name	T-8PSK1/3_S	T-QPSK1/2_S
Bandwidth (including R.O.)	4,89 MHz	4,89 MHz
Modulation	8PSK	QPSK
PHY FEC rate	1/3	1/2
UL FEC rate (recommended)	2/3	2/3
CW per SH-FRAME	79	79
Services	9	9
Bit rate/service (at TS level)	288,9 kbps	288,9 kbps
MPEG TS Total bit rate	3,900 Mbps	3,900 Mbps
Early interleaver duration	0 ms	0 ms
Late interleaver duration	190 ms	180 ms

Table A.3.4: TDM Reference configurations parameters for class 2 receivers

Parameter/Case name	T-8PSK2/9_U	T-8PSK2/9_UL	T-QPSK1/3_U	T-QPSK1/3_UL
Bandwidth (including R.O.)	4,89 MHz	4,89 MHz	4,89 MHz	4,89 MHz
Modulation	8PSK	8PSK	QPSK	QPSK
PHY FEC rate	2/9	2/9	1/2	1/3
LL-FEC rate (recommended)	Not used	Not used	Not used	Not used
CW per SH-FRAME	52	52	52	52
Services	9	9	9	9
Bit rate/Service (at TS level)	285,2 kbps	285,2 kbps	285,2 kbps	285,2 kbps
MPEG TS Total bit rate	2,567 Mbps	2,567 Mbps	2,567 Mbps	2,567 Mbps
Early interleaver duration	11 000 ms	10 000 ms	11 000 ms	10 000 ms
Late interleaver duration	0 ms	180 ms	0 ms	180 ms

Table A.3.5: System definitions for OFDM simulations

Definition	System Parameter	Comment
Bandwidth definition	Noise bandwidth	5-MHz
	Signal bandwidth	Limited to modulated carriers (data + TPS + pilots)
Pilots insertion	Pilots as defined by DVB-T	boosted by 16/9
Total Signal Power	Total power of the received signal	Normalized to 1 All Channels (TU6 inclusive) It Affects the Es/N0 calculation
Definition of C/N	ratio of useful power to noise power in the used bandwidth (= non-zero subcarriers)	
Definition of S/N	$\frac{S}{N}=\frac{C}{N}.\frac{(N_{c-U}+N_{c-TPS}+N_{c-Pilot})}{N_{FFT}}$	According to "Bandwidth definition" and "Total Signal Power"
Definition of Es/N0	$\frac{E_{S-U}}{N_0}=\frac{S}{N}.\frac{N_{FFT}}{(N_{c-U}+N_{c-TPS}+16/9.N_{c-Pilot})}$	ES-U accounts for the useful energy per symbol N_FFT is the total number of carriers (FFT-size) Nc-U is the number of useful carrier Nc-TPS is the number of TPS Nc-Pilots is the number of pilots

Table A.3.6: System definitions for TDM simulations

Definition	System Parameter	Comment
Bandwidth definition	Noise bandwidth = Symbol rate	Square-root raised cosine
	Signal bandwidth = [Symbol rate]*1,15	Roll-off 15 % is proposed
Pilots insertion	80 symbols every 1 008 data symbols	Higher than DVB-S2: 36/1 440
Total Signal Power	Total power of the received signal	Normalized to 1 All Channels (TU6 inclusive) It Affects the Es/N0 calculation
Definition of C/N	ratio of useful power to noise power in the used bandwidth (=pilot power is taken into account)	
Definition of Es/N0	Ratio of Symbol energy to noise density	To be compared with FEC simulations without pilot symbols
Ratio C/N and Es/N0	$\frac{C}{N}=\frac{E_S}{N0}$	Pilots have the same power as data symbols. So the power is conserved

A.3.1.2 Physical layer FEC and De-mapper configuration

Table A.3.7 details the algorithm and parameters chosen for the simulated decoder and de-mapper. Other implementations may use different methods and parameters that could lead to slightly different results.

Table A.3.7: Physical layer FEC configuration

Criteria	Simulation Parameters	Comment
Code	Turbo Code 3GGP2	3GPP2 C.S0002-D [i.42], Version 1.0, Date: February 13, 2004
Information Block size	12 282 bits	
Decoder engine Implementation	LOG-MAP	
Number of iterations	8	Constant
De-mapper engine Implementation	LOG-MAP	as "Decoder engine implementation" - It Affects the LLR calculation

A.3.2 Void

A.3.3 Demodulator state machine principle

A.3.3.1 States representation of a demodulator

To emulate the demodulator behaviour in fading channels with deep and long fades, a state machine model was defined, based on the fact that demodulation takes place after channel estimation and that channel estimation is an iterative process that requires some time to converge. The overall behaviour is modelled with 4 states, representing the following effects:

- loss of demodulator synchronization because of received C/N falls below operational threshold. Consequently transitions to coasting mode and after some timeout to the re-acquisition mode;
- when the demodulator is operating below threshold, it feeds erased bits (neutral values) to the Turbo decoder;
- demodulator re-acquisition time;
- slicing impact on demodulator operation.

The demodulator threshold is represented with a single value, independently of the channel behaviour, while in real implementation, it will vary according to mobile channel condition (user speed, type of mobile channel). A reference receiver with a single antenna reception has been considered for the definition of the demodulator threshold.

In the model, the threshold margin is calculated as distance between the estimated C/N of the channel and the nominal demodulator threshold, both with Rayleigh, Rice channels. In the case of the 3-state LMS model, the margin is computed relative to the LOS receive C/N.

All receivers of a multiplexing frame (simulations use a fixed pattern of services distribution) are simulated and the states of the receivers are used to determine whether the LLR estimates of all bits in a received symbol should be erased. As indicated in figure A.3.1, it should be noted that all states have a transition to the idle state when the end of a burst occurs. While the green states are driven by the modulator behaviour, the red state represents the time slicing mechanism.

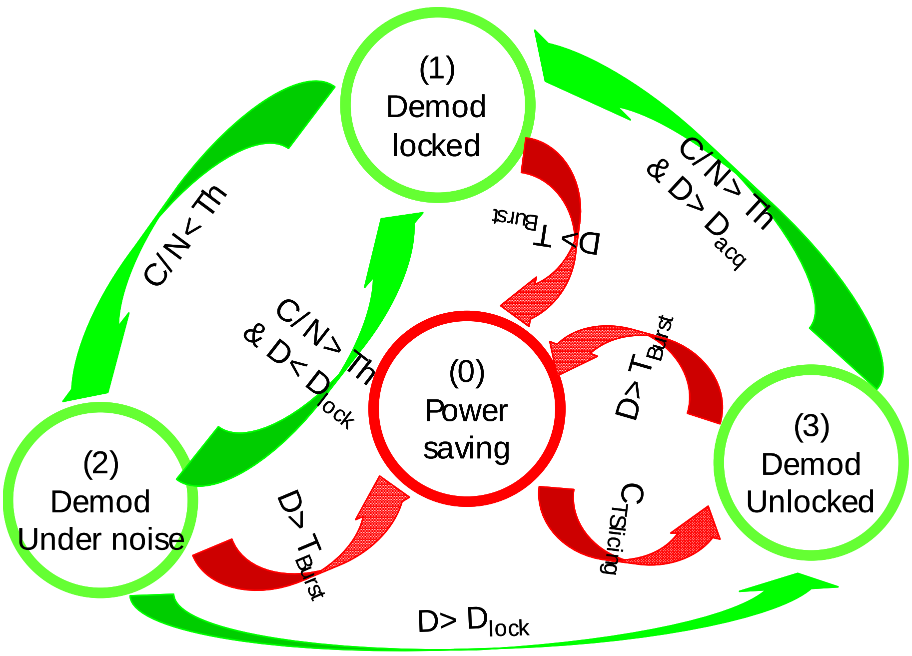


Figure A.3.1: State machine representing the demodulator process

The transitions parameters are:

- C/N: C/N estimated value at the receiver input;
- Th: receiver C/N Threshold (for staying locked not for successful frame decoding);
- D: duration of a state (origin of the transition);
- D_{lock}: maximum coasting state duration for the receiver operating under noise;
- T_{Burst}: burst duration (assumed constant);
- $C_{TSlicing}$: time interval between slices;
- D_{acq}: demodulator acquisition time (here assumed constant while in reality it will be a random variable).

The start transition stands for the condition required to wake up from the power saving mode. This is dependent on the time slicing mechanism. For example, in the current implementation of DVB-H the start condition ($C_{TSlicing}$) is as follows:

$$C_{TSlicing} = T>(Delta\_T- Delta\_T\_Margin)$$

Where Delta_T stands for the start of the next burst as it is indicated by the header of all clauses of the burst. Delta_T_Margin is the preparation time including some margin for the jitter. It is assumed that Delta_T is known as long as the demodulator has been locked for a certain period of the previous burst. However, if the demodulator has been unlocked or with C/N below the minimum required for frames detection over the whole duration of the previous burst this Delta_T remains unknown. A mechanism to overcome this limitation should then be implemented (i.e. establish a fixed time between bursts).

In any case, the $C_{TSlicing}$ condition is set in accordance with the time slicing strategy. In this way, this method is also suitable for testing the time slicing performance.

In simulations, it is assumed that a fixed and regular time-slicing is used for all programs in a multiplex and therefore, receivers go from idle state to acquisition state at a time corresponding to the time-slice start minus D_{acq}.

To represent faithfully the demodulator estimator, it is necessary to consider that a given number of symbols are affected by the channel state to lead to a channel estimator state transition. This number of symbols is $N_{estimator}$.

Impact of the demodulator behaviour in terms of performances will be show in clause A.12.2.5.

IDLE state (0)

The idle state represents the power saving mode. The idle state is the starting mode of the state machine. It is also entered each time a time slice ends.

Transition to the unlocked state occurs when a start of slice is anticipated. The start of the slice information is exploited to switch on the demodulator in advance to allow fast acquisition. This time advance is supposed to be D_{acq}.

DEMOD unlocked state (3)

The demodulator is trying to acquire the carrier. Neutral value LLRs are fed to the Turbo decoder.

If the channel estimates (A), averaged over $N_{estimator}$ symbols goes below the threshold during the acquisition time, the acquisition is restarted until a correct margin is reached for the whole duration of the acquisition time.

When the channel estimates are above threshold over the whole D_{acq} time, the transition to the locked state occurs.

DEMOD locked state (1)

Locked state occurs when the C/N estimate is above the demodulator threshold Th. Actual LLRs are delivered to the decoder.

This state lasts until averaged channel estimates are above the required threshold margin. If the averaged estimated C/N falls below the margin, the demodulator switches to the coasting state.

DEMOD coasting state (2)

This state persists as long as the channel C/N estimates averaged over $N_{estimator}$ symbols are under the demodulator threshold Th. Neutral values of LLR are fed to the Turbo decoder.

Each time the demodulator enters this state, a timer starts for a duration of D_{lock}. If the estimated demodulator C/N averaged over $N_{estimator}$ symbols remains under the threshold Th longer then D_{lock}, then the demodulator moves to the acquisition mode (3). Instead if after D_{lock} the averaged C/N estimate is above the demodulator threshold Th, then a transition to the locked mode (1) occurs and the timer is reset.

A.3.3.2 OFDM simulations

The demodulator state machine model should be activated to provide results on the LMS channel in OFDM mode. It accounts for the degradations introduced by the non-instantaneous channel synchronization in the receiver. As far as channel estimation and symbol phase synchronization are concerned, the ideal channel estimator remains, providing ideal phase and amplitude recovery to any demodulated symbol.

The demodulator state machine parameters are recalled in table A.3.8.

Table A.3.8: Demodulator state machine parameters definitions

Parameters	Definition
Bandwidth	Bandwidth of the OFDM signal, used to compute the number of subcarriers per symbol.
Mode	OFDM mode value indicating the symbol size (1K, 2K, 4K, 8K).
Constellation	Number of bits per subcarrier. Because input estimates are assumed to be on one subcarrier and output is supposed to correspond to LLR (bit by bit), the state machine generates constellation x number of subcarriers states per channel sample.
Punc_Pattern_ID	Puncturing pattern ID providing the coding rate.
TPS_CommonMultiplier TPS_NLateTaps TPS_NSlices TPS_SliceDistance TPS_NonLateIncrement	TPS parameters specifying the interleaver configuration.
MuxFrameProfile	Profile of the repetition pattern where the services are repeated on a regular pattern, each service being allocated a fixed number of code words during that period. The number of services is deduced from that profile.
ReceiverMarginThreshold_Db	Margin between LOS channel C/N level and demodulator threshold in dB. C/N = -3,5 dB is selected as the receiver threshold value.
LockDurationInSymbols	Number of OFDM symbols during which the demodulator remains locked while the signal averaged values are below threshold. This gives the duration of the "coasting" mode. 68 symbols is selected for this duration value.
AcquisitionDurationInSymbols	Number of OFDM symbols required to switch to locked mode after the demodulator is started or after it has left the locked mode. 68 symbols is selected for this duration value.
LengthOfAveragingInSymbols	Number of OFDM symbols over which the demodulator is supposed to average the received channel estimates to determine its own state. 12 symbols is selected for this duration value.

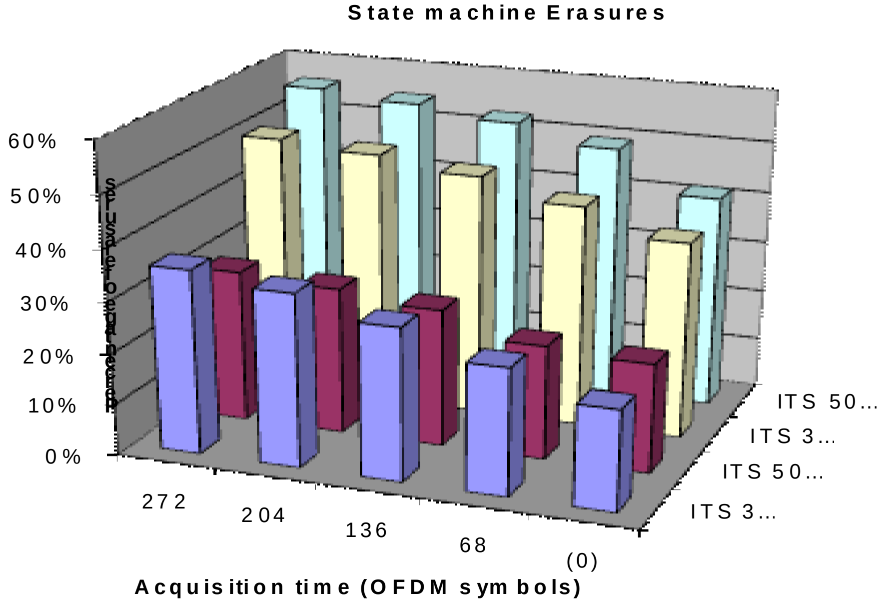
The set of demodulator state machine values corresponds to typical performance of existing DVB-H receivers. Simulation results without activation of the state machine are also provided as a benchmark.

A.3.3.3 TDM simulations

The same state machine model and parameters are used TDM simulations. The rationale is that, like for OFDM case, the TDM demodulators will rely on the "data aided" information from the pilots that are present in the TDM frame (equivalent to an OFDM symbol). Therefore, the acquisition and channel estimation algorithms will provide similar performances. This hypothesis is possibly slightly pessimistic for TDM which do not require the two dimensional (time/frequency) channel estimation of OFDM.

A.3.3.4 Analysis of acquisition time for LMS channels

The statistical analysis uses the symbols generated by the LMS channel model and feeds them into the receivers state machines model. Then, the number of occurrences of states are counted.



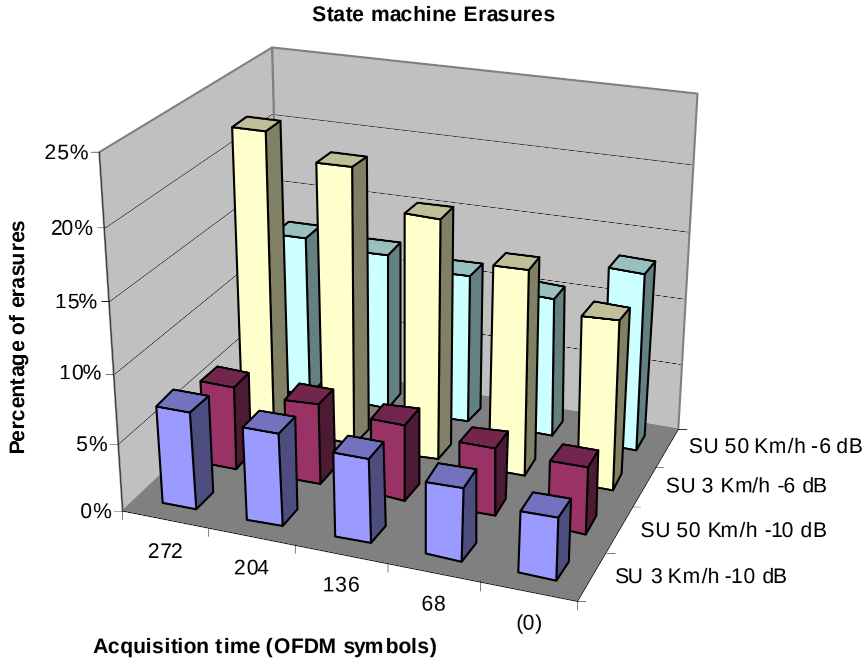


Figure A.3.2: Erasure rate for ITS and suburban channels

Simulation results reported in figure A.3.2 show that the LMS-ITS channel generates much more erasures than the LMS-SU. Also, having the acquisition time in the order of 68 OFDM symbols (one DVB-SH frame) does not increase significantly the percentage of erasures with respect to the ideal case (instantaneous acquisition).

Note that those results also depend on the averaging duration which can even decrease the number of erasures compared to the ideal acquisition case.

A.3.4 MPE-IFEC Simulation conditions

A.3.4.1 Definitions and interfaces

The following interfaces are defined and represented in figure A.3.3:

- on sender side:
  - E1 represents the IP flow coming into the IP encapsulator; these IP flows are of variable bit rates since they represent individual services;
  - E2 represents the TS interface at the output of the IP encapsulator; the TS bit rate is fixed and is a function of the waveform parameters;
  - E3 is the output of the DVB-SH modulator;
- on the receiver side:
  - R3 is the reception of the DVB-SH signal, at the input of the SH baseband decoder;
  - R2 is the output of the SH baseband receiver delivering a TS stream at continuous bit rate after all physical deinterleaving and decoding processes;
  - R1 is the output of the SH link layer receiver delivering IP flows at variable bit rates after all MPE-IFEC deinterleaving and decoding processes;
  - finally R0 is the output of the H264 video buffer ready for rendering.

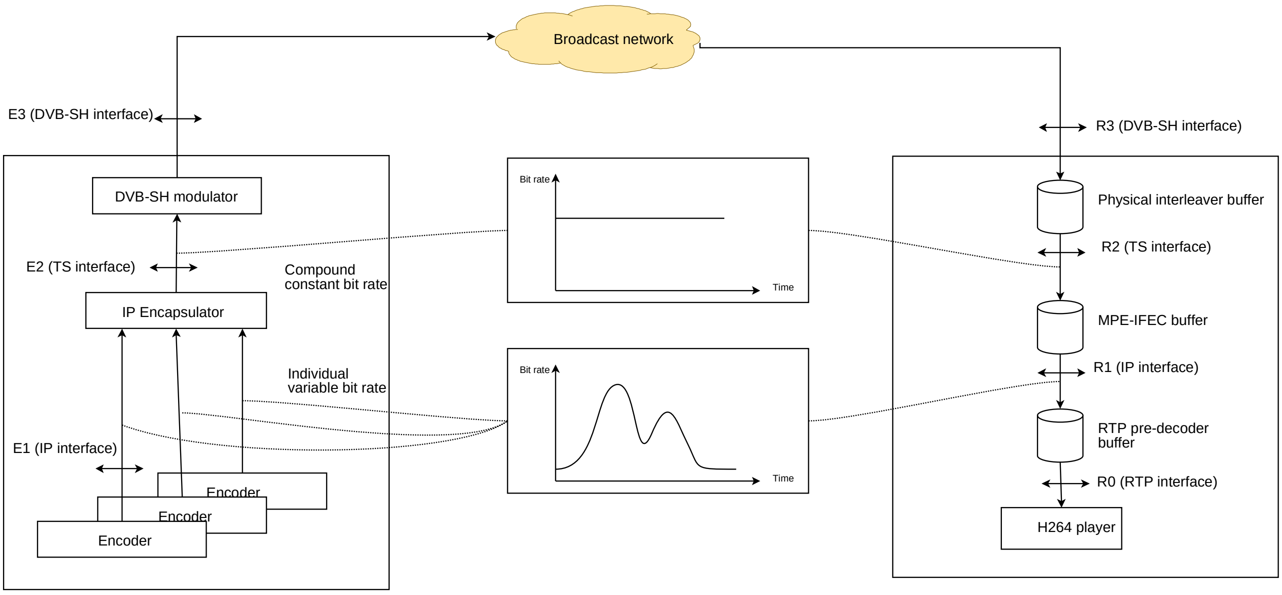


Figure A.3.3: Receiver interfaces

All simulation results have been measured at interface R1.

Some of the processes in the receiver may introduce jitter during transitions like zapping. The following performance results have been produced assuming a jitter-free interface, so assuming a constant delay in the physical and link layer buffers. Performance during transitions like zapping at non-jitter free interfaces are not scope of this implementation guidelines release. Performance including RTP buffer are found in the IPDC over DVB-SH implementation guidelines. However some aspects of zapping time improvements techniques, without performance figures, can be found in clause 6.2.5.2.

A.3.4.2 Decoder architecture

A simple decoder architecture is presented in figure A.3.4. It implements the following features:

- event driven approach:
  - time outs for the reception of a time slice burst with simple timing heuristics based on delta-t and max_burst_durations;
  - timing are based on ADST: one ADST is output at a deterministic time; this enables to output an ADST even in the case of a "starvation" due to a long erasure event;
- support of erased bursts detection:
  - when erased bursts are detected, they are sequentially mapped over the ADT to avoid having ADT with outdated columns when decoding is requested;
- "brute force approach" for the decoding:
  - the decoding is initiated every time a new information concerning the Encoding Matrix (ADT and FDT) is detected within a specific range;
  - this enables to output the ADST based on output time without requiring any new decoding since, when the ADST needs to be output, all the relevant information has already been processed by previous "brute force decoding"; the ADST is the best one with received information;
  - this enables also to support the decoding in different configurations by profiling the range of EM to be decoded; one typical configuration is to decode only at the jitter-free interface (no more information will be received so it is time to decode); another one is to decode by anticipation, especially during the zapping period without having received all parity;
  - this also enables to support a $D > 0$;
- typical configurations:
  - anticipated decoding:
    - Min_ADT = Min(ifdt_index(k,S-1); adt_index(k-D, 0));
    - Max_ADT = Min(ifdt_index(k,0); adt_index(k-D,B-1));
  - without anticipated decoding:
    - Min_ADT = Min(ifdt_index(k,S-1); adt_index(k-D, 0));
    - Max_ADT = Min(ifdt_index(k,0); adt_index(k-D,0)).

This decoder can be used for providing performance at the jitter free interface by using the non anticipated decoding range and $D = 0$.

- Min_ADT = Min(ifdt_index(k,S-1); adt_index(k, 0)) [M] = Min ((k - S- 2 + M)[M], k[M]).
- Max_ADT = Min(ifdt_index(k,0); adt_index(k,0)) [M] = Min ((k-1+M)[M], k[M]).

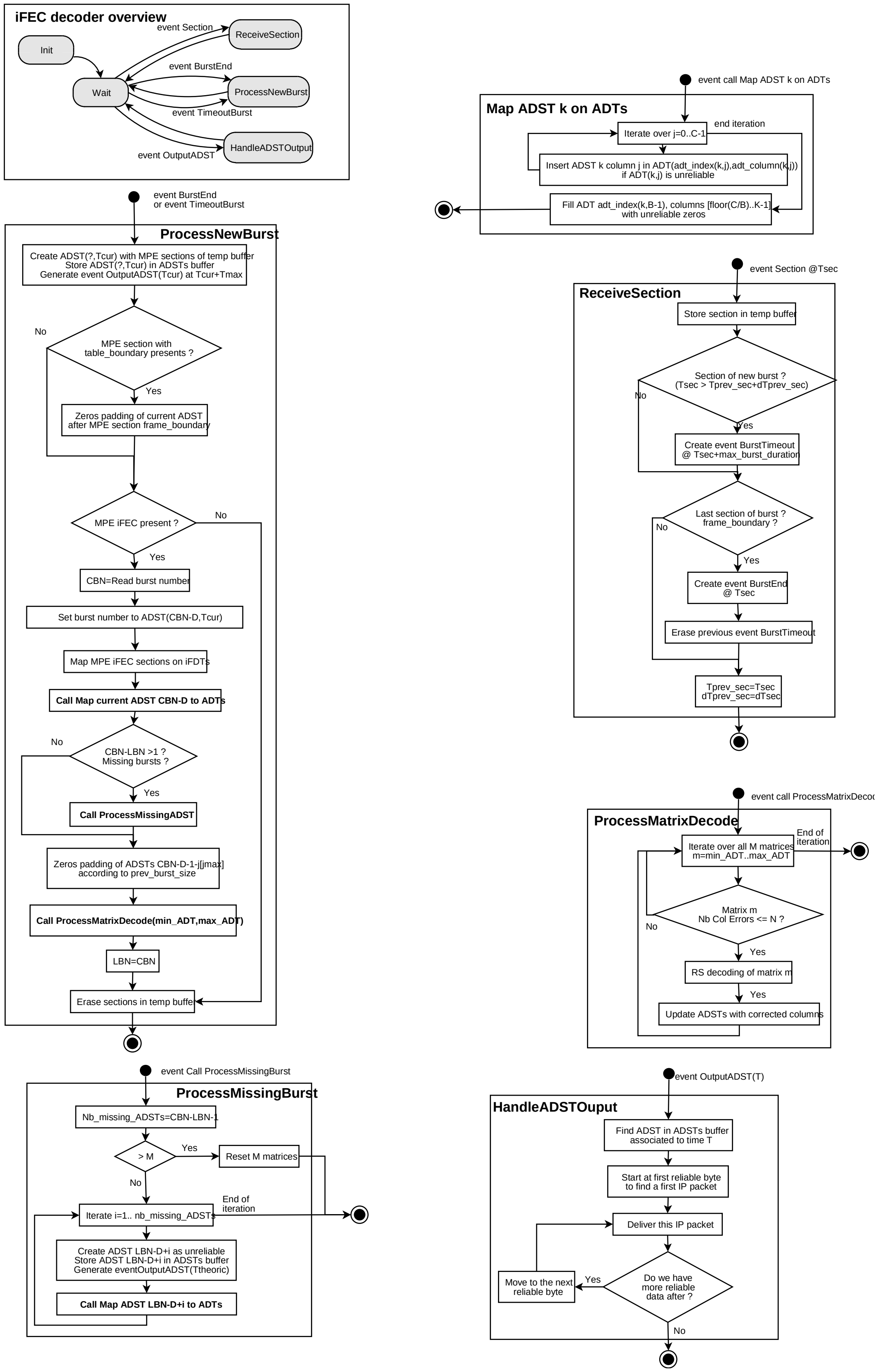


Figure A.3.4: Simple MPE-IFEC decoder architecture

A.3.4.3 Other parameters

The following parameters and options have been sought during simulation:

- IP size: set to 1 000 bytes;
- encapsulation: MPE;
- section packing: on;
- errors processing: when a TS of a section is lost: the full MPE section is considered as lost;
- the MPE-IFEC makes usage of TS level dumps which have a constant size at TP level; this fixed number of TP gives a capacity that is used to convey the IP flows using MPE sections and the RS parity using the MPE-IFEC. If the TS capacity is not sufficient to transport an IP packet or an MPE-IFEC section, the corresponding TS are set to null and not used.

A.4 Void

A.5 Void

A.6 Void

A.7 Propagation channels

It should be remarked that Rayleigh and Rice channel models definitions are also used in the context of DVB-T/H but they refer to two different contexts:

- The stationary Rice/Rayleigh channel models defined in EN 300 744 [28], annex B. The models represent **stationary** reception with antennas on the roof of a house (model "F1") or reception of signal from very high power repeater (as typical for DVB-T networks) with small antennas (model "P1"). The models are also (improperly) called RICE and RAYLEIGH in the context of DVB-T/H measurement. In reality the models represent a particular realization of TDL with stochastic fading process and TDL complex taps are fixed. These models play a minor role for the evaluation of DVB-SH and have been only used in an intermediate phase of the project.
- The uncorrelated Rice/Rayleigh channel models used in clauses 7.2.2.6.2 and 7.2.2.6.3 of the present document. The performance of the system depends highly on the speed and the interleaver length. For the evaluation of the FEC decoding a simple channel model representing an ideal interleaving (upper bound in performance compared to finite length interleaving) has been selected. A symbol-by-symbol independent Rayleigh amplitude sample distribution is used to represent typical terrestrial FEC operating conditions. In a similar way a symbol-by-symbol independent Rice amplitude sample distribution is used to represent typical satellite FEC operating conditions.

A.7.1 Terrestrial Channels

A.7.1.1 Channel Model Definitions

Maximum Doppler Frequency: $f_D = v \cdot \frac{f_0}{c}$ being v the mobile speed, f_0 the carrier frequency and c the speed of light (c=3 10^8 m/s).

Jakes (Rayleigh) Doppler Profile:

$$S(f) = \frac{1}{\pi f_D \sqrt{1-(f/f_D)^2}} \operatorname{rect}\left(\frac{f}{2 f_D}\right)$$

Gaussian Doppler Profile:

$$G(f;\sigma) = \exp\left(-\frac{f^2}{2\delta^2}\right)$$

For the terrestrial channel the state machine is not activated as it is assumed that the receiver is always in lock. Indeed, only the short term fading (TU6 model) is simulated, not the macro attenuation due to user large displacement within the cell (lognormal shadowing). For this channel, 10 minutes to 20 minutes duration were used for statistics.

A.7.1.1.1 Reference channels: AWGN and Rayleigh

Before measuring the system performance in more complex mobile channels it is important to derive the system performance under two reference channel cases: AWGN and flat Rayleigh fading. The first one will allow deriving the reference and ultimate system performance allowing easy comparison with DVB-SH IG reference values. The flat Rayleigh fading introduces the channel amplitude and phase variability allowing again equipment under test easy verification against IG reference performances. These models do not depend on terminal speed / Doppler.

A.7.1.1.2 Single transmitter: Classical TU6

The TU6 channel is widely used for terrestrial mobile networks. It is composed of a tapped delay line (TDL) with 6 taps as described in table A.7.1. To each tap is associated a delay and an average attenuation power on top of which a unit power Rayleigh complex fading process is applied. Each TDL tap Rayleigh process is independently generated and is characterized by correlated fading with a programmable power spectral density (PSD). A Rayleigh (Jakes) Doppler spectrum has been associated to the TU6 channel model. Table A.7.1 shows the TU6 reference channel parameters.

Table A.7.1: TU6 channel model parameters

Tap number	Delay (us)	Power (dB)	Doppler spectrum
1	0,0	–3	Rayleigh
2	0,2	0	Rayleigh
3	0,5	–2	Rayleigh
4	1,6	–6	Rayleigh
5	2,3	–8	Rayleigh
6	5,0	–10	Rayleigh

A.7.1.1.3 Single transmitter: TU6 plus lognormal shadowing

As previously discussed for DVB-SH, it is interesting to extend the classical TU6 by adding a lognormal shadowing on top. It is suggested to have a common lognormal shadowing process multiplying the TU6 output with zero mean and standard deviation 8 dB. The actual range of C/N in dB is to be defined as an output of the simulation exercise. The lognormal process coherence distance is 20 m [38].

A.7.1.1.4 SFN MBRAI model 1 for weak long echoes

As first SFN model, it is proposed to reuse the one proposed in the document EICTA MBRAI document [14], p.26-27 for SFN with weak long echoes.

The channel profile parameters are described in table A.7.2 Where *C/Nmode* represents the C/N required in AWGN for the selected waveform configuration. We remark that the second SFN transmitter delay is not absolute but related to the selected OFDM Guard Interval (GI). This assumes that the GI is correctly sized for the SFN geometry and waveform parameters. This is a potential limitation of the model that will be overcome by the SFN Model 4.

As for TU6 the fading PSD is Rayleigh distributed with the same relation with the maximum Doppler f_D.

Table A.7.2: Mobile SFN synchronization test channel for weak long echoes

Tap number	Delay (us)	Power (dB)	Doppler spectrum
1	0,0	–3	Rayleigh
2	0,2	0	Rayleigh
3	0,5	–2	Rayleigh
4	1,6	–6	Rayleigh
5	2,3	–8	Rayleigh
6	5,0	–10	Rayleigh
7	0,8xGI(1/4)+0,0	–3 – *C/Nmode*	Rayleigh
8	0,8xGI(1/4)+0,2	0 – *C/Nmode*	Rayleigh
9	0,8xGI(1/4)+0,5	–2 – *C/Nmode*	Rayleigh
10	0,8xGI(1/4)+1,6	–6 – *C/Nmode*	Rayleigh
11	0,8xGI(1/4)+2,3	–8 – *C/Nmode*	Rayleigh
12	0,8xGI(1/4)+5,0	–10 – *C/Nmode*	Rayleigh

A.7.1.1.5 SFN MBRAI model 2 for strong long echoes

This second SFN model is extracted from the document: EICTA MBRAI [14], p. 26-27 and corresponds to a SFN with strong long echoes. The channel profile parameters are summarized in table A.7.3.

Table A.7.3: Mobile SFN synchronization test channel for long echoes

Tap number	Delay (us)	Power (dB)	Doppler spectrum
1	0,0	–3	Rayleigh
2	0,2	0	Rayleigh
3	0,5	–2	Rayleigh
4	1,6	–6	Rayleigh
5	2,3	–8	Rayleigh
6	5,0	–10	Rayleigh
7	0,8xGI(1/4)+0,0	–3	Rayleigh
8	0,8xGI(1/4)+0,2	0	Rayleigh
9	0,8xGI(1/4)+0,5	–2	Rayleigh
10	0,8xGI(1/4)+1,6	–6	Rayleigh
11	0,8xGI(1/4)+2,3	–8	Rayleigh
12	0,8xGI(1/4)+5,0	–10	Rayleigh

A.7.1.1.6 SFN MBRAI model 3 for strong short echoes

NOTE: This model is typically less critical than SFN model 2. Therefore derivation of results is optional.

The third SFN model is extracted from the document: EICTA MBRAI document [14] p. 28 and corresponds to SFN with strong short echoes. The channel profile parameters are summarized in table A.7.4.

Table A.7.4: Mobile SFN synchronization test channel for strong short echoes

Tap number	Delay (us)	Power (dB)	Doppler spectrum
1	0,0	–3	Rayleigh
2	0,2	0	Rayleigh
3	0,5	–2	Rayleigh
4	1,6	–6	Rayleigh
5	2,3	–8	Rayleigh
6	5,0	–10	Rayleigh
7	6,0	–3	Rayleigh
8	6,2	0	Rayleigh
9	6,5	–2	Rayleigh
10	7,6	–6	Rayleigh
11	8,3	–8	Rayleigh
12	11,0	–10	Rayleigh

A.7.1.1.7 Portable Indoor and Outdoor channel model

The PI & PO model here summarised is extracted from the document: EICTA MBRAI [14], p. 26-27. The Portable Indoor (PI) and portable outdoor channel models have been developed by the Wing-TV project for describing the slowly moving hand held reception indoors and outdoors. The channel models are based on measurements in DVB-H Single Frequency Networks and have paths from two different transmitter locations. Definitions of the taps for the channels are given in sub-table 8 and sub-table 9 of table A.7.6. The indicated Doppler frequency of 1,5 Hz is corresponding to a 3 km/h velocity at mid UHF. It is suggested to scale up the original PI & PO Doppler spectra (UHF) to be representative of S-band so the f_D value should be increased to 6 Hz at 2,2 GHz. The Doppler spectra of various taps are defined in table A.7.5.

Table A.7.5: Doppler spectra of PI/PO model various taps

Table 7 Doppler Spectrum Definitions for PI and PO Channels

Spectrum for the 1st tap	Spectrum for taps 2-12
$0.1G(f;0.08f_D)+\delta(f-0.5f_D)$	$G(f;0.08f_D)$

Table A.7.6: PI/PO channel configuration

Table 8 Definition of PI channel

Path	Delay (μs)	Power (dB)	Doppler Spectrum	f_D (Hz)	STD Norm.
1	0,0	0,0	See Table 7	1,5	0,08
2	0,1	–6,4	Gauss	1,5	0,08
3	0,2	–10,4	Gauss	1,5	0,08
4	0,4	–13,0	Gauss	1,5	0,08
5	0,6	–13,3	Gauss	1,5	0,08
6	0,8	–13,7	Gauss	1,5	0,08
7	1,0	–16,2	Gauss	1,5	0,08
8	1,6	–15,2	Gauss	1,5	0,08
9	8,1	–14,9	Gauss	1,5	0,08
10	8,8	–16,2	Gauss	1,5	0,08
11	9,0	–11,1	Gauss	1,5	0,08
12	9,2	–11,2	Gauss	1,5	0,08

Table 9 Definition of PO channel

Path	Delay (μs)	Power (dB)	Doppler Spectrum	f_D (Hz)	STD Norm,
1	0,0	0,0	See Table 7	1,5	0,08
2	0,2	–1,5	Gauss	1,5	0,08
3	0,6	–3,8	Gauss	1,5	0,08
4	1,0	–7,3	Gauss	1,5	0,08
5	1,4	–9,8	Gauss	1,5	0,08
6	1,8	–3,3	Gauss	1,5	0,08
7	2,3	–5,9	Gauss	1,5	0,08
8	3,4	–20,6	Gauss	1,5	0,08
9	4,5	–19,0	Gauss	1,5	0,08
10	5,0	–17,7	Gauss	1,5	0,08
11	5,3	–18,9	Gauss	1,5	0,08
12	5,7	–19,3	Gauss	1,5	0,08

A.7.2 Satellite Channels

A.7.2.1 Propagation channels

A.7.2.1.1 AWGN Channel

This simple reference channel is considered to be very useful to see the overall demodulator performance and allows easy comparison with ideal performance.

A.7.2.1.2 Rician channel Model

Rice channel is another easy model to see the demodulator performance in dynamic conditions and allow an easy comparison with simulation results. It is proposed to test the Rice factor K = 5 dB at speeds of 3 kmph, 50 kmph and 120 kmph. Additional mobile speed of 80 kmph and 170 kmph will be required.

A.7.2.1.3 Land Mobile Satellite Channel Model

The following LMS propagation channel environments have been selected for laboratory tests.

	Satellite
URBAN	N/A
SUBURBAN	3-state LMS-SU (50 Km/h) 3-state LMS-SU (3 Km/h)
RURAL	3-state LMS-ITS (50 Km/h) Quasi-stationary Ricean, K=5 dB; 7 dB; 10 dB

The C/N and C/I levels to be used are derived from link budgets (see clause 11), with terminal characteristics given in clause 10. The used values are listed in clause A.9.

The S-band parameters for the 3-state LMS channels are obtained from [39] and [40] and are dependent to the elevation angle. In addition to the 40 degrees elevation angle used for IGv1 results for which SU and ITS parameters are available (in [41]), 30 degrees (in [42]) and 50 degrees (in [43]) elevation parameters have been added. For these two new satellite elevations, data is only available for the SU environment. Furthermore, a close look at the 50° model parameters revealed possible inaccuracy of the model parameters due to the limited data set available. So results for this elevation should be taken with caution.

For the LMS channels the simulations were performed with and without receiver state machine. The parameters for the LMS channels are obtained from [44] and [45] and are specified in tables A.7.7, A.7.8 and A.7.9. *The applicability of the selected values is not claimed to be universal but only as a starting point to differentiate between waveform configurations*. For a discussion on the applicability of such models, refer to clause 4. From experiments, it was found that the simulation length when the 3-state Markov chain is activated should be at least 1 hour of simulated time for 50 kmph and at least 3 hours for 3 kmph.

Table A.7.7: LMS model states based on measurements parameters for 30° elevation

Environment	State 1: LOS			State 2: Shadowing			State 3: Deep shadow		
	α (dB)	ψ (dB)	MP (dB)	α (dB)	ψ (dB)	MP (dB)	α (dB)	ψ (dB)	MP (dB)
Open (*)									
Suburban	-0,5	1,0	-15,0	-4,7	1,5	-19,0	-7,0	3,0	-20,0
Intermediate Tree-Shadow									
Heavy Tree-Shadow (see note 2)									

Environment	[P]			[W]	d_{corr} (m)	L_{frame} (m)	L_{trans} (m)
Open (see note 2)							
Suburban	0,9531	0,0350	0,0119	0,7467	2,3 (see note 1)	4,0 (1)	2,2 (see note 1)
	0,1891	0,6198	0,1911	0,1511		4,0 (1)	
	0,0631	0,3065	0,6304	0,1022		4,0 (1)	
Intermediate Tree-Shadow							
Heavy Tree-Shadow (see note 2)							

NOTE 1: These values have been extrapolated since they are not given in [Fontan2].
NOTE 2: Not simulated, for information only.
NOTE 3: LMS model for 30° Elevation obtained from L-Band models. Simulations with this model may be inaccurate.

Table A.7.8: LMS model states based on measurements parameters for 40° elevation

Environment	State 1: LOS			State 2: Shadowing			State 3: Deep shadow		
	α (dB)	ψ (dB)	MP (dB)	α (dB)	ψ (dB)	MP (dB)	α (dB)	ψ (dB)	MP (dB)
Open (*)	0,1	0,37	-22,0	-1,0	0,5	-22,0	-2,25	0,13	-21,2
Suburban	-1,0	0,5	-13,0	-3,7	0,98	-12,2	-15,0	5,9	-13,0
Intermediate Tree-Shadow	-0,4	1,5	-13,2	-8,2	3,9	-12,7	-17,0	3,14	-10,0
Heavy Tree-Shadow (2)	-	-	-	-10,1	2,25	-10,0	-19,0	4,0	-10,0

Environment	[P]			[W]	d_{corr} (m)	L_{frame} (m)	L_{trans} (m)
Open (2)	0,9530	0,0431	0,0039	0,5	2,5	8,9	12,4
	0,0515	0,9347	0,0138	0,375		7,5	
	0,0334	0,0238	0,9428	0,125		4,0 (1)	
Suburban	0,8177	0,1715	0,0108	0,4545	1,7	5,2	2,2
	0,1544	0,7997	0,0459	0,4545		3,7	
	0,1400	0,1433	0,7167	0,091		3,0 (1)	
Intermediate Tree-Shadow	0,7193	0,1865	0,0942	0,3929	1,5	6,3	2,6
	0,1848	0,7269	0,0883	0,3571		6,3	
	0,1771	0,0971	0,7258	0,25		4,5	
Heavy Tree-Shadow (2)	0,7792	0,0452	0,1756	0	1,7	-	3,5
	0	0,9259	0,0741	0,5		4,8	
	0	0,0741	0,9259	0,5		4,5	
NOTE 1: These values have been extrapolated since they are not given in [17]. NOTE 2: Not simulated, for information only.							

Table A.7.9: LMS model states based on measurements parameters for 50° elevation

Environment	State 1: LOS			State 2: Shadowing			State 3: Deep shadow		
	α (dB)	ψ (dB)	MP (dB)	α (dB)	ψ (dB)	MP (dB)	α (dB)	ψ (dB)	MP (dB)
Open (*)									
Suburban	-0,5	1,0	-17,0	-6,5	2,5	-17,0	-14,0	2,5	-20,0
Intermediate Tree-Shadow									
Heavy Tree-Shadow (2)									

Environment	[P]			[W]	d_{corr} (m)	L_{frame} (m)	L_{trans} (m)
Open (2)							
Suburban	0,7498	0,2462	0,0040	0,1626	2,3 (1)	4,0 (1)	2,2 (1)
	0,0479	0,9160	0,0361	0,7642		4,0 (1)	
	0,0554	0,3296	0,6150	0,0732		4,0 (1)	
Intermediate Tree-Shadow							
Heavy Tree-Shadow (2)							
NOTE 1: These values have been extrapolated since they are not given in [Fontan2]. NOTE 2: Not simulated, for information only. NOTE 3: LMS model for 50° Elevation obtained from L-Band models. Simulations with this model may be inaccurate.							

The parameters values are:

- α: Average value of the attenuation on the LOS link for a state.
- ψ: Sandard deviation of the attenuation on the LOS link for a state.

- [P]: Probability of occurrence of a transition (3x3 matrix).
- [W]: Total probability of having a given state.
- dcorr: Correlation distance of the channel.
- L_{Frame}: Minimum state frame length as defined in [17].
- L_{Trans}: Transition region length as defined in [16].

A.7.2.1.4 Satellite quasi stationary channel

For portable satellite reception it was considered important to also analyse the case of LOS with multipath but very slow user movement (e.g. 0,5 kmph or less). This channel is called quasi-stationary satellite. The results of this analysis are exploited in clause 11.7.2.2. The channel is modelled as AWGN over the FEC block size with a block-by-block slight C/N change due to the very slow fading. Hence, we can analytically compute for a given LOS C/N the probability to be below the demodulator threshold (see clause 11).

Recalling the PDF of the power for a Ricean faded signal:

$$p_D(d) = \frac{1}{2\sigma^2} \exp\left\{-\left(\frac{s^2+d}{2\sigma^2}\right)\right\} I_o\left(\frac{s\sqrt{d}}{\sigma^2}\right)$$

where d is the LOS signal power;

σ^2 the variance of the in-phase and quadrature multipath fading components; and

$I_o(\cdot)$ is the modified zero-th order Bessel function.

The carrier to multipath ratio is defined as the ratio between the power of the signal complex envelope LOS component s and the power of the multipath (random) power component $2\sigma^2$.

A.8 Evaluation criteria

A.8.1 Introduction

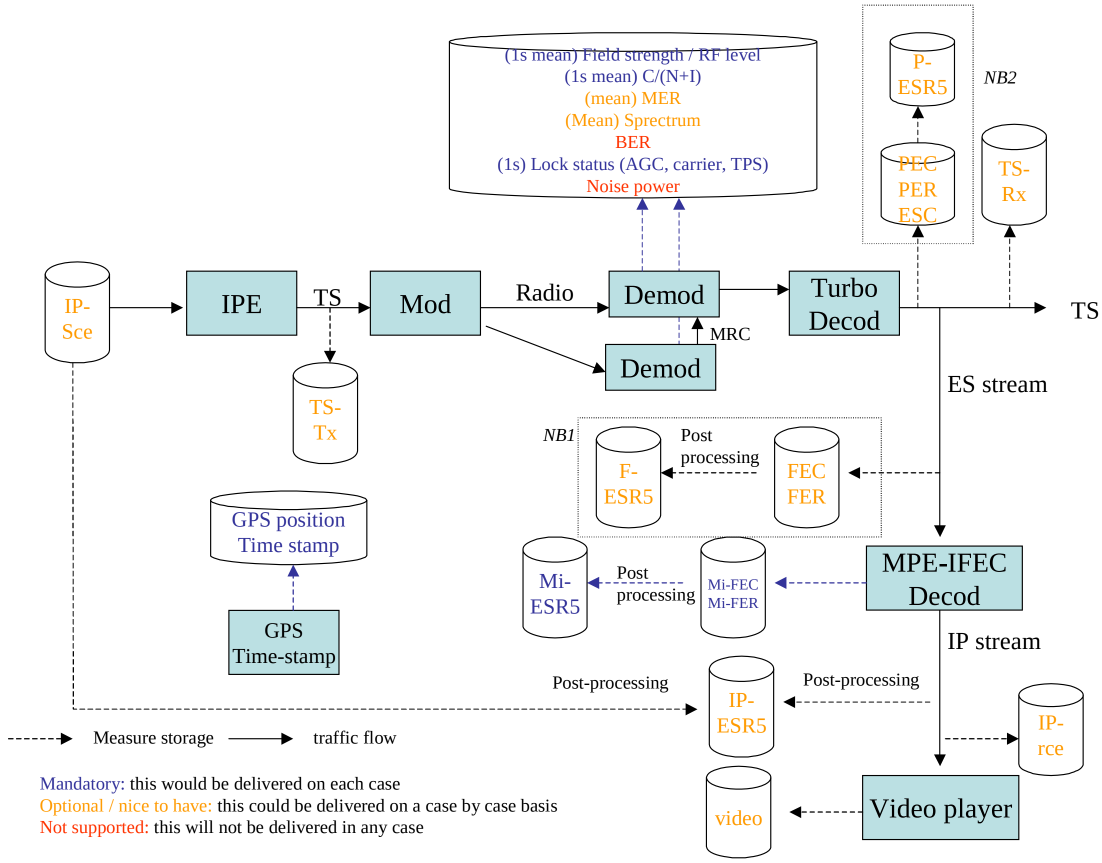
In case of long time interleaving and block fading, BER and WER alone are not representative of actual perceived quality. A long mute may be better than repetitive short error events. To take this consideration into account, we define new criteria in clause A.8.3 after having defined the logging capabilities for both field trial and laboratory environments in clause A.8.2.

A.8.2 Laboratory and field trial logging

These logging capabilities are common to terrestrial and satellite contexts.

A.8.2.1 Overview

A typical architecture of a laboratory and field trial measurement configuration is presented hereunder in figure A.8.1. The only difference between both configurations is the (x,y,z) coordinates are fixed in the laboratory case.



NOTE 1: NB1: requires at least one MPE-FEC section per burst.

NOTE 2: NB2: could be used in replacement of FEC/FER and F-ESR5.

Figure A.8.1: Field trial measurement configuration

A.8.2.2 Physical layer logging capabilities

The physical layer loggings are listed in table A.8.1. The dumps should be aligned with the GPS time stamp. This can be achieved, for instance, by creating a log file, which reports for each line the time stamp, the GPS position, and the pointer to the last written byte in each of the dumps.

Table A.8.1: Physical layer logging capabilities

Parameter	Time resolution	Used for	Derived from	Required preparation/Setup characterization	Status, implemented in version
Position	1 second	correlation between the recorded dumps and the position of the terminal	GPS		Mandatory
Time stamp	1 second	correlation between the recorded dumps and the position of the terminal	GPS, with timing interpolation		Mandatory
C/(N+I) estimation	1 second (average over at least 10 samples)	Alternative to field strength; allows simple correlation between received signal level and output of the physical layer decoder; verification of the channel status, esp. specific environments under scope of test	Demodulator		Mandatory
Lock status	1 second (reporting eventual lock losses during the past second)	detect eventual signal losses, useful for detecting unexpected behaviours of the terminal may be used to evaluate eventual gaps in the recorded dumps	Demodulator		Mandatory
Field strength	1 second (average over at least 10 samples)	Raw data for coverage analysis and time series generation	Band power measurement AGC values	AGC and antenna calibration	Desirable NB: 2 values(or more) when diversity is activated
Noise power level	1 second (average over at least 10 samples)	Estimate C/N for low field strength	Band power measurement in other band		Desirable
BER (channel BER) at demodulator output	Average over approx. 100 ms	Cross-check of demodulator performance	Analyze pilot fields or padding	cBER versus C/N for different fading channels	Desirable This can be measured with pre-configured modulator input
Strength of interference	On user request in case of un-expected behaviour	Identify critical interferer	AGC value Wideband power level Spectrum analysis	Spectrum mask allowed interference	Nice to have
MER	Stationary only	Signal quality analysis for stationary reception	Offline analysis of recorded I/Q data		Nice to have Format may differ (MER instead of I&Q)
Spectrum	One snapshot per second	Fading statistics vs. frequency axis	Offline analysis of recorded I/Q data		Nice to have Format may differ (FFT in {freq; power} coordinates
Interference level	Upon user request	Identify critical interferer	Spectrum analyzer		Nice to have

A.8.2.3 Link layer logging capabilities

The upper layer logging capabilities are listed in table A.8.2. Also here, the results should be aligned with the GPS time stamp.

Table A.8.2: Physical layer logging capabilities

Parameter	Time resolution	Used for	Derived from	Required preparation/ Setup characterization	Status, implemented in version
Burst Error Indicator	Burst ~1 second	Permits the analysis in terms of ESR5(20) service availability	Depends: - after IFEC decoding (see MiFER/MIFEC) - *after TC for Class-2 (see PER)*	Depends on the case	Mandatory
Physical layer frames/ MPEG-TS error series	MPEG2 packet	Evaluate WER and ESR5 performance Complete analysis of the performance after PHY (turbo) decoding. Map different MPE-IFEC profiles on the TS Requires additional signalling for detecting gaps (outages)	Output of turbo decoder	Special internal to MPEG2 TS signalling for detection of gaps if greater than MPEG2 CC	Desirable (at least for some sessions)
FEC/FER	Frame=burst~1 second	Frame Error Count / Frame Error Rate: measure after turbo-decoding at the level of MPE-FEC frame Gives an indication of the burst error indicator	Monitoring of the MPE-FEC frame (so at link level)	Requires at least 1 MPE-FEC section per burst because based on MPE-FEC frame	Desirable
MiFER/MiFER	Frame=burst~1 second	MPE iFec Frame Error Ratio after MPE-IFEC decoding Can be used for ESR5 computation	Logging at the output of the MPE-IFEC decoder	Needs link layer MPE-IFEC	Desirable for class 1 configurations
PER/PEC	MPEG2 packet	Packet error Rate measured after Turbo decoding; can be used to compute different MPE-IFEC schemes via post-processing	information derived from the TS_erroneous_flag set thanks to the CRC16 computed over each ST	Special internal to MPEG2 TS signalling for detection of gaps if greater than MPEG2 CC	Desirable
ADT columns status flags I/O	Each ADT column	Helps in reproducing the behaviour of different MPE-IFEC techniques, sharing the same parameters (B, S, T) of the encapsulated configuration (same results can be achieved by Meas. No. 1, but with more effort)	MPE-IFEC decoder input/output		Nice-to-have
MPE/-IFEC sections status	Section level	Permits a refined analysis of the recorded results	MPE-IFEC decoder logs	Continuity issues to manager	Nice to have
Full received (decapsulated) payload, at IP level	IP level	Permits eventual off-line play-out of the video Permits ESR5 computation at IP level based on comparison between IP input / IP output	Output of MPE/MPE-IFEC decapsulation		Nice to have

A.8.3 Performance metrics

A.8.3.1 Introduction

To define the performance metrics listed in figure A.8.2, we zoom on the receiver side on figure 5.1.

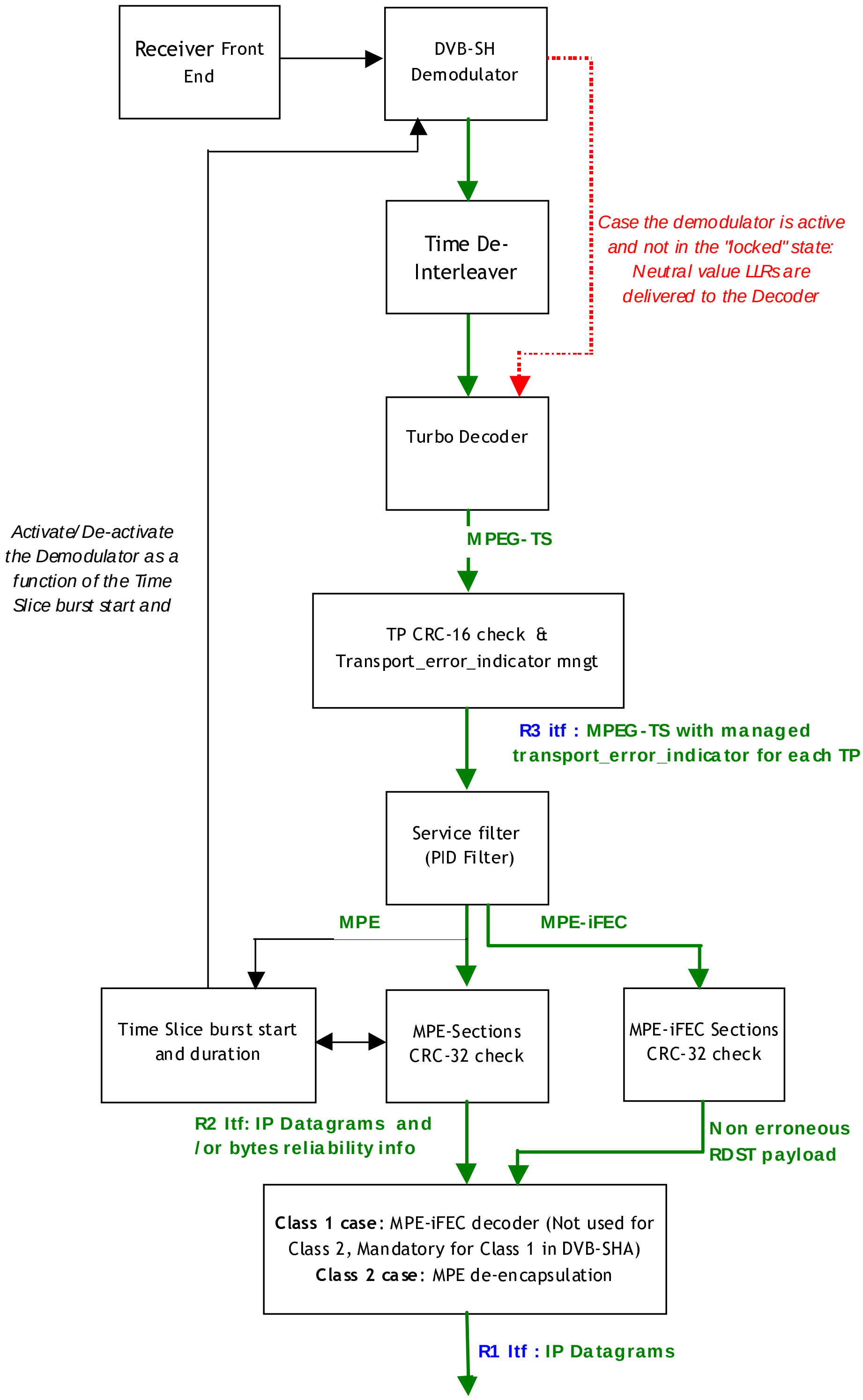


Figure A.8.2: Performance measurement points in the receiver

A.8.3.2 Output of the turbo decoder (R3)

The TS packet status is derived from the CRC16 in the turbo decode that sets the status of the transport_error_indicator bit in the TS header. From this information we can derive:

- The T_PER (Transport Packet Error Rate).
- The TP_ESC (TP Erroneous second count).
- The TP_ESR (TP Erroneous second ratio).

A.8.3.3 After PID filtering (R2)

Measurements at the R2 reference point provide:

- The FC (MPE Frame counter).
- The FEC (Frame Error Count).
- The FER (Frame Error Rate).

Depending on the receiver state (Active/Locked or Active/Not Locked), the receiver may receive the complete set, a partial set, or none of the transmitted MPE sections in a time-slice burst. MPE sections contain information on the next time slice burst used for the management of the receiver wake-up times, and that allow the detection of lost MPE frames provided that the receiver implements appropriate mechanisms.

The MPE Frame Counter (the FC) is incremented each time a time slice burst is expected. The FEC counts the number of erroneous MPE-Frames, i.e. MPE frames:

- Either containing at least one MPE section for which the CRC-32 check fails.
- Or lost MPE-frames.

The FER is simply expressed as FEC/FC, i.e. the ratio of erroneous MPE-frames.

Additionally, measurements at the R2 reference point provide in case MPE-IFEC is used:

- the MIFEC (MPE-IFEC Frame Error Count); and
- the MiFER (MPE-IFEC Frame Error Rate).

The MPE-IFEC Frame Error Counter (the MIFEC) is incremented each time an ADST table (or equivalently the IP datagram payload of a Time Slice Burst output by the MPE-IFEC decoder) is detected as erroneous. This decision relies on the RS decoder ability for errors correction (i.e. the capacity of the RS in terms of errors detection/correction).

The MPE-IFEC frame Error Rate (the MiFER) is simply expressed as MIFEC/FC, i.e. the ratio of erroneous ADST tables generated by the RS decoder.

A.8.3.4 After MPE decapsulation and MPE-IFEC decoding (R1)

Measurement at the IP level includes the IP PER (packet error rate). It is usually computed either by post processing comparison between sent and received files (a methodology that can be applied to any kind of traffic, even those encrypted) or by using packets with a specific header having a sequential number that enables to detect lost packets.

A.8.3.5 ESR5(20) metrics

A.8.3.5.1 Generic definition

The ESR5(20) definition is recalled hereafter:

"*ESR5(20) is the ratio of windows for which ESR5 is fulfilled, over the total number of windows ESR5 criterion is fulfilled when in a time interval of 20 seconds there is at most one second in error*."

The ESR5(20) evaluation formula is:

$$ESR5(20) = 1 - \frac{\sum_{w=1}^{N_w} \mathrm{ceil}\left\{ \frac{\max\left[0;\mathrm{n\_of\_erroneous\_seconds}(w) - 1\right]}{20} \right\}}{N_w}$$

where we assumed an individual observation window size of 20 s, N_w represents the number of observation windows and n_of_erroneous_seconds(w) is the number of erroneous seconds in the window w, the ceil function round the specified number up, and return the smallest number that is greater than or equal to the specified number.

The previous formula needs to be adapted to the type of measurements possible within the receiver:

- It is very important that ESR5(20) statistics are obtained over a long enough measurement time in particular at low mobile speed.
- The N_w 20 s windows used for the measurement may be disjoint or overlapped. The important is that enough statistics are accumulated to achieve reliable ESR5(20) estimate over the channel under consideration.
- The above equation can be easily generalised to different window lengths by simply replacing 20 to the actual window length in seconds and the -1 at the nominator by -5*window_length/100.

A.8.3.5.2 F-ESR5(20) and Mi-ESR5(20)

According to figure 5.1, the ESR5(20)measurement point is:

- R1 for class 1 receivers in a DVB-SH-A network configured with MPE-IFEC protection;
- R2 for class 2 and class 1 receivers in a DVB-SH network configured without MPE-IFEC protection.

So, a second is erroneous if it contains at least:

- one erroneous MPE-IFEC ADST (class 1 with MPE-IFEC), therefore called Mi_ESR5(20); or
- one erroneous MPE frame (class2 without MPE-IFEC/ class 1 without MPE-IFEC), therefore called F_ESR5(20).

The F_ESR5(20) and Mi_ESR5(20) evaluations are performed by a post processing tool that takes into account the Time Slice Burst repetition period.

A.8.3.5.3 IP_ESR5(20)

Sometimes, an additional measurement point is required, providing finer analysis and debug means. The erroneous second decision is based on the comparison of the transmitted IP datagrams carrying the traffic of one given service with the comparison of the IP datagrams output at the R1 interface. It allows the detection of:

- IP datagrams with bad checksum.
- Duplicated IP datagrams.
- Lost IP datagrams.
- IP datagrams with erroneous payload.

The ESR5(20) evaluated from the erroneous second decision based on comparison on the transmitted IP datagrams with the IP datagrams at the R1 interface in figure 5.1 is called the IP_ESR5(20).

A.8.3.6 Synthesis

Table A.8.3 gives the list of performance criteria.

Table A.8.3: Performance measurements criteria synthesis

Acronym	Full name	Measurement point (name and interface)	
T_PER	Transport Packet Error Rate	Turbo-decoder output	R3
TP_ESC	TP Erroneous second count	Turbo-decoder output	R3
TP_ESR	TP Erroneous second ratio	Turbo-decoder output	R3
FC	MPE Frame counter	In MPE decapsulation	R2
FEC	Frame Error Count	In MPE decapsulation	R2
FER	Frame Error Rate	In MPE decapsulation	R2
MIFEC	MPE-IFEC Frame Error Count	In MPE-IFEC decoder	R2
MiFER	MPE-IFEC Frame Error Rate	In MPE-IFEC decoder	R2
IP PER	IP Packet Error Rate	Output of the receiver	R3
F_ESR5(20)	Frame Erroneous Second ratio 20 s	In MPE decapsulation	R2
Mi_ESR5(20)	Mi-Frame Erroneous Second ratio 20 s	In MPE-IFEC decoder	R2
IP_ESR5(20)	IP Erroneous second ratio	Output of the receiver	R1

For a receiver nominal behaviour (after complete receiver validation tests), the IP_ESR5(20) and the F_ESR5(20) (no MPE-IFEC only) or Mi_ESR5(20) should give the same results.

A.9 Summary of simulated cases

Tables A.9.1 to A.9.3 hereafter provide all the simulation cases with the associated parameters. The satellite *C/N* and *C/I* are in line with the link budgets detailed in clause 11 (rationale with link-budgets is given at the end of this clause). In tables A.9.4 to A.9.6 for the satellite case *C/N* represents the LOS *C/N* (no channel fading/shadowing) while *C/I* represents the co-channel useful signal over interference and uplink *C/N* contribution.

Note that satellite link *C/I* and *C/N* are provided separately as the interference is suppose to fade together with the signal and is simulated like an extra noise source modulated by the same fade as the useful signal.

Table A.9.1 hereafter reports the list of simulated case in TU6 channel. The spectral efficiency value is given for information and corresponds to the case where MPE-IFEC code rate is equal to 1. The Demodulator State Machine is always OFF.

Table A.9.1: List of simulated cases in TU6 channel

Waveform configuration	Speed (Kmph)	C/I	C/N	ID	Overall Efficiency [bpsHz]	Additional parameters	Comment
O_16QAM_xx_S	3 to 50	20 dB	[2:18]	1	0,8 to 2,67	Table A.10.6	• Mobile fading bandwidth for a range from 3 Hz to 600 Hz. • All possible code rates and modulations are tested. • Demodulator State Machine OFF.
O_QPSK_xx_S	3 to 50	20 dB	[-2:14]	2	0,4 to 1,33	Table A.10.8	
O_16QAM_xx_UL	3 to 50	20 dB	[2:18]	3	0,8 to 2,67	Tables A.10.17 and A.10.18	
O_QPSK_xx_UL	3 to 50	20 dB	[-2:14]	4	0,4 to 1,33	Tables A.10.19 and A.10.20	

Tables A.9.2 and A.9.3 hereafter report the simulated cases to test different interleaver configurations. This is done for OFDM (QPSK and 16QAM) and TDM (only QPSK and 8PSK, no 16APSK). Only a subset of all possible code rates are tested for each selected modulation. And in each case two types of interleaver are tested (uniform late and uniform long). The Demodulator State Machine is always ON.

Table A.9.2: List of simulated TDM cases in LMS channel

Waveform configuration	Speed (Kmph)	C/I	C/N	ID	Overall Efficiency [bpsHz]	Additional parameters	Comment
T_QPSK_1/3_U_xx	LMS-SU - LMS-ITS; Elevation angle: 40 Speed 3 Kmph to 50 Kmph	12,3	12,0	1	0,67	Table A.10.39, IFEC CR=1	• Interleaver duration ranging from 500 ms up to 11 s. • Demodulator State Machine ON.
T_QPSK_1/3_UL_xx				2		Tables A.10.48 and A.10.49, IFEC CR=1	
T_8PSK_2/9_U_xx		11,8	12,5	3		Table A.10.43, IFEC CR=1	
T_8PSK_2/9_UL_xx				4		Table A.10.54, IFEC CR=1	
T_QPSK_1/2_U_xx		12,3	12,0	5	1,0	Table A.10.40, IFEC CR=1	
T_QPSK_1/2_UL_xx				6		Table A.10.49, IFEC CR=1	
T_8PSK_1/3_U_xx		11,8	12,5	7		Table A.10.44, IFEC CR=1	
T_8PSK_1/3_UL_xx				8		Table A.10.55, IFEC CR=1	
T_8PSK_2/5_U_xx		11,8	12,5	9	1,2	Table A.10.45, IFEC CR=1	
T_8PSK_2/5_UL_xx				10		Table A.10.56, IFEC CR=1	
T_QPSK_2/3_U_xx		12,3	12,0	11	1,33	Table A.10.41, IFEC CR=1	
T_QPSK_2/3_UL_xx				12		Table A.10.51, IFEC CR=1	

Table A.9.3: List of simulated OFDM cases in LMS channel

Waveform configuration	Channel	C/N [dB]	C/I [dB]	ID	Overall Efficiency [bpsHz]	Additional parameters	Comment
O_QPSK_xx_U_10s	LMS-SU - LMS-ITS; Elevation angle: 40 Speed 50	12,3	12,0	1	0,67	Table A.10.30, IFEC CR=1	• All coding Rates. • Demodulator State Machine ON.
O_QPSK_xx_UL_10s				2		Tables A.10.19 and A.10.20, IFEC CR=1	

Rationale between values used in Simulation and values given in clause 11 link-budgets

C/N are provided in link bugets from clause A.11.

As LMS simulations are lossless, C/N provided in link budgets are reduced from the amount of implementations losses identified in clause 10 (as well as in link budgets of clause 11).

Thus the following C/N are used in the LMS simulations.

Vehicular terminal:

Table A.9.4

Waveform	Sh-A	Sh-A	Sh-B	Sh-B
Modulation	QPSK	16QAM	QPSK	8PSK
Link budget LOS C/N (dB)	12,3	12,3	12,8	12,8
Implementation losses (dB)	1,1	1,5	0,5	1
C/N for simulations	11,2	10,8	12,3	11,8

Handheld (2b) terminal:

Table A.9.5

Waveform	Sh-A	Sh-A	Sh-B	Sh-B
Modulation	QPSK	16QAM	QPSK	8PSK
Link budget LOS C/N (dB)	6,2	6,2	6,7(see note)	6,7(see note)
Implementation losses (dB)	1,1	1,5	0,5	1
C/N for simulations	5,1	4,7	6,2	5,7
NOTE: Link budget for SH-B and handheld terminal is not provided in clause 11. However C/N may be estimated by similarly to vehicular terminal, where SH-B C/N is 0,5 dB over SH-A C/N.				

The same methodology is used for the calculation of the C/I. In link budgets, two C/I contributions are identified. Thus the following C/I are used in the LMS simulations.

Table A.9.6

Waveform	Sh-A	Sh-A	Sh-B	Sh-B
Modulation	QPSK	16QAM	QPSK	8PSK
Link budget uplink C/I (dB)	19,5	19,5	20	20
Link budget Satellite C/I (dB)	14	14	14	14
Link budget total C/I (dB)	13	13	13	13
Implementation losses (dB)	1,1	1,5	0,5	1
C/I for simulations	11,9	11,5	12,5	12

A.10 Configurations description

This clause provides the detailed reference configurations used in other places in the document (simulation, laboratory test, field test). As a summary, we precise for all the listed configurations, where it is being used. Some configurations are actually not tested in this guidelines, however they are listed for sake of completeness.

Table A.10.0: Usage of listed configuration throughout the document

Section	Title	More def	Tables	SIM	LAB	FIELD
	OFDM class 1 16QAM	variable code rate classical	A10.6	X	X	X
		variable code rate legacy	A10.7			X
	OFDM class 1 QPSK	variable code rate classical	A10.8	X	X	X
A.10.3	OFDM class 2					
	OFDM class 2 uniform late 16QAM	variable duration classical	A10.11, A10.13, A10.15			X
		variable duration optimized	A10.12, A10.14, A10.16			
		fixed duration 10s variable code rate classical	A10.17	X		X
		fixed duration 10s variable code rate optimized	A10.18	X		X
	OFDM class 2 uniform late QPSK	variable duration classical	A10.19			
		variable duration optimized	A10.20	X		
		fixed duration 10s variable code rate classical	A10.21	X		
		fixed duration 10s variable code rate optimized	A10.22	X		X
	OFDM class 2 uniform long					
	OFDM class 2 uniform long 16QAM	variable duration classical	A10.23, A10.24, A10.25			
		fixed duration 10s variable code rate	A10.26		X	X
	OFDM class 2 uniform long QPSK	variable duration classical	A10.27, A10.28, A10.29	X	X	X
		fixed duration 10s variable code rate	A10.30			X
A.10.4	TDM class 1					
	TDM class 1 T-QPSK		A10.32			
	TDM class 1 T-8PSK	classical	A10.33			X
		legacy	A10.34			
	TDM class 1 T-16APSK	classical	A10.35			
		legacy	A10.36			
A.10.5	TDM class 2					
	TDM class 2 uniform long T-QPSK	variable delay	A10.39, A10.40, A10.41	X		
		fixed duration 10s variable code rate	A10.42		X	X
	TDM class 2 uniform long T-8PSK	variable delay	A10.43, A10.44, A10.45		X	
		fixed duration 10s variable code rate	A10.46			X
	TDM class 2 uniform long T-16APSK	fixed duration 10s variable code rate	A10.47		X	X
	TDM class 2 uniform late T-QPSK	variable delay classical	A10.48	X		
		variable delay optimized	A10.49			X
		fixed duration 10s variable code rate classical	A10.50, A10.51			
		fixed duration 10s variable code rate optimized	A10.52, A10.53			
	TDM class 2 uniform late T-8PSK	variable delay classical	A10.54, A10.55, A10.56	X		
		fixed duration 10s variable code rate	A10.57			X
	TDM class 2 uniform late T-16APSK	fixed duration 10s variable code rate	A10.58			X
		TDM 8PSK1/3 uniform long variable delay (16Q	A10.59			X

The way the cases are presented in the table is detailed in clause A.10.1.

A.10.1 Introduction

In the implementation guideline, only short reference should be made so that complete description of the configurations are factorized in this clause. This 2-tables structure is exemplified below. By this way, waveform configurations are mutualised in the clause with all their details and can be reused several times in the body of the document in different circumstances.

For instance, in figure A.10.1, we can see that the body table makes a reference to the physical configuration detailed in the 16QAM short interleaver table in the clause A.10. The corresponding case in the table is the number 2 (code rate 2/9). Since the IFEC code rate selected in the body table is 1, the corresponding bit rate at IP level is 304 kbps, the corresponding TP are 240/208/0/24/8, etc.

Body table (in the body document)

Waveform configuration	Channel	C/I [dB]	C/N [dB]	ID	Additional parameters	Overall Efficiency [bpsHz]	Comment
...	AWGN and Rayleigh						Mobile fading bandwidth for a range from 3 to 600 Hz [The number
O-16QAM2o9_S		20	[-2:14]	2	Table 36 IFEC CR=1	0.89	
....							

Waveform table (in the Annex)

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	-	-	X	X	X	X
Parameter/Casename	16QAM_1/5_S	16QAM_2/9_S	16QAM_1/4_S	16QAM_2/7_S	16QAM_1/3_S	16QAM_2/5_S	16QAM_1/2_S	16QAM_2/3_S
PHYSICAL CONFIGURATION								
Bandwidth (MHz)	5							
FFT Mode	2K+GI ¼	2K+GI ¼	2K+GI ¼	2K+GI ¼	2K+GI ¼	2K+GI ¼	2K+GI ¼	2K+GI ¼
Modulation	16 QAM	16 QAM	16 QAM	16 QAM	16 QAM	16 QAM	16 QAM	16 QAM
PHY FEC rate	1/5	2/9	¼	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0
Min / late / Max interleaver duration (ms)	0 / 105 / 105	0 / 105 / 105	0 / 105 / 105	0 / 105 / 105	0 / 105 / 105	0 / 105 / 105	0 / 105 / 105	0 / 105 / 105
MPEG TS total bit rate	2,666	2,962	3,357	3,752	4,443	5,332	6,714	8,887
TS LAYER CONFIGURATION								
Services	8	8	8	8	8	8	8	8
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122	122	122	122
OFDM Symbols/ coded FEC	10	9	8	7	6	5	4	3
TP per service	216/94/94/21/7 216/125/63/21/7 216/188/0/21/7	240/104/104/24/8 240/138/70/24/8 240/208/0/24/8	272/118/118/27/9 272/157/79/27/9 272/236/0/27/9	304/132/132/30/10 304/176/88/30/10 304/264/0/30/10	360/156/157/35/12 360/208/105/35/12 360/313/0/35/12	432/188/188/42/14 432/250/126/42/14 432/376/0/42/14	544/237/237/53/17 544/316/158/53/17 544/474/0/53/17	720/313/314/70/23 720/418/209/70/23 720/627/0/70/23
MPEG2 per service bit rate (kbps at TS level)	333/145/145/32/11 333/193/97/32/11 333/290//32/11	370/160/160/37/12 370/213/108/37/12 370/321//37/12	420/182/182/42/14 420/242/122/42/14 420/364//42/14	469/204/204/46/15 469/272/136/46/15 469/407//46/15	555/241/242/54/19 555/321/162/54/19 555/483//54/19	666/290/290/65/22 666/386/194/65/22 666/580//65/22	839/366/366/82/26 839/488/244/82/26 839/731//82/26	1111/483/484/108/35 1111/645/322/108/35 1111/967//108/35
LINK LAYER CONFIGURATION								
LL-FEC rate min / recommended / max	0.5 / 0.66 / 1	0.5 / 0.66 / 1	0.5 / 0.66 / 1	0.5 / 0.66 / 1	0.5 / 0.66 / 1	0.5 / 0.66 / 1	0.5 / 0.66 / 1	0.5 / 0.66 / 1
Bitrate/service (kbps at IP level)	140 / 180 / 280	148 / 197 / 304	173 / 230 / 345	189 / 255 / 386	230 / 304 / 460	280 / 370 / 559	345 / 468 / 698	460 / 616 / 928

Figure A.10.1: Organization logic of the table configuration

Let us have a look at the waveform tables presented in the clause. For each case presented in a column, there are a number of parameters given line by line but organized in 3 groups:

- **Group 1: Physical parameters groups:** table A.10.1 basically lists physical parameters as can be given by the SH_delivery_system descriptor. There is no special information to be given here. The resulting interleaver configuration is represented by figure A.10.2 giving individual tap length from the receiver perspective (tap 0 is always fixed to 0).

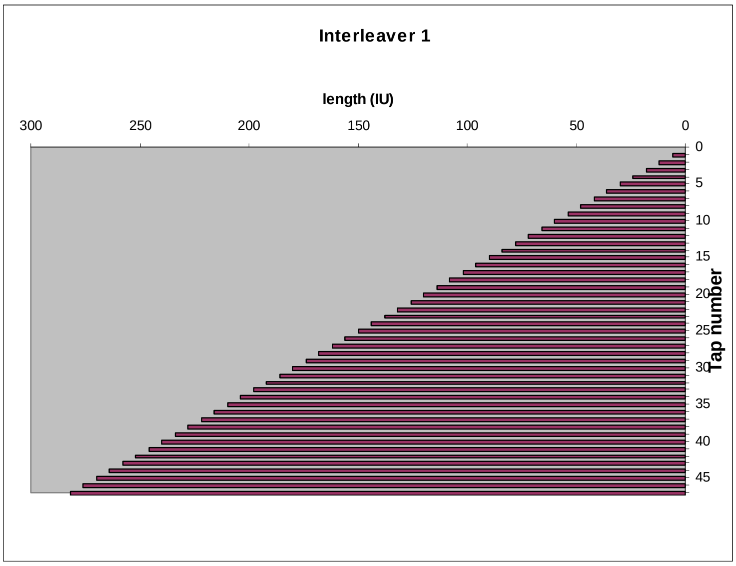


Figure A.10.2: Example of resulting physical interleaver

Table A.10.1: Physical parameters description

Parameter name	Description	Comment
Bandwidth (MHz)	∈{1.7;5;6;7;8}.	-
FFT Mode	∈{1;2;4; 8}*1024.	-
Modulation	∈{O-QPSK;16QAM} for OFDM. ∈{T-QPSK;8PSK; 16APSK} for TDM.	-
PHY FEC rate	∈{1/5;2/9;1/4;2/7;1/3;2/5;1/2;2/3}.	-
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	The 5 parameters of the interleaver configuration.	-
Minimum/late/maximum interleaver duration (ms)	The minimum, late and maximum durations of the physical interleaver with the above configuration: • minimum value is usually 0 as given by the smallest tap (0); • late value is given when there is a late part in the interleaver; • maximum value is proportional to largest tap value.	Minimum value definition could be modified to reflect the duration needed for receiving a number of parity bits ≥ 1 (fast zapping conditions).
MPEG TS total bit rate	Bit rate at the MPEG2 level.	Takes into account SH framing with EFRAMES overhead and padding.

- **Group 2: TS layer parameters group:** table A.10.2 gives the structure of the time-slice services, structure that is to be fixed before any IP service is inserted.

Table A.10.2: TS layer parameters description

Parameter name	Description	Comment
Services	Number of time-slice services.	Should be ≥ 3.
Repetition interval (ms)	Repetition interval between services.	Should be fixed value. It has to be chosen as a multiple of SH frame duration so that simulation cases are simplified.
Burst duration (ms)	Average burst duration: $\text{burst\_duration} = \frac{\text{repetiton\_interval}}{\text{services}}$.	Max burst duration can be estimated as the double of this value.
OFDM/TDM symbols/ coded FEC	Number of symbols necessary to carry the encoded word.	This value is useful check the number of coded word, and therefore the useful bandwidth, required to carry the data.
TP per service	Number of MPEG2 TS packets available in each time-slice burst. There are 5 values given: • total number of TP; • data part; • ifec part; • safety part; • psi part. $TP_{total} = TP_{data} + TP_{IFEC} + TP_{safety} + TP_{PSI}$.	The total number of TS can be used in 4 different ways: • PSI TP: TP used to carry the PSI signalling, based on the PSI ratio; • DATA TP: carry MPE sections (and IP packets); • IFEC TP: carry MPE-IFEC sections; • Safety TP: carry null TS packets as a way to cope for bit rate computation errors. This is derived from the Safety factor that can be fixed to 90 %. Note that section packing is used only in each category (MPE alone, MPE-IFEC alone) but not between MPE and MPE-IFEC. In case several IFEC configurations are possible, we give extreme possible computations.
MPEG2 per service bit rate (kbps at TS level)	Bit rate at the MPEG2 level for the individual service and the different categories of TP: $\text{MPEG2\_bitrate}_{kbps} = \frac{\text{TP}*188*8}{\text{repetition\_interval}_{ms}}$ The following values are given: • total capacity bit rate; • data capacity bit rate; • ifec capacity bit rate; • safety capacity bit rate; • PSI capacity bit rate.	The bit rate is done at MPEG2 level so that we include all overhead like MPE and MPE-IFEC headers and MPEG2 headers. In case several IFEC code rates are possible, we give extreme possible computations.

The resulting structure in terms of time-slice services with their duration is given in figure A.10.3. The repetition interval has to be chosen so that it corresponds to a multiple of SH frame duration for ease of simulation purpose (for lab and field trial cases, this is not a requirement). This representation is only an average configuration since boundaries between services may vary in the case of VBR traffic. As can be seen in figure A.10.3, the maximum number of TP (here 240) results directly from the division of the TS total capacity during a repetition interval (as given by the multiplication of the radio capacity from radio configuration and the repetition interval from the TS configuration) by the number of service. This total number of TP has to be allocated to different budgets:

- PSI ratio: a fraction of the TS capacity has to be allocated to the PSI signalling. This value can be typically set to 1/32 for lab and field environments. In simulation environments, the value is null. In the example we use 1/32 of 240, 8 TP.

- "safety TP": to avoid exceeding capacity of the link due to rounding errors in bit rate computation, a safety factor is usually fixed. A typical value of 90 % for lab and field cases is taken meaning 10 % of the capacity is useless. In the example, 10 % of 240-8=232, 24 TP are considered as "safety TP".

- The remainder of PSI and safety can be used to transport DATA and IFEC: 208 TP. In our case, we have an IFEC code rate of 2/3, meaning we have 138 and 70 TP for, respectively, DATA and IFEC sections.
- In reality, real IP traffic may be lower than this value. Therefore the actual capacity based on "real" IP traffic but not on maximum TP capacity is also shown as values inferior to the maximum.

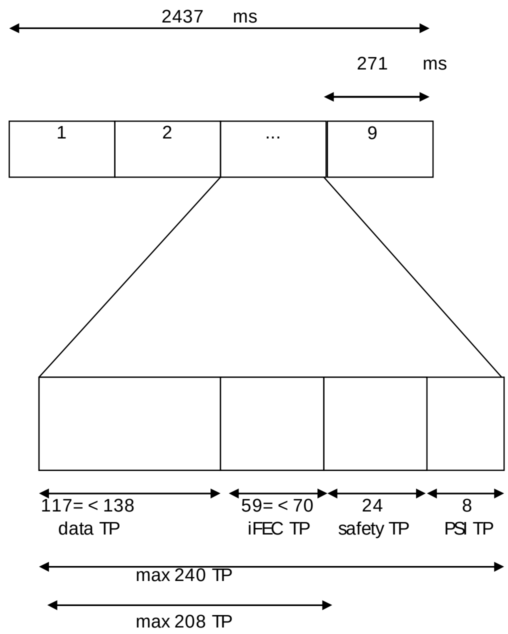


Figure A.10.3: Example of TS layer resulting configuration

- **Group 3: Link layer parameters groups:** table A.10.3 gives only the potential range of MPE-IFEC code rates and the resulting bit rates computed at MPEG2 level for the MPE sections (excluding the MPE-IFEC and excluding the safety TP). Therefore the bit rates given at TS level also exclude IFEC and safety TP and give the actual maximum throughput.

  It is acknowledged that any detailed MPE-IFEC configuration being done at the link layer should be provided in the complete configuration table that will reference those physical layer configurations. Therefore you will not find any precise MPE-IFEC parameters such as B, S, D, number of rows in these tables.

Table A.10.3: Link layer parameters description

Parameter name	Description	Comment
LL-FEC rate min/recommended/max	Range of possible code rate values	Min: best protection; recommended: typical value; Max: w/o any IFEC protection, best throughput
Bitrate/service (kbps at IP level)	Resulting bit rates at IP level	The bit rate is done at IP level, therefore is should be lower than the data bit rate computed at MPEG2 layer by a factor of $\frac{184}{188} * \frac{1000}{1017} \approx 0{,}96$ for IP packet of size 1 000 bytes

As a final reminder, we give below main parameters values for different configurations.

Table A.10.4: Typical configurations parameters values

Configuration	Name	Value	Explanation
Simulation	PSI ratio	Epsilon (~0)	We do not take into account PSI and safety at the simulation level: we can ensure perfect CBR traffic filling perfectly the burst.
	Safety	100 %	
	GI non SFN OFDM sat	1/4	For legacy purposes.
	GI OFDM terrestrial	1/4	For legacy purposes.
Lab and Field trial	PSI ratio	1/32	When using real-life equipment we cannot any more assume perfect data rates filling exactly the capacity. To compare with simulation, we should consider the total capacity bit rate at TS level that should be the same.
	Safety	90 %	
	GI non SFN OFDM sat	1/32	To optimize satellite throughput.
	GI OFDM terrestrial	1/8	To optimize terrestrial throughput.

For simplification purposes, all GI in the configuration are fixed to 1/4. However, according to real-life lab and field experiments, different values may be selected like 1/8 and 1/32. The reader will have to adjust the bitrate values accordingly.

A.10.2 OFDM class 1

In the IG release 1, we used class 1 configurations that exceeded the class 1 memory. Therefore we provide both class 1 compliant configurations that becomes the new reference case mandated for all lab and trial configurations and, for legacy purposes and comparison with IG release 1, the "old class 1 configuration" tables called "class 1 legacy".

Table A.10.5: OFDM class 1 physical layer configuration summary

Modulation	Code rate	Duration	Table
16QAM	Variable (1/5 to 2/3)	Short (official)	Table A.10.6
	Variable (1/5 to 2/3)	Short (legacy)	Table A.10.7
QPSK	Variable (1/5 to 2/3)	short	Table A.10.8

Table A.10.6: OFDM 16QAM class 1 configuration

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	-	-	X	X	X	X
Parameter/Case name	16QAM_1/5_S	16QAM_2/9_S	16QAM_1/4_S	16QAM_2/7_S	16QAM_1/3_S	16QAM_2/5_S	16QAM_1/2_S	16QAM_2/3_S
PHYSICAL CONFIGURATION								
Bandwidth (MHz)	5							
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/105/105	0/105/105	0/105/105	0/105/105	0/105/105	0/105/105	0/105/105	0/105/105
MPEG TS total bit rate	2,666	2,962	3,357	3,752	4,443	5,332	6,714	8,887
TS LAYER CONFIGURATION								
Services	8	8	8	8	8	8	8	8
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122	122	122	122
OFDM Symbols/coded FEC	10	9	8	7	6	5	4	3
total/data/ifec/safety/psi TP per time slice	216/94/94/21/7 216/125/63/21/7 216/188/0/21/7	240/104/104/24/8 240/138/70/24/8 240/208/0/24/8	272/118/118/27/9 272/157/79/27/9 272/236/0/27/9	304/132/132/30/10 304/176/88/30/10 304/264/0/30/10	360/156/157/35/12 360/208/105/35/12 360/313/0/35/12	432/188/188/42/14 432/250/126/42/14 432/376/0/42/14	544/237/237/53/17 544/316/158/53/17 544/474/0/53/17	720/313/314/70/23 720/418/209/70/23 720/627/0/70/23
total/data/ifec/safety/psi kbps@TP per time slice	333/145/145/32/11 333/193/97/32/11 333/290//32/11	370/160/160/37/12 370/213/108/37/12 370/321//37/12	420/182/182/42/14 420/242/122/42/14 420/364//42/14	469/204/204/46/15 469/272/136/46/15 469/407//46/15	555/241/242/54/19 555/321/162/54/19 555/483//54/19	666/290/290/65/22 666/386/194/65/22 666/580//65/22	839/366/366/82/26 839/488/244/82/26 839/731//82/26	1111/483/484/108/35 1111/645/322/108/35 1111/967//108/35
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	140/180/280	148/197/304	173/230/345	189/255/386	230/304/460	280/370/559	345/468/698	460/616/928

Table A.10.7: OFDM 16QAM class 1 legacy configuration

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	-	-	X	X	X	X
Parameter/Case name	16QAM_1/5_S	16QAM_2/9_S	16QAM_1/4_S	16QAM_2/7_S	16QAM_1/3_S	16QAM_2/5_S	16QAM_1/2_S	16QAM_2/3_S
PHYSICAL CONFIGURATION								
Bandwidth (MHz)	5							
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/ nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	10/48/1/0/0	10/48/1/0/0	10/48/1/0/0	10/48/1/0/0	10/48/1/0/0	10/48/1/0/0	10/48/1/0/0	10/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/211/211	0/211/211	0/211/211	0/211/211	0/211/211	0/211/211	0/211/211	0/211/211
MPEG TS total bit rate	2,666	2,962	3,357	3,752	4,443	5,332	6,714	8,887
TS LAYER CONFIGURATION								
Services	8	8	8	8	8	8	8	8
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122	122	122	122
OFDM Symbols/coded FEC	10	9	8	7	6	5	4	3
total/data/ifec/safety/psi TP per time slice	216/94/94/21/7 216/125/63/21/7 216/188/0/21/7	240/104/104/24/8 240/138/70/24/8 240/208/0/24/8	272/118/118/27/9 272/157/79/27/9 272/236/0/27/9	304/132/132/30/10 304/176/88/30/10 304/264/0/30/10	360/156/157/35/12 360/208/105/35/12 360/313/0/35/12	432/188/188/42/14 432/250/126/42/14 432/376/0/42/14	544/237/237/53/17 544/316/158/53/17 544/474/0/53/17	720/313/314/70/23 720/418/209/70/23 720/627/0/70/23
total/data/ifec/safety/psi kbps@TP per time slice	333/145/145/32/11 333/193/97/32/11 333/290//32/11	370/160/160/37/12 370/213/108/37/12 370/321//37/12	420/182/182/42/14 420/242/122/42/14 420/364//42/14	469/204/204/46/15 469/272/136/46/15 469/407//46/15	555/241/242/54/19 555/321/162/54/19 555/483//54/19	666/290/290/65/22 666/386/194/65/22 666/580//65/22	839/366/366/82/26 839/488/244/82/26 839/731//82/26	1111/483/484/108/35 1111/645/322/108/35 1111/967//108/35
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	140/180/280	148/197/304	173/230/345	189/255/386	230/304/460	280/370/559	345/468/698	460/616/928

Table A.10.8: OFDM QPSK class 1 configuration

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	-	-	X	X	-	-
Parameter/Case name	QPSK_1/5_S	QPSK_2/9_S	QPSK_1/4_S	QPSK_2/7_S	QPSK_1/3_S	QPSK_2/5_S	QPSK_1/2_S	QPSK_2/3_S
PHYSICAL CONFIGURATION								
Bandwidth (MHz)	5							
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/211/211	0/211	0/211	0/211	0/211	0/211	0/211	0/211
MPEG TS total bit rate (Mbps)	1,333	1,481	1,679	1,876	2,222	2,666	3,357	4,443
TS LAYER CONFIGURATION								
Services	8	8	8	8	8	8	8	8
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122	122	122	122
OFDM Symbols/coded FEC	20	18	16	14	12	10	8	6
total/data/ifec/safety/psi TP per time slice	108/46/47/11/4 108/62/31/11/4 108/93/0/11/4	120/52/52/12/4 120/69/35/12/4 120/104/0/12/4	136/58/59/14/5 136/78/39/14/5 136/117/0/14/5	152/66/66/15/5 152/88/44/15/5 152/132/0/15/5	180/78/78/18/6 180/104/52/18/6 180/156/0/18/6	216/94/94/21/7 216/125/63/21/7 216/188/0/21/7	272/118/118/27/9 272/157/79/27/9 272/236/0/27/9	360/156/157/35/12 360/208/105/35/12 360/313/0/35/12
total/data/ifec/safety/psi kbps@TP per time slice	167/71/73/17/6 167/96/48/17/6 167/143//17/6	185/80/80/19/6 185/106/54/19/6 185/160//19/6	210/89/91/22/8 210/120/60/22/8 210/181//22/8	235/102/102/23/8 235/136/68/23/8 235/204//23/8	278/120/120/28/9 278/160/80/28/9 278/241//28/9	333/145/145/32/11 333/193/97/32/11 333/290//32/11 333	420/182/182/42/14 420/242/122/42/14 420/364//42/14	555/241/242/54/19 555/321/162/54/19 555/483//54/19
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	66/91/132	74/99/148	83/115/173	91/124/189	115/148/230	140/181/280	173/230/345	230/304/460

A.10.3 OFDM class 2

In order to synchronize the long physical interleaver with the slice structure, some rules should be followed.

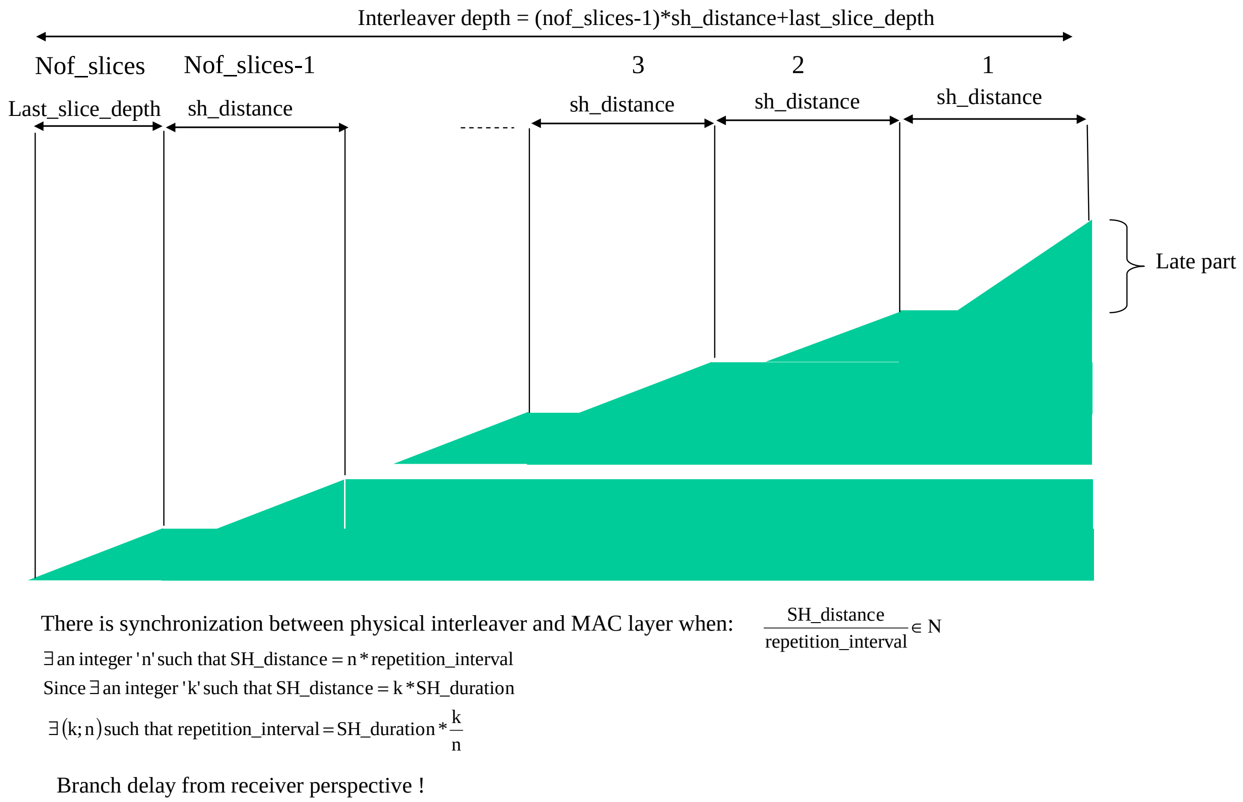


Figure A.10.4: Correspondence between SH physical interleaver and TS configuration

When such synchronization is performed between link and physical layers, this is signaled in one of the informative lines. In fact the only requirement is that SH distance is an exact multiple of repetition interval; therefore repetition interval is also a multiple of SH duration.

The following logic is therefore thought while fixing the interleaver parameters:

1) search a physical interleaver configuration matching expected duration => this leads to a specific SH_distance value;

2) fix the number of time slices so that individual slice can accommodate an individual IP flow;

3) define repetition_interval as an integer fraction of SH_distance best approaching a1 second interval;

4) compute the burst duration as the division of repetition_interval by the number of time slices.

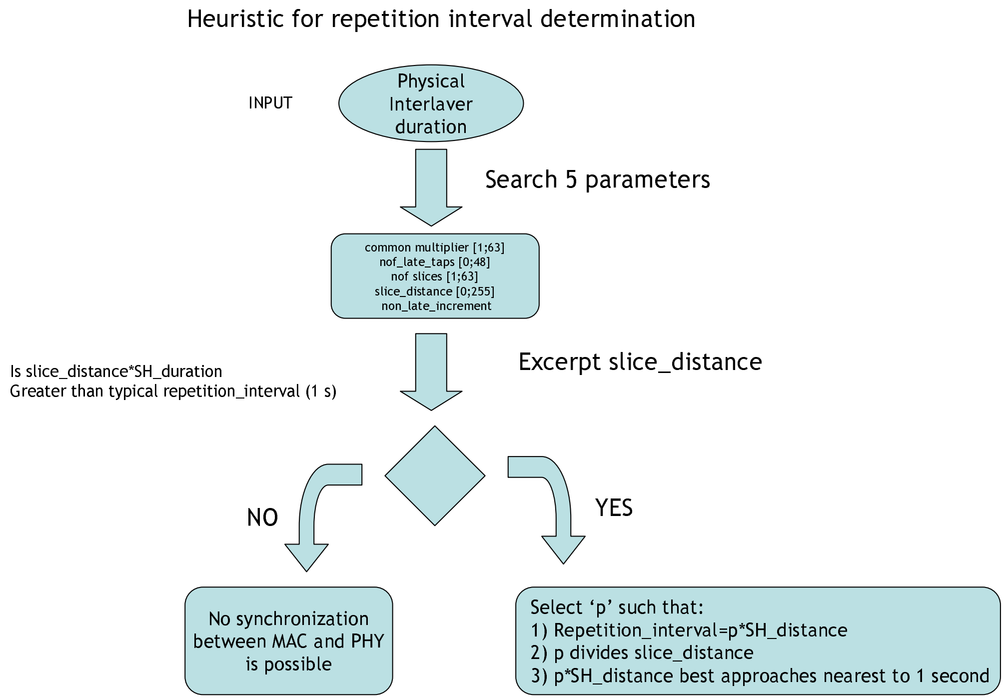


Figure A.10.5: Heuristic for repetition_interval determination

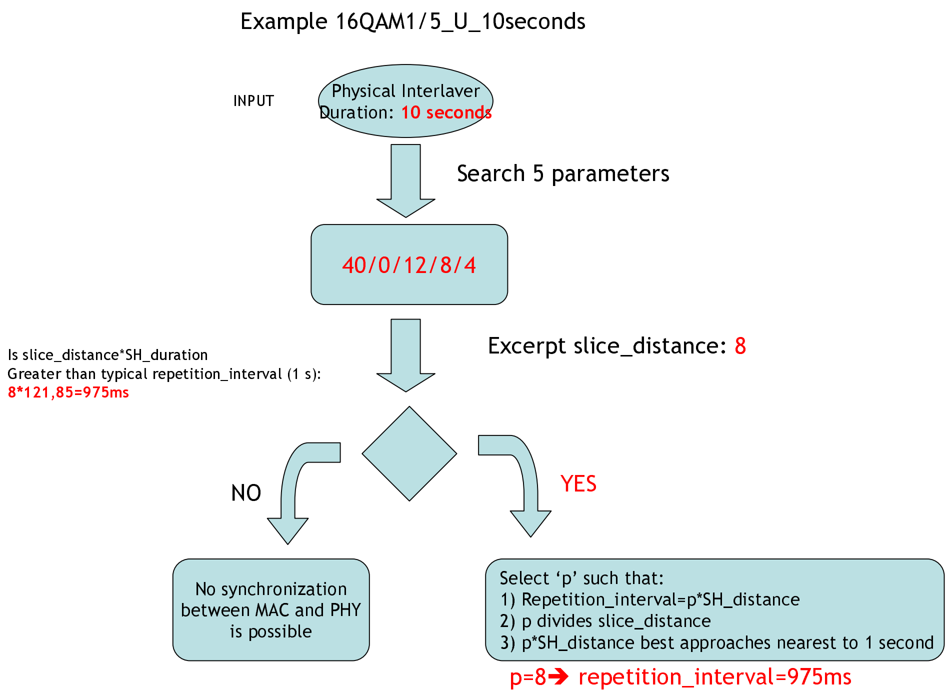


Figure A.10.6: Heuristic example in the case of 16QAM 1/4_U_10s

For uniform-late, two types of modulations are investigated, QPSK and 16QAM. For each modulation we give either {fixed code rate; variable duration} or {variable code rate; fixed duration} configurations. In addition, some investigations on the size of the late part have led to consider different configurations. These are called "optimized UL configuration". They do not replace legacy ones but are new configurations added for the sake of referencing, in particular by some field trials.

Table A.10.9: OFDM class 2 uniform late physical layer configuration summary

Modulation	Code rate	Duration	Table
16QAM	1/5	Variable (0,5 s to 10 s)	Table A.10.11
		Optimized interleaver Variable (5 s to 10 s)	Table A.10.12
	1/4	Variable (0,5 s to 10 s)	Table A.10.13
		Optimized interleaver Variable (5 s to 10 s)	Table A.10.14
	1/3	Variable (0,5 s to 10 s)	Table A.10.15
		Optimized interleaver Variable (5 s to 10 s)	Table A.10.16
	Variable (1/5 to 2/5)	10 s Uniform late	Table A.10.17
		Optimized interleaver 10 seconds Uniform late CR up to 1/3 only	Table A.10.18
QPSK	Variable (1/5 to 2/5)	10 s Uniform late	Table A.10.19
		Optmized 10 s Uniform late CR up to 1/3 only	Table A.10.20
	1/3	Variable (0,5 s to 10 s)	Table A.10.21
		Optimized Variable (5 s to 10 s)	Table A.10.22

For uniform-long, two types of modulations are investigated, QPSK and 16QAM. For each modulation we give either {fixed code rate; variable duration} or {variable code rate; fixed duration} configurations.

Table A.10.10: OFDM class 2 uniform long physical layer configuration summary

Modulation	Code rate	Duration	Table
16QAM	1/5	Variable (0,5 s to 10 s)	Table A.10.23
	1/4	Variable (0,5 s to 10 s)	Table A.10.24
	1/3	Variable (0,5 s to 10 s)	Table A.10.25
	Variable (1/5 to 2/3)	10 s Uniform late	Table A.10.26
QPSK	1/3	Variable (0,5 s to 10 s)	Table A.10.27
	1/2	Variable (0,5 s to 10 s)	Table A.10.28
	2/3	Variable (0,5 s to 10 s)	Table A.10.29
	Variable (1/5 to 2/3)	10 s Uniform late	Table A.10.30

Table A.10.11: OFDM 16QAM1/5 Uniform Late variable duration

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	16QAM_1/5_UL_9,9s	16QAM_1/5_UL_5s	16QAM_1/5_UL_2,5s	16QAM_1/5_UL_1s	16QAM_1/5_UL_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	1/5	1/5	1/5	1/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/24/9/10/12	20/24/9/5/12	18/24/4/5/12	9/24/2/1/10	10/24/2/1/4
Min/late/max interleaver duration (ms)	0/9 964	0/5 089	0/2 505	0/1 049	0/534
MPEG TS total bit rate	2,666	2,666	2,666	2,666	2,666
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	YES	NO	NO	NO	NO
Repetition interval (ms)	1218,6	1218,6	1218,6	1218,6	1218,6
Burst duration (ms)	135	135	135	135	135
OFDM Symbols/coded FEC	10	10	10	10	10
total/data/ifec/safety/psi TP per time slice	240/208/0/24/8	240/208/0/24/8	240/208/0/24/8	240/208/0/24/8	240/208/0/24/8
total/data/ifec/safety/psi kbps@TP per time slice	296/257//30/10	296/257//30/10	296/257//30/10	296/257//30/10	296/257//30/10
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	243	243	243	243	243

Table A.10.12: OFDM 16QAM1/5 optimized Uniform Late variable duration

ID	1	2
new to IG rel 1	-	X
Parameter/Case name	16QAM_1/5_UL_9,9s	16QAM_1/5_UL_5s
PHYSICAL CONFIGURATION		
Bandwidth (MHz)	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM
PHY FEC rate	1/5	1/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/12/10/9/4	10/12/7/7/2
Min/late/max interleaver duration (ms)	0/49/9 924	0/49/5 163
MPEG TS total bit rate	2,666	2,666
TS LAYER CONFIGURATION		
Services	9	9
Synchronization LINK/PHY	YES	NO
Repetition interval (ms)	1097	1219
Burst duration (ms)	122	135
OFDM Symbols/coded FEC	10	10
total/data/ifec/safety/psi TP per time slice	216/125/63/21/7	240/138/70/24/8
total/data/ifec/safety/psi kbps@TP per time slice	296/258//29/10	296/257//30/10
LINK LAYER CONFIGURATION		
LL-FEC rate min/recommended / max	1/1/1	1/1/1
Max bitrate/service (kbps at IP level)	243	243

Table A.10.13: OFDM 16QAM1/4 Uniform Late variable duration

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	16QAM_1/4_UL_9,9s	16QAM_1/4_UL_5s	16QAM_1/4_UL_2,5s	16QAM_1/4_UL_1s	16QAM_1/4_UL_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/4	1/4	1/4	1/4	1/4
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/24/9/10/12	20/24/9/5/12	18/24/4/5/12	9/24/2/1/10	10/24/2/1/4
Min/Late/Max interleaver duration (ms)	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964
MPEG TS total bit rate	3,357	3,357	3,357	3,357	3,357
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	YES	NO	NO	NO	NO
Repetition interval (ms)	1218,6	1218,6	1218,6	1218,6	1218,6
Burst duration (ms)	135	135	135	135	135
OFDM Symbols/coded FEC	8	8	8	8	8
total/data/ifec/safety/psi TP per time slice	302/262/0/30/10	302/262/0/30/10	302/262/0/30/10	302/262/0/30/10	302/262/0/30/10
total/data/ifec/safety/psi kbps@TP per time sliceTP	373/323//37/12	373/323//37/12	373/323//37/12	373/323//37/12	373/323//37/12
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	309	309	309	309	309

Table A.10.14: OFDM 16QAM1/4 optimized Uniform Late variable duration

ID	1	2
new to IG rel 1	X	X
Parameter/Case name	16QAM_1/4_UL_9,9s	16QAM_1/4_UL_5s
PHYSICAL CONFIGURATION		
Bandwidth (MHz)	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM
PHY FEC rate	1/4	1/4
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/16/9/10/6	10/16/9/5/10
Min/Late/Max interleaver duration (ms)	0/67/9 829	0/67/4 955
MPEG TS total bit rate	3,357	3,357
TS LAYER CONFIGURATION		
Services	9	9
Synchronization LINK/PHY	YES	NO
Repetition interval (ms)	1 218,6	1 218,6
Burst duration (ms)	135	135
OFDM Symbols/coded FEC	8	8
total/data/ifec/safety/psi TP per time slice	302/262/0/30/10	302/262/0/30/10
total/data/ifec/safety/psi kbps@TP per time slice	373/323//37/12	373/323//37/12
LINK LAYER CONFIGURATION		
LL-FEC rate min/recommended/max	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	309	309

Table A.10.15: OFDM 16QAM1/3 Uniform Late variable duration

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	16QAM_1/3_UL_9,9s	16QAM_1/3_UL_5s	16QAM_1/3_UL_2,5s	16QAM_1/3_UL_1s	16QAM_1/3_UL_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/24/9/10/12	20/24/9/5/12	18/24/4/5/12	9/24/2/1/10	10/24/2/1/4
Min/Late/Max interleaver duration (ms)	0/206/9 964	0/5 089	0/2 505	0/1 049	0/534
MPEG TS total bit rate	4,443	4,443	4,443	4,443	4,443
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	YES	NO	NO	NO	NO
Repetition interval (ms)	1 218,6	1 218,6	1 218,6	1 218,6	1 218,6
Burst duration (ms)	135	135	135	135	135
OFDM Symbols/coded FEC	6	6	6	6	6
total/data/ifec/safety/psi TP per time slice	400/348/0/39/13	400/348/0/39/13	400/348/0/39/13	400/348/0/39/13	400/348/0/39/13
total/data/ifec/safety/psi kbps@TP per time slice	494/430//48/16	494/430//48/16	494/430//48/16	494/430//48/16	494/430//48/16
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	408	408	408	408	408

Table A.10.16: OFDM 16QAM1/3 optimized Uniform Late variable duration

ID	1	2
new to IG rel 1	X	X
Parameter/Case name	16QAM_1/3_UL_9,9s	16QAM_1/3_UL_5s
PHYSICAL CONFIGURATION		
Bandwidth (MHz)	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM
PHY FEC rate	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/18/11/8/9	10/18/6/8/4
Min/Late/Max interleaver duration (ms)	0/76/9 829	0/76/4 964
MPEG TS total bit rate	4,443	4,443
TS LAYER CONFIGURATION		
Services	9	9
Synchronization LINK/PHY	NO	NO
Repetition interval (ms)	1 218,6	1 218,6
Burst duration (ms)	135	135
OFDM Symbols/coded FEC	6	6
total/data/ifec/safety/psi TP per time slice	400/348/0/39/13	400/348/0/39/13
total/data/ifec/safety/psi kbps@TP per time slice	494/430//48/16	494/430//48/16
LINK LAYER CONFIGURATION		
LL-FEC rate min/recommended/max	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	408	408

Table A.10.17: OFDM 16QAM variable code rate Uniform Late 10 s

ID	1	2	3	4	5	6
new to IG rel 1	-	X	X	X	X	X
Parameter/Case name	16QAM_1/5_UL	16QAM_2/9_UL	16QAM_1/4_UL	16QAM_2/7_UL	16QAM_1/3_UL	16QAM_2/5_UL
PHYSICAL CONFIGURATION						
Bandwidth (MHz)	5					
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/24/9/10/12	20/24/9/10/12	20/24/9/10/12	20/24/9/10/12	20/24/9/10/12	20/24/9/10/12
Min/Late/Max interleaver duration (ms)	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964	0/9 964
MPEG TS total bit rate	2,666	2,962	3,357	3,752	4,443	5,332
TS LAYER CONFIGURATION						
Services	9	9	9	9	9	9
Synchronization PHY/LINK	YES	YES	YES	YES	YES	YES
Repetition interval (ms)	1218,6	1219	1219	1219	1219	1219
Burst duration (ms)	135	135	135	135	135	135
OFDM Symbols/coded FEC	10	9	8	7	6	5
total/data/ifec/safety/psi TP per time slice	240/208/0/24/8	266/231/0/26/9	302/262/0/30/10	337/293/0/33/11	400/348/0/39/13	480
total/data/ifec/safety/psi kbps@TP per time slice	296/257//30/10	328/285//32/11	373/323//37/12	416/362//41/14	494/430//48/16	592
LINK LAYER CONFIGURATION						
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	243	270	309	348	408	480
NOTE: Code rates 1/2 and 2/3 are not included since the UL configuration is not useful because late part code rate would be ≥ 1.						

Table A.10.18: OFDM 16QAM variable code rate optimized Uniform Late 10 s

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	16QAM_1/5_UL	16QAM_2/9_UL	16QAM_1/4_UL	16QAM_2/7_UL	16QAM_1/3_UL
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5				
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	2/9	1/4	2/7	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	10/12/10/9/4	10/15/12/7/8	10/16/9/10/6	10/16/9/10/6	10/18/11/8/9
Min/Late/Max interleaver duration (ms)	0/49/9 924	0/63/9 455	0/67/9 829	0/67/9 829	0/76/9 829
MPEG TS total bit rate	2,666	2,962	3,357	3,752	4,443
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization PHY / LINK	NO	NO	YES	YES	NO
Repetition interval (ms)	1 218,6	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135
OFDM Symbols/coded FEC	10	9	8	7	6
total/data/ifec/safety/psi TP per time slice	240/208/0/24/8	266/231/0/26/9	302/262/0/30/10	337/293/0/33/11	400/348/0/39/13
total/data/ifec/safety/psi kbps@TP per time slice	296/257//30/10	328/285//32/11	373/323//37/12	416/362//41/14	494/430//48/16
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	243	270	309	348	408

NOTE: Code rates 2/5, 1/2 and 2/3 are not included since the UL configuration is not useful because late part code rate would be ≥ 1.

Table A.10.19: OFDM QPSK variable code rate Uniform Late 10 s

ID	1	2	3	4	5	6
new to IG rel 1	X	X	X	X	-	X
Parameter/Case name	QPSK_1/5_UL	QPSK_2/9_UL	QPSK_1/4_UL	QPSK_2/7_UL	QPSK_1/3_UL	QPSK_2/5_UL
PHYSICAL CONFIGURATION						
Bandwidth (MHz)	5					
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12
Min/Late/Max interleaver duration (ms)	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964	0/206/9 964
MPEG TS total bit rate (Mbps)	1,333	1,481	1,679	1,876	2,222	2,666
TS LAYER CONFIGURATION						
Services	8	8	8	8	8	8
Synchronization PHY / LINK	NO	NO	NO	NO	NO	NO
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	152	152	152	152	152	152
OFDM Symbols/coded FEC	20	18	16	14	12	10
total/data/ifec/safety/psi TP per time slice	135/117/0/13/5	150/130/0/15/5	170/147/0/17/6	190/165/0/19/6	225/195/0/22/8	270/234/0/27/9
total/data/ifec/safety/psi kbps@TP per time slice	167/144//16/6	185/160//19/6	210/181//21/7	235/204//23/7	278/241//27/10	333/289//33/11
LINK LAYER CONFIGURATION						
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	138	185	171	191	230	276
NOTE: Code rates 1/2 and 2/3 are not included since the UL configuration is not useful because late part code rate would be ≥ 1.						

Table A.10.20: OFDM QPSK variable code rate optimized Uniform Late 10 s

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	-
Parameter/Case name	QPSK_1/5_UL	QPSK_2/9_UL	QPSK_1/4_UL	QPSK_2/7_UL	QPSK_1/3_UL
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5				
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/5	2/9	1/4	2/7	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	5/12/10/5/4	5/15/12/4/7	5/16/9/5/6	5/16/9/5/6	5/18/11/4/6
Min/Late/Max interleaver duration (ms)	0/49/11 021	0/63/10 786	0/67/9 829	0/67/9 829	0 / 76 / 9802
MPEG TS total bit rate (Mbps)	1,333	1,481	1,679	1,876	2,222
TS LAYER CONFIGURATION					
Services	8	8	8	8	8
Synchronization PHY / LINK	NO	NO	NO	NO	NO
Repetition interval (ms)	1219	1219	1219	1219	1219
Burst duration (ms)	152	152	152	152	152
OFDM Symbols/coded FEC	20	18	16	14	12
total/data/ifec/safety/psi TP per time slice	135/117/0/13/5	150/130/0/15/5	170/147/0/17/6	190/165/0/19/6	225/195/0/22/8
total/data/ifec/safety/psi kbps@TP per time slice	167/144//16/6	185/160//19/6	210/181//21/7	235/204//23/7	278/241//27/10
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	138	185	171	191	230

Table A.10.21: OFDM QPSK1/3 Uniform Late variable duration

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	QPSK_1/3_UL_9,9s	QPSK_1/3_UL_5,1s	QPSK_1/3_UL_2,1s	QPSK_1/3_UL_1s	QPSK_1/3_UL_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/24/9/5/12	10/24/3/8/12	10/24/9/1/10	23/48/1/0/0	12/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/206/9 964	0/206/5 082	0/206/2 129	0/969/969	0/505/505
MPEG TS total bit rate (Mbps)	2,222	2,222	2,222	2,222	2,222
TS LAYER CONFIGURATION					
Services	8	8	8	8	8
Synchronization PHY/LINK	NO	NO	NO	NO	NO
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	152	152	152	152	152
OFDM Symbols/coded FEC	12	12	12	12	12
total/data/ifec/safety/psi TP per time slice	225/195/0/22/8	225/195/0/22/8	225/195/0/22/8	225/195/0/22/8	225/195/0/22/8
total/data/ifec/safety/psi kbps@TP per time slice	278/241//27/10	278/241//27/10	278/241//27/10	278/241//27/10	278/241//27/10
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	230	230	230	230	230

Table A.10.22: OFDM QPSK1/3 optimized Uniform Late variable duration

ID	1	2
new to IG rel 1	-	X
Parameter/Case name	QPSK_1/3_UL_9,9s	QPSK_1/3_UL_5,1s
PHYSICAL CONFIGURATION		
Bandwidth (MHz)	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK
PHY FEC rate	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	5/18/11/4/6	5/18/6/4/4
Min/Late/Max interleaver duration (ms)	0/76/9 802	0/76/4 964
MPEG TS total bit rate (Mbps)	2,222	2,222
TS LAYER CONFIGURATION		
Services	8	8
Synchronization PHY/LINK	NO	NO
Repetition interval (ms)	1 219	1 219
Burst duration (ms)	152	152
OFDM Symbols/coded FEC	12	12
total/data/ifec/safety/psi TP per time slice	225/195/0/22/8	225/195/0/22/8
total/data/ifec/safety/psi kbps@TP per time slice	278/241//27/10	278/241//27/10
LINK LAYER CONFIGURATION		
LL-FEC rate min/recommended/max	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	230	230

Table A.10.23: OFDM 16QAM1/5 Uniform Long variable duration

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	16QAM_1/5_U_11s	16QAM_1/5_U_5s	16QAM_1/5_U_2,5s	16QAM_1/5_U_1s	16QAM_1/5_U_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	1/5	1/5	1/5	1/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/8/4	40/0/6/8/4	10/0/4/6/7	48/48/1/0/0	24/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/5 376	0/0/2 538	0/0/1 011	0/505/505
MPEG TS total bit rate	2,666	2,666	2,666	2,666	2,666
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization PHY/LINK	YES	YES	NO	YES	YES
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
OFDM Symbols/coded FEC	10	10	10	10	10
total/data/ifec/safety/psi TP per time slice	192/167/0/19/6	192/167/0/19/6	192/167/0/19/6	192/167/0/19/6	192/167/0/19/6
total/data/ifec/safety/psi kbps@TP per time slice	296/258//29/9	296/258//29/9	296/258//29/9	296/258//29/9	296/258//29/9
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	247	247	247	247	247

Table A.10.24: OFDM 16QAM1/4 Uniform Long variable duration

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	16QAM_1/4_U_11s	16QAM_1/4_U_5s	16QAM_1/4_U_2,5s	16QAM_1/4_U_1s	16QAM_1/4_U_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/4	1/4	1/4	1/4	1/4
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/8/4	40/0/6/8/4	10/0/4/6/7	48/48/1/0/0	24/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/5 376	0/0/2 538	0/1 011/1 011	0/505/505
MPEG TS total bit rate	3,357	3,357	3,357	3,357	3,357
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	YES	YES	NO	YES	YES
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
OFDM Symbols/coded FEC	8	8	8	8	8
total/data/ifec/safety/psi TP per time slice	241/209/0/24/8	241/209/0/24/8	241/209/0/24/8	241/209/0/24/8	241/209/0/24/8
total/data/ifec/safety/psi kbps@TP per time slice	372/322//37/12	372/322//37/12	372/322//37/12	372/322//37/12	372/322//37/12
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended / max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	304	304	304	304	304

Table A.10.25: OFDM 16QAM1/3 Uniform Long variable duration

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	16QAM_1/3_U_9,9s	16QAM_1/3_U_5s	16QAM_1/3_U_2,5s	16QAM_1/3_U_1s	16QAM_1/3_U_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/8/4	40/0/6/8/4	10/0/4/6/7	48/48/1/0/0	24/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/5 376	0/0/2 538	0/1 011/1 011	0/505/505
MPEG TS total bit rate	4,443	4,443	4,443	4,443	4,443
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	YES	YES	NO	YES	YES
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
OFDM Symbols/coded FEC	6	6	6	6	6
total/data/ifec/safety/psi TP per time slice	320/186/93/31/10	320/186/93/31/10	320/186/93/31/10	320/186/93/31/10	320/186/93/31/10
total/data/ifec/safety/psi kbps@TP per time slice	494/287/143/48/15	494/287/143/48/15	494/287/143/48/15	494/287/143/48/15	494/287/143/48/15
LINK LAYER CONFIGURATION					
LL-FEC rate min / recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	411	411	411	411	411

Table A.10.26: OFDM 16QAM variable code rate Uniform Long 10 s

ID	1	2	3	4	5	6	7	8
new to IG rel 1	-	X	X	X	X	X	X	X
Parameter/Case name	16QAM_1/5_U	16QAM _2/9_U	16QAM_1/4_U	16QAM_2/7_U	16QAM_1/3_U	16QAM_2/5_U	16QAM_1/2_U	16QAM_2/3_U
PHYSICAL CONFIGURATION								
Bandwidth (MHz)	5							
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM	16QAM
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/8/4	40/0/12/8/4	40/0/12/8/4	40/0/12/8/4	40/0/12/8/4	40/0/12/8/4	40/0/12/8/4	40/0/12/8/4
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938
MPEG TS total bit rate (Mbps)	2,666	2,962	3,357	3,752	4,443	5,332	6,714	8,887
TS LAYER CONFIGURATION								
Services	9	9	9	9	9	9	9	9
Synchronization PHY/LINK	YES	YES	YES	YES	YES	YES	YES	YES
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108	108	108	108
OFDM Symbols/coded FEC	10	9	8	7	6	5	4	3
total/data/ifec/safety/psi TP per time slice	192/167/0/19/6	213/185/0/21/7	241/209/0/24/8	270/234/0/27/9	320/279/0/31/10	384/334/0/38/12	483/420/0/47/16	640/558/0/62/20
total/data/ifec/safety/psi kbps@TP per time slice	296/258//29/9	329/285//32/11	372/322//37/12	417/361//42/14	494/430//48/15	592/515//59/19	745/648//73/25	987/861//96/31
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	247	271	304	345	411	493	616	821

Table A.10.27: OFDM QPSK1/3 Uniform Long variable duration

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	QPSK_1/3_U_11s	QPSK_1/3_U_5,5s	QPSK_1/3_U_2,3s	QPSK_1/3_U_1s	QPSK_1/3_U_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/4/2	40/0/12/2/2	10/0/4/3/2	24/48/1/0/0	12/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/5 577	0/0/2 391	0/1 011/1 011	0/505/505
MPEG TS total bit rate (Mbps)	2,222	2,222	2,222	2,222	2,222
TS LAYER CONFIGURATION					
Services	8	8	8	8	8
Synchronization LINK PHY	YES	NO	NO	YES	YES
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122
OFDM Symbols/coded FEC	12	12	12	12	12
total/data/ifec/safety/psi TP per time slice	180/156/0/18/6	180/156/0/18/6	180/156/0/18/6	180/156/0/18/6	180/156/0/18/6
total/data/ifec/safety/psi kbps@TP per time slice	278/241//28/9	278/241//28/9	278/241//28/9	278/241//28/9	278/241//28/9
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	230	230	230	230	230

Table A.10.28: OFDM QPSK1/2 Uniform Long variable duration

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	QPSK_1/2_U_9,9s	QPSK_1/2_U_5,1s	QPSK_1/2_U_2,1s	QPSK_1/2_U_1s	QPSK_1/2_U_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/2	1/2	1/2	1/2	1/2
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/4/2	40/0/12/2/2	10/0/4/3/2	24/48/1/0/0	12/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/5 577	0/0/2 391	0/1 011/1 011	0/505/505
MPEG TS total bit rate (Mbps)	3,357	3,357	3,357	3,357	3,357
TS LAYER CONFIGURATION					
Services	8	8	8	8	8
Synchronization PHY/LINK	YES	NO	NO	YES	YES
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122
OFDM Symbols/coded FEC	8	8	8	8	8
total/data/ifec/safety/psi TP per time slice	272/236/0/27/9	272/236/0/27/9	272/236/0/27/9	272/236/0/27/9	272/236/0/27/9
total/data/ifec/safety/psi kbps@TP per time slice	420/364//42/14	420/364//42/14	420/364//42/14	420/364//42/14	420/364//42/14
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	345	345	345	345	345

Table A.10.29: OFDM QPSK2/3 Uniform Long variable duration

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	QPSK_2/3_U_9,9s	QPSK_2/3_U_5,1s	QPSK_2/3_U_2,1s	QPSK_2/3_U_1s	QPSK_2/3_U_0,5s
PHYSICAL CONFIGURATION					
Bandwidth (MHz)	5	5	5	5	5
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	2/3	2/3	2/3	2/3	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/4/2	40/0/12/2/2	10/0/4/3/2	24/48/1/0/0	12/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/5 577	0/0/2 391	0/1 011/1 011	0/505/505
MPEG TS total bit rate (Mbps)	4,443	4,443	4,443	4,443	4,443
TS LAYER CONFIGURATION					
Services	8	8	8	8	8
Synchronization PHY/LINK	YES	NO	NO	YES	YES
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122
OFDM Symbols/coded FEC	6	6	6	6	6
total/data/ifec/safety/psi TP per time slice	360/313/0/35/12	360/313/0/35/12	360/313/0/35/12	360/313/0/35/12	360/313/0/35/12
total/data/ifec/safety/psi kbps@TP per time slice	555/483//54/19	555/483//54/19	555/483//54/19	555/483//54/19	555/483//54/19
MPEG2 per service bit rate (kbps at TS level)	555	555	555	555	555
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	460	460	460	460	460

Table A.10.30: OFDM QPSK variable code rate Uniform Long 10 s

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	-	X	X	X
Parameter/Case name	QPSK_1/5_U	QPSK_2/9_U	QPSK_1/4_U	QPSK_2/7_U	QPSK_1/3_U	QPSK_2/5_U	QPSK_1/2_U	QPSK_2/3_U
PHYSICAL CONFIGURATION								
Bandwidth (MHz)	5							
FFT Mode	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4	2K+GI 1/4
Modulation	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK	QPSK
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2
Min/Late/Max interleaver duration (ms)	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938	0/0/10 938
MPEG TS total bit rate (Mbps)	1,333	1,481	1,679	1,876	2,222	2,666	3,357	4,443
TS LAYER CONFIGURATION								
Services	8	8	8	8	8	8	8	8
Synchronization PHY/LINK	YES	YES	YES	YES	YES	YES	YES	YES
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122	122	122	122
OFDM Symbols/coded FEC	20	18	16	14	12	10	8	6
total/data/ifec/safety/psi kbps@TP per time slice	108/93/0/11/4	120/104/0/12/4	136/117/0/14/5	152/132/0/15/5	180/156/0/18/6	216/188/0/21/7	272/236/0/27/9	360/313/0/35/12
MPEG2 per service bit rate (kbps at TS level)	167/143//17/6	185/160//19/6	210/181//22/8	235/204//23/8	278/241//28/9	333/290//32/11	420/364//42/14	555/483//54/19
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	132	148	173	189	230	280	345	460

A.10.4 TDM class 1

We have the same issue as for OFDM regarding the memory available for the class 1 with two types of interleaver, the official one respecting the class 1 memory and the legacy one corresponding to the implementation guidelines version 1. Please refer to OFDM class 1 explanation.

Table A.10.31: TDM class 1 physical layer configuration summary

Modulation	Code rate	Duration	Table
T-QPSK	Variable (1/5 to 2/3)	Short (official)	Table A.10.32
8PSK	Variable (1/5 to 2/3)	Short (official)	Table A.10.33
		Short (legacy)	Table A.10.34
16APSK	Variable (1/5 to 2/3)	Short (official)	Table A.10.35
		Short (legacy)	Table A.10.36

Table A.10.32: T-QPSK short interleaver

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	X	X	-	X
Parameter/Case name	T_QPSK_1/5_S	T_QPSK_2/9_S	T_QPSK_1/4_S	T_QPSK_2/7_S	T_QPSK_1/3_S	T_QPSK_2/5_S	T_QPSK_1/2_S	T_QPSK_2/3_S
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/185/185	0/182/182	0/184/184	0/182/182	0/184/184	0/182/182	0/181/181	0/185/185
MPEG TS total bit rate (Mbps)	1,530	1,728	1,925	2,222	2,567	3,110	3,900	5,184
TS LAYER CONFIGURATION								
Services	8	8	8	8	8	8	8	8
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	122	122	122	122	122	122	122	122
TDM Symbols/coded FEC	30 240	27 216	24 192	21 168	18 144	15 120	12 096	9 072
total/data/ifec/safety/psi kbps@TP per time slice	124/54/54/12/4 124/72/36/12/4 124/108/0/12/4	140/60/61/14/5 140/80/41/14/5 140/121/0/14/5	156/67/68/16/5 156/90/45/16/5 156/135/0/16/5	180/78/78/18/6 180/104/52/18/61 80/156/0/18/6	208/90/90/21/7 208/120/60/21/7 208/180/0/21/7	252/109/110/25/8 252/146/73/25/8 252/219/0/25/8	316/137/138/31/1 0 316/183/92/31/10 316/275/0/31/10	420/182/183/41/14 420/243/122/41/14 420/365/0/41/14
MPEG2 per service bit rate (kbps at TS level)	191/83/83/19/6 191/111/56/19/6 191/167//19/6	216/93/94/22/8 216/123/63/22/8 216/187//22/8	241/103/105/25/8 241/139/69/25/8 241/208//25/8	278/120/120/28/9 278/160/80/28/9 278/241//28/9	321/139/139/32/11 321/185/93/32/11 321/278//32/11	389/168/170/39/12 389/225/113/39/12 389/338//39/12	488/211/213/48/1 5 488/282/142/48/1 5 488/424//48/15	648/281/282/63/22 648/375/188/63/22 648/563//63/22
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	05/0,66/1
Bitrate/service (kbps at IP level)	74/107/156	83/115/173	99/132/197	115/148/230	132/173/263	156/214/321	197/271/403	263/353/542

Table A.10.33: T-8PSK short interleaver

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	X	X	X-	X
Parameter/Case name	T_8PSK_1/5_S	T_8PSK_2/9_S	T_8PSK_1/4_S	T_8PSK_2/7_S	T_8PSK_1/3_S	T_8PSK_2/5_S	T_8PSK_1/2_S	T_8PSK_2/3_S
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/122/122	0/122/122	0/121/121	0/122/122	0/121/121	0/121/121	0/121/121	0/121/121
MPEG TS total bit rate (Mbps)	2,320	2,567	2,913	3,308	3,900	4,690	5,826	7,800
TS LAYER CONFIGURATION								
Services	9	9	9	9	9	9	9	9
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108	108	108	108
TDM Symbols/coded FEC	20 160	18 144	16 128	14 112	12 096	10 080	8 064	6 048
total/data/ifec/safety/psi kbps@TP per time slice	167/72/72/17/6 167/96/48/17/6 167/144/0/17/6	184/80/80/18/6 184/106/54/18/6 184/160/0/18/6	209/90/91/21/7 209/120/61/21/7 209/181/0/21/7	238/103/104/23/8 238/138/69/23/8 238/207/0/23/8	280/121/122/28/9 280/162/81/28/9 280/243/0/28/9	337/146/147/33/11 337/195/98/33/11 337/293/0/33/11	419/182/182/41/14 419/242/122/41/14 419/364/0/41/14	561/244/244/55/18 561/325/163/55/18 561/488/0/55/18
MPEG2 per service bit rate (kbps at TS level)	258/111/111/26/9 258/148/74/26/9 258/222//26/9	284/123/123/28/9 284/164/83/28/9 284/247//28/9	322/139/140/32/11 322/185/94/32/11 322/279//32/11	367/159/160/35/12 367/213/106/35/12 367/319//35/12	432/187/188/43/14 432/250/125/43/14 432/375//43/14	520/225/227/51/17 520/301/151/51/17 520/452//51/17	646/281/281/63/22 646/373/188/63/22 646/562//63/22	866/376/376/85/28 866/501/251/85/28 866/753//85/28
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	107/140/214	0,5/0,66/1	132/173/263	148/197/304	173/238/353	214/288/435	263/353/534	362/476/723

Table A.10.34: T-8PSK short interleaver legacy

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	- (see note)	X	X-	X
Parameter/Case name	T_8PSK_1/5_S	T_8PSK_2/9_S	T_8PSK_1/4_S	T_8PSK_2/7_S	T_8PSK_1/3_S	T_8PSK_2/5_S	T_QPSK_1/2_S	T_QPSK_2/3_S
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15	5 MHz, QPSK; 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	9/48/1/0/0	9/48/1/0/0	9/48/1/0/0	9/48/1/0/0	9/48/1/0/0	9/48/1/0/0	9/48/1/0/0	9/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/219/219	0/220/220	0/218/218	0/220/200	0/217/217	0/217/217	0/218/218	0/217/217
MPEG TS total bit rate (Mbps)	2,320	2,567	2,913	3,308	3,900	4,690	5,826	7,800
TS LAYER CONFIGURATION								
Services	9	9	9	9	9	9	9	9
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108	108	108	108
TDM Symbols/coded FEC	20 160	18 144	16 128	14 112	12 096	10 080	8 064	6 048
total/data/ifec/safety/psi kbps@TP per time slice	167/72/72/17/6 167/96/48/17/6 167/144/0/17/6	184/80/80/18/6 184/106/54/18/6 184/160/0/18/6	209/90/91/21/7 209/120/61/21/7 209/181/0/21/7	238/103/104/23/8 238/138/69/23/8 238/207/0/23/8	280/121/122/28/9 280/162/81/28/9 280/243/0/28/9	337/146/147/33/11 337/195/98/33/11 337/293/0/33/11	419/182/182/41/14 419/242/122/41/14 419/364/0/41/14	561/244/244/55/18 561/325/163/55/18 561/488/0/55/18
MPEG2 per service bit rate (kbps at TS level)	258/111/111/26/9 258/148/74/26/9 258/222//26/9	284/123/123/28/9 284/164/83/28/9 284/247//28/9	322/139/140/32/11 322/185/94/32/11 322/279//32/11	367/159/160/35/12 367/213/106/35/12 367/319//35/12	432/187/188/43/14 432/250/125/43/14 432/375//43/14	520/225/227/51/17 520/301/151/51/17 520/452//51/17	646/281/281/63/22 646/373/188/63/22 646/562//63/22	866/376/376/85/28 866/501/251/85/28 866/753//85/28
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	107/140/214	0,5/0,66/1	132/173/263	148/197/304	173/238/353	214/288/435	263/353/534	362/476/723

NOTE: ID 5 configuration differs slightly from the IG rel 1 one which was (8:48;1;0;0); however its status is set to be already covered by IG rel 1.

Table A.10.35: T-16PSK short interleaver legacy

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	X	X	X-	X
Parameter/Case name	16APSK_1/5_S	16APSK_2/9_S	16APSK_1/4_S	16APSK_2/7_S	16APSK_1/3_S	16APSK_2/5_S	16APSK_1/2_S	16APSK_2/5_S
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0	5/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/91/91	0/91/91	0/91/91	0/91	0/91	0/91/91	0/91/91	0/90
MPEG TS total bit rate (Mbps)	3,110	3,456	3,900	4,443	5,184	6,221	7,800	10,417
TS LAYER CONFIGURATION								
Services	12	12	12	12	12	12	12	12
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	81	81	81	81	81	81	81	81
TDM Symbols/coded FEC	15 120	13 608	12 096	10 584	9 072	7 560	6 048	4 536
total/data/ifec/safety/psi kbps@TP per time slice	168/72/73/17/6 168/96/49/17/6 168/145/0/17/6	186/81/81/18/6 186/108/54/18/6 186/162/0/18/6	210/91/91/21/7 210/121/61/21/7 210/182/0/21/7	240/104/104/24/8 240/138/70/24/8 240/208/0/24/8	280/121/122/28/9 280/162/81/28/9 280/243/0/28/9	336/146/146/33/11 336/194/98/33/11 336/292/0/33/11	421/183/183/41/14 421/244/122/41/14 421/366/0/41/14	562/244/245/55/18 562/326/163/55/18 562/489/0/55/18
MPEG2 per service bit rate (kbps at TS level)	259/111/113/26/9 259/148/76/26/9 259/224//26/9	287/125/125/28/9 287/167/83/28/9 287/250//28/9	324/140/140/32/11 324/187/94/32/11 324/281//32/11	370/160/160/37/12 370/213/108/37/12 370/321//37/12	432/187/188/43/14 432/250/125/43/14 432/375//43/14	518/225/225/51/17 518/299/151/51/17 518/450//51/17	650/282/282/63/22 650/376/188/63/22 650/565//63/22	867/376/378/85/28 867/503/251/85/28 867/754//85/28
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	107/140/214	115/156/238	132/173/263	148/197/304	173/238/353	214/288/427	271/362/542	362/476/723

Table A.10.36: T-16APSK short interleaver legacy

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	X	X	X-	X
Parameter/Case name	16APSK_1/5_S	16APSK_2/9_S	16APSK_1/4_S	16APSK_2/7_S	16APSK_1/3_S	16APSK_2/5_S	16APSK_1/2_S	16APSK_2/5_S
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15	5 MHz, 16APSK; 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	12/48/1/0/0	12/48/1/0/0	12/48/1/0/0	12/48/1/0/0	12/48/1/0/0	12/48/1/0/0	12/48/1/0/0	12/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/218	0/218	0/217	0/218	0/218	0/218	0/217	0/217
MPEG TS total bit rate (Mbps)	3,110	3,456	3,900	4,443	5,184	6,221	7,800	10,417
TS LAYER CONFIGURATION								
Services	12	12	12	12	12	12	12	12
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	81	81	81	81	81	81	81	81
TDM Symbols/coded FEC	15 120	13 608	12 096	10 584	9 072	7 560	6 048	4 536
total/data/ifec/safety/psi kbps@TP per time slice	168/72/73/17/6 168/96/49/17/6 168/145/0/17/6	186/81/81/18/6 186/108/54/18/6 186/162/0/18/6	210/91/91/21/7 210/121/61/21/7 210/182/0/21/7	240/104/104/24/8 240/138/70/24/8 240/208/0/24/8	280/121/122/28/9 280/162/81/28/9 280/243/0/28/9	336/146/146/33/11 336/194/98/33/11 336/292/0/33/11	421/183/183/41/14 421/244/122/41/14 421/366/0/41/14	562/244/245/55/18 562/326/163/55/18 562/489/0/55/18
MPEG2 per service bit rate (kbps at TS level)	259/111/113/26/9 259/148/76/26/9 259/224//26/9	287/125/125/28/9 287/167/83/28/9 287/250//28/9	324/140/140/32/11 324/187/94/32/11 324/281//32/11	370/160/160/37/12 370/213/108/37/12 370/321//37/12	432/187/188/43/14 432/250/125/43/14 432/375//43/14	518/225/225/51/17 518/299/151/51/17 518/450//51/17	650/282/282/63/22 650/376/188/63/22 650/565//63/22	867/376/378/85/28 867/503/251/85/28 867/754//85/28
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1	0,5/0,66/1
Bitrate/service (kbps at IP level)	107/140/214	115/156/238	132/173/263	148/197/304	173/238/353	214/288/427	271/362/542	362/476/723

A.10.5 TDM class 2

For uniform-long, three types of modulations are investigated: T-QPSK, 8PSK and 16APSK. For each modulation we give either {fixed code rate; variable duration} or {variable code rate; fixed duration} configurations.

In class 2 configurations, OFDM modulation choice has a deep impact on the the interleaver duration because the SH frame duration is doubled. Therefore, although most cases are given with O-QPSK modulation, some specific cases are also given with 16QAM modulation.

Table A.10.37: TDM class 2 uniform long physical layer configuration summary

Terrestrial modulation	Modulation	Code rate	Duration	Table
O-QPSK	T-QPSK	1/3	Variable (0,5 s to 10 s)	Table A.10.39
		1/2	Variable (0,5 s to 10 s)	Table A.10.40
		2/3	Variable (0,5 s to 10 s)	Table A.10.41
		Variable (1/5 to 2/3)	10 s Uniform late	Table A.10.42
	8PSK	2/9	Variable (0,5 s to 10 s)	Table A.10.43
		1/3	Variable (0,5 s to 10 s)	Table A.10.44
		2/5	Variable (0,5 s to 10 s)	Table A.10.45
		Variable (1/5 to 2/3)	10 s Uniform late	Table A.10.46
	16APSK	Variable (1/5 to 2/3)	10 s Uniform late	Table A.10.47
16QAM	8PSK	1/3	Variable (0,5 s to 10 s)	Table A.10.57

For uniform-late, three types of modulations are investigated: T-QPSK, 8PSK and 16APSK. For each modulation we give either {fixed code rate; variable duration} or {variable code rate; fixed duration} configurations.

Table A.10.38: TDM class 2 uniform late physical layer configuration summary

Modulation	Code rate	Duration	Table
T-QPSK	1/3	Variable (0,5 s to 10 s)	Table A.10.48
		Variable (5 s to 10 s) Optimized interleaver	Table A.10.49
	Variable (1/5 to 2/5)	10 s Uniform late	Table A.10.50
		10 s Uniform late Optimized interleave only upto 1/3	Table A.10.51
8PSK	2/9	Variable (0,5 s to 10 s)	Table A.10.52
		Variable (5 s to 10 s) Optimized interleaver	Missing
	1/3	Variable (0,5 to 10 s)	Table A.10.53
		Variable (5 s to 10 s) Optimized interleaver	Missing
	2/5	Variable (0,5 s to 10 s)	Table A.10.54
		Variable (5 s to 10 s) Optimized interleaver	Missing
	Variable (1/5 to 2/5)	10 s Uniform late	Table A.10.55
		10 s Uniform late Optimized interleave only upto 1/3	Missing
16APSK	Variable (1/5 to 2/3)	10 s Uniform late	Table A.10.56
		10 s Uniform late Optimized interleave only upto 1/3	Missing

Table A.10.39: TDM QPSK1/3 Uniform Long variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	T_QPSK_1/3_U_11 s	T_QPSK_1/3_U_5,5 s	T_QPSK_1/3_U_2,7 s	T_QPSK_1/3_U_1,2 s	T_QPSK_1/3_U_0,5 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/12/4/2	40/0/12/2/2	20/0/12/1/2	10/0/6/1/1	5/0/3/1/1
Min/Late/Max interleaver duration (ms)	0/0/10 911	0/0/5 549	0/2 775	0/1 273	0/546
MPEG TS total bit rate (Mbps)	2,567	2,567	2,567	2,567	2,567
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
PHY / MAC synchronization	Yes	No	No	No	No
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	18 144	18 144	18 144	18 144	18 144
total/data/ifec/safety/psi kbps@TP per time slice	184/160/0/18/6	184/160/0/18/6	184/160/0/18/6	184/160/0/18/6	184/160/0/18/6
MPEG2 per service bit rate (kbps at TS level)	284/247//28/9	284/247//28/9	284/247//28/9	284/247//28/9	284/247//28/9
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	230	230	230	230	230

Table A.10.40: TDM QPSK1/2 Uniform Long variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_QPSK_1/2_U_11 s	T_QPSK_1/2_U_5 s	T_QPSK_1/2_U_2,7 s	T_QPSK_1/2_U_1,2 s	T_QPSK_1/2_U_0,6 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/2	1/2	1/2	½	1/2
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/0/12/4/2	20/0/12/2/2	10/0/12/1/2	5/0/6/1/2	3/0/3/1/2
Min/Late/Max interleaver duration (ms)	0/0/10 816	0/0/5 454	0/0/2 727	0/0/1 273	0/0/557
MPEG TS total bit rate (Mbps)	3,900	3,900	3,900	3,900	3,900
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
PHY / MAC synchronization	Yes	No	No	No	No
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	12 096	12 096	12 096	12 096	12 096
total/data/ifec/safety/psi kbps@TP per time slice	280/243/0/28/9	280/243/0/28/9	280/243/0/28/9	280/243/0/28/9	280/243/0/28/9
MPEG2 per service bit rate (kbps at TS level)	432/375//43/14	432/375//43/14	432/375//43/14	432/375//43/14	432/375//43/14
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	353	353	353	353	353

Table A.10.41: TDM QPSK2/3 Uniform Long variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_QPSK_2/3_U_9,9 s	T_QPSK_2/3_U_4 s	T_QPSK_2/3_U_2 s	T_QPSK_2/3_U_1 s	T_QPSK_2/3_U_0,4 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	2/3	2/3	2/3	2/3	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/0/12/4/2	20/0/12/2/2	10/0/12/1/2	5/0/6/1/2	3/0/3/1/2
Min/Late/Max interleaver duration (ms)	0 / 0 / 10 816	0 / 0 / 5 455	0 / 0 / 2 727	0 / 0 / 1 273	0 / 0 / 557
MPEG TS total bit rate (Mbps)	5 184	5 184	5 184	5 184	5 184
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
PHY / MAC synchronization	YES	NO	NO	NO	NO
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	9072	9072	9072	9072	9072
total/data/ifec/safety/psi kbps@TP per time slice	373/324/0/37/12	373/324/0/37/12	373/324/0/37/12	373/324/0/37/12	373/324/0/37/12
MPEG2 per service bit rate (kbps at TS level)	575/500//57/19	575/500//57/19	575/500//57/19	575/500//57/19	575/500//57/19
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	476	476	476	476	476

Table A.10.42: TDM 8PSK2/9 Uniform Long variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	T_8PSK_2/9_U_11 s	T_8PSK_2/9_U_6 s	T_8PSK_2/9_U_2,7 s	T_8PSK_2/9_U_1,2 s	T_8PSK_2/9_U_0,5 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	2/9	2/9	2/9	2/9	2/9
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	60/0/12/4/2	60/0/12/2/2	30/0/12/1/1	15/0/6/1/1	5/0/3/1/1
Min/Late/Max interleaver duration (ms)	0/0/10 911	0/0/5 549	0/0/2 728	0/0/1 273	0/0/526
MPEG TS total bit rate (Mbps)	2,567	2,567	2,567	2,567	2,567
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	18 144	18 144	18 144	18 144	18 144
total/data/ifec/safety/psi kbps@TP per time slice	184/160/0/18/6	184/160/0/18/6	184/160/0/18/6	184/160/0/18/6	184/160/0/18/6
MPEG2 per service bit rate (kbps at TS level)	284/247//28/9	284/247//28/9	284/247//28/9	284/247//28/9	284/247//28/9
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	230	230	230	230	230

Table A.10.43: TDM QPSK variable code rate Uniform Long 10 s (O-QPSK)

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	-	X	X	X
Parameter/Case name	T_QPSK_1/5_U	T_QPSK_2/9_U	T_QPSK_1/4_U	T_QPSK_2/7_U	T_QPSK_1/3_U	T_QPSK_2/5_U	T_QPSK1/2_U	T_QPSK2/3_U
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2	40/0/12/4/2
Min/Late/Max interleaver duration (ms)	0/0/10 912	0/0/10 909	0/0/10 911	0/0/10 909	0/0/10 911	0/0/10 909	0/0/10 908	0/0/10 909
MPEG TS total bit rate (Mbps)	1,530	1,728	1,925	2,222	2,567	3,110	3,900	5,184
TS LAYER CONFIGURATION								
Services	9	9	9	9	9	9	9	9
PHY/MAC synchronization	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108	108	108	108
TDM Symbols/coded FEC	30 240	27 216	24 192	21 168	18 144	15 120	12 096	9 072
total/data/ifec/safety/psi kbps@TP per time slice	110/95/0/11/4	124/108/0/12/4	138/119/0/14/5	160/139/0/16/5	184/160/0/18/6	224/195/0/22/7	280/243/0/28/9	373/324/0/37/12
MPEG2 per service bit rate (kbps at TS level)	170/147//17/6	191/167//19/6	213/184//22/8	247/214//25/8	284/247//28/9	346/301//34/11	432/375//43/14	575/500//57/19
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	140	156	173	206	230	288	353	476

Table A.10.44: TDM 8PSK1/3 Uniform Long variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_8PSK_1/3_U_9,9 s	T_8PSK_1/3_U_4 s	T_8PSK_1/3_U_2 s	T_8PSK_1/3_U_1 s	T_8PSK_1/3_U_0,5 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	60/0/12/4/2	30/0/12/2/2	15/0/12/1/2	13/0/6/1/1	5/0/3/1/1
Min/Late/Max interleaver duration (ms)	0/0/10 908	0/0/5 454	0/0/2 727	0/0/1 265	0/0/526
MPEG TS total bit rate (Mbps)	3,900	3,900	3,900	3,900	3,900
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	12 096	12 096	12 096	12 096	12 096
total/data/ifec/safety/psi kbps@TP per time slice	280/243/0/28/9	280/243/0/28/9	280/243/0/28/9	280/243/0/28/9	280/243/0/28/9
MPEG2 per service bit rate (kbps at TS level)	432/375//43/14	432/375//43/14	432/375//43/14	432/375//43/14	432/375//43/14
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	353	353	353	353	353

Table A.10.45: TDM 8PSK2/5 Uniform Long variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_8PSK_2/5_U_20 s	T_8PSK_2/5_U_16 s	T_8PSK_2/5_U_10 s	T_8PSK_2/5_U_4 s	T_8PSK_2/5_U_2 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	2/5	2/5	2/5	2/5	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	60/0/12/8/2	60/0/12/6/2	60/0/12/4/2	30/0/12/2/2	15/0/12/1/2
Min/Late/Max interleaver duration (ms)	0/0/21 631	0/0/16 270	0/0/10 908	0/0/5 454	0/0/2 727
MPEG TS total bit rate (Mbps)	4,690	4,690	4,690	4,690	4,690
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	No	Yes	No	No
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	10 080	10 080	10 080	10 080	10 080
total/data/ifec/safety/psi kbps@TP per time slice	337/293/0/33/11	337/293/0/33/11	337/293/0/33/11	337/293/0/33/11	337/293/0/33/11
MPEG2 per service bit rate (kbps at TS level)	520/452//51/17	520/452//51/17	520/452//51/17	520/452//51/17	520/452//51/17
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	435	435	435	435	435

Table A.10.46: TDM 8PSK variable code rate Uniform Long 10 s (O-QPSK)

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	-	X	X	X	X	X	X
Parameter/Case name	8PSK_1/5_U	8PSK_2/9_U	8PSK_1/4_U	8PSK_2/7_U	8PSK_1/3_U	8PSK_2/5_U	8PSK1/2_U	8PSK2/3_U
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	60/0/12/4/2	60/0/12/4/2	60/0/12/4/2	60/0/12/4/2	60/0/12/4/2	60/0/12/4/2	60/0/12/4/2	60/0/12/4/2
Min/Late/Max interleaver duration (ms)	0/0/10 910	0/0/10 910	0/0/10 910	0/0/10 910	0/0/10 910	0/0/10 910	0/0/10 910	0/0/10 910
MPEG TS total bit rate (Mbps)	2,320	2,567	2,913	3,308	3,900	4,690	5,826	7,800
TS LAYER CONFIGURATION								
Services	9	9	9	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108	108	108	108
TDM Symbols/coded FEC	20 160	18 144	16 128	14 112	12 096	10 080	8 064	6 048
total/data/ifec/safety/psi kbps@TP per time slice	167/144/0/17/6	184/160/0/18/6	209/181/0/21/7	238/207/0/23/8	280/243/0/28/9	337/293/0/33/11	419/364/0/41/14	561/488/0/55/18
MPEG2 per service bit rate (kbps at TS level)	258/222//26/9	284/247//28/9	322/279//32/11	367/319//35/12	432/375//43/14	520/452//51/17	646/562//63/22	866/753//85/28
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	214	230	263	304	353	435	534	723

Table A.10.47: TDM 16APSK variable code rate Uniform Long 10 s (O-QPSK)

ID	1	2	3	4	5	6	7	8
new to IG rel 1	X	X	X	X	X	X	X	X
Parameter/Case name	16APSK_1/5_U	16APSK_2/9_U	16APSK_1/4_U	16APSK_2/7_U	16APSK_1/3_U	16APSK_2/5_U	16APSK1/2_U	16APSK2/3_U
PHYSICAL CONFIGURATION								
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5	1/2	2/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/ non_late_increment)	63/0/12/4/2	63/0/12/4/2	63/0/12/4/2	63/0/12/4/2	63/0/12/4/2	63/0/12/4/2	63/0/12/4/2	63/0/12/4/2
Min/Late/Max interleaver duration (ms)	0/0/10 870	0/0/10 870	0/0/10 869	0/0/10 870	0/0/10 870	0/0/10 870	0/0/10 869	0/0/10 870
MPEG TS total bit rate (Mbps)	3,110	3,456	3,900	4,443	5,184	6,221	7,800	10,417
TS LAYER CONFIGURATION								
Services	9	9	9	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	975	975	975	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108	108	108	108
TDM Symbols/coded FEC	15 120	13 608	12 096	10 584	9 072	7 560	6 048	4 536
total/data/ifec/safety/psi kbps@TP per time slice	224/195/0/22/7	248/216/0/24/8	280/243/0/28/9	320/279/0/31/10	373/324/0/37/12	448/390/0/44/14	561/488/0/55/18	750/653/0/73/24
MPEG2 per service bit rate (kbps at TS level)	346/301//34/11	383/333//37/12	432/375//43/14	494/430//48/15	575/500//57/19	691/602//68/22	866/753//85/28	1157/1007//113/37
LINK LAYER CONFIGURATION								
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at TS level)	288	321	353	411	476	575	723	969

Table A.10.48: TDM QPSK1/3 Uniform Late variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	T_QPSK_1/3_UL_9,9 s	T_QPSK_1/3_UL_4 s	T_QPSK_1/3_UL_2 s	T_QPSK_1/3_UL_1 s	T_QPSK_1/3_UL_0,4 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/24/9/5/12	10/24/9/2/6	10/24/9/1/3	10/24/5/1/3	12/48/1/0/0
Min/Late/Max interleaver duration (ms)	0/180/9 936	0/3 993	0/1 997	0/1 092	0/441
MPEG TS total bit rate (Mbps)	2,567	2,567	2,567	2,567	2,567
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	No	No	No	No
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135
CW per SH frame	52	52	52	52	52
total/data/ifec/safety/psi TP per time slice	231/200/0/23/8	231/200/0/23/8	231/200/0/23/8	231/200/0/23/8	231/200/0/23/8
total/data/ifec/safety/psi kbps@TP per time slice	285/247//28/10	285/247//28/10	285/247//28/10	285/247//28/10	285/247//28/10
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	237	237	237	237	237

Table A.10.49: TDM QPSK1/3 optimized Uniform Late variable delay (O-QPSK)

ID	1	2
new to IG rel 1	-	X
Parameter/Case name	T_QPSK_1/3_UL_9,9 s	T_QPSK_1/3_UL_4 s
PHYSICAL CONFIGURATION		
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	5/18/11/4/9	5/18/6/4/4
Min/Late/Max interleaver duration (ms)	0/66/9 819	0/66/4 952
MPEG TS total bit rate (Mbps)	2,567	2,567
TS LAYER CONFIGURATION		
Services	9	9
Synchronization LINK/PHY	Yes	No
Repetition interval (ms)	1 219	1 219
Burst duration (ms)	135	135
CW per SH frame	52	52
total/data/ifec/safety/psi TP per time slice	231/200/0/23/8	231/200/0/23/8
total/data/ifec/safety/psi kbps@TP per time slice	285/247//28/10	285/247//28/10
LINK LAYER CONFIGURATION		
LL-FEC rate min/recommended/max	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	237	237

Table A.10.50: TDM QPSK variable code rate Uniform Late 10 s (O-QPSK)

ID	1	2	3	4	5	6
new to IG rel 1	X	X	X	X	-	X
Parameter/Case name	T_QPSK_1/5_UL	T_QPSK_2/9_UL	T_QPSK_1/4_UL	T_QPSK_2/7_UL	T_QPSK_1/3_UL	T_QPSK_2/5_UL
PHYSICAL CONFIGURATION						
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12	10/24/9/5/12
Min/Late/Max interleaver duration (ms)	0/9 937	0/9 934	0/9 936	0/9 934	0/9 936	0/9 934
MPEG TS total bit rate (Mbps)	1,530	1,728	1,925	2,222	2,567	3,110
TS LAYER CONFIGURATION						
Services	9	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135	135
CW per SH frame	31	35	39	45	52	63
total/data/ifec/safety/psi TP per time slice	137/118/0/14/5	155/135/0/15/5	173/150/0/17/6	200/173/0/20/7	231/200/0/23/8	280
total/data/ifec/safety/psi kbps@TP per time slice	169/146//17/6	191/167//19/6	214/185//21/7	247/214//25/9	285/247//28/10	346
LINK LAYER CONFIGURATION						
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	138	158	178	204	237	346

Table A.10.51: TDM QPSK variable code rate optimized Uniform Late 10 s (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	-
Parameter/Case name	T_QPSK_1/5_UL	T_QPSK_2/9_UL	T_QPSK_1/4_UL	T_QPSK_2/7_UL	T_QPSK_1/3_UL
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15	5 MHz, QPSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	5/12/10/5/4	5/15/12/4/7	5/16/9/5/6	5/16/9/5/6	5/18/11/4/6
Min/Late/Max interleaver duration (ms)	0/43/11 014	0/55/10 777	0/59/9 819	0/59/9 818	0/66/9 795
MPEG TS total bit rate (Mbps)	1,530	1,728	1,925	2,222	2,567
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135
CW per SH frame	31	35	39	45	52
total/data/ifec/safety/psi TP per time slice	137/118/0/14/5	155/135/0/15/5	173/150/0/17/6	200/173/0/20/7	231/200/0/23/8
total/data/ifec/safety/psi kbps@TP per time slice	169/146//17/6	191/167//19/6	214/185//21/7	247/214//25/9	285/247//28/10
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	138	158	178	204	237

Table A.10.52: TDM 8PSK2/9 Uniform Late variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	-	X	X	X	X
Parameter/Case name	T_8PSK_2/9_UL_9,9 s	T_8PSK_2/9_UL_6 s	T_8PSK_2/9_UL_4 s	T_8PSK_2/9_UL_1 s	T_8PSK_2/9_UL_0,5 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	2/9	2/9	2/9	2/9	2/9
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	15/24/9/5/12	15/24/9/3/12	8/24/9/2/6	5/24/5/1/3	3/24/3/1/1
Min/Late/Max interleaver duration (ms)	0/180/9 936	0/180/6 037	0/96/3 949	0/60/1 014	0/36/505
MPEG TS total bit rate (Mbps)	2,567	2,567	2,567	2,567	2,567
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	No	No	No	No
Repetition interval (ms)	1219	1219	1219	1219	1219
Burst duration (ms)	135	135	135	135	135
TDM Symbols/coded FEC	18 144	18 144	18 144	18 144	18 144
total/data/ifec/safety/psi TP per time slice	231/200/0/23/8	231/200/0/23/8	231/200/0/23/8	231/200/0/23/8	231/200/0/23/8
total/data/ifec/safety/psi kbps@TP per time slice	285/247//28/10	285/247//28/10	285/247//28/10	285/247//28/10	285/247//28/10
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	237	237	237	237	237

Table A.10.53: TDM 8PSK1/3 Uniform Late variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_8PSK_1/3_UL_9,9 s	T_8PSK_1/3_UL_4 s	T_8PSK_1/3_UL_2 s	T_8PSK_1/3_UL_1 s	T_8PSK_1/3_UL_0,5 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	15/24/9/5/12	12/24/9/2/6	12/24/9/1/3	5/24/5/1/1	3/24/3/1/1
Min/Late/Max interleaver duration (ms)	0/177/9 934	0/142/3 973	0/142/1 987	0/59/988	0/35/504
MPEG TS total bit rate (Mbps)	3,900	3,900	3,900	3,900	3,900
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	No	No	No	No
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135
TDM Symbols/coded FEC	12 096	12 096	12 096	12 096	12 096
total/data/ifec/safety/psi TP per time slice	351/306/0/34/11	351/306/0/34/11	351/306/0/34/11	351/306/0/34/11	351/306/0/34/11
total/data/ifec/safety/psi kbps@TP per time slice	433/378//42/14	433/378//42/14	433/378//42/14	433/378//42/14	433/378//42/14
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	362	362	362	362	362

Table A.10.54: TDM 8PSK2/5 Uniform Late variable delay (O-QPSK)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_8PSK_2/5_UL_20 s	T_8PSK_2/5_UL_16 s	T_8PSK_2/5_UL_10 s	T_8PSK_2/5_UL_4 s	T_8PSK_2/5_UL_2 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	2/5	2/5	2/5	2/5	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	15/24/9/10/12	15/24/9/8/12	15/24/9/5/12	15/24/9/2/6	10/24/9/1/4
Min/Late/Max interleaver duration (ms)	0/177/19 682	0/177/15 782	0/177/9 933	0/177/3 992	0/118/1 991
MPEG TS total bit rate (Mbps)	4,690	4,690	4,690	4,690	4,690
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	Yes	No	No	No	No
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135
TDM Symbols/coded FEC	10 080	10 080	10 080	10 080	10 080
total/data/ifec/safety/psi TP per time slice	422/367/0/41/14	422/367/0/41/14	422/367/0/41/14	422/367/0/41/14	422/367/0/41/14
total/data/ifec/safety/psi kbps@TP per time slice	521/453//51/17	521/453//51/17	521/453//51/17	521/453//51/17	521/453//51/17
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	434,00	434,00	434,00	434,00	434,00

Table A.10.55: TDM 8PSK variable code rate Uniform Late 10 s (O-QPSK)

ID	1	2	3	4	5	6
new to IG rel 1	X	-	X	X	X	X
Parameter/Case name	8PSK_1/5_UL	8PSK_2/9_UL	8PSK_1/4_UL	8PSK_2/7_UL	8PSK_1/3_UL	8PSK_2/5_UL
PHYSICAL CONFIGURATION						
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	15/24/9/5/12	15/24/9/5/12	15/24/9/5/12	15/24/9/5/12	15/24/9/5/12	15/24/9/5/12
Min/Late/Max interleaver duration (ms)	0/179/9 935	0/179/9 935	0/179/9 935	0/179/9 935	0/179/9 935	0/179/9 935
MPEG TS total bit rate (Mbps)	2,320	2,567	2,913	3,308	3,900	4,690
TS LAYER CONFIGURATION						
Services	9	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135	135
TDM Symbols/coded FEC	20 160	18 144	16 128	14 112	12 096	10 080
total/data/ifec/safety/psi TP per time slice	208/180/0/21/7	231/200/0/23/8	262/227/0/26/9	297/258/0/29/10	351/306/0/34/11	422/367/0/41/14
total/data/ifec/safety/psi kbps@TP per time slice	257/222//26/9	285/247//28/10	323/280//32/11	367/318//36/12	433/378//42/14	521/453//51/17
LINK LAYER CONFIGURATION						
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	211	237	270	302	362	434

Table A.10.56: TDM 16APSK variable code rate Uniform Late 10 s (O-QPSK)

ID	1	2	3	4	5	6
new to IG rel 1	X	X	X	X	X	X
Parameter/Case name	16APSK_1/5_UL	16APSK_2/9_UL	16APSK_1/4_UL	16APSK_2/7_UL	16APSK_1/3_UL	16APSK_2/5_UL
PHYSICAL CONFIGURATION						
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15	5 MHz, 16APSK, 0,15
PHY FEC rate	1/5	2/9	1/4	2/7	1/3	2/5
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	20/24/9/5/12	20/24/9/5/12	20/24/9/5/12	20/24/9/5/12	20/24/9/5/12	20/24/9/5/12
Min/Late/Max interleaver duration (ms)	0/178/9 934	0/178/9 936	0/177/9 934	0/178/9 936	0/178/9 934	0/178/9 933
MPEG TS total bit rate (Mbps)	3,110	3,456	3,900	4,443	5,184	6,221
TS LAYER CONFIGURATION						
Services	9	9	9	9	9	9
Synchronization LINK/PHY	Yes	Yes	Yes	Yes	Yes	Yes
Repetition interval (ms)	1 219	1 219	1 219	1 219	1 219	1 219
Burst duration (ms)	135	135	135	135	135	135
CW per SH frame	63	70	79	90	105	126
TDM Symbols/coded FEC	15120	13608	12096	10584	9072	7560
total/data/ifec/safety/psi TP per time slice	280/243/0/28/9	311/270/0/31/10	351/306/0/34/11	400/348/0/39/13	466/405/0/46/15	560/487/0/55/18
total/data/ifec/safety/psi kbps@TP per time slice	346/300//35/11	384/333//38/12	433/378//42/14	494/430//48/16	575/500//57/19	691/601//68/22
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	283	316	362	408	480	578

Table A.10.57: TDM 8PSK1/3 Uniform Long variable delay (O-16QAM)

ID	1	2	3	4	5
new to IG rel 1	X	X	X	X	X
Parameter/Case name	T_8PSK_1/3_U_9,9 s	T_8PSK_1/3_U_5 s	T_8PSK_1/3_U_2 s	T_8PSK_1/3_U_1 s	T_8PSK_1/3_U_0,5 s
PHYSICAL CONFIGURATION					
OFDM related parameters (BW, GI, OFDM modulation order)	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK	5 MHz, GI=1/4, QPSK
TDM modulation parameters (TDM bandwidth, TDM_modulation; roll-off)	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15	5 MHz, 8PSK, 0,15
PHY FEC rate	1/3	1/3	1/3	1/3	1/3
Interleaver configuration (common_multiplier/nof_late_taps/ nof_slices/slice_distance/non_late_increment)	40/0/16/5/2	60/0/12/4/2	20/0/18/1/2	13/0/9/1/1	5/0/5/1/1
Min/Late/Max interleaver duration (ms)	0/0/9 223	0/0/5 549	0/0/2 342	0/0/1 022	0/0/516
MPEG TS total bit rate (Mbps)	3,851	3,851	3,851	3,851	3,851
TS LAYER CONFIGURATION					
Services	9	9	9	9	9
Synchronization LINK/PHY	No	No	No	No	No
Repetition interval (ms)	975	975	975	975	975
Burst duration (ms)	108	108	108	108	108
TDM Symbols/coded FEC	12 096	12 096	12 096	12 096	12 096
total/data/ifec/safety/psi kbps@TP per time slice	277/276/0/0/1	277/276/0/0/1	277/276/0/0/1	277/276/0/0/1	277/276/0/0/1
MPEG2 per service bit rate (kbps at TS level)	427/426///2	427/426///2	427/426///2	427/426///2	427/426///2
LINK LAYER CONFIGURATION					
LL-FEC rate min/recommended/max	1/1/1	1/1/1	1/1/1	1/1/1	1/1/1
Bitrate/service (kbps at IP level)	403	403	403	403	403

A.11 Reference Demodulator

A further refinement has been performed by defining reference demodulator algorithms for investigating the impact of imperfect channel estimation. The demodulator algorithms that are described below are based on the data-aided approach and are considered representative of practical implementations (enhancements are possible). Carrier frequency and symbol clock extraction have not been included in the model as their impact are considered negligible. The models have been implemented in floating point.

A.11.1 TDM Case

A.11.1.1 TDM Reference Demodulator

The proposed reference demodulator heavily exploits the TDM pilot symbols. The DVB-SH TDM slot structure is depicted in figure A.11.1. For easing the acquisition and tracking a regular pilot spacing within each PL slot has been included in the DVB-SH standard. Each TDM slot of length L_{TOT} is subdivided in *two* data sub-slots and pilot fields.

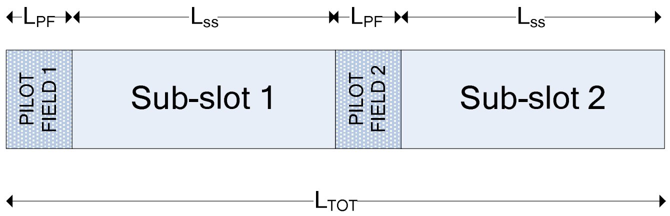


Figure A.11.1: Slot pilot insertion in DVB-SH TDM frame

Carrier Phase Estimation:

The DVB-S2 carrier phase estimator described in clause 3.6 of [i.16] has been adapted to the DVB-SH case. Following [i.16], the phase estimator consists of a ML feed-forward phase estimator operating on each pilot block belonging to the slot. The ML phase estimator algorithm adapted from [i.16] for the pilot group #n is given by:

$$z_n = \sum_{k=1}^{L_{PF}} \left[C\left[k+(n-1)(L_{SS}+L_{PF})\right] \right]^* r\left[k+(n-1)(L_{SS}+L_{PF})\right],\ n=1,..\ N_p$$
$$\hat{\theta}_{PF}(n) = \arg\{z_n\},\ n=1,..\ N_p$$

where r (k) represents the k-th on-time baseband demodulator square-root raised-cosine chip matched filter output complex samples after frequency offset removal.

The phase unwrapping allows to implement a simple linear phase interpolator between consecutive pilot estimations.

The final unwrapped pilot estimates $\hat{\theta}_{PFU}(n)$ are computed from $\hat{\theta}_{PF}(n)$ as:

$$\hat{\theta}_{PFU}(n) = \hat{\theta}_{PFU}(n-1) + SAW\left[\hat{\theta}_{PF}(n) - \hat{\theta}_{PFU}(n-1)\right]$$

where $SAW[\Phi] \equiv [\Phi]_{-\pi}^{+\pi}$ is a saw tooth nonlinearity that reduces Φ to the interval (-π, π). Linear interpolation between consecutive pilot blocks n and n+1 for sub-slice carrier phase removal then follows trough the following equation:

$$\hat{\theta}_{PFUI}(l,n) = \hat{\theta}_{PFU}(n) + \frac{l}{L_{SS}}\left[\hat{\theta}_{PFU}(n+1) - \hat{\theta}_{PFU}(n)\right]$$

where l = 1, 2, …, L_{ss}, is the symbol index within the sub-slice n.

Carrier amplitude estimation:

By reusing the pilot-aided estimated signal phasor z_n one can compute the signal amplitude simply as:

$$A(n) = |z_n|, \; n = 1,..\, N_p$$

Linear amplitude interpolation between consecutive pilot blocks *n* and *n*+1 for sub-slice then follows trough the following equation:

$$A(l,n) = A(n) + \frac{l}{L_{SS}}\left[A(n+1) - A(n)\right]$$

where l = 1, 2, …, L_{ss}, is the symbol index within the sub-slice *n*.

Signal-to-Noise-Ratio Estimation:

The signal to noise ratio estimation is required for the correct turbo decoder operation. For the SNR estimator the data-aided version of the SNORE [SNORE] algorithm based on pilot symbols have been adopted. The algorithmic details can be found in [i.16].

A.11.1.2 TDM Reference Demodulator Simulation Results

To assess the performance of the DVB-SH TDM option a physical layer simulator inclusive of the above described channel estimation algorithms has been implemented.

First a Ricean fading channel with *K* = 5 dB, mobile speed of *v* = 50 kmph and worst-case for single antenna reception SNR = -3,5 dB has been considered. Channel estimation results are plotted in figures A.11.2 to A.11.4. Some high error peaks are present corresponding to deep fading conditions. In these conditions, the channel estimation has no impact on BER as the Turbo decoder will not be able to decode even with perfect channel estimation. The BER curve confirms this analysis.

Another set of channel estimation results have been obtained for the LMS intermediate tree shadowed channel with mobile speed of *v* = 70 kmph and a line-of-sight SNR = 10 dB (considered to be a representative demodulator operating point in this channel). The pilot-aided phase estimation show to perform well also in this difficult type of channel. Again large phase error peaks correspond to deep channel fading conditions.

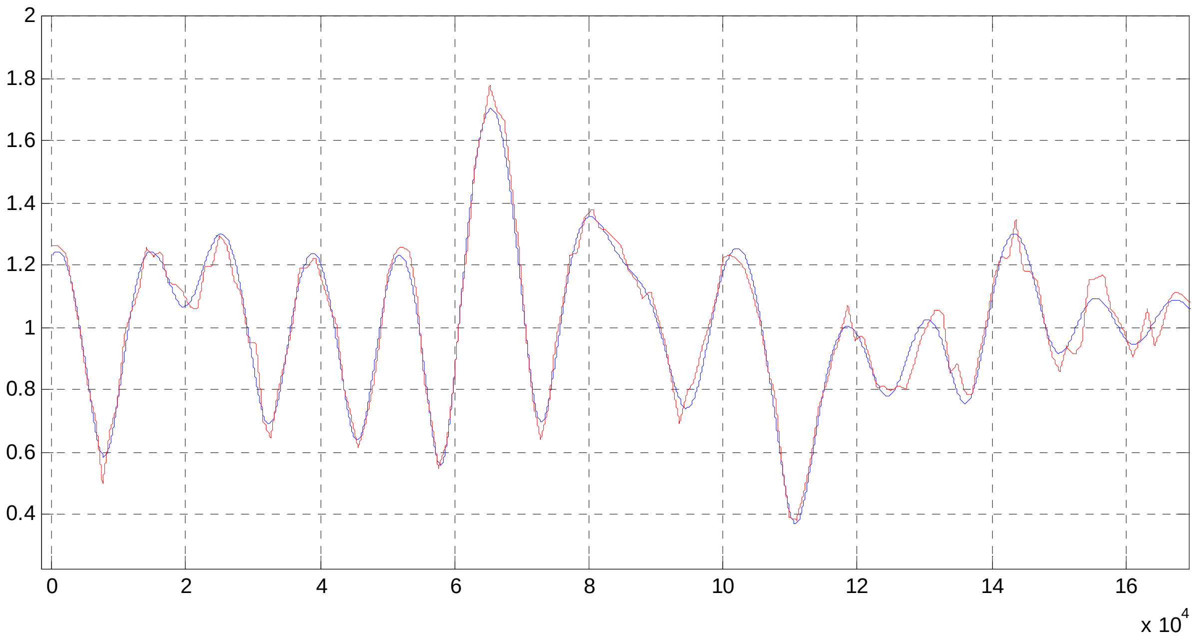


Figure A.11.2: Simulated TDM reference demodulator channel estimation amplitude for Rice K = 5 dB, mobile v = 50 kmph channel, SNR = -3,5 dB

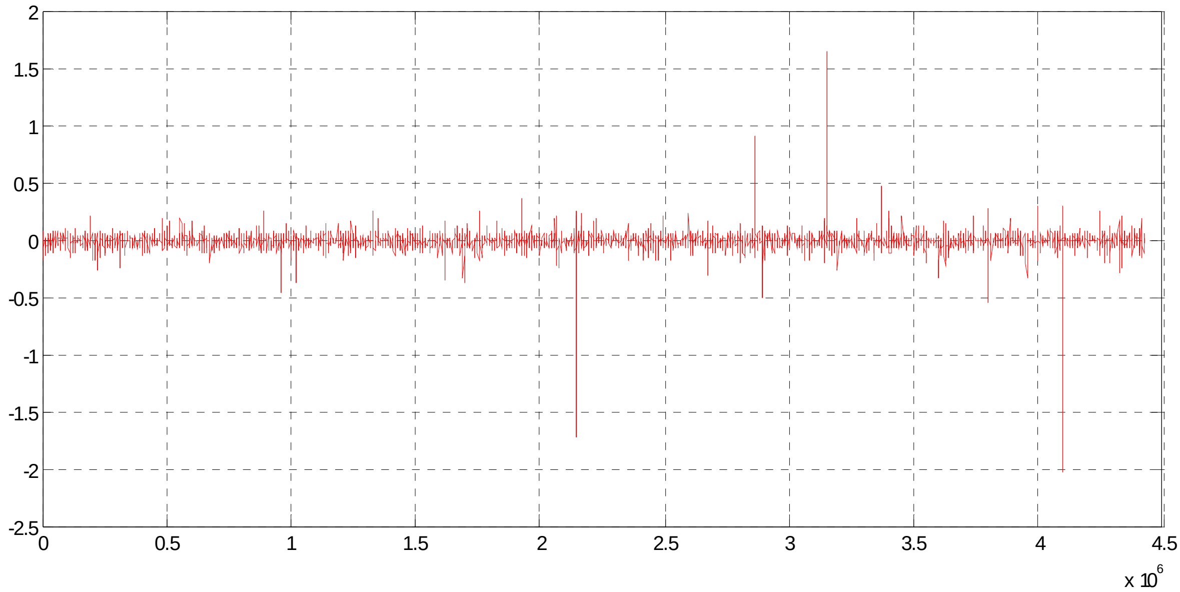


Figure A.11.3: Simulated TDM reference demodulator channel estimation phase error in radians for Rice K = 5 dB, mobile v = 50 kmph channel, SNR = -3,5 dB

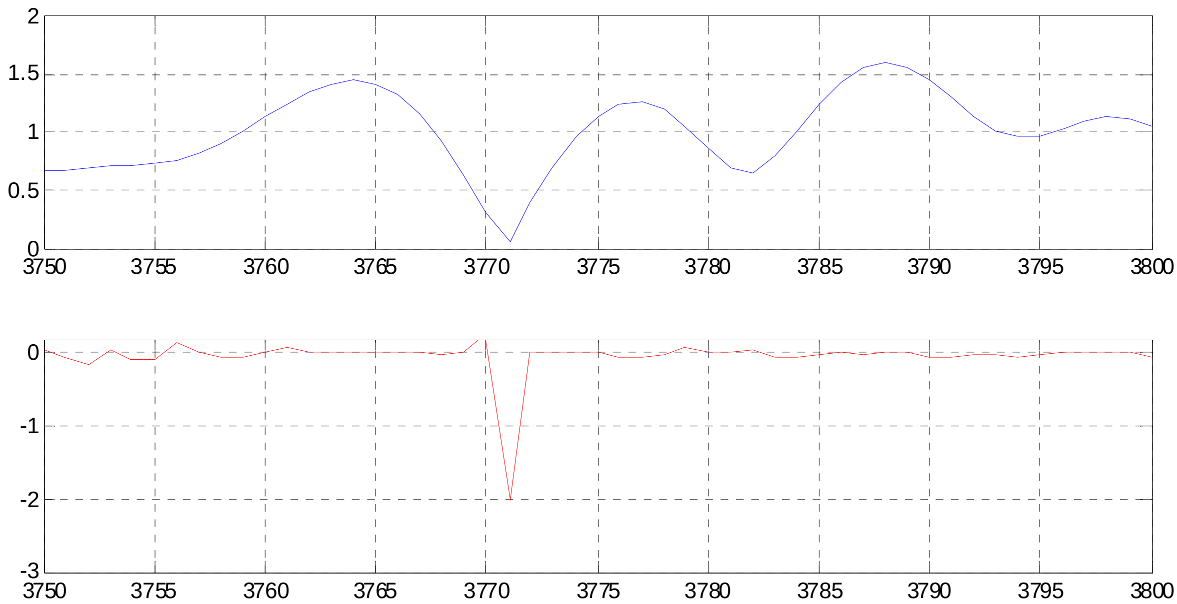


Figure A.11.4: Simulated TDM reference demodulator channel estimation for Rice K = 5 dB, mobile v = 50 kmph channel, SNR = -3,5 dB: zoom of fading channel amplitude and estimated carrier phase error relation

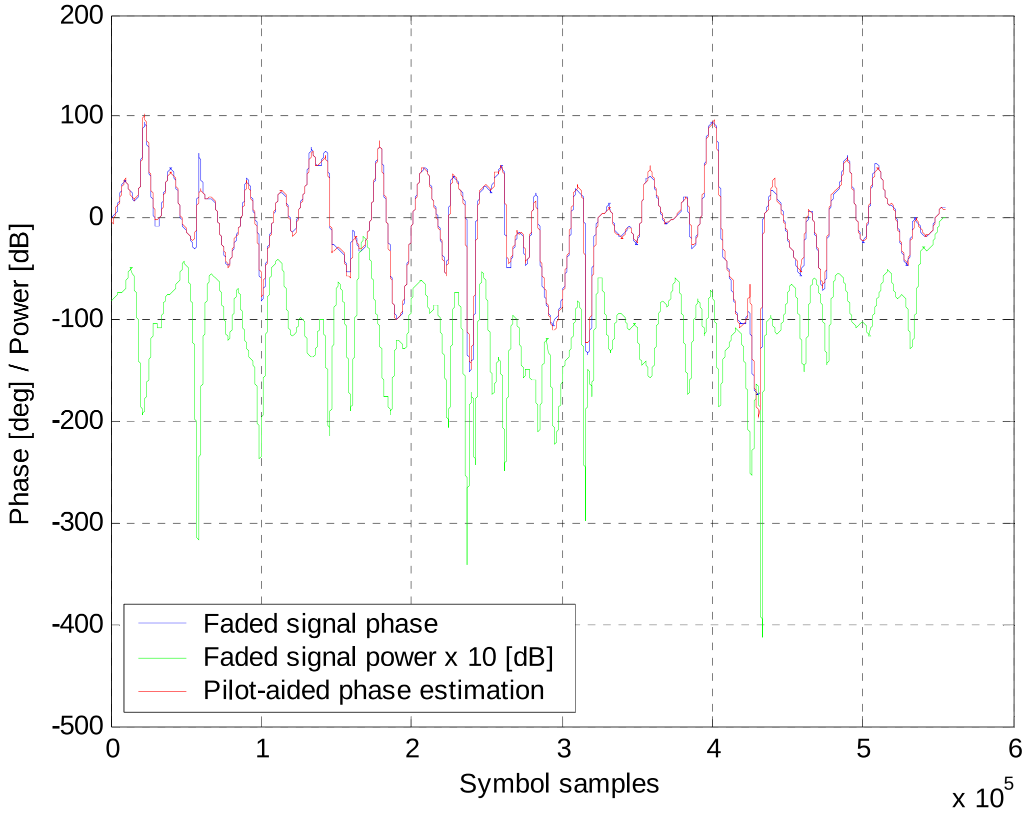


Figure A.11.5: Real and simulated TDM reference demodulator channel carrier phase estimation for v = 70 kmph, SNR = 10 dB and a LMS intermediate state of the intermediate tree shadowing environment

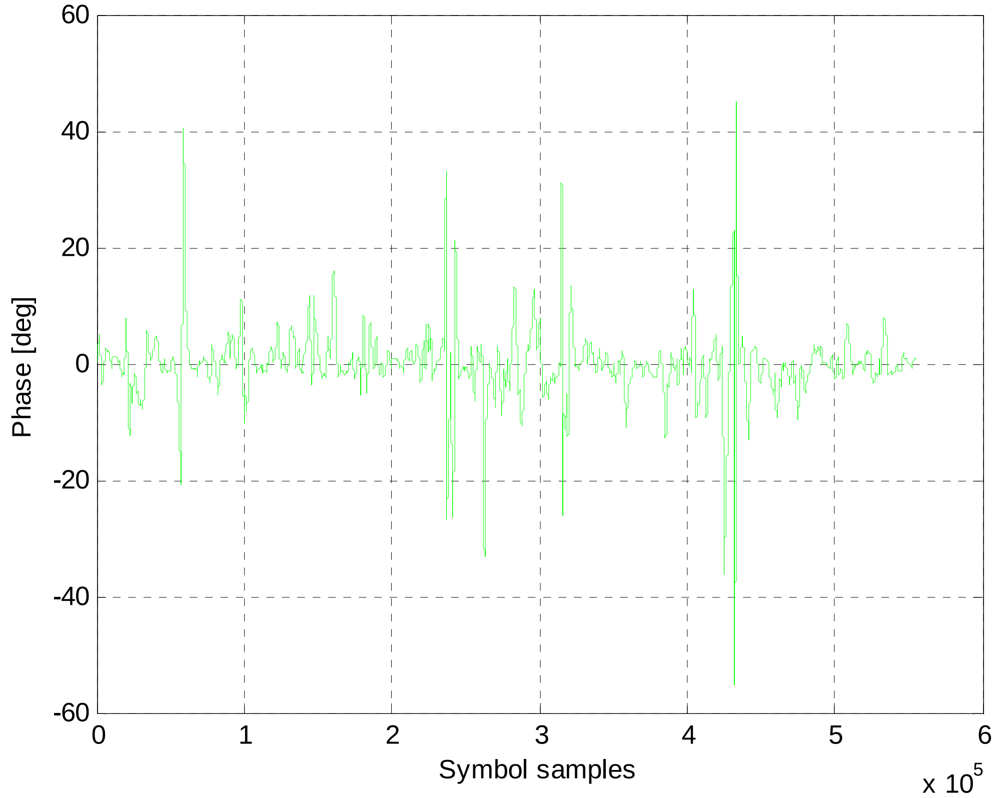


Figure A.11.6: TDM reference demodulator channel carrier phase estimation error for v = 70 kmph, SNR = 10 dB and a LMS intermediate state of the intermediate tree shadowing environment

A more comprehensive analysis of the channel estimation impact has been performed looking at the BER performance versus E_s / N_0 in Ricean *K*=5 dB channels. Figure A.11.7 shows the corresponding results for a short PL interleaving configuration. AWGN and Rice reference performance with ideal channel estimation (for short 200 ms interleaver and infinite length interleaver (uncorrelated fading)) are also reported for convenience. It can be concluded that the proposed reference demodulator channel estimation has negligible impact with Ricean channel *K* = 5 dB.

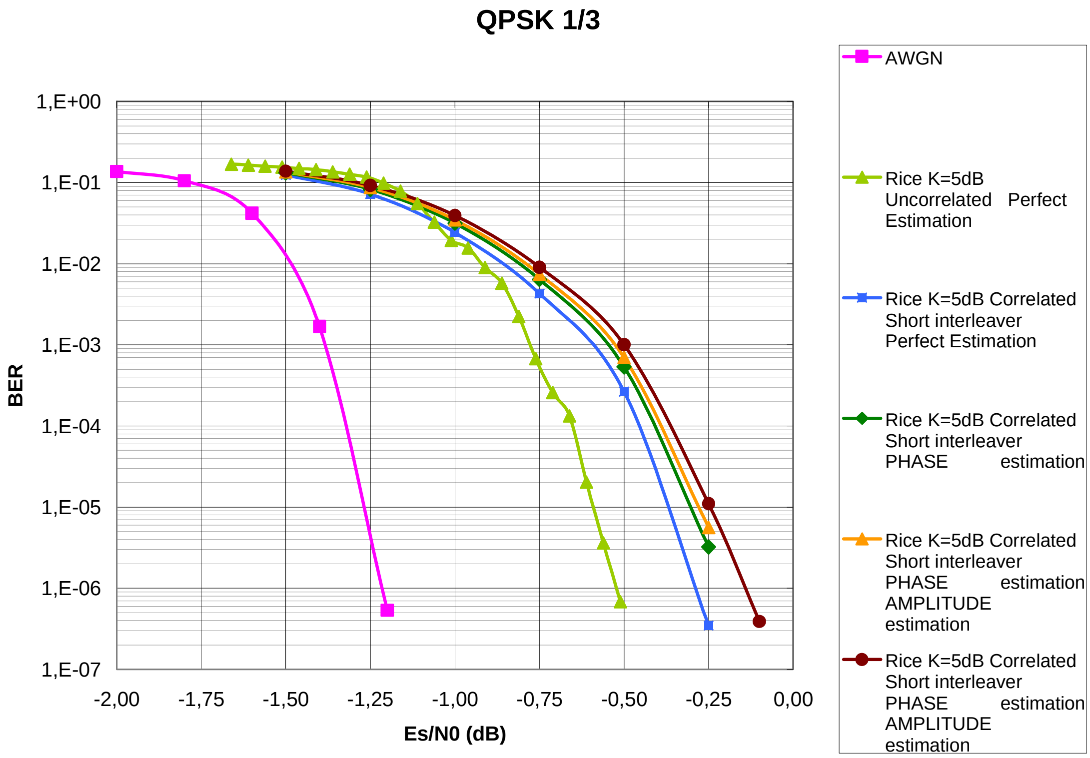


Figure A.11.7: Simulated BER TDM reference demodulator BER performance for Rice *K*=5 dB, mobile *v*= 50 kmph channel, as a function of E_s / N_0

Reference demodulator performance have also been assessed for the LMS 3-state channel case with the demodulator state machine on. The results at C/N = 10 dB are summarized in table A.11.1. It is apparent that the TDM reference demodulator has no practical impact on the ESR5 performance.

The final verification performed on the TDM reference demodulator is related to the carrier phase noise impact. For this a synthetic phase noise nuisance following the mask of clause has been included in the simulation chain. We remark that the low frequency component of the DVB-SSP phase noise is less than the DVB-S2 one (about 30 dB at 100 Hz) but higher-frequency component (above 1E5 Hz) is larger. The mild slope of the DVB-SSP phase noise PSD makes the higher frequency component contribution dominating. This makes the phase noise time series much faster than the fading/shadowing ones. The simulated BER performance results are reported in figure A.11.8. It can be concluded that for Rice channel with K = 5 dB and mobile speed of 50 kmph the carrier phase noise impact for QPSK r = 1/3 is less than 0,05 dB at BER = 1E-5.

Table A.11.1: Summary of ESR5 results at C/N = 10 dB with TDM state machine on and ideal or reference demodulator channel estimation

Reference case	FEC/Interleaver	ESR5 satisfaction % ideal channel estimation	ESR5 satisfaction % reference demodulator channel estimation
SubUrban (SU) V = 3 kmph	Short PL interleaver	45,6	48,9
	Non uniform long PL interleaver	88,5	87,6
	Uniform long PL interleaver	89,3	88,5
	Short PL interleaver plus upper layer FEC	87,6	Not available
Intermediate Tree Shadowing (ITS) V = 50 kmph	Short PL interleaver	0,0	0,0
	Non uniform long PL interleaver	25,2	25,6
	Uniform long PL interleaver	80,4	78,2
	Short PL interleaver plus upper layer FEC	19,5	Not available

Table A.11.2: S-band DVB-SH phase noise mask (Satellite contribution only)

	10 Hz	100 Hz	1 000 Hz	10 kHz	100 kHz	1 MHz	10 MHz
Phase noise (dBc/Hz)	-29	-59	-69	-74	-83	-95	-101

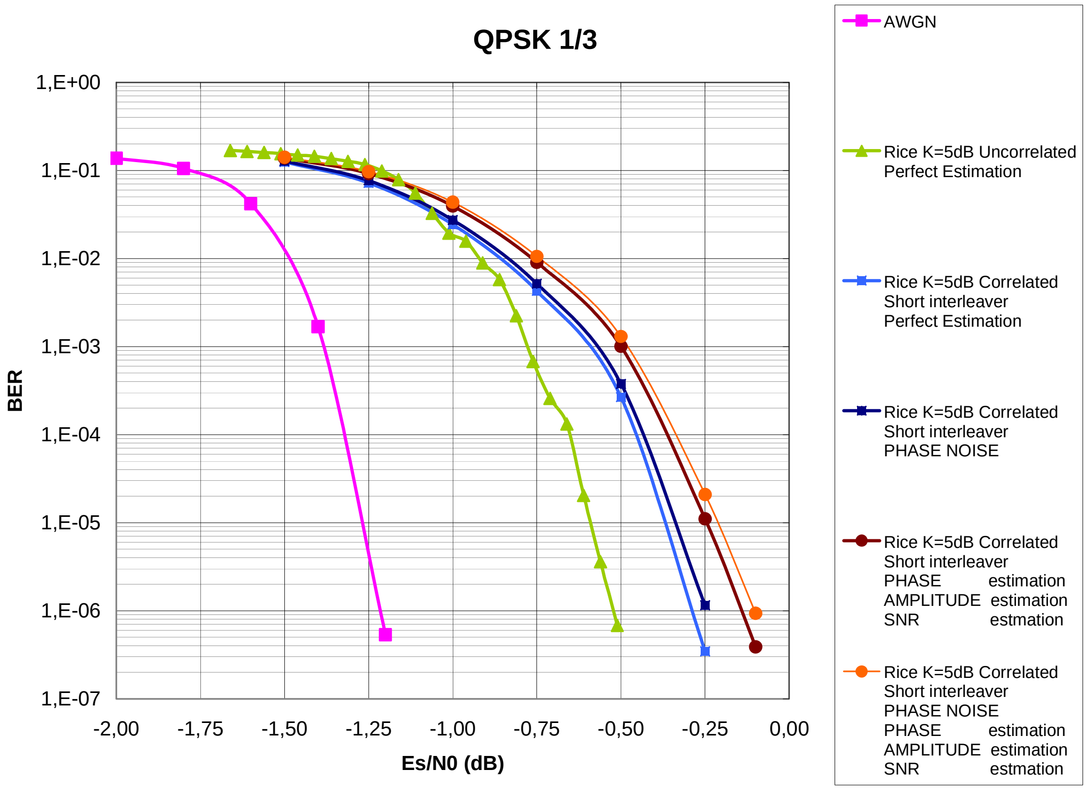


Figure A.11.8: Simulated BER TDM reference demodulator BER performance for Rice *K* = 5 dB, mobile *v* = 50 kmph channel, as a function of E_s / N_0 with and without phase noise

A.11.2 OFDM Case

A.11.2.1 OFDM Reference Demodulator (see note)

Channel transfer estimation is performed through adaptive Least Mean Square (LMS) interpolation both on time and frequency domain thanks to the knowledge of the channel at pilot positions. Firstly time-domain adaptive interpolation is performed to calculate the channel transfer function at scattered pilot subcarriers. Based on these estimates, frequency-domain adaptive interpolation is then performed at the non-pilot subcarriers. The following scheme is based on the assumption of a DVB-SH compliant OFDM transmission.

At the pilot symbols location, the transfer function of the channel is calculated as:

$$H(n,k) = \frac{Y(n,k)}{X(n,k)} \quad \text{(A.1)}$$

where $Y(n,k)$ represents the received complex symbol at for the *k*-th subcarrier of the *n*-th OFDM symbol and $X(n,k)$ represents the transmitted pilot known at the demodulator corresponding to the same position in time and frequency.

Time domain interpolation

a) Filter coefficients update.

The interpolator coefficient update exploits the information carried by the continual pilot subcarriers, where the channel transfer function is known at each OFDM symbol.

The result from (see above equation) is then compared to the estimate of the channel in the same position (n,k) and performed through the interpolator filter. The filter coefficients are updated thanks to the estimation error, calculated as the distance between the estimated channel and the calculated one.

The time-domain interpolator filter is designed to be $4(M_1+M_2)$-OFDM symbol long, where $4M1$ and $4M_2$ represent the number of exploited OFDM symbols before and after the instant to be estimated.

The procedure is divided into three separate steps.

Step 1: Channel transfer function estimation

The channel estimation is performed filtering the pilots of the tone under analysis, using the set of coefficients calculated thanks to the previous $(n-1)$-th OFDM symbol.

The estimation is repeated three times employing three different set of coefficients and pilot symbols belonging to continual subcarrier C_i (a graphical interpretation can be find in figure A.11.9):

$$H_L(n,C_i) = \sum_{m=-M_1+1}^{M_2} W_L(n-1,m)\,H(n-L+4m,C_i) \qquad \text{(A.2)}$$

where $W_L(n-1,m)$ represents the m-th coefficient of the L-th filter (L=1,2,3), updated during the previous $(n-1)$-th OFDM symbol.

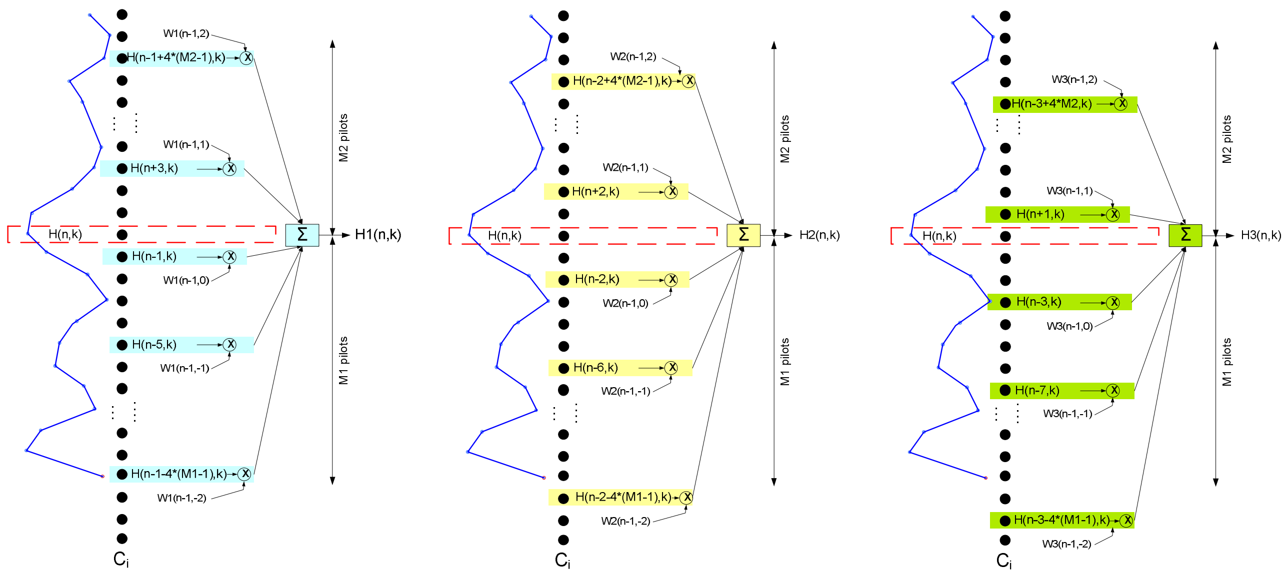


Figure A.11.9: Graphical representation of the time domain channel estimation algorithm on continual pilot C_i

Each filter performs the channel estimation for the same cell of the n-th OFDM symbol. Only one pilot every four is exploited and each set consists of a different pilot group. This redundancy is justified by the regular distribution of the scattered-pilot cells (see figure A.11.10). The column C_i represents the (time) continual pilots while the column S_j represents the (time) scattered pilots.

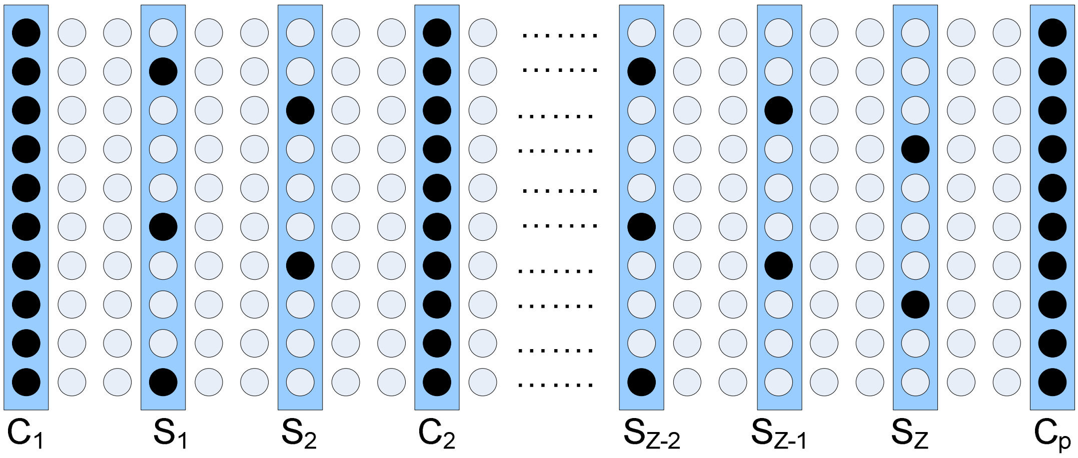


Figure A.11.10: Graphical representation of the DVB-SH OFDM pilot distribution in time (vertical axis) and frequency (horizontal axis)

In general, the parameters M_1 and M_2 can be independently defined for each filter. Furthermore, when M_2 is set to zero the filter is behaving as a predictor instead as an interpolator.

This procedure is applied to each continual pilot tone (C_i) and repeated for each received OFDM symbol (n).

Step 2: Estimation error calculation

The estimation error for the filter updating is then derived from the estimates performed in Step 1 (A.2) and the knowledge of the channel thanks to the pilot symbol transmission, according to the following equations for L=1,2,3:

$$E_L(n,C_i) = H(n,C_i) - H_L(n,C_i) \tag{A.3}$$

The error calculation is performed on each continual pilot tone (C_i) and repeated for each received OFDM symbol (n).

Step 3: Update of the time domain interpolation filter coefficients

The interpolator filter coefficients are then updated using the errors calculated in Step 2. The following summation are over all continual pilot tones (1..P) and repeated for every m ranging in [$-M_1$+1:M_2] for L=1,2,3:

$$W_L(n,m) = W_L(n-1,m) + \mu \sum_{i=1}^{P} E_L^{*}(n,C_i)\,H(n-L+4m,C_i) \tag{A.4}$$

where μ represents the time interpolator filter LMS adaptation step.

a) Scattered pilot interpolation

The filter coefficients calculated exploiting the subcarriers containing continual pilot symbols are then employed to estimate the channel in non-pilot symbols position for the subcarriers containing scattered pilot symbols.

The regular distribution of the scattered pilots ensures pilots to be transmitted every 4 OFDM symbols (then the factor 4 in Eq.A.2). The distance between a non-pilot carrier-symbol and the previously transmitted pilot is not higher then 3 OFDM symbols. According to which OFDM symbol is transmitting the pilot (n-1, n-2, n-3), the proper filter coefficient set W_1, W_2 or W_3 should be used for the interpolation. More specifically the interpolated channel estimate for symbol n-L belonging to the scattered pilot subcarrier S_i is computed as:

$$H(n,S_i) = \sum_{m=-M_1+1}^{M_2} W_L(n,m)\,H(n-L+4m,S_i) \tag{A.5}$$

The channel estimation $H\left(n,S_i\right)$ is based on the knowledge of the channel at $H\left(n-L+4m,S_i\right)$ with L=1, 2, 3 where scattered pilot symbols are transmitted. This clarifies the necessity of three distinct sets of filter coefficients. This procedure is applied to each scattered-pilot subcarrier (S_i) and repeated for each received OFDM symbol (n).

Frequency domain interpolation

The interpolation in the frequency domain is then performed to derive the channel estimation for the OFDM subcarriers not containing any pilot symbol. The channel transfer function estimation is based on the knowledge of the channel at the continual and scattered subcarriers derived during the previous time-domain interpolation phase. These stones are highlighted in figure A.11.10. It clearly appears that two subcarriers are not carrying at all pilot symbols.

The frequency-domain interpolator filter is designed to be $3(J_1+J_2)$ subcarrier long, where $3*J_1$ and $3*J_2$ represent the number of exploited OFDM subcarriers before and after the subcarrier to be estimated.

b) Channel interpolation for pilot-less subcarriers

Following an approach similar to the one applied for the time-domain interpolation, two distinct interpolation filters V_I with I=1, 2 are defined in the frequency-domain:

$$H\left(n,k\right)=\sum_{m\,=\,-J_1+1}^{J_2} V_I\left(n-1,j\right)H\left(n,k-I+3j\right),\ \text{for } k=3K+I \tag{A.6}$$

where K is an integer and $V_I\,(n\text{-}1,j)$ represents the j -th coefficient of the L-th filter (L=1,2), updated during the previous *(n-1)*-th OFDM symbol.

Each filter performs the channel estimation for the k-th subcarrier. The indices relation $k=3K+I$ ensures that only the not-continual and not-scattered subcarriers are considered. The channel estimation $H\left(n,k\right)$ is based on the knowledge of the channel cells $H\left(n,k-I+3j\right)$ *I=1, 2,* calculated during the time-domain interpolation.

c) Filter Coefficients Update

Step 1: Estimation error calculation

In the absence of pilot symbols, the error signals for the coefficient update are calculated based on a decision-direct approach. After the computation of the channel estimate $H(n,k)$, the estimated transmitted symbol $X\left(n,k\right)$ can be derived as:

$$X\left(n,k\right)=H_D\left\{\frac{Y\left(n,k\right)}{H\left(n,k\right)}\right\} \tag{A.7}$$

being $H_D\{\ \cdot\ \}$ the function representing the complex symbol hard decision. For QSPK modulation the function $H_D\{\ \cdot\ \}$ is simply given by $H_D\left\{z_n\right\}=\frac{1}{\sqrt{2}}\left\{\text{sign}\left[\text{Re}(z_n)\right]+j\ \text{sign}\left[\text{Im}(z_n)\right]\right\}$.

Assuming the symbol hard decision reliable, it can be used as a pilot symbol so that the channel transfer function can be re-calculated according to:

$$H\left(n,k\right)=\frac{Y\left(n,k\right)}{X\left(n,k\right)} \tag{A.8}$$

The Estimation Error is defined as:

$$D\left(n,k\right)=H\left(n,k\right)-H\left(n,k\right) \tag{A.9}$$

Step 2: Update of the Frequency Domain Interpolation Filter Coefficients

The interpolator filter coefficients are then updated using the errors calculated in Step 1. The following summations are over all possible tones (k) to be updated (not-scattered and not-continual pilot subcarriers) i.e. for k = 3K+I, I=1, 2:

$$V_I(n,j) = V_I(n-1,j) + \gamma \sum D^*(n,k) H(n,k-L+3j) \quad \text{(A.10)}$$

where γ represents the frequency interpolator filter LMS adaptation step.

NOTE: The algorithms described here are inferred from the descriptions given in US patent US 2006/0269016 A1, Nov. 2006.

A.11.2.2 OFDM Reference Demodulator Simulation Results

A.11.2.2.1 Terrestrial Channel

First the OFDM demodulator has been tested over the TU6 terrestrial reference channel, as described in TS 145 005 [29]. A fading Doppler spread range spanning from f_d = 50 Hz up to f_d = 400 Hz (roughly equivalent to a mobile speed from v = 25 kmph up to v = 200 kmph at S-band) and an SNR=-2,0 dB have been used. Demodulator channel estimation simulation results are plotted in figures A.11.11 and A.11.12 where the OFDM subcarrier estimates are serialized in time (the frequency domain is read first). Figure A.11.11 represents the signal amplitude estimation (orange curve) versus the real one (black curve) as a function of the received symbols. Figure A.11.12 shows the correlation between large carrier phase estimation errors and deep channel fading conditions. These large phase errors have no practical impact on the end results. Turbo decoder will be anyway in error due to the very low SNR. Figures A.11.14 to A.11.16 confirm this analysis.

Simulated OFDM reference demodulator channel estimation for TU6 with fading Doppler spread fd = 50 Hz, SNR = -2,0 dB.

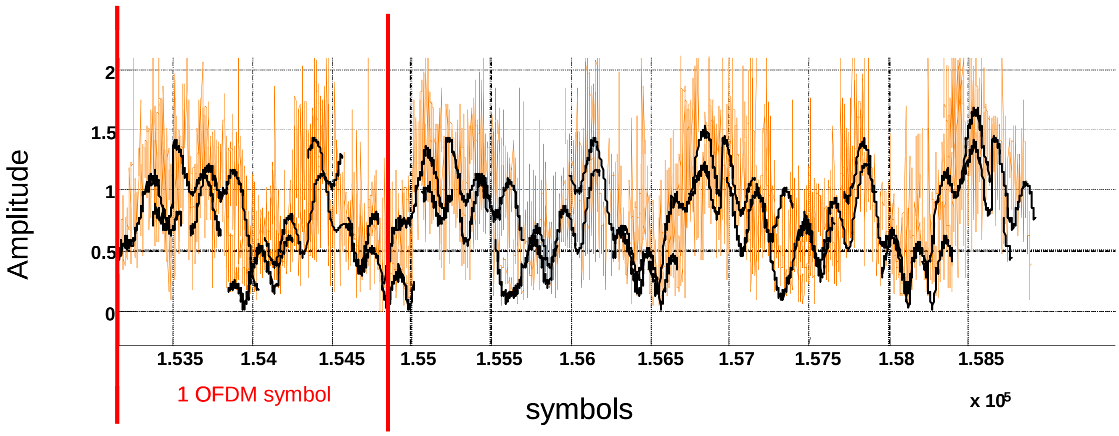


Figure A.11.11: Channel amplitude (black real, estimate orange)

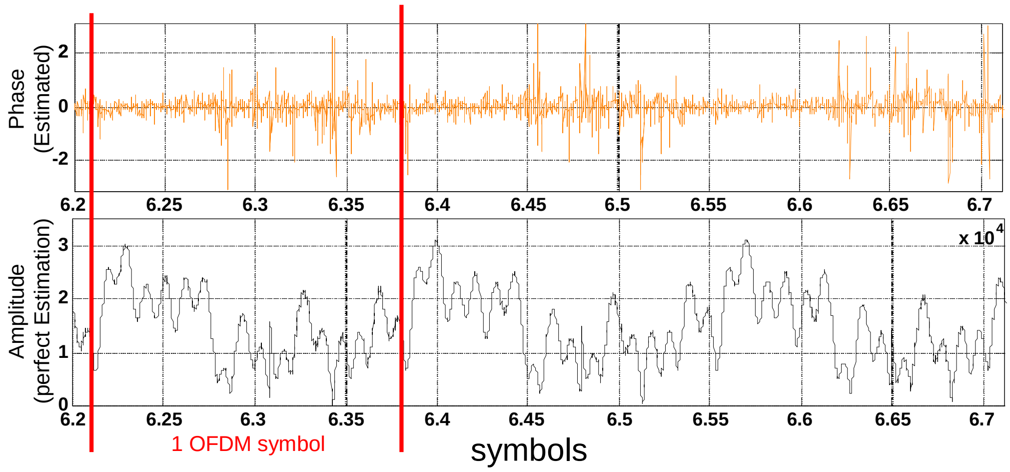


Figure A.11.12: Fading channel amplitude (black) and estimated carrier phase error (orange)

Another way to analyse the adaptive channel estimation algorithm performance is to look at its capability to estimate the fading bandwidth (Doppler spread). Figure A.11.13 compares the simulated TU6 fading process bandwidth (blue curve) with the reference demodulator fading bandwidth estimate (red curve). It is remarked the good fading bandwidth estimation performed by the selected LMS adaptive algorithm.

Example of filter adaptation at a continual-pilot-tone position.

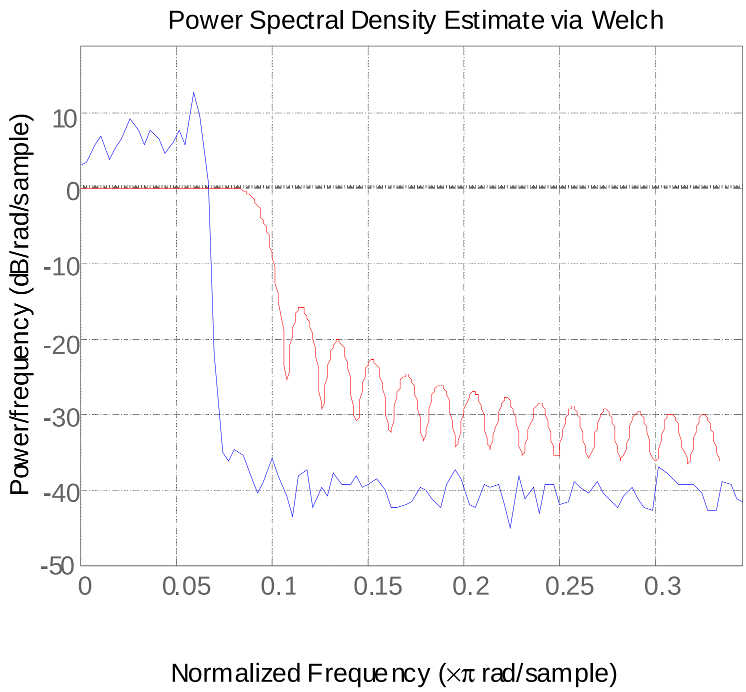


Figure A.11.13: Channel response of LMS-filter (red plot) to a 50 Hz Doppler TU6 channel (blue plot)

A more comprehensive evaluation of the OFDM reference demodulator channel estimation impact over terrestrial channels has been performed simulating the BER performance versus E_s / N_0 in the TU6 channel. Figures A.11.14 to A.11.16 and figures A.11.17 to A.11.19 show the corresponding results for a short (±200 ms) PL interleaving configuration at different Doppler spread values. AWGN reference performances with ideal channel estimation are also reported for convenience. Figure A.11.14 (A.11.17) and figure A.11.15 (A.11.18) show results with perfect and real channel estimation respectively. Figure A.11.16 (A.11.19) show the dependency of demodulator losses to the fading Doppler frequency when the target BER is set to 10^{-5}. The losses of the an ideal demodulator w.r.t. AWGN reference performance (blue curve) are decreasing with increasing Doppler frequency as the PL interleaver is increasingly able to decorrelate the fading. Instead losses are increasing with the Doppler spread in case of real channel estimation as the channel estimation errors are growing with the user (and channel) speed (dynamic). Finally the black curve in figure A.11.12 shows the implementation losses of the reference demodulator compared to ideal channel estimation. In the QPSK-1/3 case, the channel estimation losses are below 1 dB for f_d = 50 Hz and goes up to about 4 dB for f_d = 300 Hz. In the QPSK-1/2 case, the losses even lower at low Doppler frequency, below 0,5 dB for f_d = 50 Hz, but remarkably higher, about 15 dB, for f_d = 300. These results are considered representative of a typical OFDM demodulator.

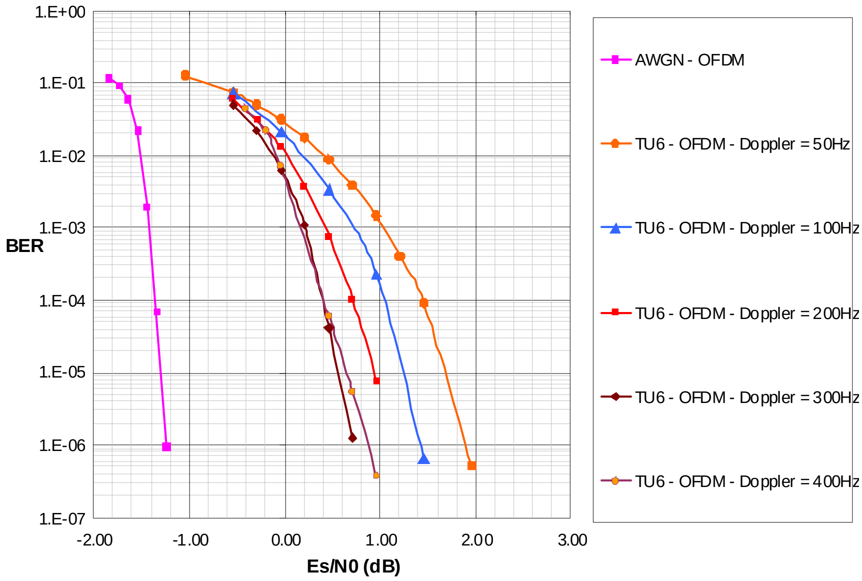


Figure A.11.14: Simulated BER OFDM reference demodulator as a function of Es/N$_0$, for QPSK-1/3 in TU6 channel with several Doppler frequencies: perfect channel estimation

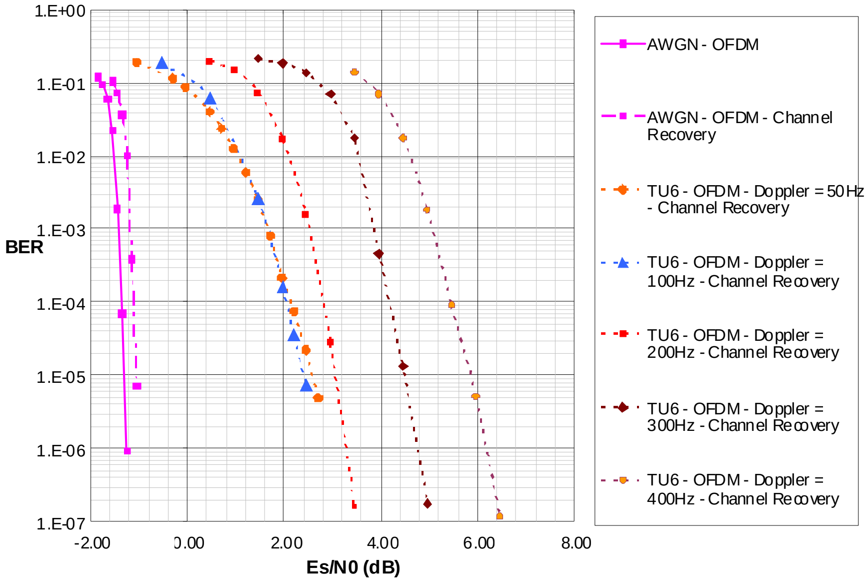


Figure A.11.15: Simulated BER OFDM reference demodulator as a function of Es/N$_0$, for QPSK-1/3 in TU6 channel with several Doppler frequencies: real channel estimation

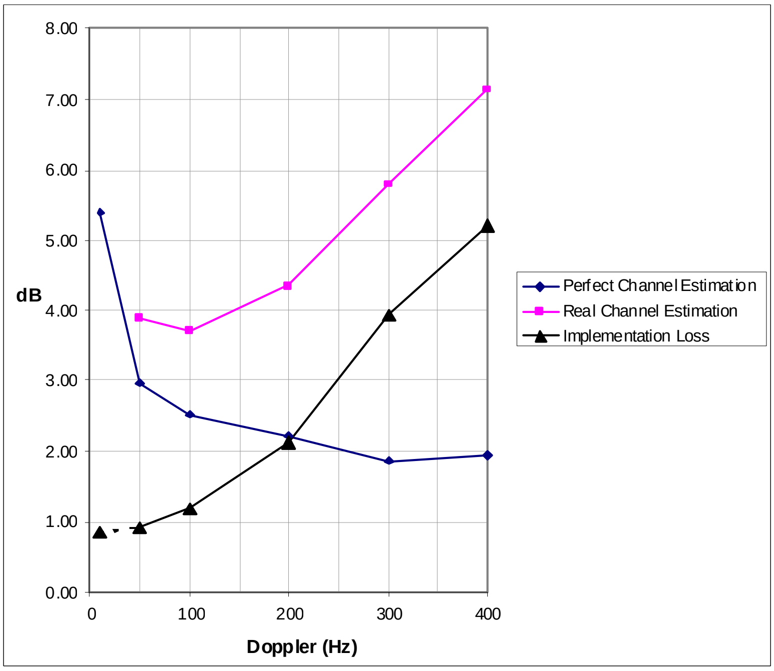


Figure A.11.16: Losses w.r.t. AWGN reference (perfect and real channel estimation) and Implementation losses of a real w.r.t. perfect channel estimation (at BER = 10^{-5})

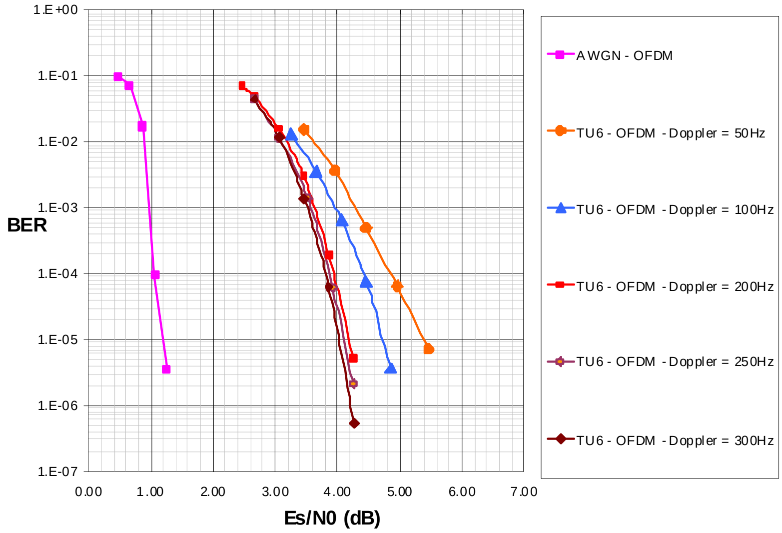


Figure A.11.17: Simulated BER OFDM reference demodulator as a function of Es/N$_0$, for QPSK rate 1/2 in TU6 channel with several Doppler frequencies: perfect channel estimation

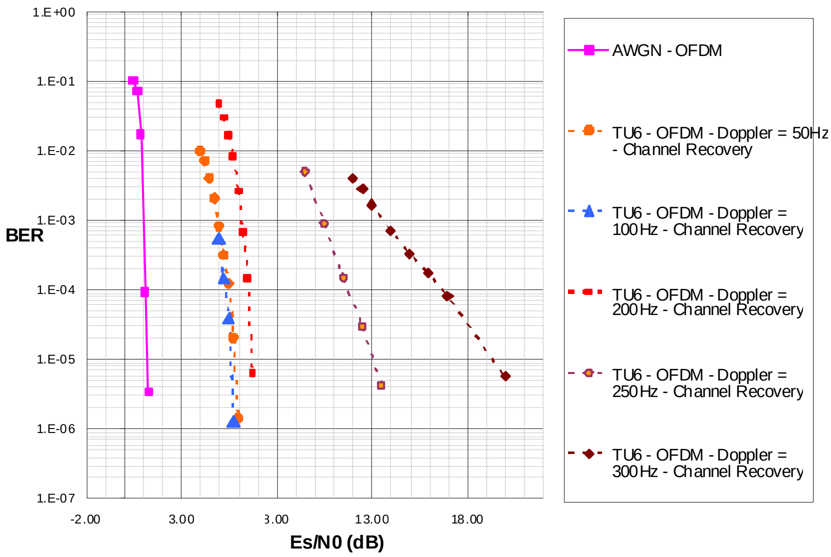


Figure A.11.18: Simulated BER OFDM reference demodulator as a function of Es/N$_0$, for QPSK rate 1/2 in TU6 channel with several Doppler frequencies: real channel estimation

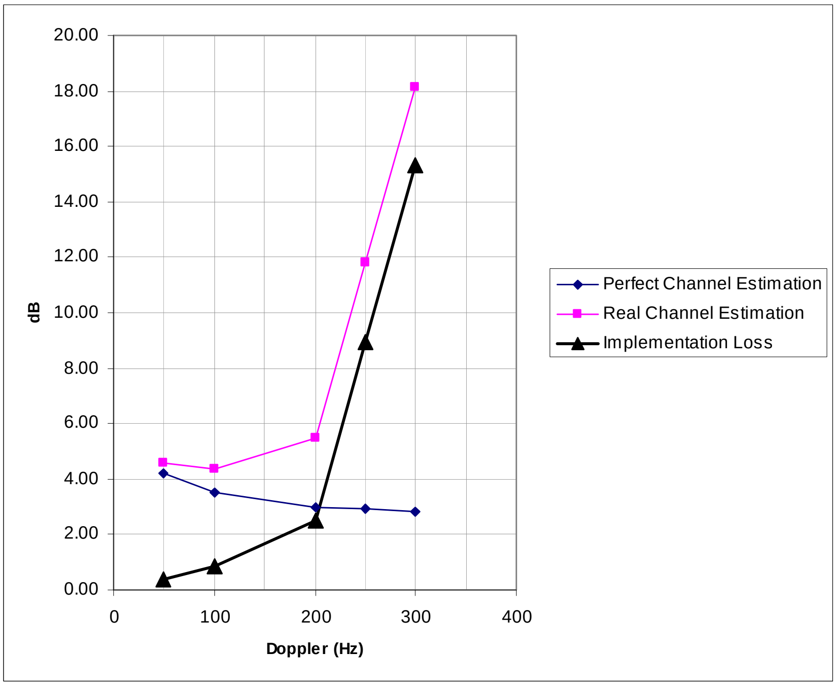


Figure A.11.19: Losses w.r.t. AWGN reference (Perfect and Real channel estimation) and Implementation losses of a real w.r.t. perfect channel estimation (at BER = 10^{-5})

A.11.2.2.2 Satellite Channel

Another set of channel estimation results have been obtained for a satellite flat Ricean fading channel with Rice factor K = 5 dB, mobile speed of v = 50 kmph and SNR = -2,0 dB (the same algorithm with the same parameters as above is used here). As described in clause A.10, the reference demodulator algorithm are the same for the terrestrial and satellite case. In case of satellite only operations, being the fading channel typically flat, there is no need for frequency interpolation of the channel estimates. No attempt has been done to optimize the channel estimation for the satellite-only type of operations.

The reference demodulator channel estimation results are plotted in figures A.11.20 to A.11.21: BER performances versus E_s/N_0 are plotted in figure A.11.22 for both perfect and real channel estimation. The impact of the proposed reference demodulator amount to 0,7 dB with Ricean channel and a mobile speed of 100 kmph compared to ideal channel estimation.

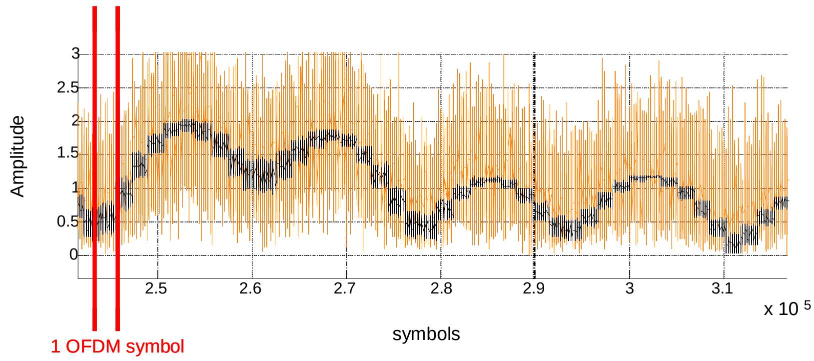


Figure A.11.20: Simulated OFDM reference demodulator channel amplitude estimation (orange line) versus real amplitude (black line) for Ricean, mobile speed 50 kmph, SNR = -2,0 dB

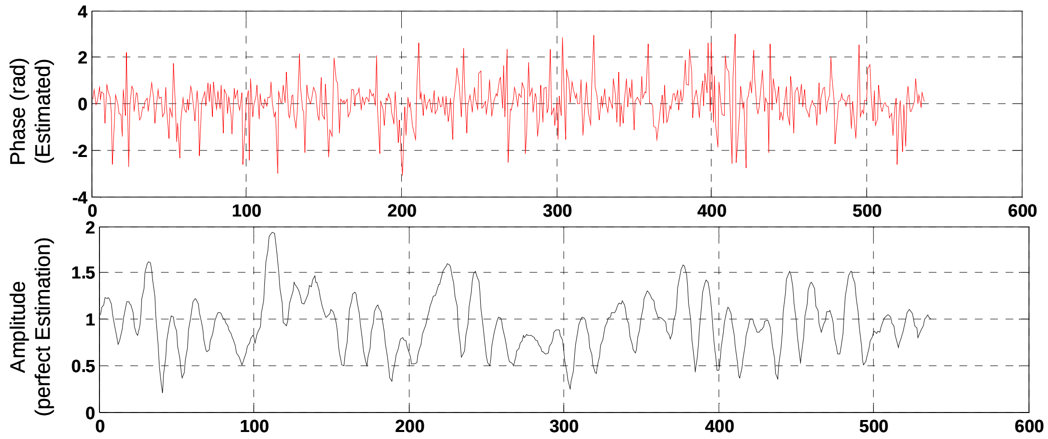


Figure A.11.21: Simulated OFDM reference demodulator channel estimation for Ricean, mobile speed 50 kmph, SNR = -2,0 dB: zoom of amplitude, zoom of fading channel amplitude and estimated carrier phase error relation and zoom of fading channel amplitude and estimated carrier phase error relation for a single FFT-tone

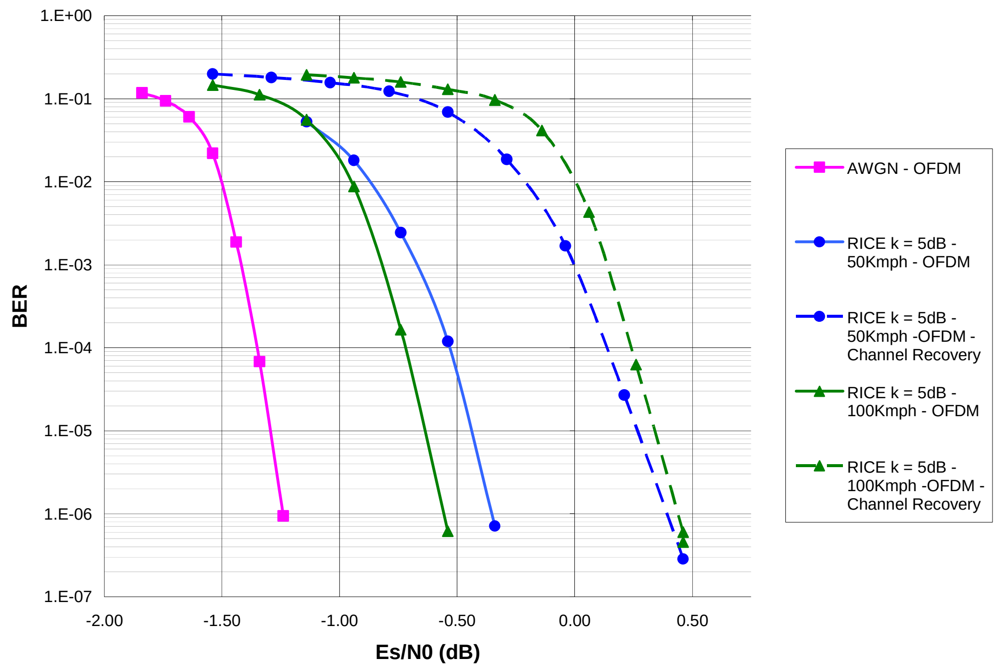


Figure A.11.22: Simulated BER OFDM reference demodulator as a function of E_S/N_0, for Ricean channel with mobile speed 50 kmph and 100 kmph: perfect channel estimation versus real channel estimation

A.11.3 TDM Signalling Channel Performance Results

In this clause we report some results obtained by simulation on the DVB-SH TDM signalling channel performance. At the terminal switch on it is important to receive the SH network key PL configuration parameters through correct signalling channel detection.

The performance have been derived under the following conditions:

- dynamic 3-state LMS-ITS, LMS-SU channel;
- 400 000 signalling field simulated;
- distance between two consecutive signalling fields (SH frame payload) is taken into account;

- accumulation based on:
    - equal gain combining;
    - maximal ratio combining.

The computed performances are:

- signalling Block Word Error Rate;
- probability of successful detection versus number of received signalling fields.

Figure A.11.23 shows the simulated signalling field Word Error Rate as a function of the soft accumulated signalling fields for ITS LMS channel with a mobile speed of 50 kmph and E_s / N_0 = -3,5 dB. Two way of performing soft combining have been considered the maximal ratio (MRC) and the equal gain (EGC) ones. As expected, the MRC provides superior performance compared to the ECG.

Figure A.11.24 shows the probability of correct TDM signalling field detection as a function of the number of received signalling fields for the 3-state LMS-ITS mobile channel v = 50 kmph, E_s / N_0 = -3,5 dB. The results is of course dependent on how many signalling fields are softly combined, but in the worst case of no field combining after 30 signalling fields the signalling information is successfully detected with 99 % probability. With 5 fields soft combined 15 fields are sufficient to have the same detection probability.

Figure A.11.25 reports the probability of correct TDM signalling field detection as a function of the number of received signalling fields for the 3-state LMS-SU mobile channel v = 3 kmph, E_s / N_0 = -3,5 dB. The results is of course dependent on how many signalling fields are softly combined, but in the worst case of no field combining after 30 signalling fields the signalling information is successfully detected with 99 % probability. With 5 fields soft combined 10 fields are sufficient to have the same detection probability.

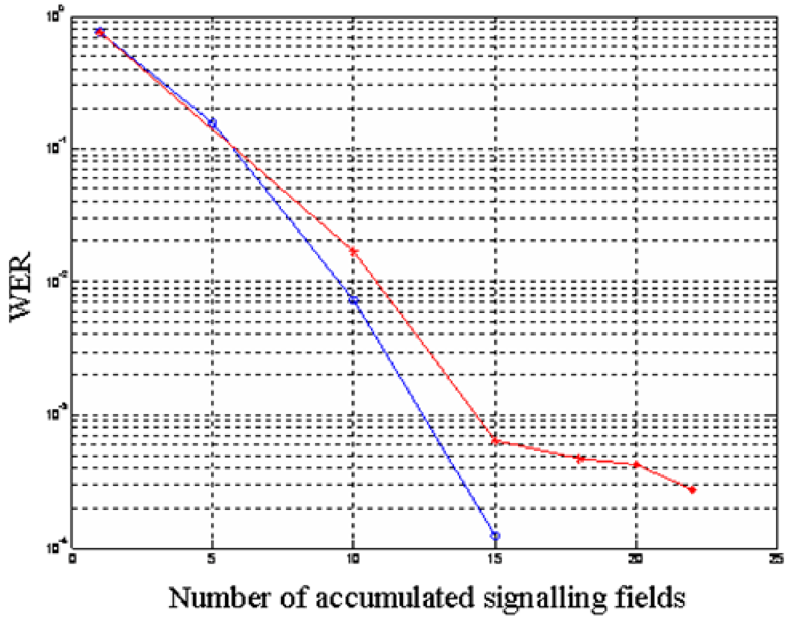


Figure A.11.23: TDM signalling Word Error Rate as a function of the number of soft accumulated signalling fields for the 3-state LMS-ITS mobile channel *v* = 50 kmph, E_s / N_0 = -3,5 dB

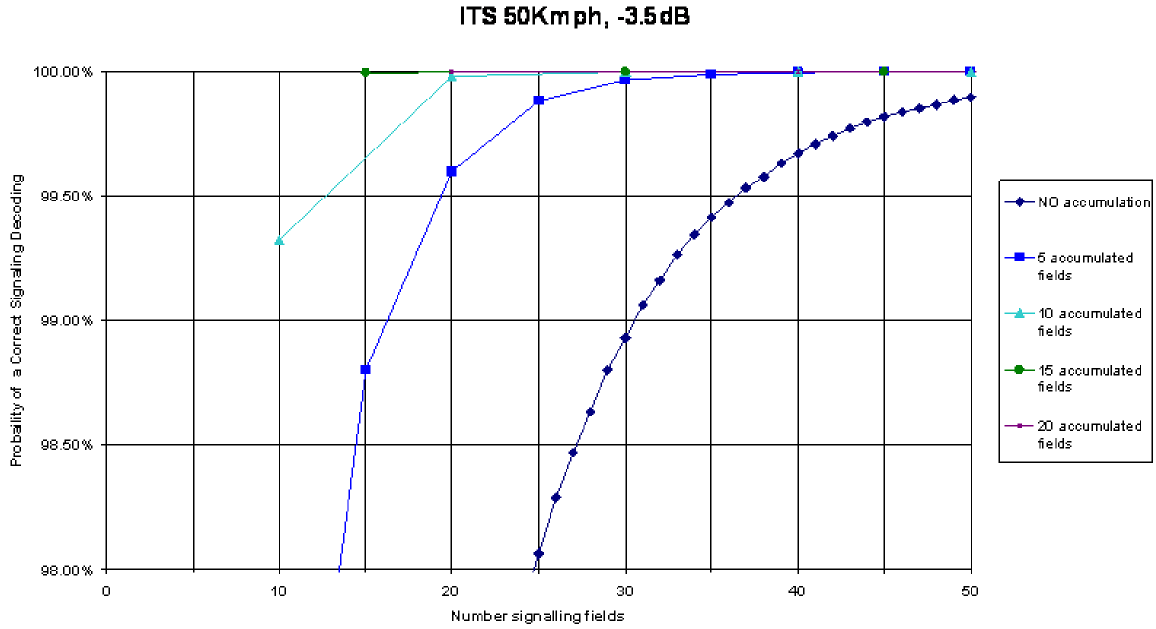


Figure A.11.24: Probability of correct TDM signalling field detection as a function of the number of received signalling fields for the 3-state LMS-ITS mobile channel *v* = 50 kmph, E_s / N_0 = -3,5 dB

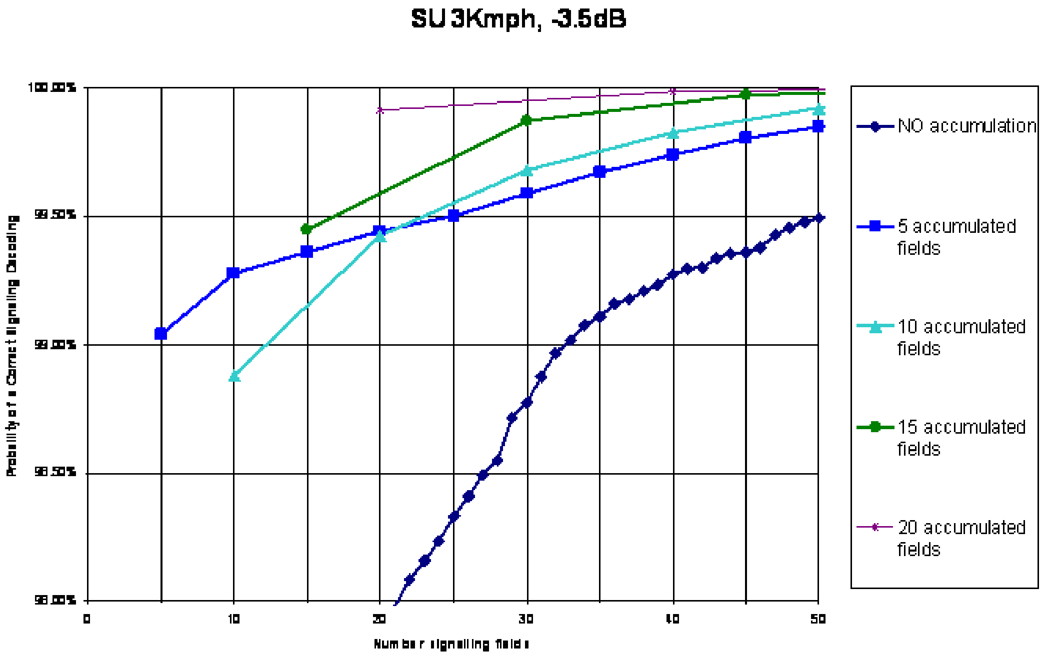


Figure A.11.25: Probability of correct TDM signalling field detection as a function of the number of received signalling fields for the 3-state LMS-SU mobile channel *v* = 3 kmph, E_s / N_0 = -3,5 dB

A.12 Simulation results

The simulation results reported here are first based on the system configurations described above, referred to as the "Reference Cases" and briefly recalled below. They allow comparison of different physical configurations **under the same high level technical parameter settings (spectrum efficiency, satellite and terminal characteristics or interleaver length), irrespective of the receiver complexity trade-off, such as memory size**. In addition, supplementary cases have also been performed, referred to as "Sensitivity Analysis", that give complementary information on respective capabilities of different physical configurations, when specific constraints are relaxed/changed. Because the constraints are not relaxed/changed in the same way, direct comparison of these additional results are not straightforward.

Simulated environments

The characteristics of the main terminal Categories are:

Table A.12.1

	Handheld	**Handheld**	**Portable**	**Vehicular**
	Category 3	category 2b	category 2a	category 1
Antenna polarization	L or C	L or C	L or C	C
G/T (dB/K)	**-32,1**	**-29,1**	**-24,9**	**-21,0**

Service delivery environments/use cases/terminal categories lead to a large combination of simulations targets. The simulations reported here are restricted to the following targets:

1) satellite coverage, Category 1 terminals, LMS-SU model (Suburban);

2) satellite coverage, Category 1 terminals, LMS-ITS model (Rural and forested);

3) satellite coverage, Category 2b terminals, LMS-SU (Suburban) model;

4) terrestrial coverage, any terminal category, TU6 propagation model (fast fading).

Target QoS criteria for the simulations

Two QoS parameters have been computed:

5) for a global approach to QoS the ESR5 fulfilment is used, according to definition in clause A.3.8.5;

6) for easy reference and reuse of terrestrial network planning efforts conducted on other broadcasting services for terrestrial coverage the FER/MFER5 are also used.

Simulation parameters

The following parameters have been defined to identify uniquely each case:

- waveform configuration, covering all main settings of the DVB-SH waveform:
  - physical layer parameters:
    - transmission Mode: OFDM or TDM for LMS environment, and OFDM only for TU6 environment;
    - physical layer Modulation: QPSK and 16QAM for OFDM, QPSK and 8PSK for TDM;
    - physical layer FEC code rate;
    - physical layer Interleaver Profile: Short (S), 200 ms long and Uniform (U) or Uniform Late (UL) (see clause A.10) with a reference total duration in the range of 10 s;
    - OFDM simulations have been performed using a Guard Interval of 1/4;
  - link Layer parameters (when applicable):
    - link Layer code rate;

    - link Layer Length of the Interleaving, defined as the B+S value (see clauses 6 and A.6 for details) with a reference total duration in the range of 10 s (clause A.3.1 for the delay breakdown);

- channel propagation types as described in clause A.7;
- state Machine Activation (clause A.5), simulating demodulator synchronization behaviour in LMS channel;
- two reference terminal speeds: 3 kmph for Handheld, and 50 kmph for vehicular;
- Carrier-to-Interference ratio (C/I) and Carrier-to-Noise ratio (C/N) according to the link budget calculations (clause A.9);
- two types of terminals are considered for LMS simulations, each associated with a specific satellite EIRP:
    - category 1 terminals, with circular polarized antenna, receiving signal from a 63 dBW EIRP satellite;
    - category 2b terminals, with linear polarized antenna, receiving from a 68 dBW EIRP satellite;
- TU6 performance is simulated independently of the terminal category, as it characterizes the required C/N in terrestrial fast fading environment;
- for LMS environments, class 1 is always used together with MPE-IFEC.

The complete list of Reference Cases is given in clause A.10 (at Physical Layer level) for both terrestrial and satellite applications.

Simulation results presentation

The following parts of this clause are organized as follows:

- summary of the results for LMS is presented in clause A.12;
- presentation of the simulated performance in LMS environment, including vehicular reception in LMS-Suburban, vehicular reception in LMS-ITS, and handheld reception in LMS Suburban is reported in clause A.12.2;
- presentation of the simulated performance in TU6 environment is reported in clause A.12.3.

A.12.1 Summary of the Results for LMS channel

The results for the Reference Cases show the following main performance differences between the configurations:

1) Physical-Layer (class 2) vs Link-Layer (class 1+ MPE-IFEC) protection

The reference configuration results show that protection provided exclusively at physical layer (class 2 receivers) performs better than an "equivalent" combination of protections at Physical Layer and Link Layer (class 1 receivers with MPE-IFEC), under the same constraints.

More precisely:

- when the constraint of handheld and portable terminals is removed, keeping only vehicular terminals (Category 1):
    - in LMS-SU, ESR5 fulfillment is above 99 % for class 2, whereas for class 1 it is from 88 % to 98 %;
    - in LMS-ITS, ESR5 fulfillment is from 86 % to 100 % for class 2, whereas class 1 can never achieve 90 %;
    - class 2 often exceeds the targeted performance, even for the lowest satellite EIRP considered, both in LMS-ITS and LMS-SU. Therefore, C/N reduction for class 2 is addressed in the "Sensitivity Analysis" clause below;
    - class 1 with MPE-IFEC, most likely require interleavers longer than 10 s (see clause 10 of the guidelines). Therefore, ESR5 fulfillment improvement versus increased interleaver length are addressed in the "Sensitivity Analysis" clause below;

- for what concerns handheld reception (Category 2b), in LMS-SU and at the highest satellite EIRP considered, the results indicate that ESR5 fulfillment is below target for all physical configurations. However, class 2 is able to reach 88 % while class 1+ MPE-IFEC reaches only 62 %.

2) **Uniform Late (UL) and Uniform (U) long physical layer interleavers**

The Uniform Late interleaver profile is specifically designed to allow fast zapping in good reception condition. Also, it allows coexistence of class 1 and class 2 in the same network. However, these advantages entail some performance penalty. For the targeted 90 % of ESR5 fulfillment, the Uniform Late interleaver losses with respect to Uniform Long interleaver are:

- up to 4 dB in the LMS-SU, for Category 1 terminal at 50 kmph;
- less than 2 dB in the LMS-ITS, for Category 1 terminal at 50 kmph;
- about 1,5 dB in LMS-SU, for Category 2b terminal at 3 kmph.

3) **Usage of Higher Order Modulation with Lower coding rate**

When a constant overall spectral efficiency is considered, several configurations (*Modulation* / PHY-*CodeRate* / IFEC-*CodeRate*) can be envisaged.

With reference to the class 2, the results for the reference scenarios (cases 22/23-25/26 and 35/36-38/39) are inconclusive since these cases show 100 % fulfil of the ESR5.

The results for class 1 show a slight advantage for the Lower Order Modulation with a less robust FEC protection, in particular for the LMS-ITS (cases 14 vs. 18 and 21 vs. 24). For the analyzed configurations, the results show that the Higher Order Modulation with Lower Coding Rate obtains equivalent performances only when operating in high ESR5 fulfillment regions.

A.12.2 Detailed results in LMS environments: Reference Cases and Sensitivity analysis

A.12.2.1 Reference Cases results

The results for the Reference Cases and the LMS environments are provided in the 3 tables below, corresponding respectively to the following 3 scenarios:

- table A.12.2, Scenario (a): Category 1 terminal in LMS-SU, speed of 50 kmph and 63 dBW EIRP Satellite;
- table A.12.3, Scenario (b): Category 1 terminal in LMS-ITS, speed of 50 kmph and 63 dBW EIRP Satellite;
- table A.12.4, Scenario (c): Category 2b terminal in LMS-SU, speed of 3 kmph and 68 dBW EIRP Satellite.

The tables give the main parameters of the waveform, in particular the type of Physical Layer interleaver, the code rates at Physical Layer and, when used, the code rates at Link Layer level. The total code rate is the product of these two code rates. The total capacity is the equivalent capacity provided after all FEC decoding and computed at the MPEG2 TS interface. The total interleaving length is about 10 s for all cases.

Table A.12.2: ESR5 fulfilment in Scenario (a)
LMS-SU environment, about 10 s of interleaving,
63 dBW EIRP Satellite, Category 1 terminal, speed = 50 kmph, State Machine = "ON"

ID	Waveform configuration						TOTAL capacity (Mbps)	ESR5 fulfilment
	transmission Mode	Modulation	PHY code-rate	Link code-rate	TOTAL code-rate	PHY Interleaver Profile		
27	OFDM	16QAM	1/4	2/3	1/6	S	2.24	88.20%
28	OFDM	16QAM	1/5	-	1/5	U	2.67	100.00%
29	OFDM	16QAM	1/5	-	1/5	UL	2.67	99.00%
30	OFDM	16QAM	2/7	7/12	1/6	S	2.19	93.30%
31	OFDM	QPSK	1/2	2/3	1/3	S	2.24	89.70%
32	OFDM	QPSK	1/3	-	1/3	U	2.22	100.00%
33	OFDM	QPSK	1/3	-	1/3	UL	2.22	100.00%
34	TDM	8PSK	1/3	2/3	2/9	S	2.60	96.70%
35	TDM	8PSK	2/9	-	2/9	U	2.57	100.00%
36	TDM	8PSK	2/9	-	2/9	UL	2.57	100.00%
37	TDM	QPSK	1/2	2/3	1/3	S	2.60	95.00%
38	TDM	QPSK	1/3	-	1/3	U	2.57	100.00%
39	TDM	QPSK	1/3	-	1/3	UL	2.57	100.00%

Only two configurations (ID=27 and ID=31), corresponding to class 1 and MPE-IFEC, have ESR5 fulfillment slightly below 90 %. When class 2 is used, performance is above 99 %.

Table A.12.3: ESR5 fulfillment in Scenario (b)
LMS-ITS environment, about 10 s of interleaving,
63 dBW EIRP Satellite, Category 1 terminals, speed = 50 kmph, State Machine ="ON"

ID	Waveform configuration						TOTAL capacity (Mbps)	ESR5 fulfilment
	transmission Mode	Modulation	PHY code-rate	Link code-rate	TOTAL code-rate	PHY Interleaver Profile		
14	OFDM	16QAM	1/4	2/3	1/6	S	2.24	6.80%
15	OFDM	16QAM	1/5	-	1/5	U	2.67	95.50%
16	OFDM	16QAM	1/5	-	1/5	UL	2.67	86.00%
17	OFDM	16QAM	2/7	7/12	1/6	S	2.19	6.40%
18	OFDM	QPSK	1/2	2/3	1/3	S	2.24	13.30%
19	OFDM	QPSK	1/3	-	1/3	U	2.22	100.00%
20	OFDM	QPSK	1/3	-	1/3	UL	2.22	99.00%
21	TDM	8PSK	1/3	2/3	2/9	S	2.60	52.80%
22	TDM	8PSK	2/9	-	2/9	U	2.57	100.00%
23	TDM	8PSK	2/9	-	2/9	UL	2.57	100.00%
24	TDM	QPSK	1/2	2/3	1/3	S	2.60	64.70%
25	TDM	QPSK	1/3	-	1/3	U	2.57	100.00%
26	TDM	QPSK	1/3	-	1/3	UL	2.57	100.00%

Most configurations with class 2 have an ESR5 fulfillment above 90 %. Configurations using 16QAM OFDM and code rate 1/5 have slightly lower performance because of a higher spectrum efficiency. Results for class 1 and IFEC show that a 10 s IFEC gives an insufficient protection in LMS-ITS. Results for class 2 show that for most of the cases, performance is above 99 %.

Table A.12.4: ESR5 fulfillment for Scenario (c) LMS-SU environment, with about 10 s of interleaving, 68 dBW EIRP Satellite, Category 2b terminals, speed = 3 kmph, State Machine = "ON"

ID	Waveform configuration						TOTAL capacity (Mbps)	ESR5 fulfilment
	transmission Mode	Modulation	PHY code-rate	Link code-rate	TOTAL code-rate	PHY Interleaver Profile		
72	OFDM	16QAM	1/5	-	1/5	U	2.67	51.00%
73	OFDM	16QAM	1/5	-	1/5	UL	2.67	42.00%
75	OFDM	QPSK	1/3	-	1/3	U	2.22	84.00%
76	OFDM	QPSK	1/3	-	1/3	UL	2.22	75.00%
77	TDM	8PSK	1/3	2/3	2/9	S	2.60	48.90%
78	TDM	8PSK	2/9	-	2/9	U	2.57	87.57%
79	TDM	8PSK	2/9	-	2/9	UL	2.57	84.43%
80	TDM	QPSK	1/2	2/3	1/3	S	2.60	62.10%
81	TDM	QPSK	1/3	-	1/3	U	2.57	87.98%
82	TDM	QPSK	1/3	-	1/3	UL	2.57	86.89%

In all configurations, ESR5 fulfillment is below 90 %. QPSK with PHY coding rate of 1/3 provide performance very close to this target (88 %). Possible improvements are either the use of longer interleavers or the increase of the transmission link margin, through the reduction of the capacity, or the increase of the satellite transmit power:

- Interleaver impact is analyzed for IFEC cases in clause 1.2.2.2 (figure 9).
- Link margin impact is analyzed for Long Physical Interleaver case in clause 1.2.2.2 (figure 10).

To the reference cases reported in the 3 tables above:

NOTE 1: When compared to OFDM QPSK, the Spectrum Efficiency for OFDM-16QAM code rate 1/5 (ID# 15-16, 28-29,72-73), is slightly higher (2,7 Mbps capacity instead of 2,2 Mbps). The rationale of this difference is due to the minimum allowed coding rate, equal to 1/5. This reflects both in an higher bit-rate and in a slight performance degradation.

NOTE 2: The total channel capacity refers to the aggregate bit rate of the available services; target net bit-rate per service is roughly 280 kbps, then, depending on the system configuration, the number of services can be either 8 or 9.

NOTE 3: In general it can be noticed that even given the same TOTAL EFFICIENCY, the system configurations can give slightly different net bit rates. The reason is related to a different Physical Layer configuration (modulation and/or coding rate) which could end up in a slightly different waveform configuration (e.g. CU-padding in the SH-Frame composition).

NOTE 4: The exact C/N for each Scenario is reported in the link-budgets in clause 1.7.6. However, the results reported here already take into account the implementation losses, i.e for typical receivers (clause 10.4.3): 1,1 dB for OFDM QPSK, 0,5 dB for TDM QPSK, 1,5 dB for OFDM 16QAM and 1 dB for TDM 8PSK.

A.12.2.2 Sensitivity Analysis

This clause provides with performances in the LMS environments when different system configurations with respect to the Reference Cases are considered, relaxing or changing some constraints as appropriate.

As noted, the performance of class 1 receiver using MPE-IFEC with interleaving length up to 10 s do not always meet the 90 % ESR5 fulfillment target. Therefore, a sensitivity analysis to the length of the protection period and to the code rate is performed with the aim of finding the optimal IFEC parameters in each scenario, keeping the same reference *C/N*. The results are presented in a graphical format, reporting the minimum required (Link-Layer) interleaver length as a function of the supported aggregated net-bit rate. It is important to notice that, for each graphic, the curves represent the boundary conditions: all the system configurations which belong to the left-bottom part of the plane are viable solutions. Indeed, for a given interleaver length, all the bit-rates up to the point reported in the curve are guaranteed with 90 % of ESR5 fulfillment as minimum.

On the contrary, class 2 receivers in Category 1 terminals generally exceed the 90 % ESR5 fulfillment. Therefore, a parametric analysis of the ESR5 as a function of the C/N is performed for a set of cases when class 2 is considered. The same *C/I* as reported in the reference cases is kept. Influence of the type of long Physical Layer interleaver (Uniform versus Uniform Late) is also addressed.

The reader is reminded to be cautious when comparing in detail the OFDM versus TDM performances from the results presented, due to:

a) the guard interval value chosen for OFDM;

b) the different net bit rates;

c) the different implementation losses;

d) the different noise bandwidth assumption made.

A.12.2.3 Vehicular reception in LMS environments

A.12.2.3.1 Vehicular Reception in LMS-SU and SH-A Waveform (OFDM)

The simulation performances reported in this clause have been obtained in LMS-SU environment and for SH-A waveform with a 63 dBW EIRP satellite (reference cases IDs 27 to 33 in table A.12.2). These cases are considered for vehicular applications; therefore a typical speed of 50 kmph has been analyzed.

Figure A.12.1 shows the sensitivity analysis for class 1 terminals. The supported bit rate for the reference case is indicated by the dashed line. Under these working conditions, it is possible to increase the supported bit rate while satisfying the 90 % ESR5 fulfillment target. This is achieved by increasing the MPE-IFEC code rate and interleaving depth. To be remarked that even assuming ideal channel estimation 16QAM combined with low FEC rate does not bring any advantage compared to QPSK. On the contrary performance and generally slightly worse for 16QAM.

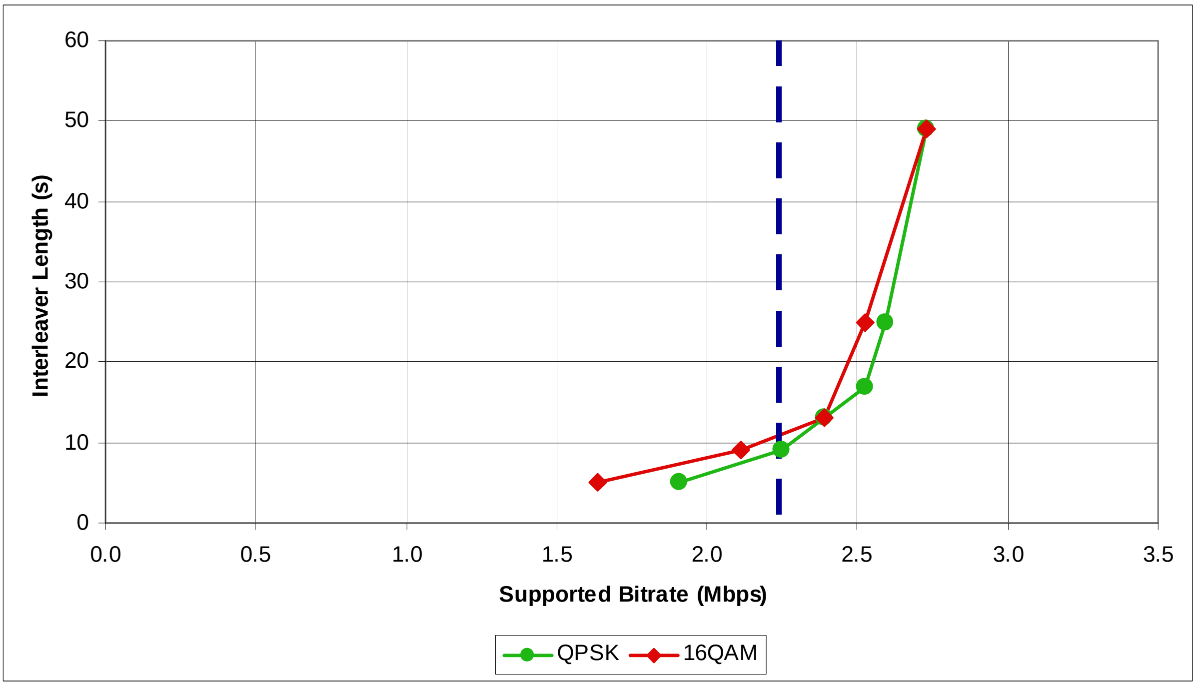


Figure A.12.1: SH-A, class 1 - QPSK 1/2 and 16QAM 1/4 - LMS-SU - 50 kmph - 63 dBW EIRP Satellite Different Link-Layer configuration fulfilling the ESR5(20) criterion at 90 % (corresponding reference cases IDs 27 to 31)

Figure A.12.2 shows the sensitivity analysis for class 2 terminals. The *C/N* setting used for the reference case was 11,2 dB (shown by the vertical dashed line). The required *C/N* to guarantee the 90 % fulfillment of ESR5 is found to be below 5 dB, i.e. 7 dB below the reference configuration *C/N* setting.

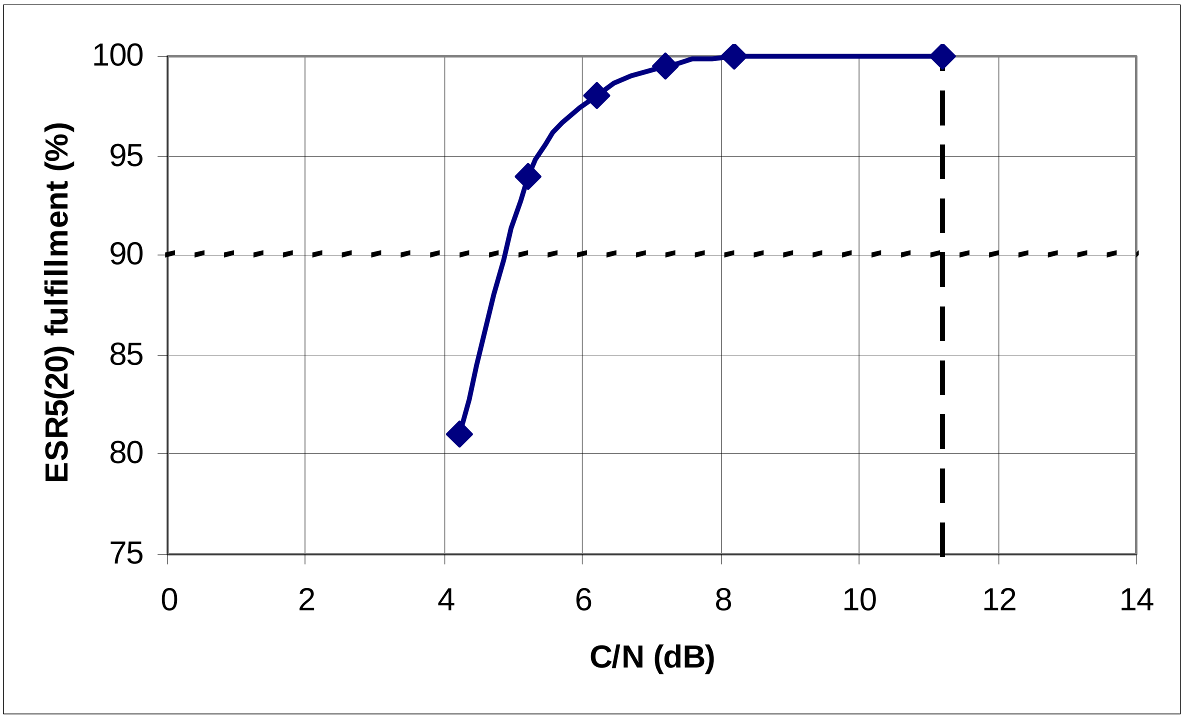


Figure A.12.2: SH-A, class 2: Uniform Long Interleaver Profile - QPSK 1/3 - LMS-SU - 50 kmph Sensitivity Analysis to C/N value for an OFDM modulation (corresponding reference cases IDs 31 and 32)

A.12.2.3.2 Vehicular Reception in a LMS-SU and SH-B Waveform (TDM)

The simulation performances reported in this clause have been obtained in an LMS SU channel and for a SH-B waveform with a 63 dBW EIRP satellite (reference cases IDs 34 to 39 in table A.12.2). These cases are considered for vehicular applications; therefore a typical speed of 50 kmph has been analyzed.

Figure A.12.3 shows the sensitivity analysis for class 1 terminals. The same conclusions as for the corresponding case of SH-A can be drawn.

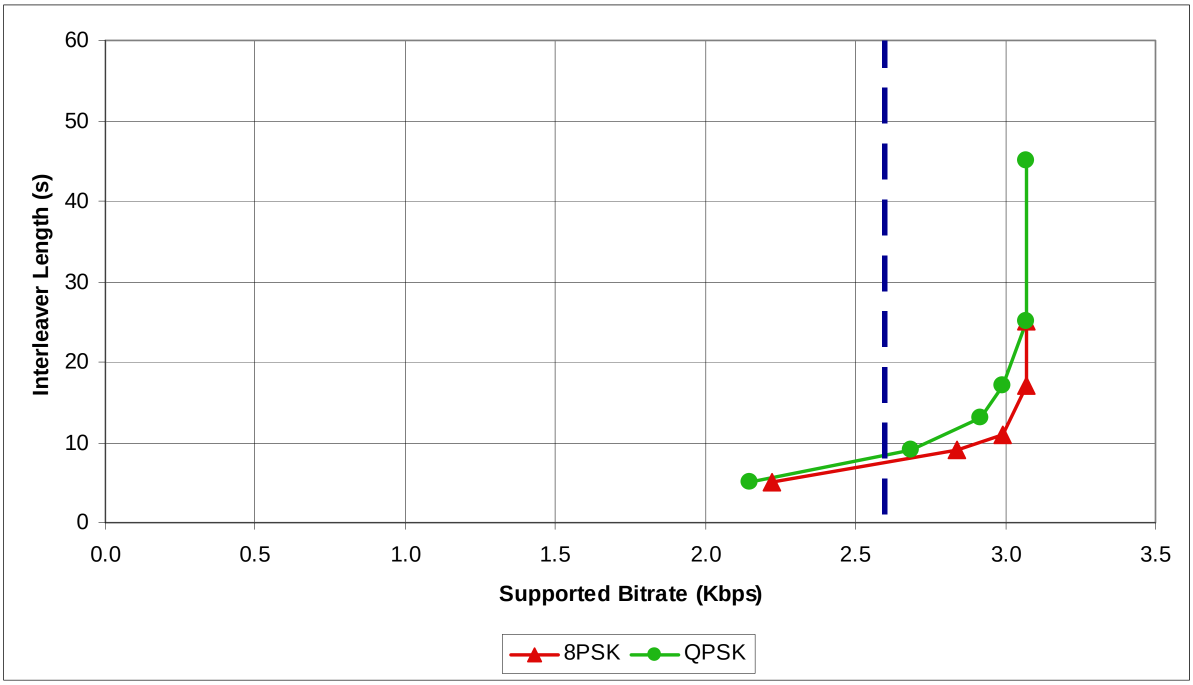


Figure A.12.3: SH-B, class 1: - QPSK ½ and 8PSK1/3 - LMS-SU - 50 kmph - 63 dBW EIRP Satellite Different Link-Layer configuration fulfilling the ESR5(20) criterion at 90 % (corresponding reference cases IDs 34 to 37)

Figure A.12.4 shows the sensitivity analysis for class 2. The C/N setting used for the reference case was 12,3 dB (shown by the vertical dashed line). The required C/N to guarantee the 90 % fulfillment of ESR5 is found to be below 4 dB, i.e. 8 dB below the reference configuration setting.

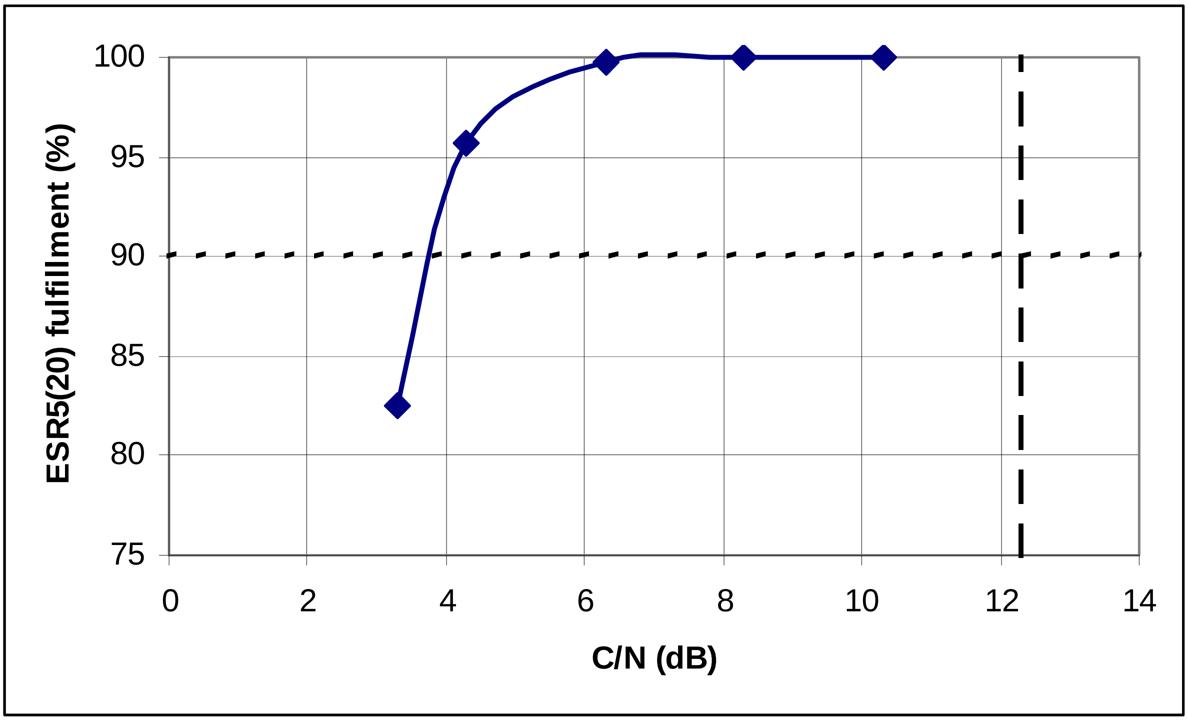


Figure A.12.4: SH-B, class 2: with Uniform Long Interleaver Profile - QPSK 1/3 - LMS-SU - 50 kmph Sensitivity Analysis to C/N value for an OFDM modulation (corresponding reference cases IDs 37 and 38)

A.12.2.3.3 Vehicular Reception in a LMS ITS and DVB SH-A Waveform (OFDM)

The simulation performances reported in this clause have been obtained in LMS-ITS environment and for SH-A waveform with a 63 dBW EIRP satellite (reference cases IDs 14 to 20 in table A.12.3). These cases are considered for vehicular applications; therefore a typical speed of 50 kmph has been analyzed.

As can be seen in figure A.12.5, in LMS-ITS, an SH-A class 1 receiver cannot reach the target capacity at 63 dBW satellite EIRP for the reference case (represented as vertical dashed line).

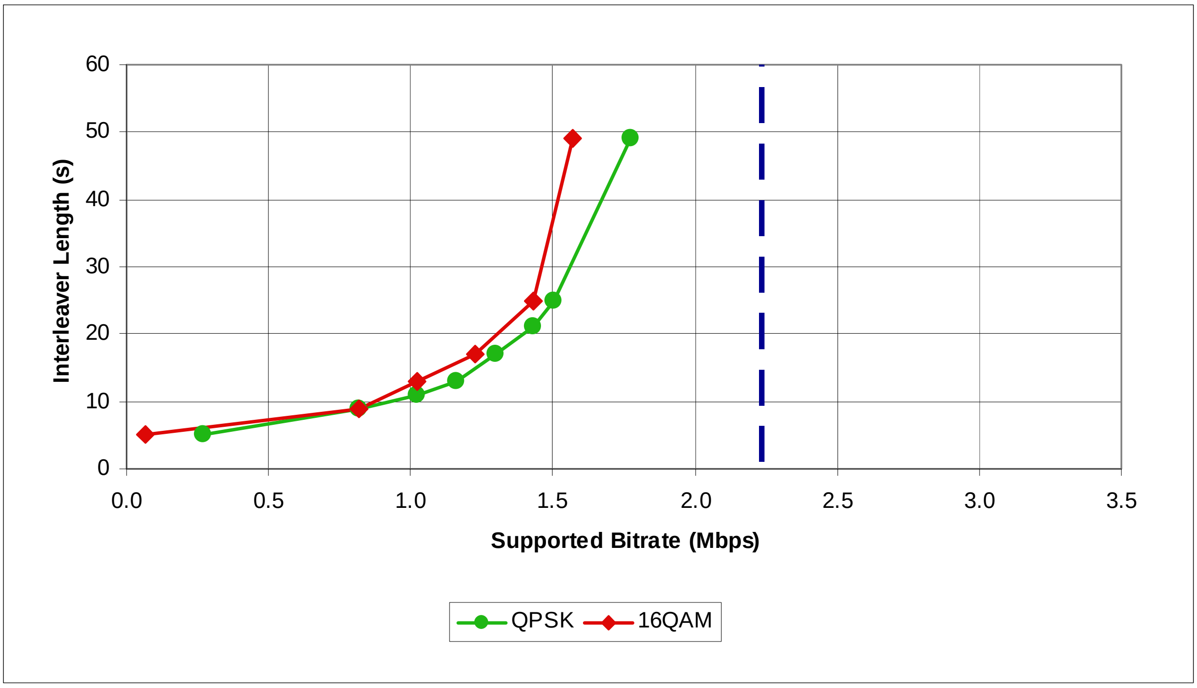


Figure A.12.5: SH-A, class 1: - 16QAM 1/4 and QPSK 1/2- LMS-ITS - 50 kmph - 63 dBW EIRP Satellite Different Link-Layer configuration fulfilling the ESR5(20) criterion at 90 % (corresponding reference cases IDs 14 to 18)

Figure A.12.6 shows the sensitivity analysis for class 2. The C/N setting used for the reference case was 11,2 dB (shown by the vertical dashed line). The required C/N to guarantee the 90 % fulfillment of ESR5 is found to be 9 dB, i.e. 2 dB below the reference configuration setting.

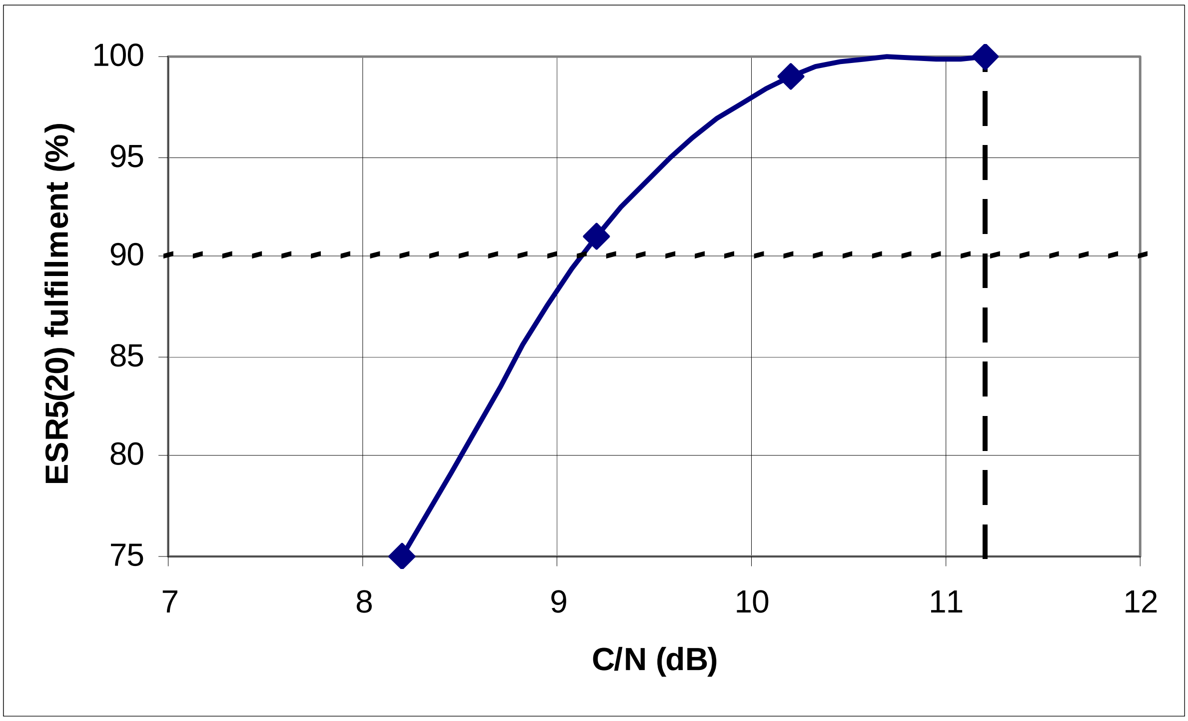


Figure A.12.6: SH-A, class 2 with Uniform Long Interleaver Profile- QPSK 1/3 - LMS-ITS - 50 kmph Sensitivity Analysis to C/N value for a OFDM modulation (corresponding reference cases IDs 18 and 19)

A.12.2.3.4 Vehicular Reception in a LMS ITS and DVB SH-B Waveform (TDM)

The simulation performances reported in this clause have been obtained in an LMS ITS channel and for a SH-B waveform with a 63 dBW EIRP satellite (reference cases IDs 21 to 26 in table A.12.3). These cases are considered characteristic for vehicular applications; in particular a typical speed of 50 kmph has been analyzed.

From figure A.12.7, it can be noticed that a SH-B class 1 receiver is able to guarantee an aggregate traffic slightly lower than the target capacity for the reference case (vertical dashed line in the following figure) up to 2,5 Mbps with roughly 25 s of interleaver length at the reference C/N of 12,8 dB. The performances for the QPSK modulation (with a code rate of 1/2 at physical layer) are close to those for an 8PSK modulation (with a code rate of 1/3 at physical layer). Indeed, under this condition, the advantage for the QPSK for the reference 10 s of interleaver length is limited to about 10 % in terms of bit-rate or interleaver-length.

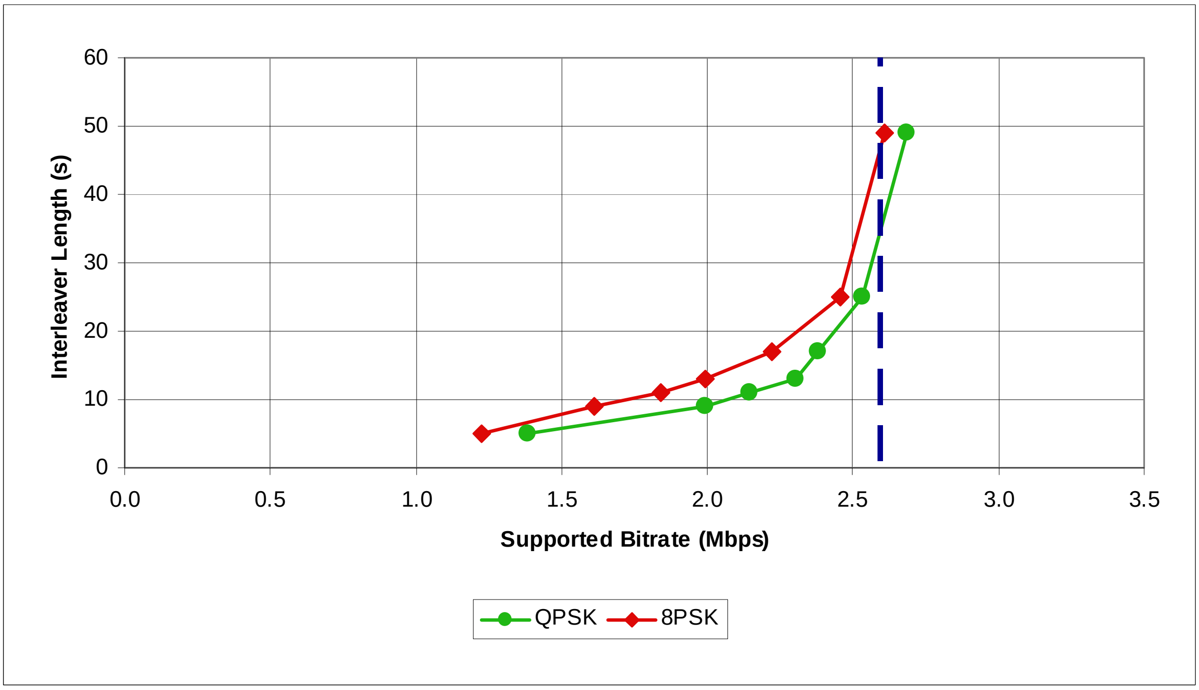


Figure A.12.7: TDM, class 1 - QPSK ½ and 8PSK 1/3 - LMS-ITS - 50 kmph - 63 dBW EIRP Satellite Different Link-Layer configuration fulfilling the ESR5(20) criterion at 90 % (corresponding reference cases IDs 21 to 24)

Figure A.12.8 shows the same sensitivity analysis also for class 2 under the same reference C/N simulation condition of 12,8 dB. Also in this case QPSK modulation performances are slightly better to those for an 8PSK modulation. Under this condition, both QPSK and 8PSK modulations guarantee high throughput for the reference 10 s of interleaver length or equivalently the possibility to reduce the interleaver length to get the target capacity up to 2,5 Mbps.

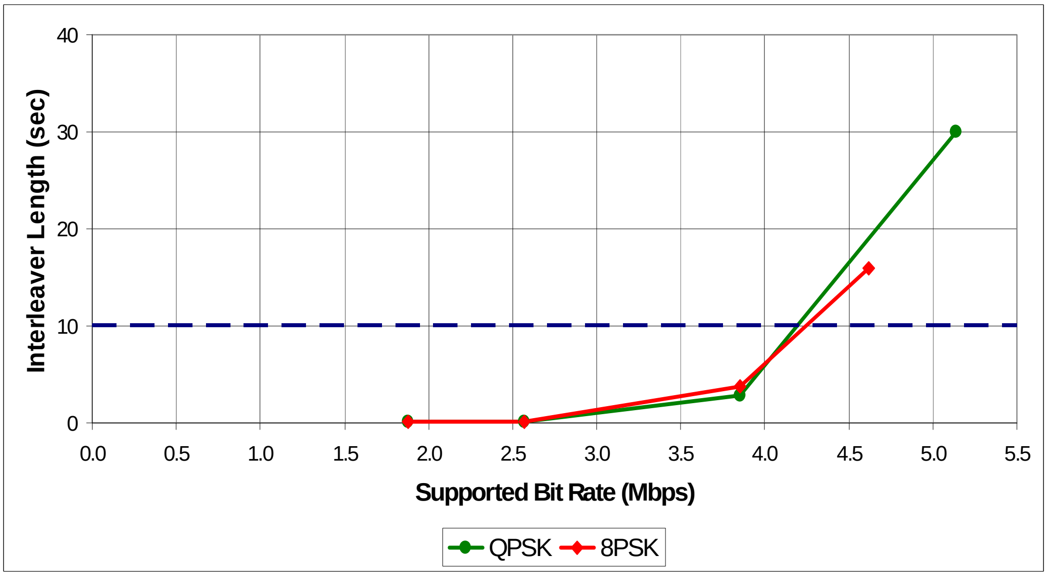


Figure A.12.8: TDM, class 2- QPSK - LMS-ITS - 50 kmph - 63 dBW EIRP Satellite Different Physical-Layer configuration fulfilling the ESR5(20) criterion at 90 % (corresponding reference cases IDs 21 to 24)

Figure A.12.9 shows the sensitivity analysis for class 2. The *C/N* setting used for the reference case was 12,3 dB (shown by the vertical dashed line). The required *C/N* to guarantee the 90 % fulfillment of ESR5 is found to be 8 dB, i.e. 4 dB below the reference configuration setting.

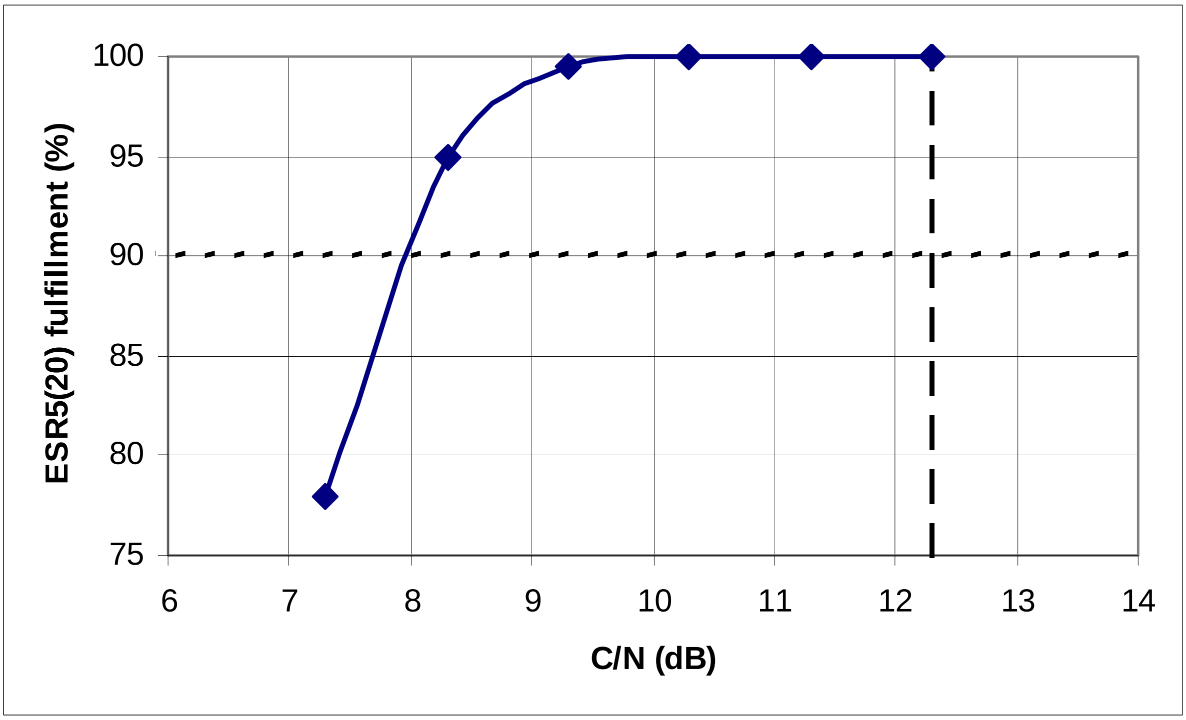


Figure A.12.9: SH-B, class 2 with Uniform Long Interleaver Profile - QPSK - LMS-ITS - 50 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference cases IDs 24 and 25)

A.12.2.3.5 Category 2b reception in LMS-SU at 68 dBW satellite EIRP

The simulation performances reported in this clause have been obtained in an LMS SU channel with a 68 dBW EIRP satellite (reference cases IDs 72 to 82 in table A.12.4). These cases are considered for handheld applications; therefore a typical speed of 3 kmph has been analyzed.

As can be seen in figure A.12.10, in LMS-ITS, an SH-A class 1 receiver cannot reach the target capacity of the reference case (represented as vertical dashed line) at 68 dBW satellite EIRP. QPSK performance are better than 8PSK.

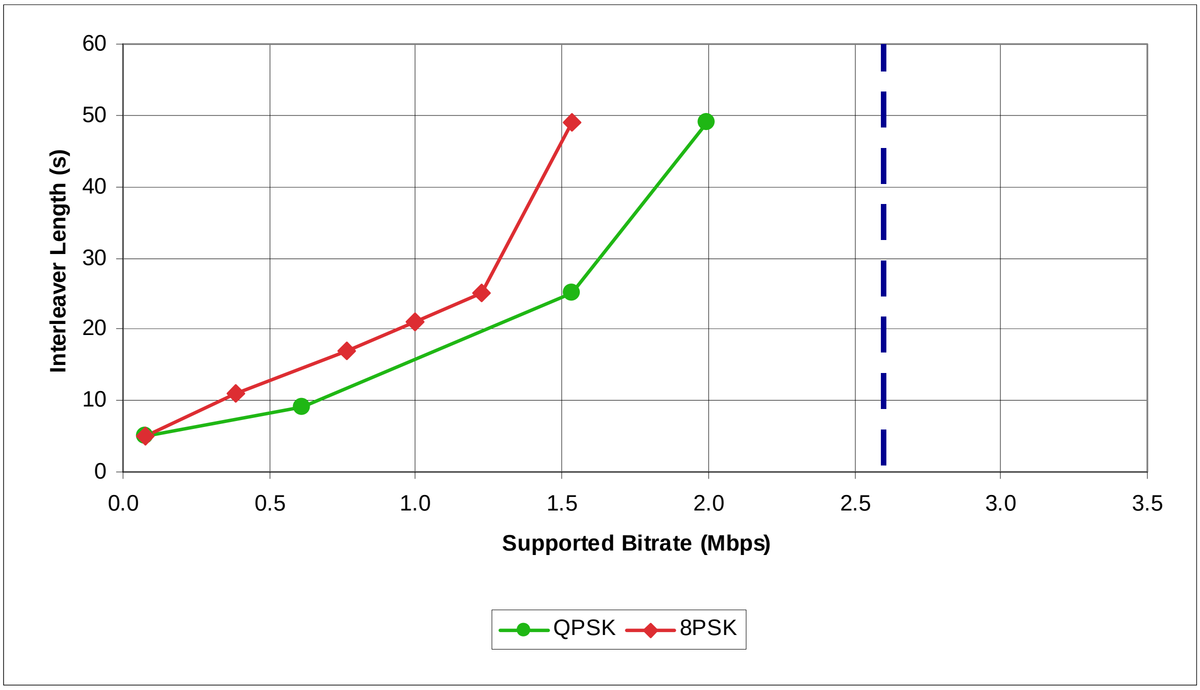


Figure A.12.10: SH-B, class 1: - QPSK ½ and 8PSK 1/3 - LMS-SU - 3 kmph - 68 dBW EIRP Satellite Different Link-Layer configuration fulfilling the ESR5(20) criterion at 90 % (corresponding reference cases IDs 77 and 78)

Figures A.12.11 and A.12.12 give the sensitivity analysis for class 2. The *C/N* has been increased with respect to the reference value of 5,1 dB and 6,2 dB for the reference OFDM and TDM cases respectively. For ID75 configuration the required *C/N* to guarantee the 90 % fulfillment of ESR5 criterion is 8 dB i.e. 3 dB above the reference *C/N* value. For ID81 and ID82 configurations the required *C/N* to guarantee the 90 % fulfillment of ESR5 criterion is 8 dB i.e. 2 dB above the reference *C/N* value.

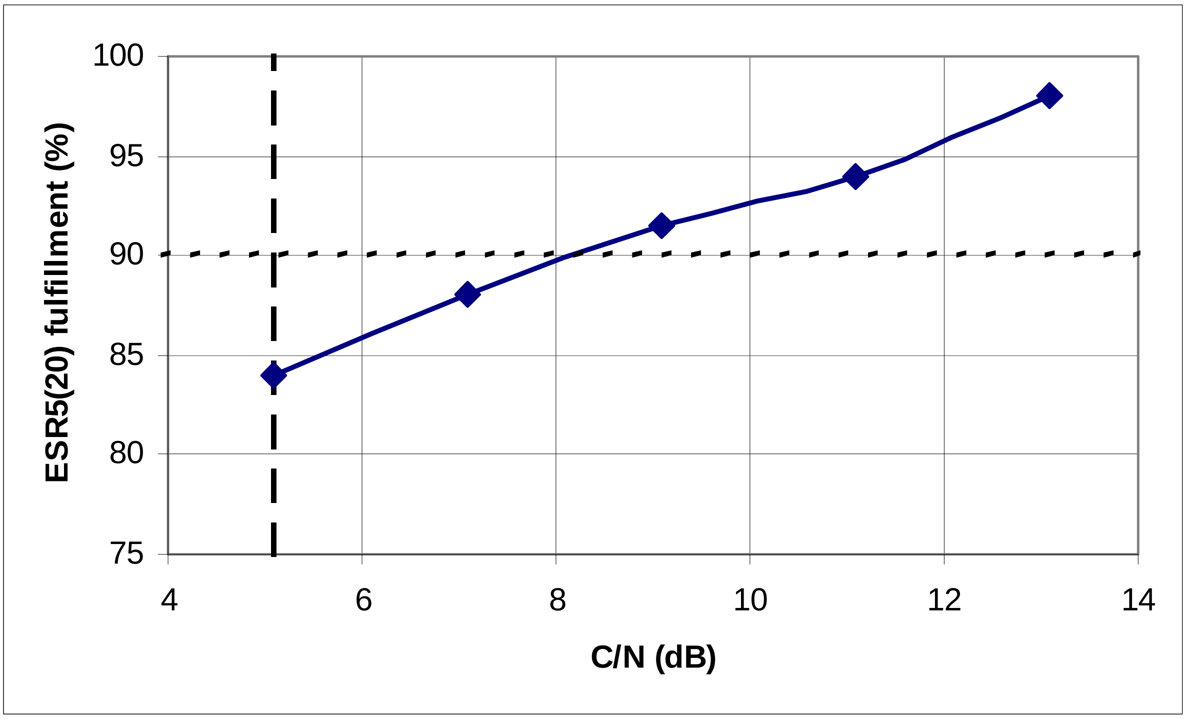


Figure A.12.11: SH-A, class 2: Uniform Interleaver Profile - QPSK 1/3 - LMS-ITS - 3 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference cases ID 75)

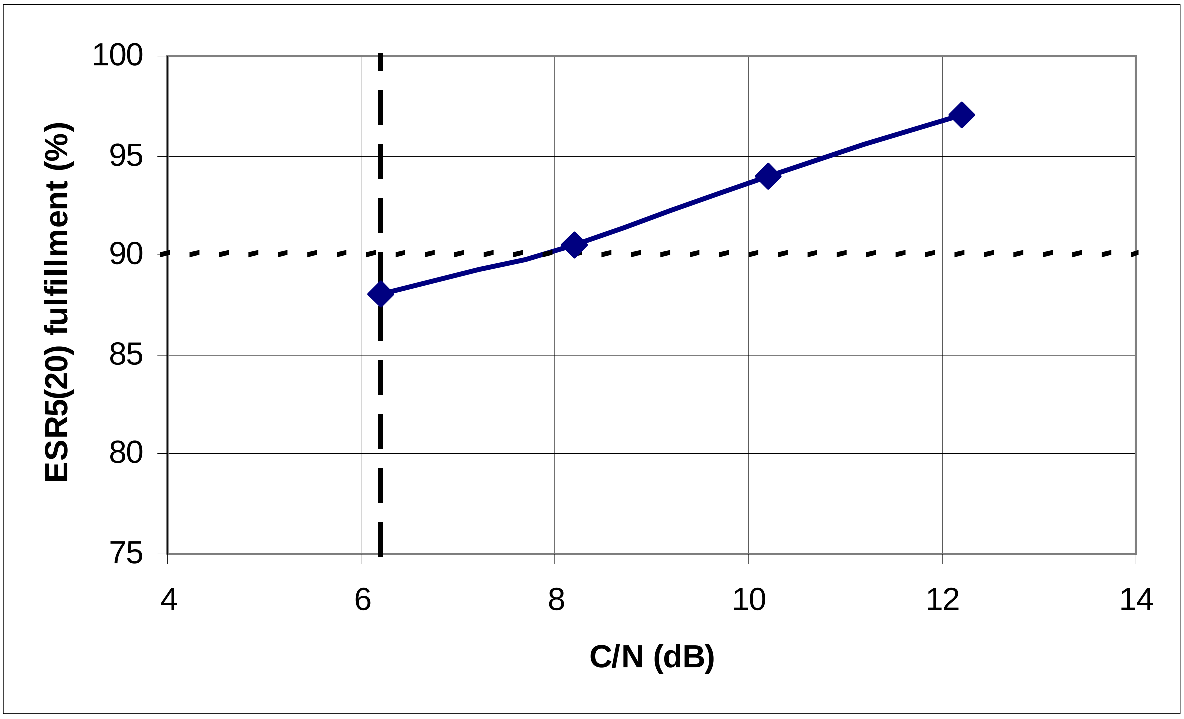


Figure A.12.12: SH-B, class 2: with Uniform Interleaver Profile - QPSK - LMS-ITS - 3 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference cases IDs 80 and 81)

A.12.2.4 Uniform Late (UL) and Uniform (U) long interleavers

This clause provides with performances comparison between the interleaver profiles Uniform Long (U) and Uniform Late (UL) as defined in clause A.4 and used for the simulation campaign for the class 2 receiver performance evaluation. Only SH-B waveform is considered, but the results are considered of general validity and applicable also to SH-A profile.

Both interleaver profiles show excellent performances in the reference system configurations, if typical satellite EIRP is considered and vehicular terminals Category 1 are addressed (e.g. cases 19 and 20 or 25 and 26 in table A.12.3). When the system is operating in such high ESR5 fulfillment regions, the performances of the two different profiles are almost equivalent and it is also possible to exploit the fast-zapping feature allowed by the UL profile.

On the contrary, when the target 90 % of ESR5 fulfillment is considered, the U interleaver profile is characterized by a lower C/N working point compared to UL thus providing extra link margin for possible system optimization in terms of satellite EIRP, terminal RF performances and spectral efficiency is possible. In particular, this clause quantifies this UL interleaver advantage for the reference scenarios.

Three different cases are analyzed in this clause:

a) LMS-SU scenario at 50 kmph for vehicular reception (figure A.12.13);

b) LMS-ITS scenario at 50 kmph for vehicular reception (figures A.12.14 and A.12.16)

c) LMS-SU scenario at 3 kmph for handheld reception (figure A.12.15).

With reference to a), the Uniform Long interleaver meets the 90 % ESR5 fulfillment at roughly 4 dB of C/N with a saving of 4 dB with respect the Uniform Late. This represents the worst case within the selected cases, indeed this difference reduces below 1 dB for both b) and c).

Figure A.12.16 shows the comparison results when different Physical-Layer configuration fulfilling the ESR5 criterion at 90 % are considered. When the interleaver length is set to the reference duration of 10 s (shown by the horizontal dashed line), the U interleaver allows supporting a bit rate higher than 4 Mbps with an increase of rougly 1 Mbps with respect the UL. Both configurations ensure a supported bit rate higher than the reference throughput (shown by the vertical dashed line), allowing for a interleaver duration reduction.When the same throughput is considered, the U interleaver requires a duration almost half of the UL one.

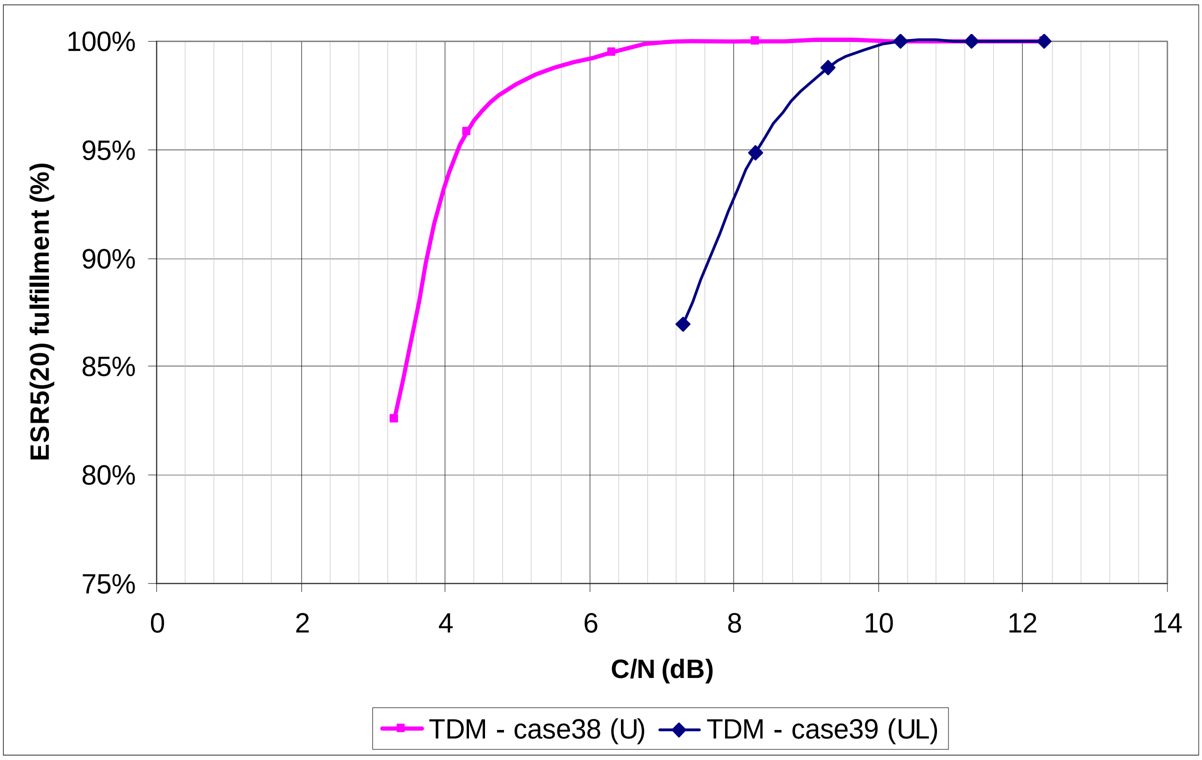


Figure A.12.13: SH-B, class 2: - QPSK 1/3 - LMS-SU - 50 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference case IDs 38 and 39)

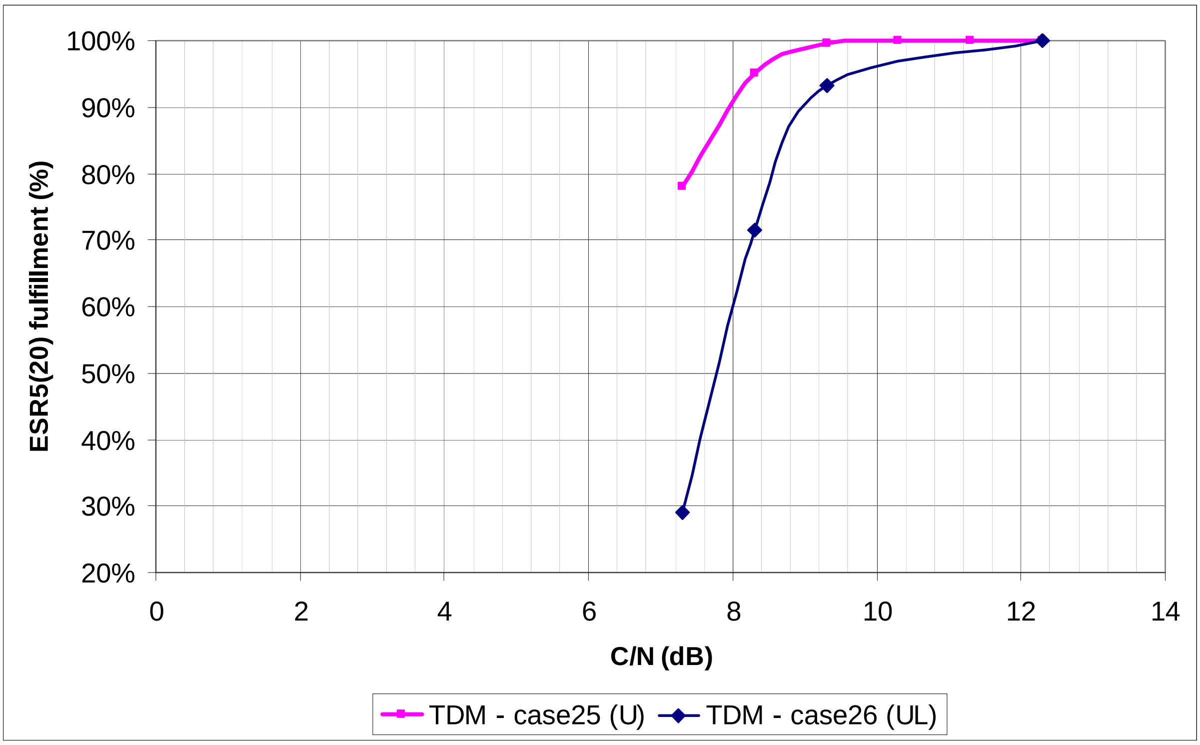


Figure A.12.14: SH-B, class 2: - QPSK 1/3 - LMS-ITS - 50 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference case IDs 25 and 26)

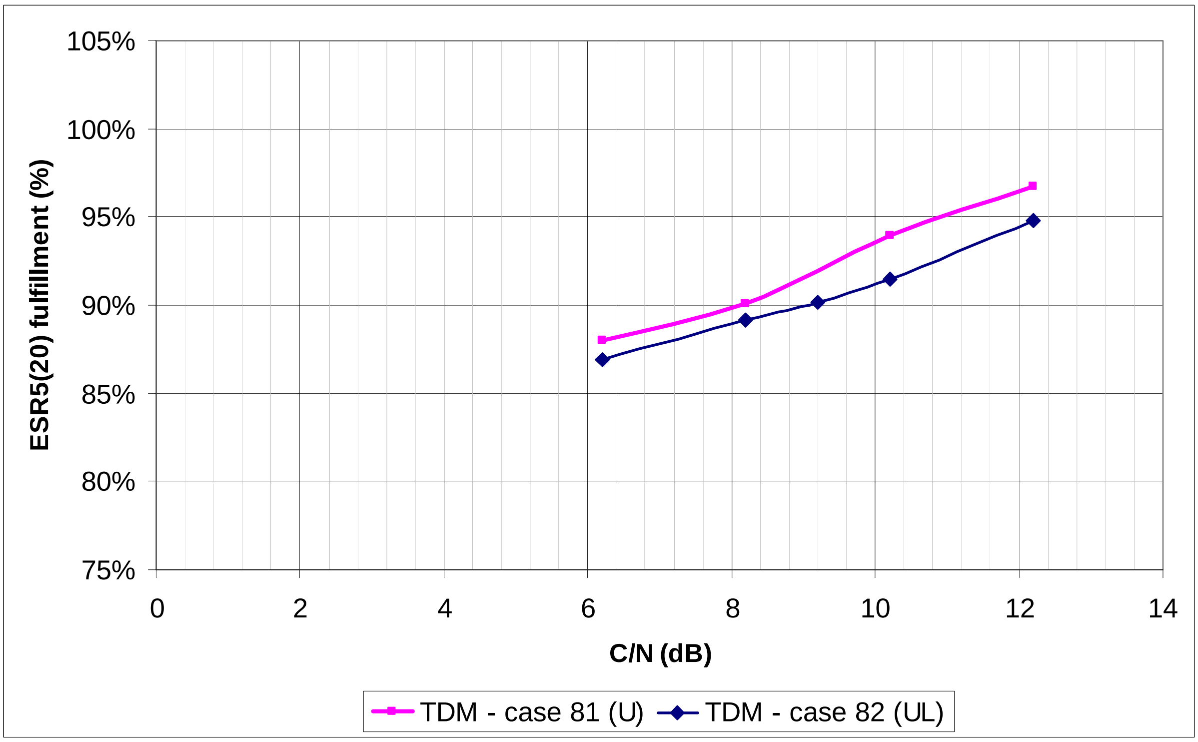


Figure A.12.15: TDM, class 2: - QPSK 1/3 - LMS-SU - 3 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference case IDs 81 and 82)

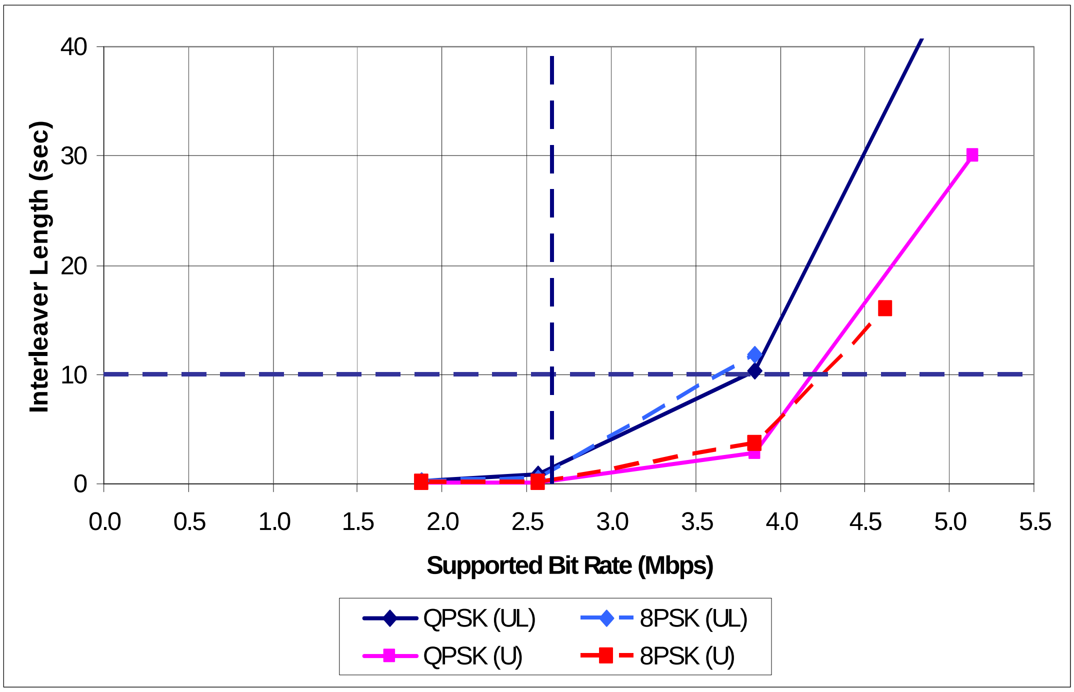


Figure A.12.16: TDM, class 2: - QPSK and 8PSK - LMS-ITS - 50 kmph
Different Physical-Layer configuration fulfilling the ESR5(20) criterion at 90 %

A.12.2.5 Demodulator state machine performance impact

This clause provides with performances assessment when the demodulator state machine is not modelled (clause A.5). The results are to be considered as performances for an ideal receiver with zero re-acquisition time.

In this clause only SH-B waveform is considered and for a specific case (case 25 in table A.12.3), but the results are considered of general validity and applicable also to different environment and SH-A profile.

The case analyzed case is the LMS-ITS scenario at 50 kmph for vehicular reception (figure A.12.17). When the ideal demodulator is considered (without state machine), the target 90 % of ESR5 fulfilment is achieved at less than 7 dB of *C/N* with a saving of more than 1 dB with respect to the reference demodulator. This represents the best case where the behaviour of a real demodulator is not considered and corresponding losses not accounted for.

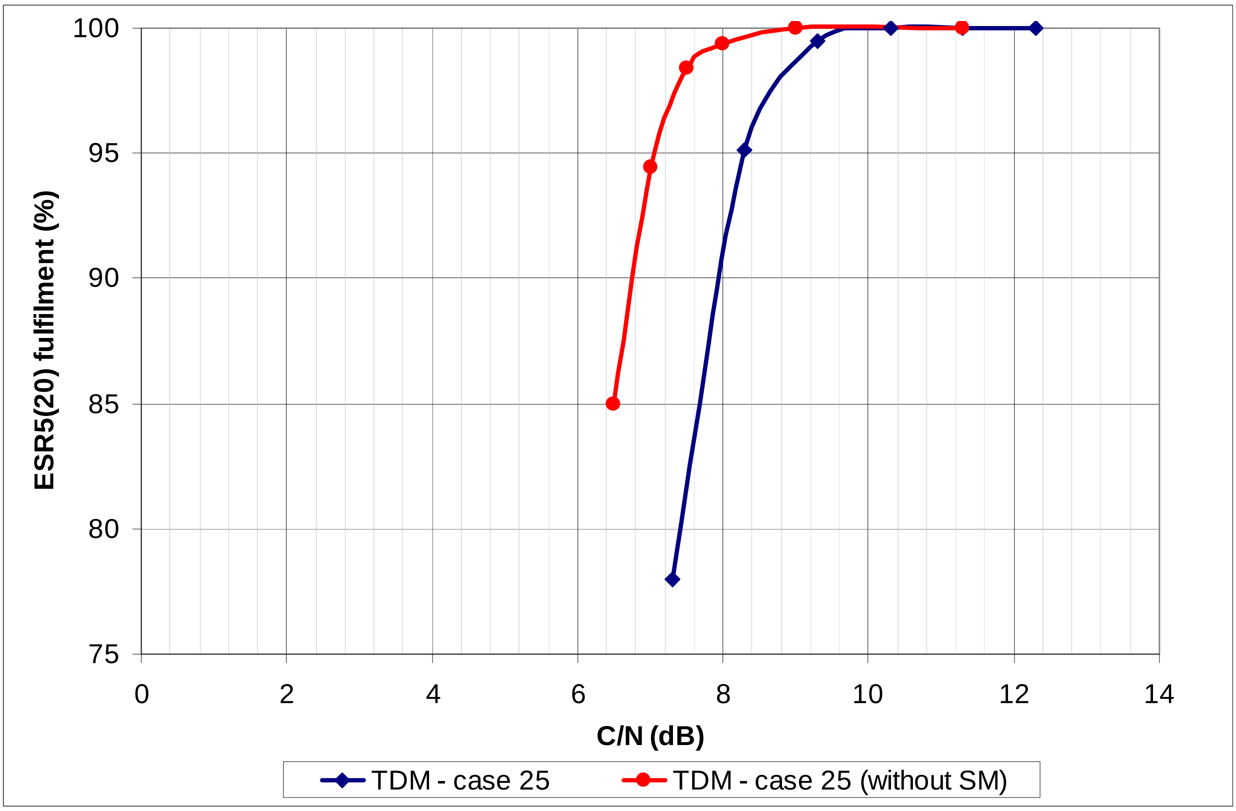


Figure A.12.17: SH-B, class 2: - QPSK 1/3 - LMS-ITS - 50 kmph Sensitivity Analysis to C/N value for a TDM modulation (corresponding reference case ID 25)

A.12.2.6 C/I performance impact

This clause provides with performances assessment when different values of co-channel useful signal over interference are considered (C/I as defined in clause A.9). It is to recall that satellite link *C/I* and *C/N* are provided separately as the interference is suppose to fade together with the signal and is simulated like an extra independent noise source modulated by the same fade as the useful signal.

In this clause only the SH-B waveform is considered but the findings can be considered of general validity and applicable to different environment and SH-A profile with comparable coding and modulation formats.

For the selected cases the ESR5performances are almost independent from the *C/I* value when typical satellite multibeam antenna values are considered. The very limited sensitivity to the *C/I* is noted, thus confirming that the performances of the selected configurations are not co-channel-interference-limited.

Two different physical layer configuration are analyzed in this clause:

a) LMS- ITS scenario at 50 kmph for vehicular reception, QPSK-1/3, 63 dBW EIRP Satellite (case 25 in table A.12.3, figure A.12.18).

b) LMS-ITS scenario at 50 kmph for vehicular reception, 8PSK-2/5, 63 dBW EIRP Satellite (figure A.12.19).

c) In the case a) for an ESR5 90 % fulfillment the *C/I* reduction from 16 to 12 dB causes only a required *C/N* increase of 0,1 dB. This is because we are using a highly protected physical layer configuration (QPSK r=1/3), operating with quite high link margin for counteracting the LMS channel and because interference follows the LMS channel fluctuations.

d) In the case b) for an ESR5 90 % fulfillment the *C/I* reduction from 16 dB to 12 dB causes only a required *C/N* increase of 0,2 dB. The slight higher *C/I* sensitivity observed in this case is due to the use of a less protected physical layer configuration (8PSK r=2/5). In any case for a satellite mobile broadcasting system operating with spectral efficiencies less than 1 bps/Hz the antenna *C/I* is not a very critical parameter up to 10 dB isolation.

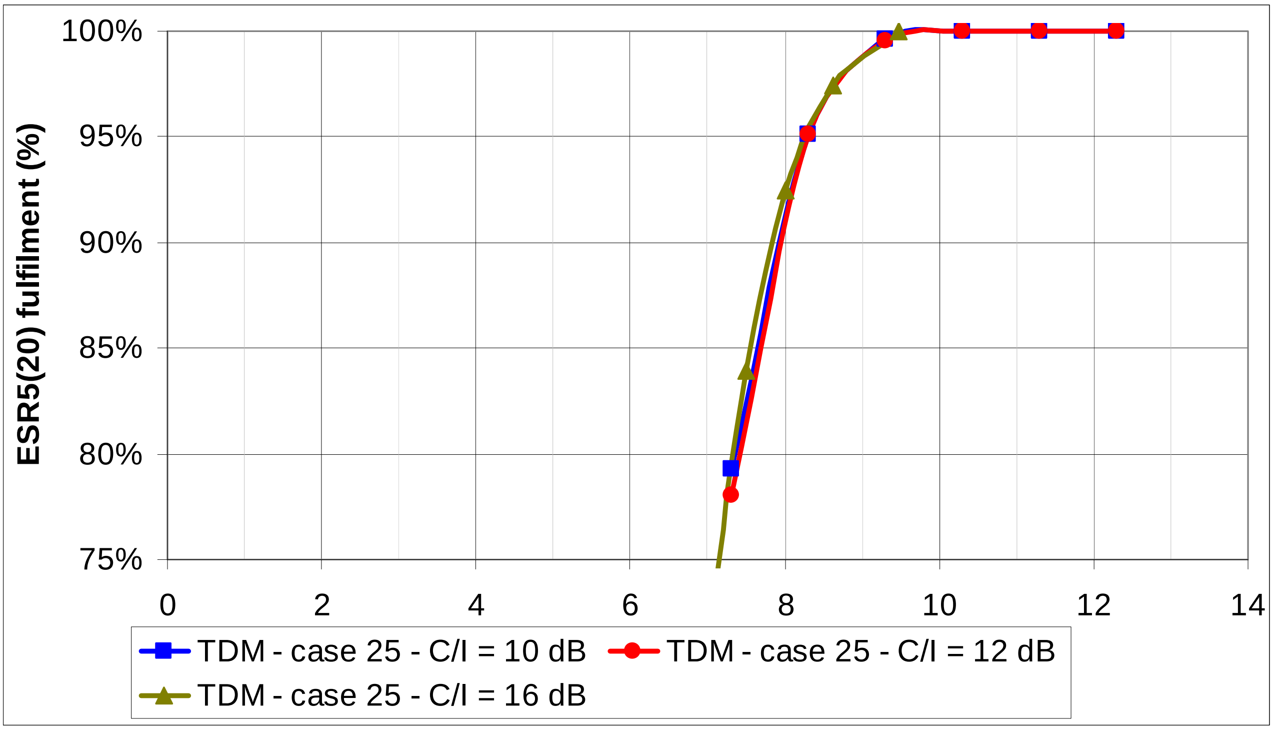


Figure A.12.18: SH-B, class 2: - QPSK 1/3 - LMS-ITS - 50 kmph Sensitivity Analysis to C/I value for a TDM modulation (corresponding reference case ID 25)

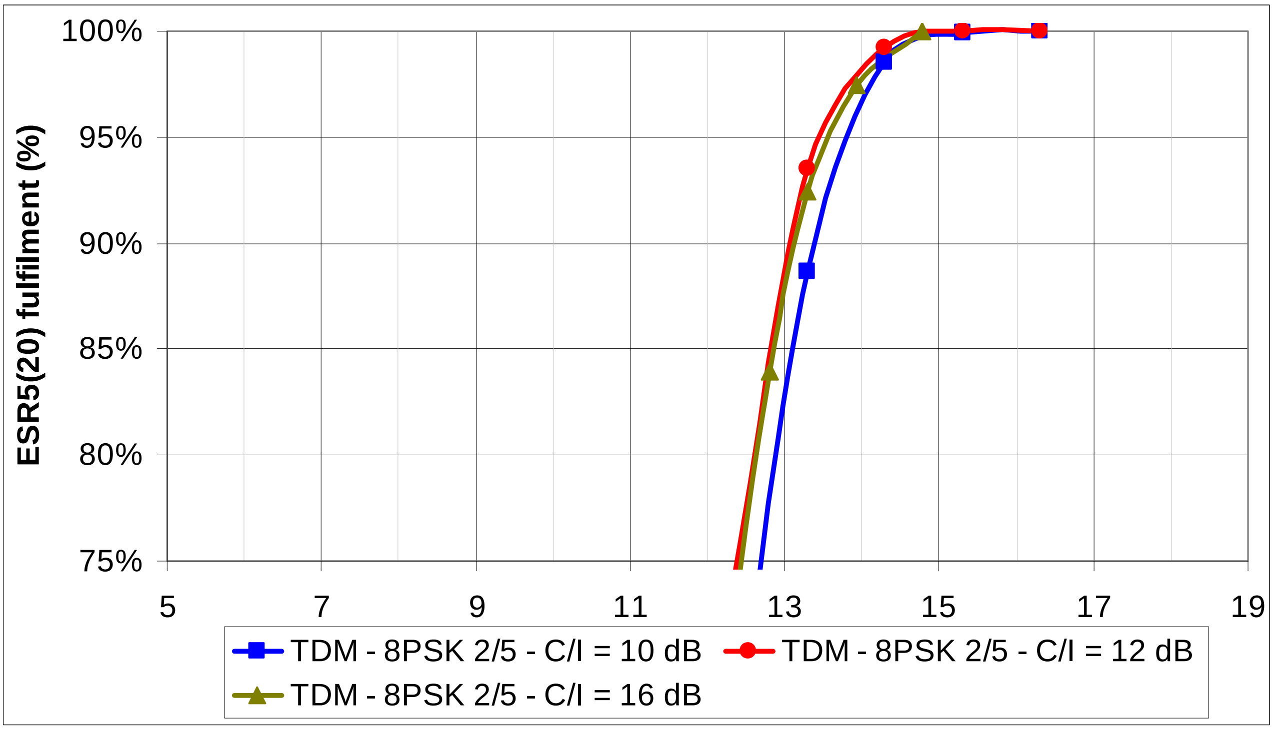


Figure A.12.19: SH-B, class 2: - 8PSK 2/5 - LMS-ITS - 50 kmph Sensitivity Analysis to C/I value for a TDM modulation

A.12.2.7 ESR5 fulfilment at different spectral efficiencies

This clause provides with DVB-SH ESR5 performance assessment when different spectral efficiencies are considered.

Only SH-B waveform results are reported but the findings are of general validity and also applicable with good approximation to different LMS environments and to the SH-A profile.

The case analyzed corresponds to the LMS-ITS scenario for vehicular reception at 50 kmph with a 68 dBW EIRP satellite (figure A.12.20). An Uniform Long interlaver with the reference duration of 10 s has been considered as well an ideal demodulator. QPSK and 8PSK modulation have been simulated for different physical layer FEC coding rates configurations corresponding to different supported bit rates. It is remarked that the performance of QPSK and 8PSK are essentially identical in terms of ESR5 fullfillemnt ratio as a function of the bit rate.

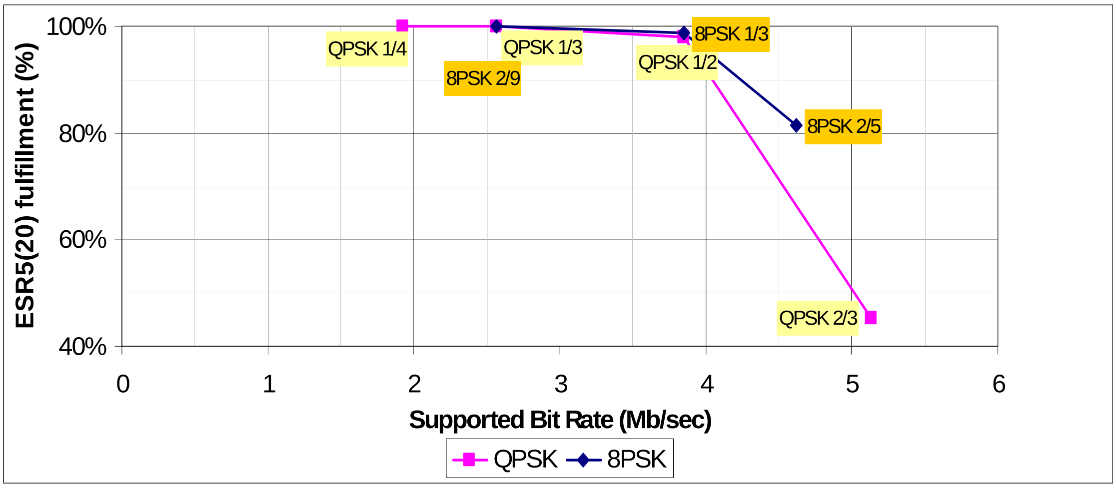


Figure A.12.20: SH-B, class 2: - QPSK and 8PSKLMS-ITS - 50 kmph Sensitivity Analysis to supported bit rate for a TDM modulation

A.12.3 Reception in TU6 environment

TU6 propagation model is used as reference for characterization of the small scale fading effects encountered in terrestrial environment reception. Thus, it only applies to the DVB-SH OFDM waveform. Performance are defined in terms of required *C/N*, to be used for Terrestrial Network Planning (clause 11).

Because of the hybrid structure of the DVB-SH system, two cases of terrestrial transmission are possible:

- terrestrial retransmission of the satellite signal of an SH-A (SFN) system, where in particular long interleavers, at Physical Layer level, or at IFEC level, may be implemented;
- dedicated terrestrial transmission, in case of retransmission of the satellite content for SH-A (non-SFN) or SH-B system, or in case of terrestrial-only local content broadcasting.

This clause gives the simulation performance in a TU6 channel, for an OFDM waveform, with a terminal speed of 3 kmph and 50 kmph. They apply to a reference service of 280 kbps. Overall capacity is calculated on the basis of a 1/8 guard interval, as FER/ESR5 simulation results in TU6 are not sensitive to guard interval larger than 1/8.

A.12.3.1 FER and ESR5 relationship

The receiver threshold performance is a function of the quality criteria. Two criteria are commonly used in broadcasting systems, which are the FER and the ESR5 (see clause A.8).

From the various simulations performed with the DVB-SH waveform in TU6, the relationship between ESR5 and FER reported in figure A.12.21 is observed, whatever the code rate and the speed of the receiver.

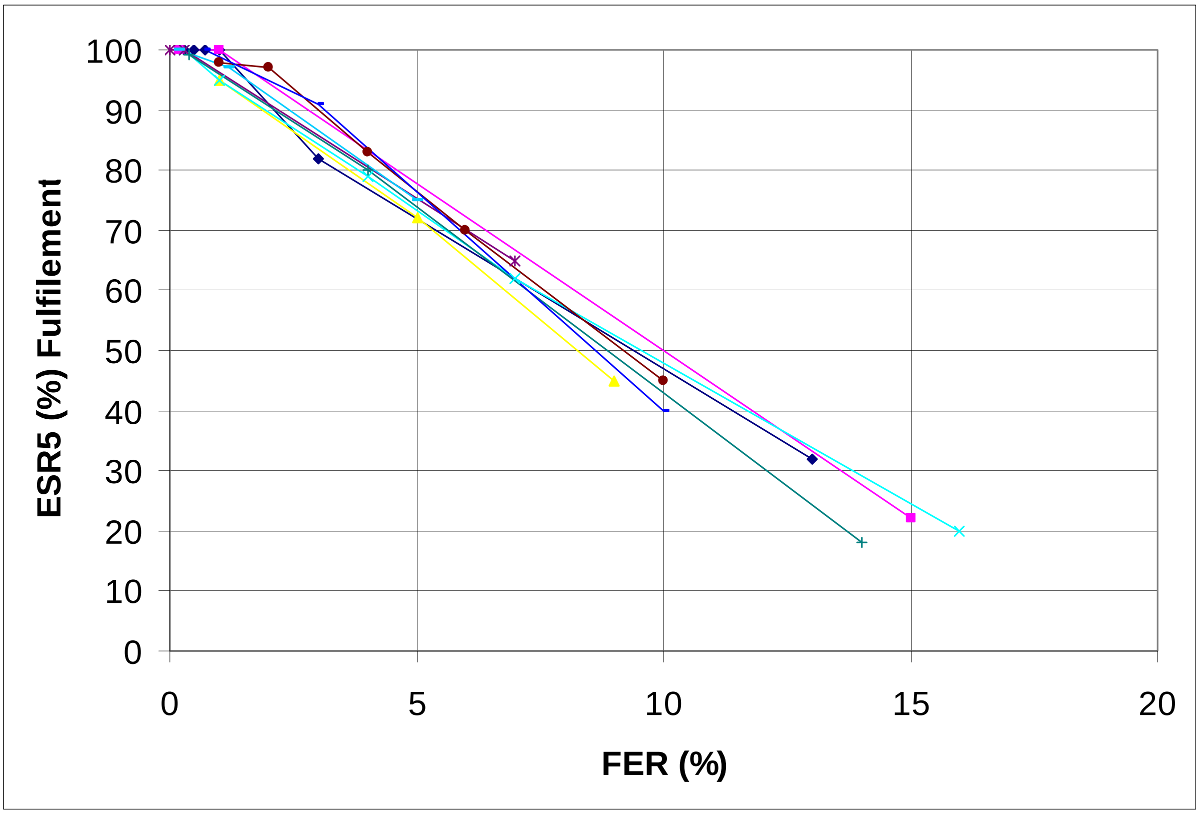


Figure A.12.21: ESR5 fulfillment performance versus FER performance for TU6 environment, for various speed, code rate and modulation

Each plot on the figure refers to a different order of modulation (QPSK or 16QAM) and code rate (from ½ to 1/5).

From this plot, 1-ESR5 = 99 % is equivalent to an FER = 1 %. In terms of *C/N* threshold difference between FER = 5 % and 1-ESR5 = 99 % (or 1 % FER), is typically less than 0,5 dB at 50 kmph, and is about 1 dB at 3 kmph. Exact values are provided in tables A.12.5 and A.12.6 as results of simulations. According to above results on FER and ESR relationship, FER 1 % performance can be considered as equivalent to 1-ESR5(20) 99 %.

A.12.3.2 FER performance in TU6

Tables A.12.5 and A.12.6 give the FER=5 % and FER 1 % performances of the DVB-SH OFDM waveform for all possible DVB-SHA physical layer coding rates and modulation (with short interleaver (200 ms QPSK and 100 ms for 16QAM with CM = 5 for both cases) and Uniform Late interleaving of 10 s for both QPSK and 16QAM cases 1-4, table A.12.6). The reference configuration of 5 MHz, FFT 2K and GI of 1/4 and two speeds (3 km/h and 50 km/h) have been taken into account.

Table A.12.5: FER performance in TU6 (Short)

Code rate PHY	Capacity (Mbps)	C/N(dB) @ FER=5 %	C/N(dB) @ FER=1% or 1-ESR5(20) 99
TU6 OFDM QPSK - 50 kmph			
1/5	1,333	-2,4	-2,2
2/9	1,481	-1,9	-1,5
1/4	1,679	-1,1	-0,8
2/7	1,876	-0,4	0
1/3	2,222	0,7	1
2/5	2,666	2,2	2,8
1/2	3,357	4	4,6
2/3	4,443	7,5	8,1
TU6 OFDM 16QAM - 50 kmph			
1/5	2,666	2,8	3,2
2/9	2,962	3,5	4,1
1/4	3,357	4,4	5,1
2/7	3,752	5,1	5,8
1/3	4,443	6,5	7
2/5	5,332	8,5	9,5
1/2	6,714	10,8	11,5
2/3	8,887	15,1	16,5
TU6 OFDM QPSK - 3 kmph			
1/5	1,333	-1	-0,1
2/9	1,481	-0,6	0,6
1/4	1,679	0,3	1,2
2/7	1,876	1,1	2,1
1/3	2,222	2	3
2/5	2,666	3,7	4,6
1/2	3,357	5,2	6,3
2/3	4,443	8,6	9,8
TU6 OFDM 16QAM - 3 kmph			
1/5	2,666	4,5	5,7
2/9	2,962	5,3	6,7
1/4	3,357	6,2	7,5
2/7	3,752	6,9	8,2
1/3	4,443	8,1	9,4
2/5	5,332	10,1	11,3
1/2	6,714	12,2	13,7
2/3	8,887	15,8	18

Table A.12.6: FER performance in TU6 with Uniform Late interleaving

Code rate PHY	Capacity (Mbps)	C/N (dB)@ FER=5 %	C/N(dB @ FER=1%) or 1-ESR5(20) 99
TU6 OFDM QPSK - 50 kmph			
1/5	1,333	-2,7	-2,4
2/9	1,481	-2,2	-1,9
1/4	1,679	-1,3	-1
2/7	1,876	-0,5	-0,2
1/3	2,222	0,5	0,8
2/5	2,666	2,2	2,5
1/2	3,357	3,8	4,4
2/3	4,443	7,5	8
TU6 OFDM 16QAM - 50 kmph			
1/5	2,666	2,3	2,7
2/9	2,962	3	3,4
1/4	3,357	4	4,5
2/7	3,752	4,5	5,2
1/3	4,443	6,1	6,6
2/5	5,332	8,2	8,6
1/2	6,714	10,4	11
2/3	8,887	14,5	15,5
TU6 OFDM QPSK - 3 kmph			
1/5	1,333	-2,1	-1,3
2/9	1,481	-1,2	-0,7
1/4	1,679	-0,5	0,1
2/7	1,876	0,4	1
1/3	2,222	1,4	2,1
2/5	2,666	2,9	3,6
1/2	3,357	4,5	5,2
2/3	4,443	7,8	8,6
TU6 OFDM 16QAM - 3 kmph			
1/5	2,666	3,2	4
2/9	2,962	3,9	4,7
1/4	3,357	5	5,7
2/7	3,752	5,8	6,4
1/3	4,443	6,6	7,6
2/5	5,332	8,9	9,6
1/2	6,714	10,8	11,8
2/3	8,887	14,5	15,5

From tables A.12.5 and A.12.6, it may be noted that:

- with short interleaving (class 1 demodulator), *C/N* thresholds for ESR5 fulfilment at 99 % at 3 kmph are typically 1,5 dB and 2,0 dB worse than those at 50 kmph when QPSK is considered; between 2,0 dB and 2,5 dB with 16QAM;
- with Uniform Late interleaving of 10 s (class 2 demodulator), the improvement at 3 km/h is around 1 dB for QPSK, and 2 dB for 16QAM, while, as expected, at 50 km/h, the improvement is less significant and in average limited to 0,5 dB for both QPSK and 16QAM.

A.12.3.3 ESR5 Performance in TU

This clause reports the ESR5 performance in TU, arranged in terms of Physical interleaver length, and IFEC definition. The following configurations have been considered:

a) Long Physical layer interleaving, without IFEC;

b) Short Physical layer interleaving, with and without IFEC.

Performance versus Long Physical layer interleaving length

Short and Long Physical layer interleaving configurations are analyzed first. No IFEC is considered. The short interleaver is 100 ms long for 16QAM, and 200 ms long for QPSK. The UL interleaver is 10 s long, with half of the interleaver paths located within 200 ms.

Table A.12.7 gives the ESR5 simulated performance based on FER simulation measurement in a TU6 channel, for an OFDM waveform, with a terminal speed of 3 kmph and 50 kmph. In both interleaver configurations, the same capacity is provided, as no IFEC is used. Guard Interval of 1/8 has been considered

The results are reported for a ESR5 fulfilment of 99 %. The improvement on the *C/N* threshold between short physical interleaver and UL interleaver is up to 0,5 dB at 50 kmph with both modulations, 1 dB at 3 kmph with QPSK, and about 1,8 dB at 3 kmph with 16APSK.

Table A.12.7: C/N threshold in TU6 for ESR5 fulfilment of 99 %, UL versus S interleaver (GI=1/8)

Code Rate PHY	Interleaver PHY	Capacity (Mbps)	C/N (dB) @ 3 kmph	C/N (dB) @ 50 kmph
TU6 OFDM 16QAM				
1/3	S	4,9	9,4	7,0
1/3	UL	4,9	7,6	6,6
1/4	S	3,7	7,5	5,1
1/4	UL	3,7	5,7	4,5
1/5	S	3,0	5,7	3,2
1/5	UL	3,0	4,0	2,7
TU6 OFDM QPSK				
1/2	S	3,7	6,3	4,6
1/2	UL	3,7	5,2	4,4
1/3	S	2,5	3,0	1,0
1/3	UL	2,5	2,1	0,8

Performance with and without IFEC

Performance of configurations with different IFEC setting are analyzed in this clause. In all cases class 1 demodulator with short physical interleaver has been considered.

Tables A.12.8 give the ESR5 simulation performance in a TU6 channel, for an OFDM waveform, with a terminal speed of 3 kmph and 50 kmph. In most configurations, the same capacity after IFEC decoding is provided. Related effect on the required *C/N* of IFEC code rate together with IFEC interleaving length (B+S) is analyzed. The QoS target is 99 % of ESR5 fulfilment.

At 3 kmph, which is somewhat representative of handheld reception in terrestrial environment, it is shown that:

- In case of QPSK, same capacity is achieved with FEC code rate 1/2 plus IFEC rate 2/3 and FEC with code rate 1/3 and no IFEC. As expected a longer IFEC interleaver allows to improve the *C/N* threshold but the performance remains inferior to the only-phisical layer approach with code rate 1/3 and short interleaver (200 ms).

- In case of 16QAM it is not possible a direct comparison: slightly higher capacity is achieved in the case of only-phisical layer approach, with code rate 1/5 and no IFEC, at the expense of a larger required C/N. This is due to the fact that the selected 16QAM 1/5 configuration is deliberately more spectrum efficient.

In a SH-A SFN configuration, IFEC settings defined for the satellite link will marginally penalize the reception in terrestrial environment for low speed users. However in case of SH-B or SH-A MFN it will be possible to get lower C/N threshold with low zapping time using physical layer FEC only and short interleaver (200 ms).

In case of dedicated terrestrial transmission, the introduction of IFEC will offer fine tuning system capability. Typically, addition of IFEC will provide the possibility to control of the quality of service for each service based on high flexibility capability.

Table A.12.8: C/N threshold and ESR5 fulfilment in TU6 @ 3 Kmph, with and without IFEC

Code Rate PHY	Interleaver PHY	Code Rate IFEC	Capacity (Mbps)	C/N (dB)	Interleaving length (s)	1-ESR5 (%)
TU6 OFDM QPSK - 3 kmph						
1/3	S	-	2,5	3,0	0,200	99,00
1/2	S	0,67	2,5	5,5	10	99,50
1/2	S	0,67	2,5	5,0	16	99,20
1/2	S	0,67	2,5	4,5	30	99,00
TU6 OFDM 16QAM - 3 kmph						
1/5	S	-	3,0	5,7	0,100	99,00
1/4	S	0,67	2,5	5,0	10 to 16	> 99,20
1/4	S	0,60	2,2	5,0	10	99,30
1/4	S	0,42	1,5	4,0	20 to 30	> 99,10

IFEC Synthetic overall Performance

Figure A.12.22 gives a synthetic overview of the performance achieved at ESR5(20) @ 99 % criteria for all configurations, in TU6 at 3 kmph, and some IFEC configurations:

a) the different configurations are represented according to the legends;

b) UL configurations and short configurations are represented by the different lines;

c) IFEC configurations are represented with larges squares.

DVB-SH includes 8 code rate settings at physical layer, from 1/5 to 2/3. All of them are presented in figure A.12.22 with QPSK modulation, and 3 with 16QAM modulation. The full set of code rates, together with the flexibility of IFEC, allows to provide a very fine system capacity optimization capability. It can be seen that class 1 demodulator with all protection at physical layer outperforms class 1 + IFEC demodulator for all the configurations reported. The C/N threshold gain is in the order of 1 dB for QPSK and 1,5 dB for 16QAM. The other advantage is that the FEC only configuration has a much lower zapping time than the one using the IFEC.

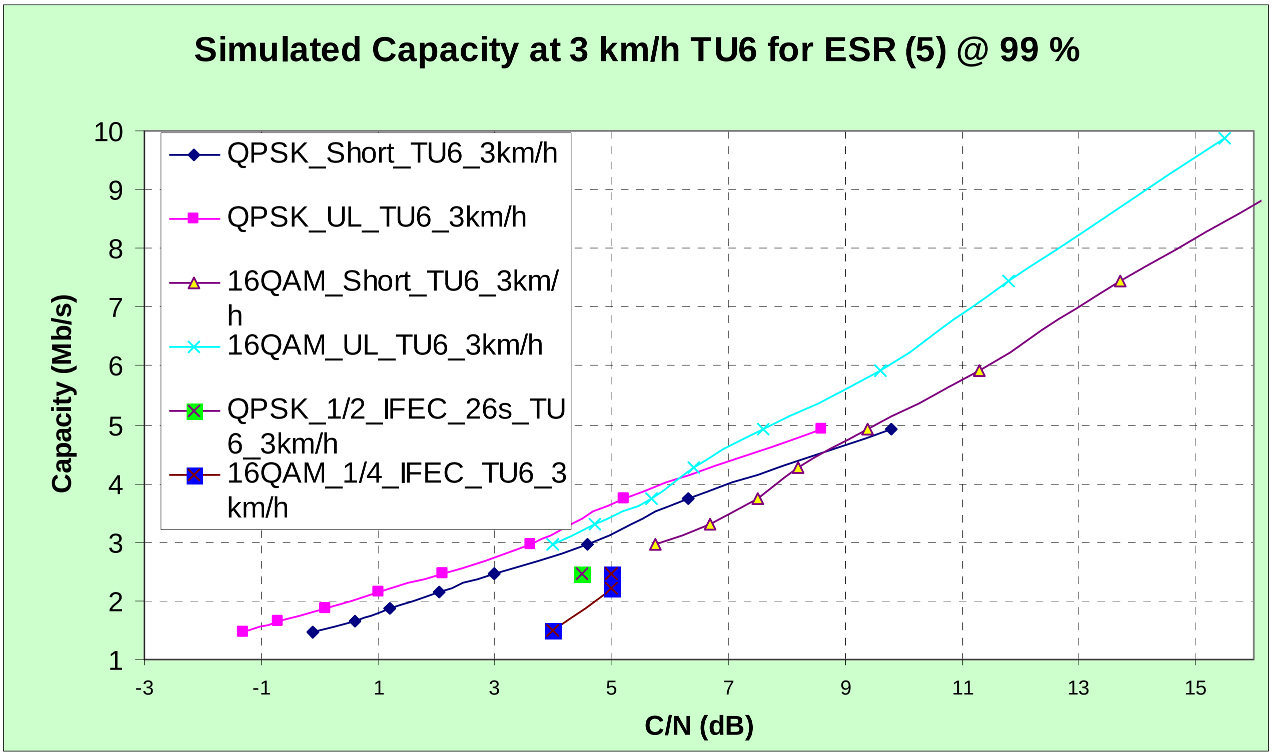


Figure A.12.22: C/N performances for ESR (5) @ 99 % in TU6 3 km/h

Figure A.12.23 gives a synthetic overview of the performance achieved at ESR5(20) @99 % criteria for all above configurations, in TU6 at 50 kmph.

Available dumps covered the IFEC performance for QPSK 1/2.

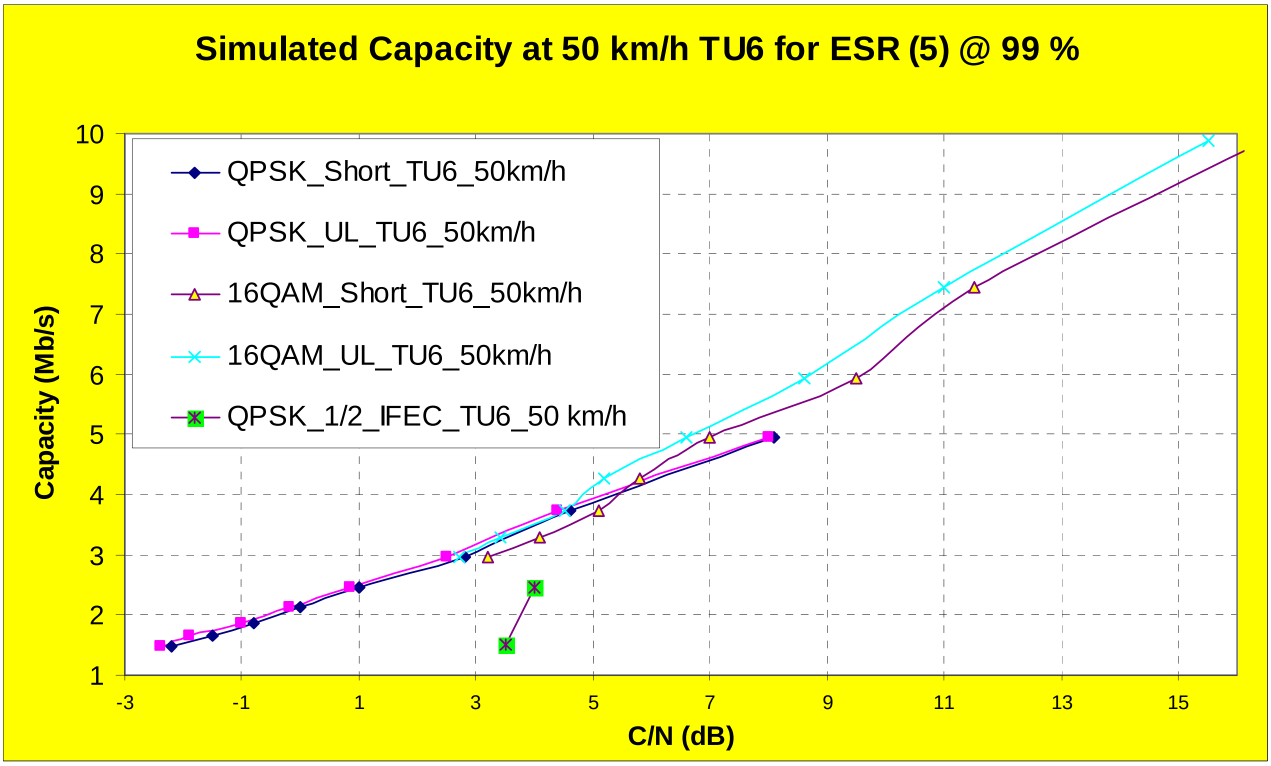


Figure A.12.23: C/N performances for ESR5(20) @ 99 % in TU6 50 km/h

A.13 Experimental results

A.13.1 Laboratory results

A.13.1.1 Terrestrial channels

A.13.1.1.1 Configurations

For the terrestrial channel tests, the baseline configurations tested are described in table A.13.1. A TU6 channel model (see clause A.7) has been used. For all configurations the impact of speed on required C/N for a QoS criterion corresponding to FER 5 % has been analysed.

Table A.13.1: Baseline Terrestrial channels configuration

BW (MHz)	Constellation	Code rate	FFT size	GI	Short Interleaving	class 2 length	IFEC length and rate	Capacity (Mb/s)	Capacity with IFEC
5	QPSK	1/5	2k	1/8	265 ms	U 9s	N/A	1,48	1,037
5	QPSK	2/9	2k	1/8	265 ms	U 9s	N/A	1,65	1,152
5	QPSK	1/4	2k	1/8	265 ms	U 9s	N/A	1,87	1,306
5	QPSK	2/7	2k	1/8	265 ms	U 9s	N/A	2,14	1,498
5	QPSK	1/3	2k	1/8	265 ms	U 9s	Same as LMS	2,47	1,728
5	QPSK	2/5	2k	1/8	265 ms	U 9s	N/A	2,96	2,074
5	QPSK	1/2	2k	1/8	265 ms	U 9s	N/A	3,73	2,611
5	QPSK	2/3	2k	1/8	265 ms	U 9s	N/A	4,94	3,456
5	16QAM	1/5	2k	1/8	132 ms	U 9s	N/A	2,96	2,074
5	16QAM	2/9	2k	1/8	132 ms	U 9s	N/A	3,29	2,304
5	16QAM	1/4	2k	1/8	132 ms	U 9s	N/A	3,73	2,611
5	16QAM	2/7	2k	1/8	132 ms	U 9s	N/A	4,28	2,995
5	16QAM	1/3	2k	1/8	132 ms	U 9s	Same as LMS	4,94	3,456
5	16QAM	2/5	2k	1/8	132 ms	U 9s	N/A	5,92	4,147
5	16QAM	1/2	2k	1/8	132 ms	U 9s	N/A	7,46	5,222
5	16QAM	2/3	2k	1/8	132 ms	U 9s	N/A	9,87	6,912

NOTE: The different short interleaving lengths provided in the table correspond to a value compatible with approx. 4 Mbits on chip memory as minimum requirement in clause 10 for receivers class 1.

A part from these configurations, the impact of different parameters has been studied for some modulation-coding schemes. More precisely the following parameters have been considered:

- Different bandwidth.
- Different FFT sizes.
- Different GI.
- IFEC length.
- Impact of Long Interleaving.
- Uniform Long interleaving length.

A.13.1.1.2 Synthetic results

Table A.13.2 summarizes the various C/N values required for achieving a FER=5 %.

Table A.13.2: Summary of results

Mode	Code Rate	C/N (dB) @ Low speed	C/N (dB) @ Medium speed	C/N (dB)	Low Doppler frequency (Hz)	Medium Doppler frequency (Hz)	Doppler frequency (Hz) @ 3 dB loss wrt medium speed	Max Doppler frequency (Hz) @ > 30 dB loss wrt medium speed
QPSK	1/5	0,5	0,0	3,0	10	334	639	669
	2/9	0,8	0,3	3,3	10	325	620	650
	1/4	1,3	0,8	3,8	10	316	601	631
	2/7	2,0	1,3	4,3	10	306	583	613
	1/3	2,8	2,0	5,0	10	288	545	575
	2/5	4,3	3,5	6,5	10	278	526	556
	1/2	5,8	5,3	8,3	10	259	489	519
	2/3	9,5	9,3	12,3	10	222	414	444
16QAM	1/5	5,9	3,8	6,8	10	275	520	550
	2/9	6,3	4,4	7,4	10	269	508	538
	1/4	7,2	5,0	8,0	10	256	483	513
	2/7	7,8	5,9	8,9	10	244	458	488
	1/3	9,1	7,2	10,2	10	238	445	475
	2/5	10,9	9,1	12,1	10	219	408	438
	1/2	12,8	11,3	14,3	10	206	383	413
	2/3	17,5	16,3	19,3	10	175	328	350

Figures A.13.1 and A.13.2 represent the graphical representation of table A.13.2.

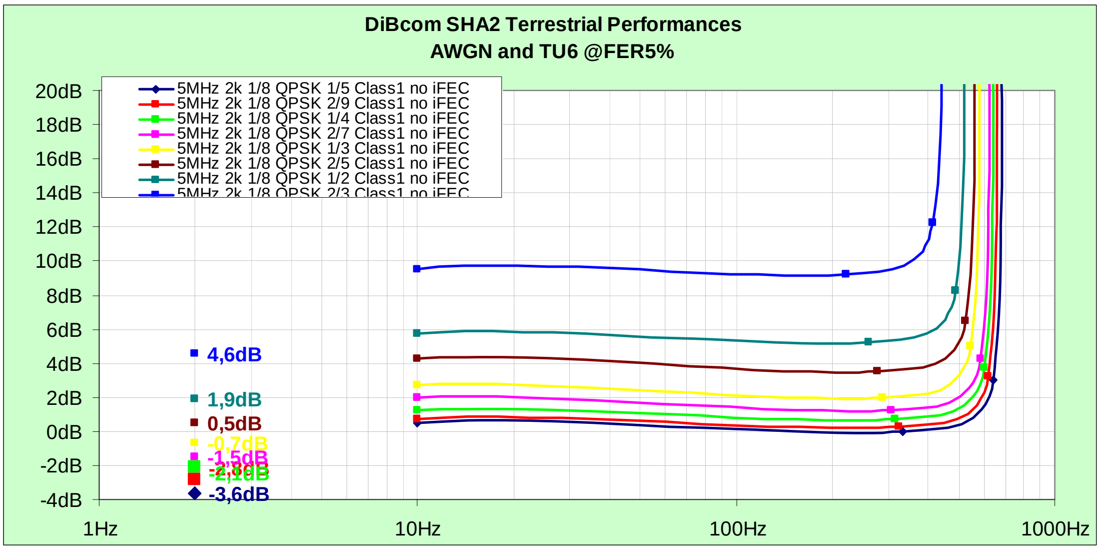


Figure A.13.1: QPSK AWGN and terrestrial performances

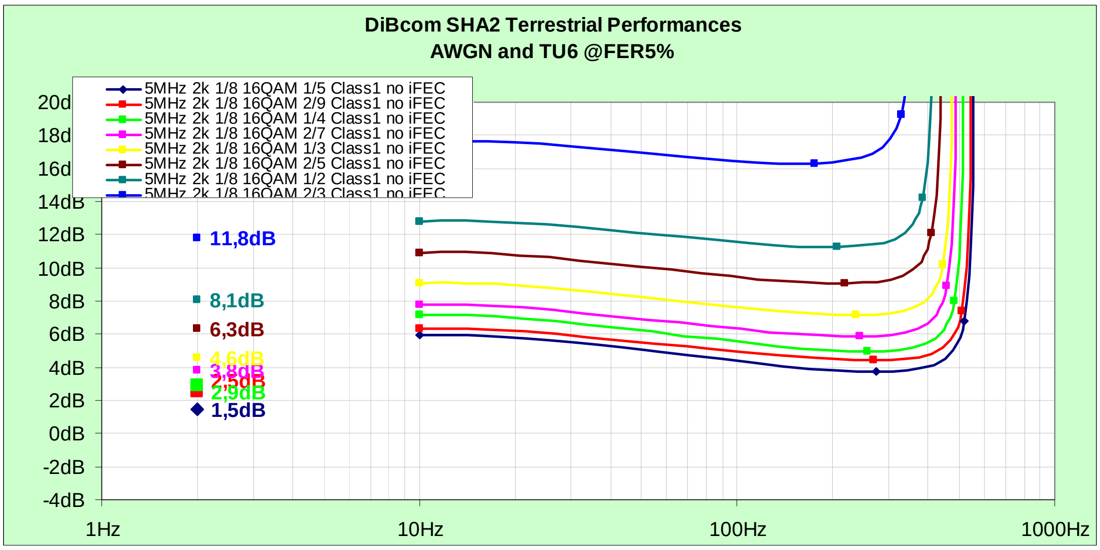


Figure A.13.2: 16QAM AWGN and terrestrial performances

A.13.1.1.3 Impact of different parameters

In this clause the impact of the different parameters listed in clause A.13.1.1.1 is presented in graphical format.

Different bandwidth impact

The chipset used for the laboratory tests supports 5 MHz, 6 MHz, 7 MHz and 8 MHz bandwidth. Expected performance dependency on the OFDM signal bandwidth is the following:

- No difference on the required AWGN C/N for FER=5 %.
- No significant difference for the required medium-speed C/N (time interleaving depth in class1 is reduced but should be still sufficient for medium speed).
- Small penalty on the 10 Hz required C/N in class 1 due to time interleaving duration reduction.
- Linear increase of the maximum supported Doppler frequency with the signal bandwidth.

Nevertheless, performances of the commercial chip used for the test are not in line with these expectations. Due to throughput turbo-decoder limitations, 7 MHz and 8 MHz OFDM bandwidth configurations underperform compared to 5 MHz and 6 MHz. Observed degradations are variable and depend on the actual bit rate: higher is the bit rate, higher is the measured degradation. This finding is not related to the standard itself but rather on the specific ASIC implementation.

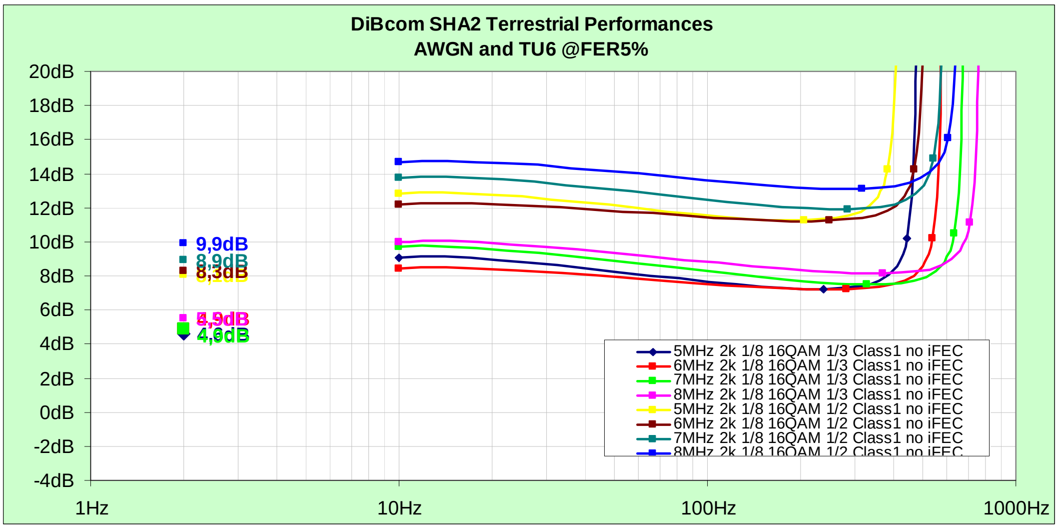


Figure A.13.3: Bandwidth influence

Different FFT sizes

Expected performance dependency on the OFDM FFT size variation is the following:

- No difference on the required AWGN C/N for FER=5 %.
- No difference on the required TU6 C/N for FER=5 %.
- Linear increase of the maximum supported Doppler frequency with the OFDM FFT size.

Only one modulation/coding case has been studied; indeed the maximum Doppler frequency for different cases can be derived applying the simple rule: $Fdmax_{2K}$ = 2 x $Fdmax_{4K}$ = 4 x $Fdmax_{8K}$.

As it can be seen in figure A.13.4, performance expectations are fulfilled by tested receiver.

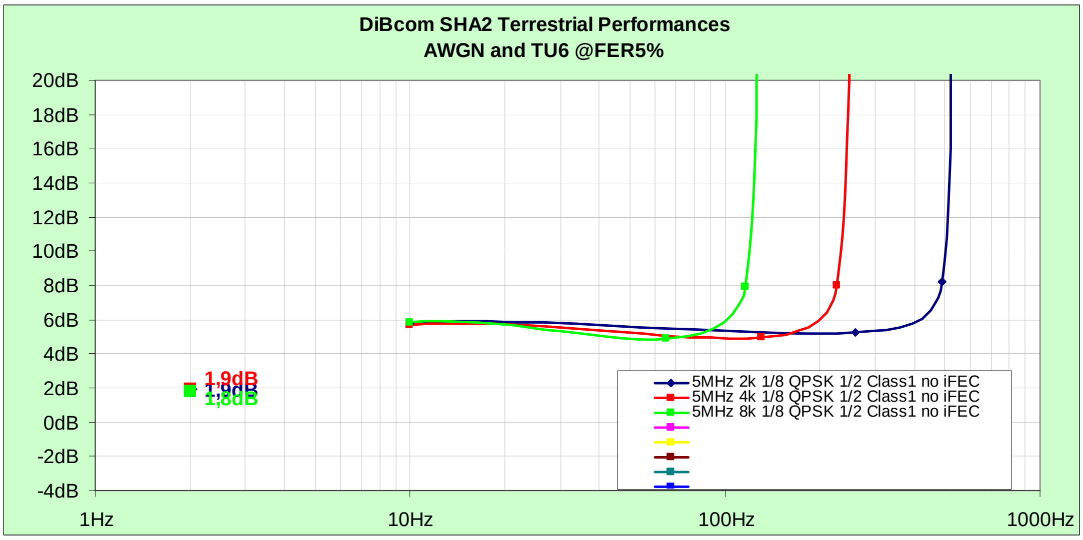


Figure A.13.4: Impact of FFT size

Impact of different Guard Interval

Expected behaviour for guard interval size variation is the following:

- No difference on the required AWGN C/N for FER=5 %.
- No difference on the required TU6 C/N for FER=5 %.

- Increase of Doppler-bandwidth when reducing the guard interval size.

As it can be seen in figure A.13.5, performance expectations are fulfilled by tested receiver.

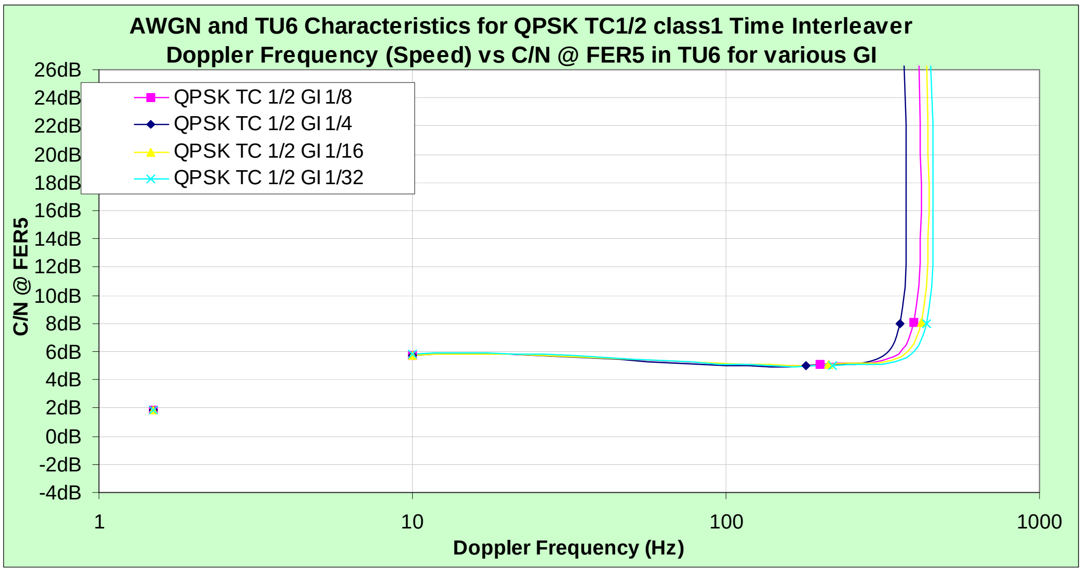


Figure A.13.5: Guard interval size influence

According to these results, the GI value has no impact on the C/N performances at any speed, except when very close to the maximum Doppler at the C/N + 3 dB point. This result legitimates the use of GI 1/8 instead of 1/4 in all measurements.

Impact of MPE-IFEC length

Different B+S values have been tested in TU6 at 110 Hz with 16QAM 1/3.

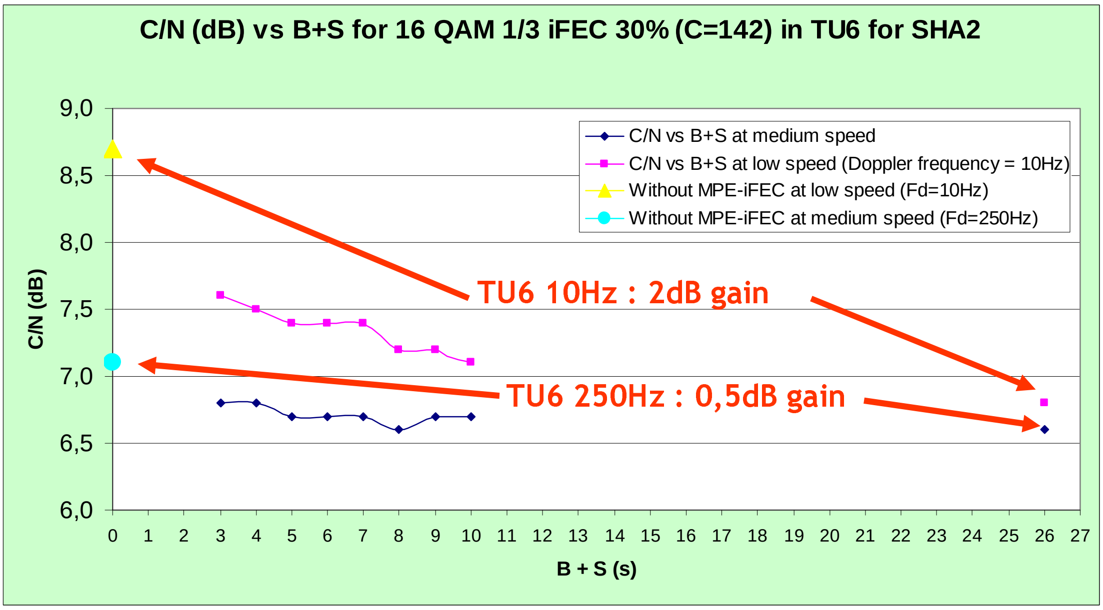


Figure A.13.6: Impact of B+S in MPE-IFEC

- The results summarised in figure A.13.6 confirm that:
  - The performances are improving when the B+S value increases.
  - The gain of performance is about 0,5 dB for the low speed test mode (Fd is about 10Hz) and about 0,2 dB for medium speed (Fd is about 250 Hz) for B+S varying from 3 s to 10 s.

- At low speed, the minimum C/N is obtained for the highest B+S value (26 s). B+S at about 8 s presents good performance as well with a C/N required just 0,4 dB higher than for B+S=26 s.
- At medium speed, the performances are not really improved when B+S varies (0,2 dB).
- Concerning tests without MPE-IFEC, the gain brought by the MPE-IFEC at least is about 1,1 dB at 10 Hz and about 0,3 dB at 250 Hz.
- The gain brought by B+S=26 s is about 0,3 dB at low speed and about 0,1 dB at medium speed. This gain can be considered as insufficient compared to the length of the IFEC interleaver. A shorter B+S value (for example 8 s) is therefore considered sufficient for the terrestrial mode although when shadowing is present on top of TU6 fading a longer interleaver size may be beneficial.

Impact of long interleaving in TU6

This clause reports the C/N performances in TU6 with a uniform long interleaver versus speed; sensitivity analysis to C/N is shown in clause 7.3.1.2.2.

It is observed that as expected the long interleaving impact is to uniform the required C/N for achieving FER=5 % versus the maximum Doppler-frequency. This is because the long interleaver breaks the fading correlation at low user speed (moderate to small fading bandwidth) reducing the corresponding loss observed in its absence.

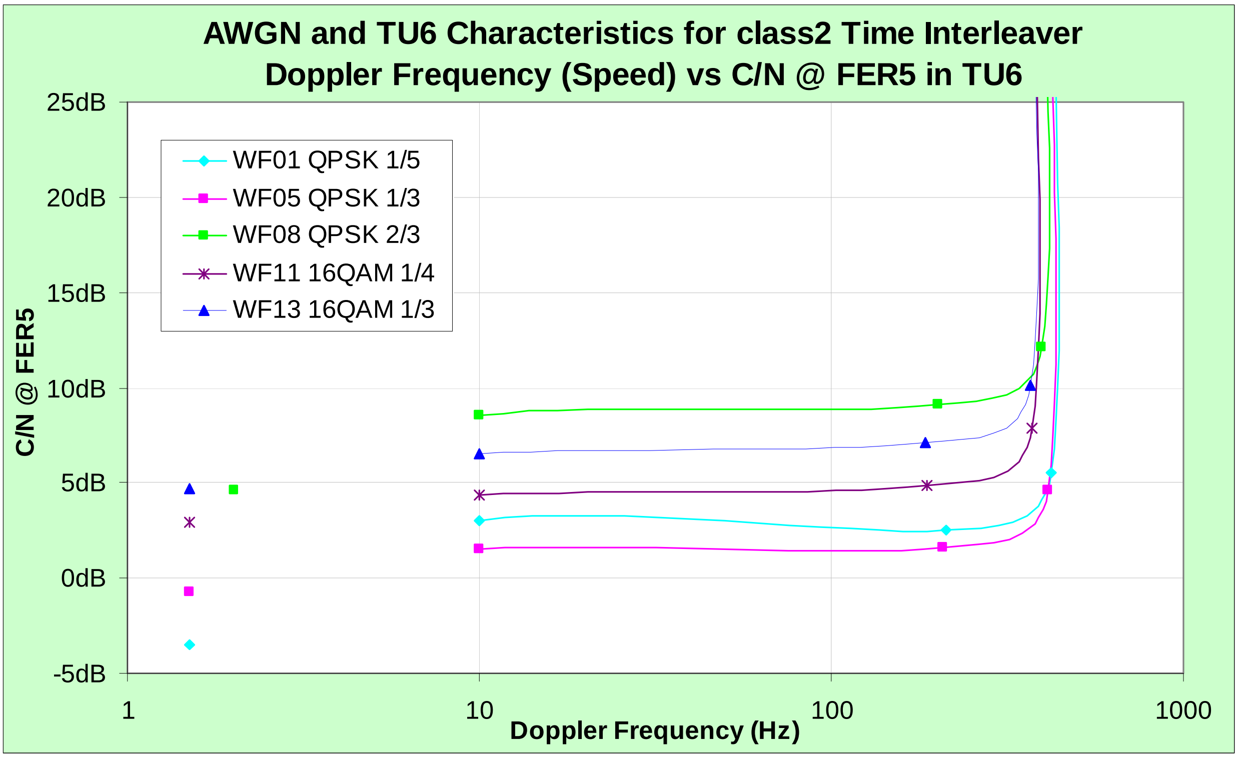


Figure A.13.7: Impact of long duration interleaving

Impact of interleaving length in class 2

Using the same demodulator chipset, various measurements have been performed for 5 MHz signal bandwidth, 2k FFT and GI =1/8; both QPSK 1/3 and 16QAM 1/3 are considered. The channel model is TU6 at 10 Hz (5 km/h). The different configurations used during the tests are described in table A.13.3 and C/N @ FER 5 % performances reported in figure A.13.8.

Table A.13.3: Configurations summary

-	Duration	Exact duration	Configuration	Receiver memory
16QAM 1/3	9 s	8 641 ms	8 0 1 0 57	247 MBits
	5 s	5 003 ms	6 0 1 0 44	143 MBits
	2,5 s	2 501 ms	6 0 1 0 22	72 MBits
	1 s	985 ms	4 0 1 0 13	28 MBits
	0,5 s	493 ms	2 0 1 0 13	14 MBits
	class 1	152 ms	8 48 1 0 0	4,3 MBits
QPSK 1/3	9 s	9 551 ms	4 0 1 0 63	137 MBits
	5 s	5 003 ms	3 0 1 0 44	72 MBits
	2,5 s	2 501 ms	3 0 1 0 22	28 MBits
	1 s	985 ms	2 0 1 0 13	14 MBits
	0,5 s	493 ms	1 0 1 0 13	7 MBits
	class 1	303 ms	8 48 1 0 0	4,3 MBits

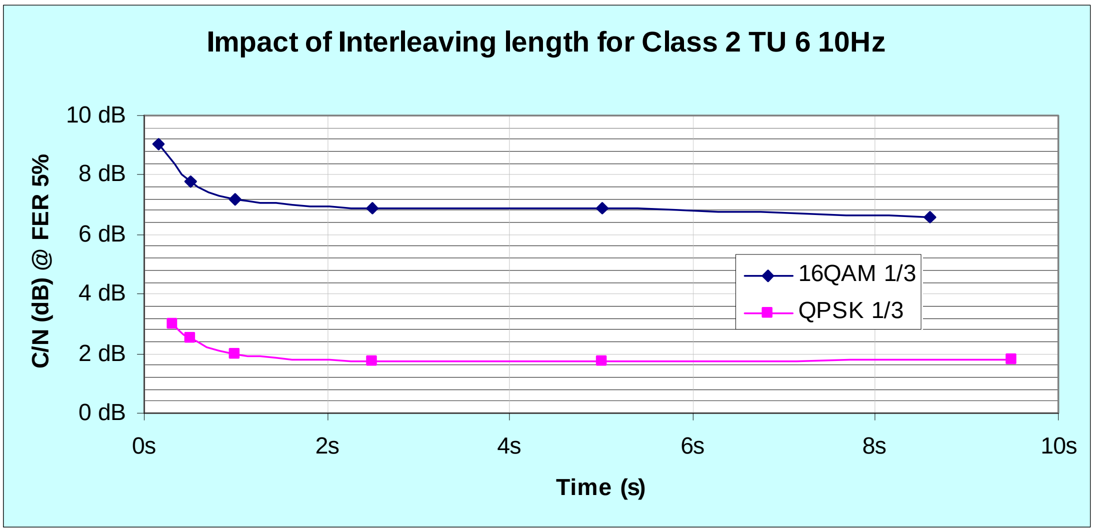


Figure A.13.8: Interleaving Length Impact

Main outcomes are the following:

- The overall long interleaver gain is 1,2 dB for QPSK and 2,5 dB for 16QAM.
- The minimum C/N value of 1,8 dB for QPSK is already reached for a 2,5 s interleaver length.
- For 16QAM at 2,5 s and 5 s, the required C/N is only 0,3 dB above the minimum value of 6,6 dB measured for an interleaver length of 9 s.
- For both mod-codes a 2,5 s long interleaver seems an optimal value although when shadowing is present on top of TU6 fading a longer interleaver size may be beneficial.

NOTE: The used configurations are not listed in clause A.10. This is because the measurements reported in this clause had the main scope to show the trends when suing different interleaving depths. The results are in line with other simulations results (not presented here) using the clause A.10 configurations.

A.13.1.1.4 Conclusions

The different measurements presented above show that, with a single exception, the different simulations are consistent with the measured performances during laboratory tests with real chipsets.

The introduction of a long interleaving at physical or link layer provides a real improvement at low mobile speed. Depending on the used configuration (Uniform Late and Long Uniform), the observed gain in TU6 channel is between 1 dB and 2 dB. At medium and higher speed, this gain is reduced at around 0,5 dB. In the presence of shadowing on top of TU6 fading the interleaver gain may be higher. The measurements show that the interleaving length could be reduced in TU6 to 1 s or 2 s, but at the price of losing signal under bridges or small tunnels. The impact of fading bandwidth on the (user speed) has been analysed in detail showing the good system resilience to Doppler fading bandwidth. For S-band and typical OFDM parameters the maximum mobile speed supported by commercial demodulators exceeds the standard technical requirements.

A.13.1.2 Satellite channels measurements results

A.13.1.2.1 Measurements with receiver 1

A.13.1.2.1.1 Configurations

For the satellite channels, different configurations have been tested:

- Class 1 with "standard" interleaving length, that is to say 132 ms for 16QAM and 265 ms for QPSK.
- Class 1 with MPE-IFEC at 70 % code rate.
- Class 2 with 9 s interleaving length.

The different propagation channels are the following (see clause A.7 for details):

- Sub-Urban (SU) at 50 km/h.
- Intermediate Tree Shadowing (ITS) at 50 km/h.

The different configurations are summarised in next table, with maximum burst duration is 200 ms.

Table A.13.4: LMS measurements configurations

BW (MHz)	Constellation	Code rate	FFT size	GI	Class 1 Interleaving (132/265 ms)	Class 2 Configuration (9 s)	IFEC Configuration
5	**QPSK**	1/5	2k	1/8	8/48/1/0/0	4/0/1/0/59	B=17, S=8, D=0, C=66, R=64 CR=2/3, 512 rows, Max Burst Duration: 200 ms
5	**QPSK**	2/9	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**QPSK**	1/4	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**QPSK**	2/7	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**QPSK**	1/3	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**QPSK**	2/5	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**QPSK**	1/2	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**QPSK**	2/3	2k	1/8	8/48/1/0/0	4/0/1/0/59	
5	**16QAM**	1/5	2k	1/8	8/48/1/0/0	8/0/1/0/59	B=17, S=8, D=0, C=118, R=64 CR=2/3, 768 rows, Max Burst Duration: 160 ms
5	**16QAM**	2/9	2k	1/8	8/48/1/0/0	8/0/1/0/59	
5	**16QAM**	1/4	2k	1/8	8/48/1/0/0	8/0/1/0/59	
5	**16QAM**	2/7	2k	1/8	8/48/1/0/0	8/0/1/0/59	
5	**16QAM**	1/3	2k	1/8	8/48/1/0/0	8/0/1/0/59	
5	**16QAM**	2/5	2k	1/8	8/48/1/0/0	8/0/1/0/59	
5	**16QAM**	1/2	2k	1/8	8/48/1/0/0	8/0/1/0/59	
5	**16QAM**	2/3	2k	1/8	8/48/1/0/0	8/0/1/0/59	

NOTE: The measurements are using configurations allowed by the 256 Mbits memory. The maximum obtained time interleaver length is 9,9 s. The different results were in line with simulations results and consistent with other receiver results when using equivalent channel emulator. This gain justifies the use of alternate configurations.

A.13.1.2.1.2 Examples of Results

We focus in this clause on the results corresponding to the configurations defined in clause A.10.

- QPSK 1/3 Long Uniform.
- QPSK 1/2 Short plus IFEC.
- 16QAM 1/4 Long Uniform.
- 16QAM 2/5 Short plus IFEC.
- 16QAM 1/3 Long Uniform.
- 16QAM 1/2 Short plus IFEC.

Figures A.13.9 and A.13.10 provide performances in SU and ITS 50Km/h for QPSK and an overall equivalent coding rate of 1/3:

- Class 2 corresponds to QPSK 1/3 Long Uniform.
- Class 1 corresponds to QPSK 1/3 Short.
- Class 1 plus IFEC corresponds to QPSK 1/2 plus IFEC.

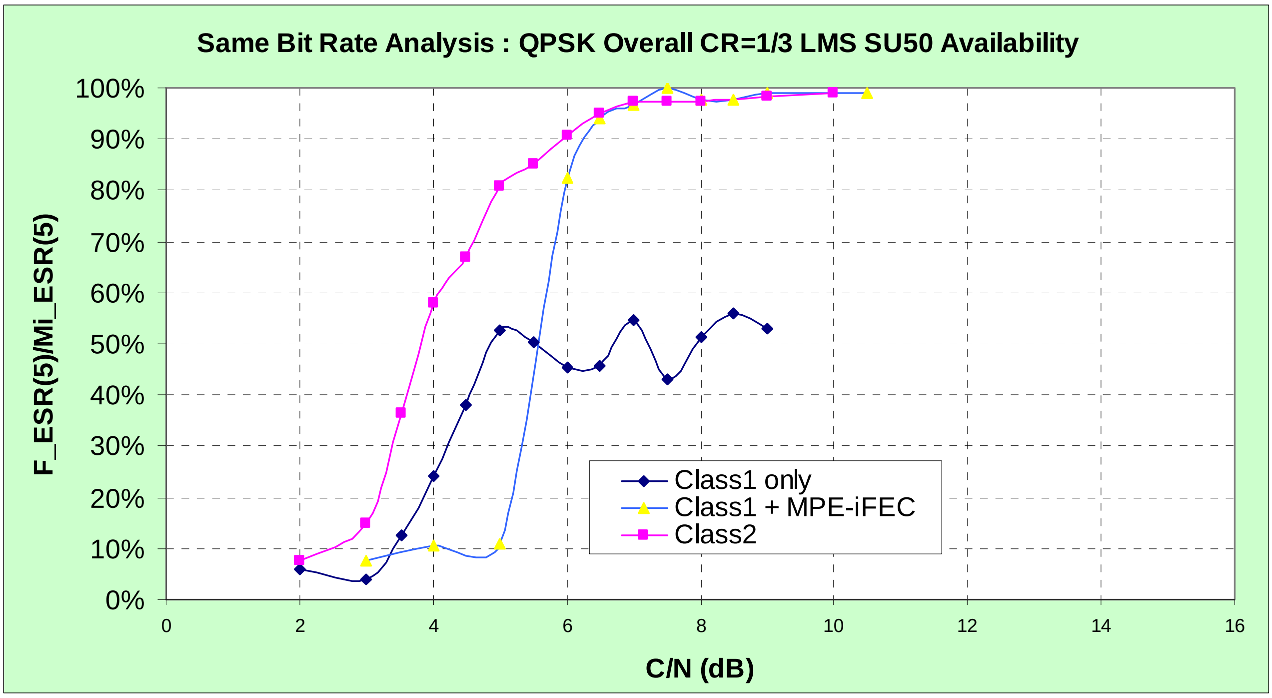


Figure A.13.9: QPSK 1/3 equivalent LMS SU 50 results

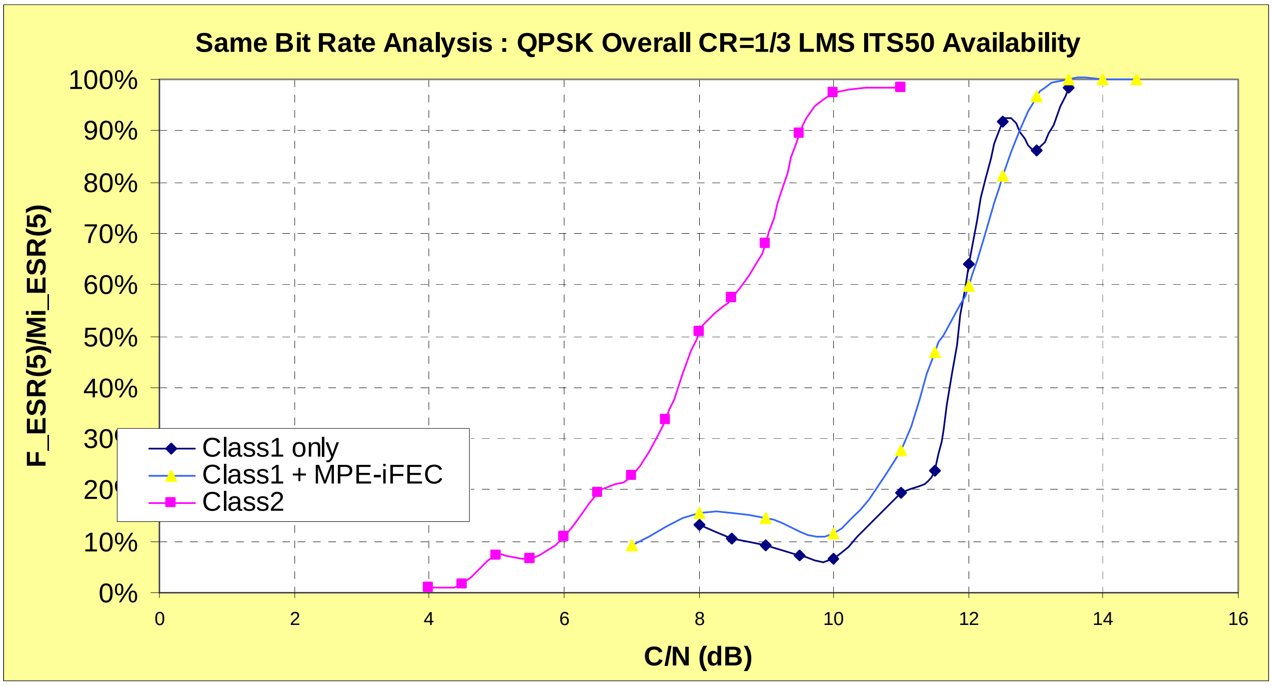


Figure A.13.10: QPSK 1/3 equivalent LMS ITS 50 results

Figures A.13.11 and A.13.12 correspond to 16QAM and an overall equivalent coding rate of 1/4:

- Class 2 corresponds to 16QAM 1/4 Uniform.
- Class 1 corresponds to 16QAM 1/4 Short.
- Class 1 plus IFEC corresponds to 16QAM 2/5 plus IFEC.

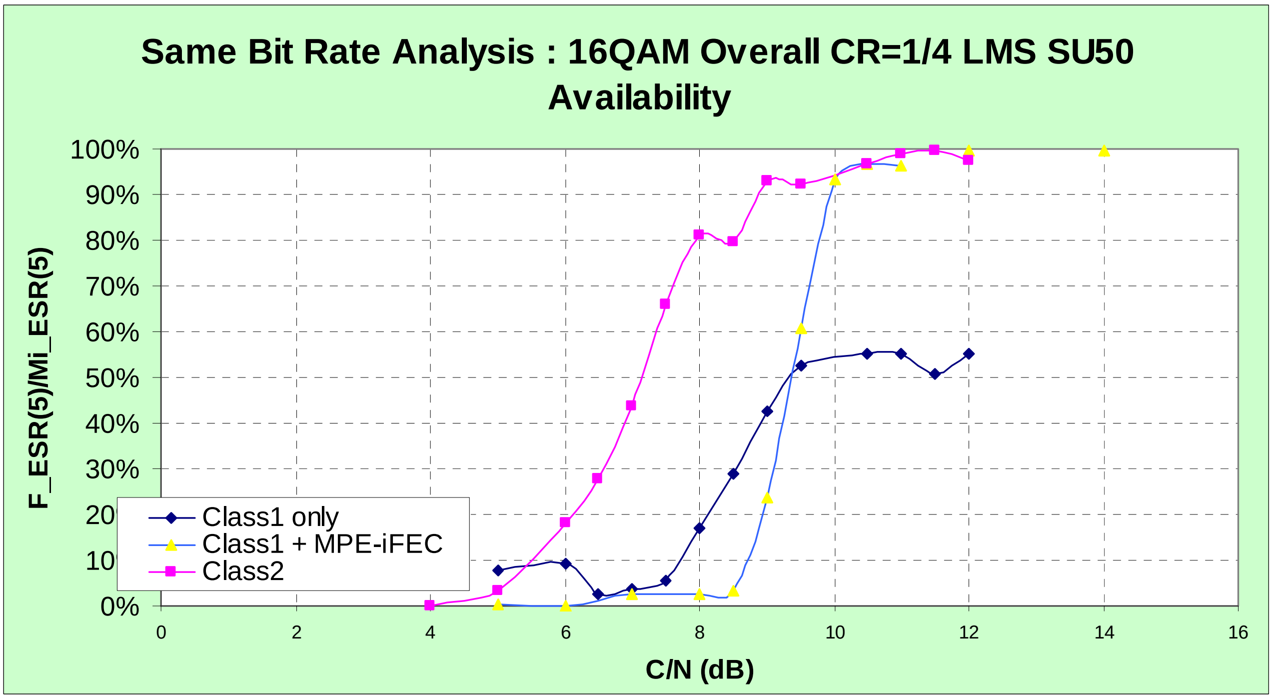


Figure A.13.11: 16QAM 1/4 equivalent LMS SU 50 results

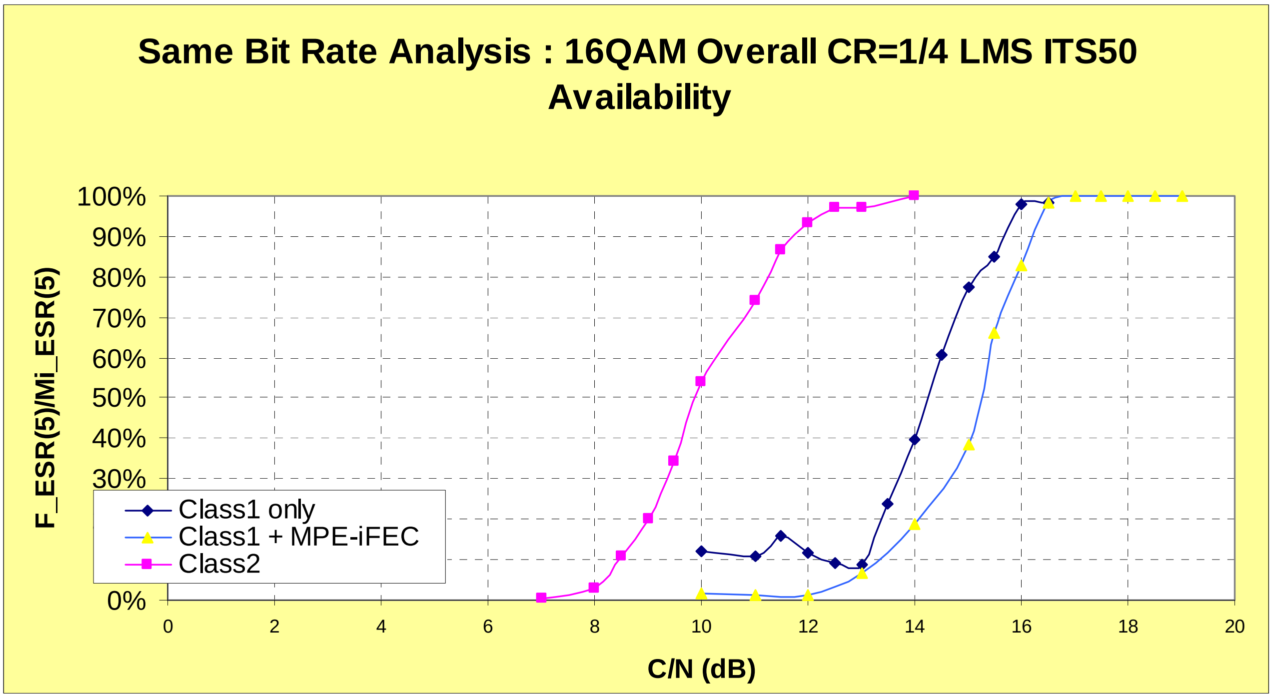


Figure A.13.12: 16QAM 1/4 equivalent LMS ITS 50 results

Figures A.13.13 and A.13.14 correspond to 16QAM and an overall equivalent coding rate of 1/3:

- Class 2 corresponds to 16QAM 1/4 Uniform.
- Class 1 corresponds to 16QAM 1/3 Short.
- Class 1 plus IFEC corresponds to 16QAM 1/2 plus IFEC.

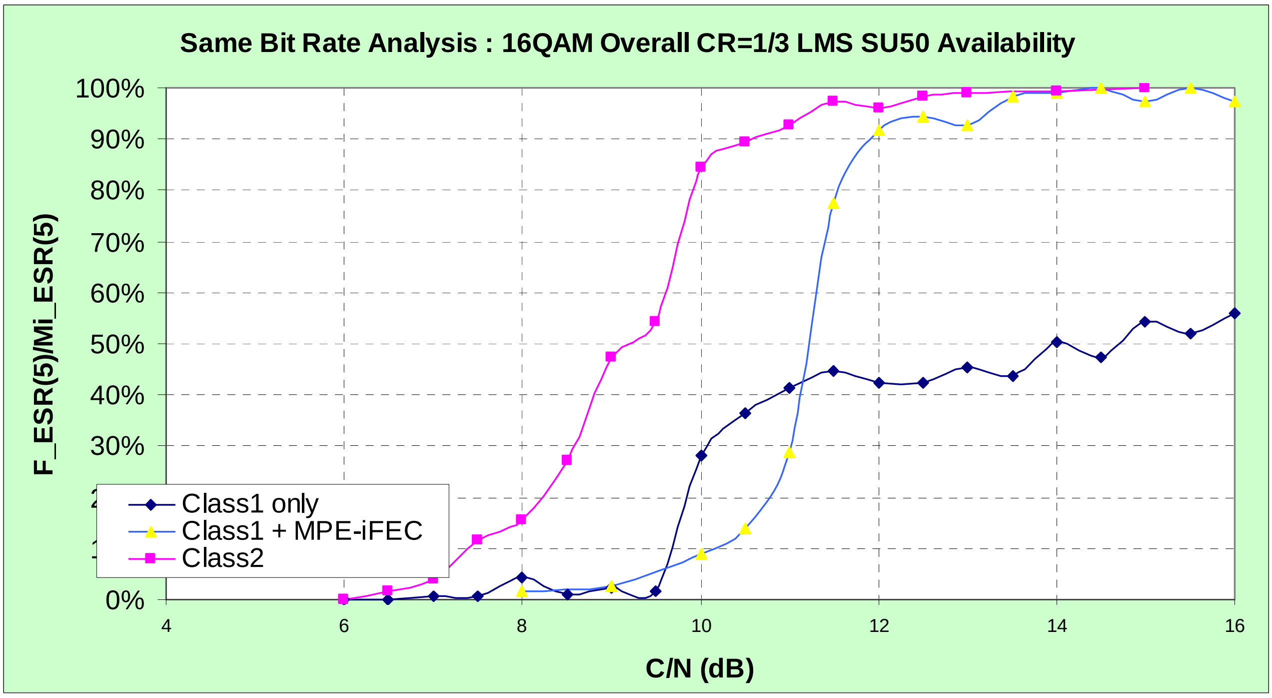


Figure A.13.13: 16QAM 1/3 equivalent LMS SU 50 results

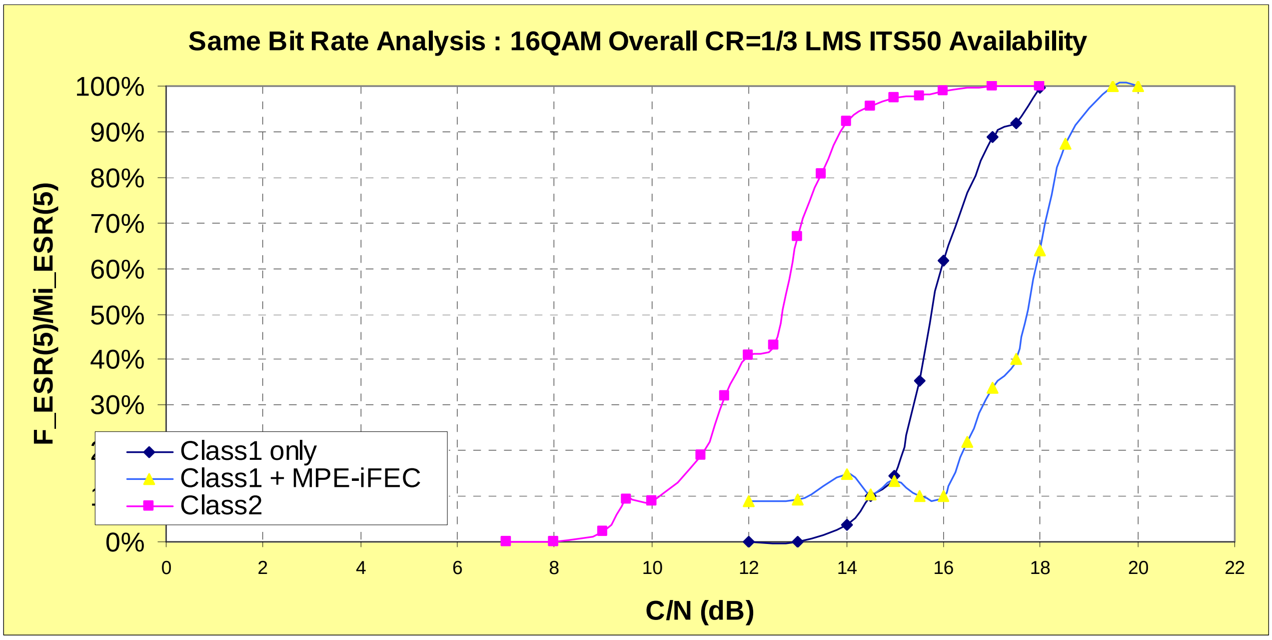


Figure A.13.14: 16QAM 1/3 equivalent LMS ITS 50 results

A.13.1.2.1.3 Conclusions

When comparing the measurement results reported above to the simulation findings in clause A.12, we have the following results for SH-A (OFDM) class 2 (long time interleaver):

- QPSK 1/3 ITS 50 km/h: in clause A.12 the required simulated C/N is 8,5 dB for ESR5(20) 90 % fulfilment without implementation losses, versus 9,5 dB measured in laboratory tests.
- QPSK 1/3 SU 50 km/h: in clause A.12 the required simulated C/N is 5 dB for ESR5(20) 90 % fulfilment without implementation losses, versus 6 dB in laboratory tests.
- 16QAM 1/5 50 km/h: in the IG the required simulated C/N is 10,8 dB for ESR5(20) 90 % fulfilment without implementation losses, versus 11 dB in the laboratory tests.

We can therefore conclude that there is a good matching between the clause A.12 simulations results and the laboratory measurements. Implementation losses over LMS channels are 1 dB or less.

In the case of SH-A class 1 configurations with MPE-IFEC, there are some discrepancies between simulations and laboratory tests results due to the different configurations adopted.

From the previous curves above, we can derive the following conclusions:

- Whatever the modulation and coding, in sub-urban environment, there is a need for long interleaving scheme: at physical or at link layer.
- In sub-urban channel, class 2 gain is about 1 dB compared to class 1+ MPE-IFEC.
- In ITS channel, class 2 performs always better (2 dB or more) than class 1 or class 1 plus IFEC.
- IT appears that class 1 alone has comparable performances with class 1 + IFEC, but in pure ITS channel: if the vehicle goes under a highway bridge, class 1 will be useless, and it is thus generally preferable to have long interleaving i.e. class 1 + IFEC.

NOTE: All the previous results have been obtained with a satellite channel emulator named CE1 implementing the 3-state Perez-Fontan LMS channel model described in clause A.7 with some approximation. Different results were obtained with a channel emulator named CE2 fully compliant with clause A.7 model. The measurements have been performed with QPSK 1/3 and 16QAM 1/3. The different channel emulators are referenced CE1 and CE2.

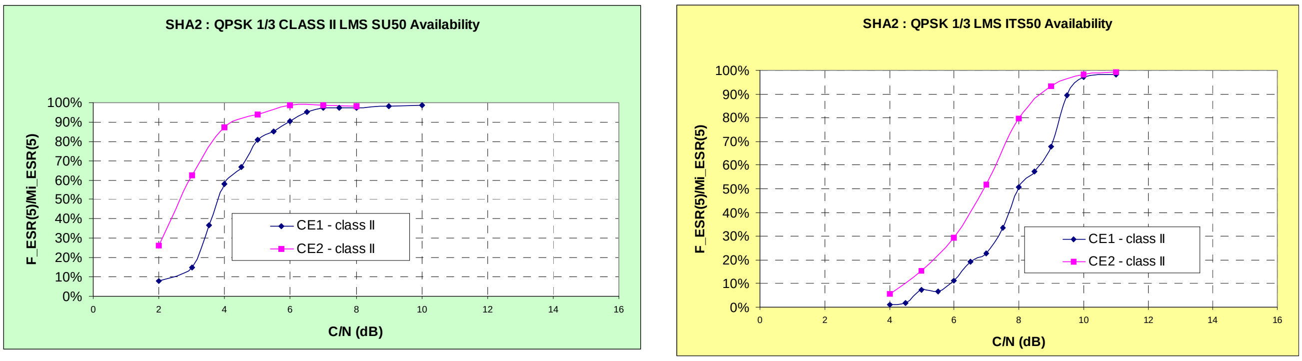


Figure A.13.15: Comparison of QPSK 1/3 performances in SU and ITS with CE1 and CE2 channel emulators

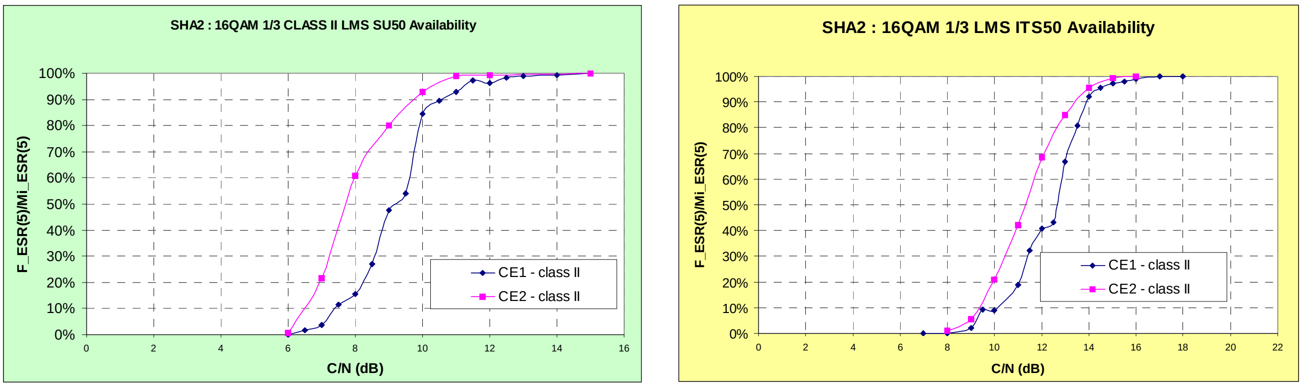


Figure A.13.16: Comparison of 16QAM 1/3 performances in SU and ITS with CE1 and CE2 channel emulators

Considering all cases, the curves show between 0,8 dB and 1,7 dB of performance difference in favour of CE2. We can conclude that the results reported in this clause are slightly pessimistic.

A.13.1.2.2 Measurements with receiver 2

A.13.1.2.2.1 Configurations

The second receiver has been used for both SH-B (TDM) and SH-A (OFDM) waveforms under LMS channel conditions. Only class 2 with long time interleaving has been considered.

Table A.13.5: Measurements configurations for Receiver 2

BW (MHz)/ Roll off	Constellation	Code rate	FFT	GI	Class 2 Configuration (9 s)
OFDM					
5	**QPSK**	1/3	2k	1/4	40/0/12/4/2
5	**16QAM**	1/3	2k	1/4	40/0/12/8/4
TDM					
5/0,15	**QPSK**	1/3	N/A	N/A	40/0/12/4/2
5/0,15	**16 APSK**	1/3	N/A	N/A	63/0/12/4/2

The measurements were made in ITS and SU channels at 50 km/h during 3 000 s. The channel emulator used in this case (named CE3) is closely following the LMS model described in clause A.7.

A.13.1.2.2.2 Results

First, the results are given **for TDM.**

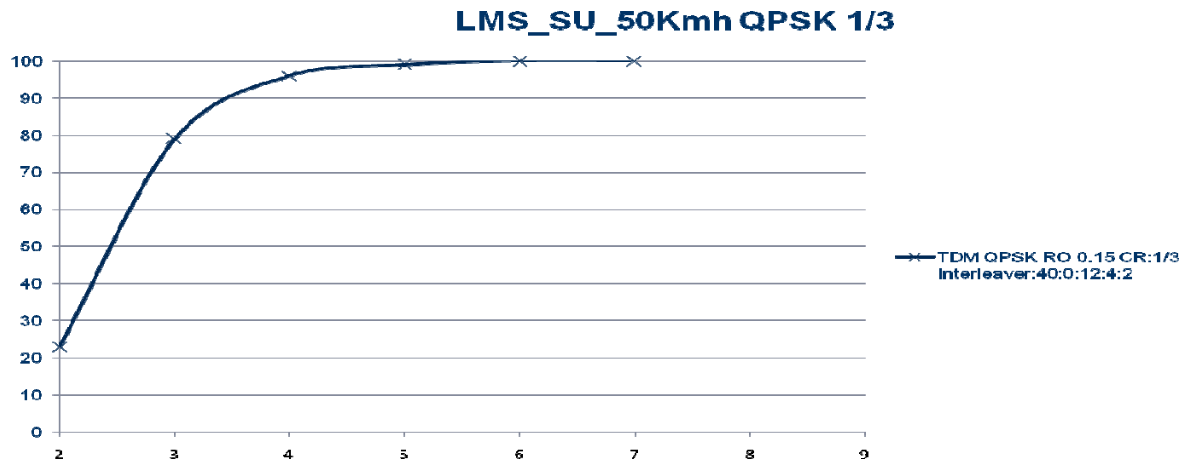


Figure A.13.17: TDM QPSK 1/3 results in LMS-SU 50 km/h

The ESR5(20) 90 % is obtained at C/N = 3,5 dB.

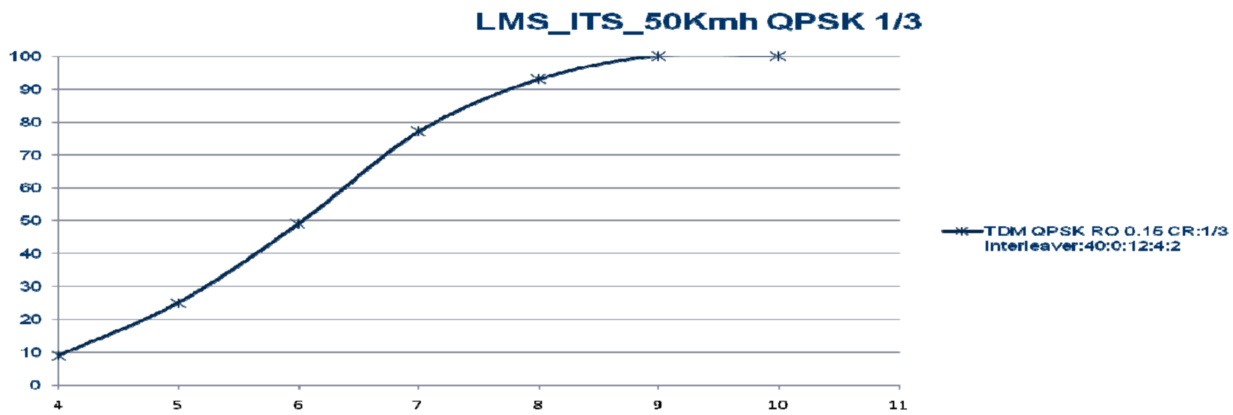


Figure A.13.18: TDM QPSK 1/3 results in LMS-ITS 50 km/h

The ESR5(20) 90 % is obtained at C/N = 7,7 dB.

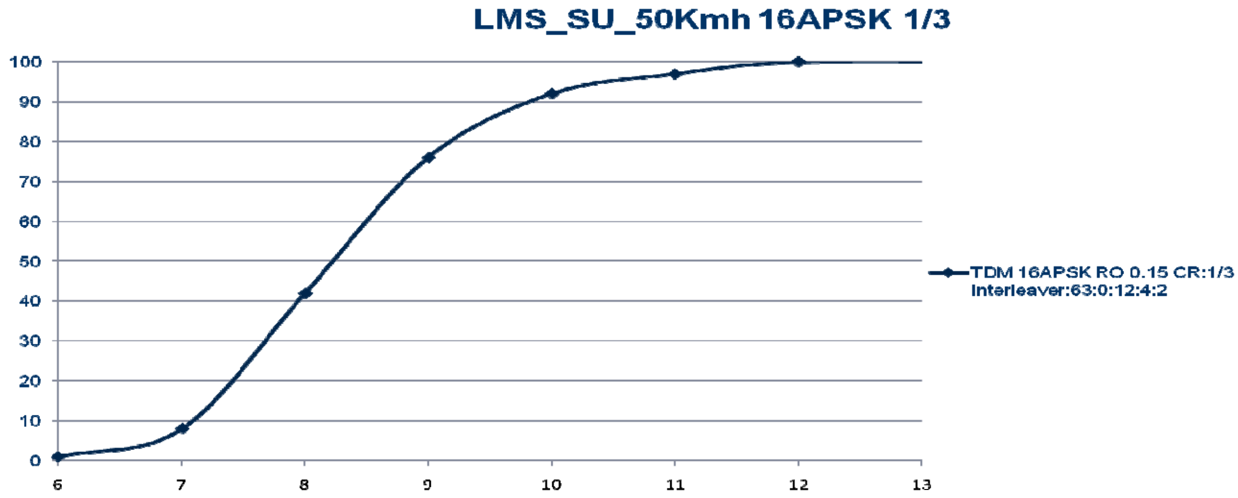


Figure A.13.19: TDM 16 APSK 1/3 results in LMS-SU 50 km/h

The ESR5(20) 90 % is obtained at C/N = 9,5 dB.

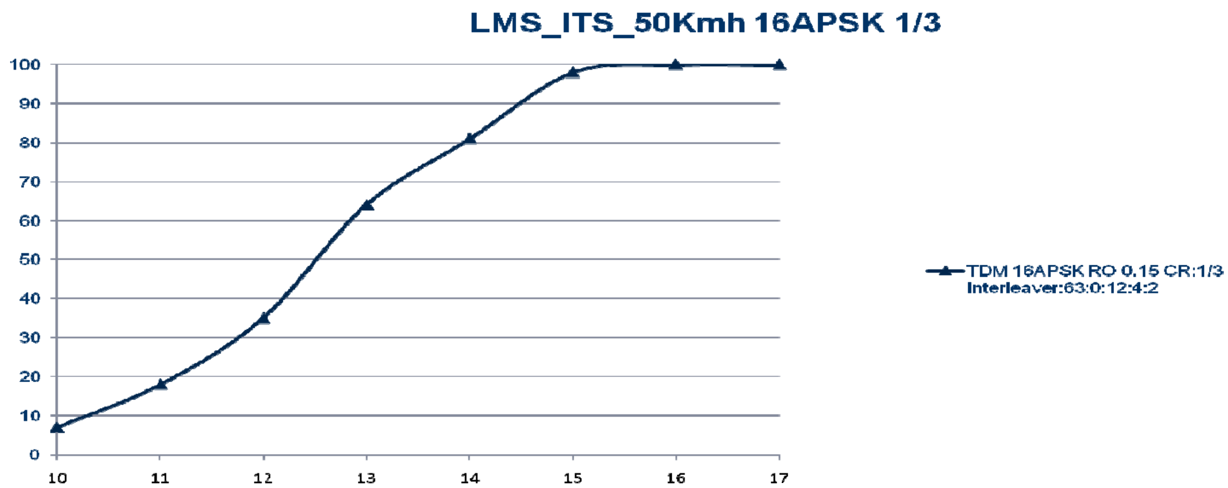


Figure A.13.20: TDM 16 APSK 1/3 results in LMS-ITS 50 km/h

The ESR5(20) 90 % is obtained at C/N = 14,5 dB.

For OFDM

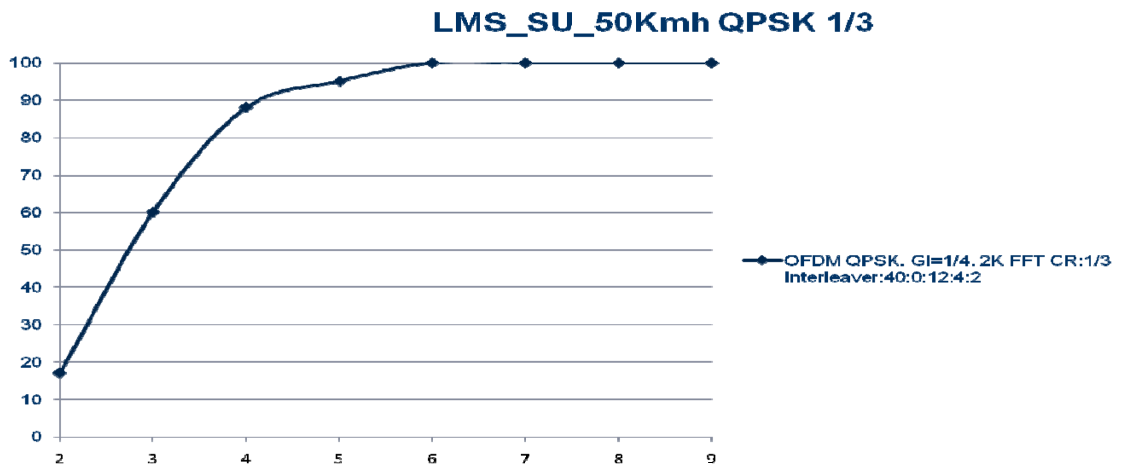


Figure A.13.21: OFDM QPSK 1/3 results in LMS-SU 50 km/h

The ESR5(20) 90 % is obtained at C/N = 4 dB.

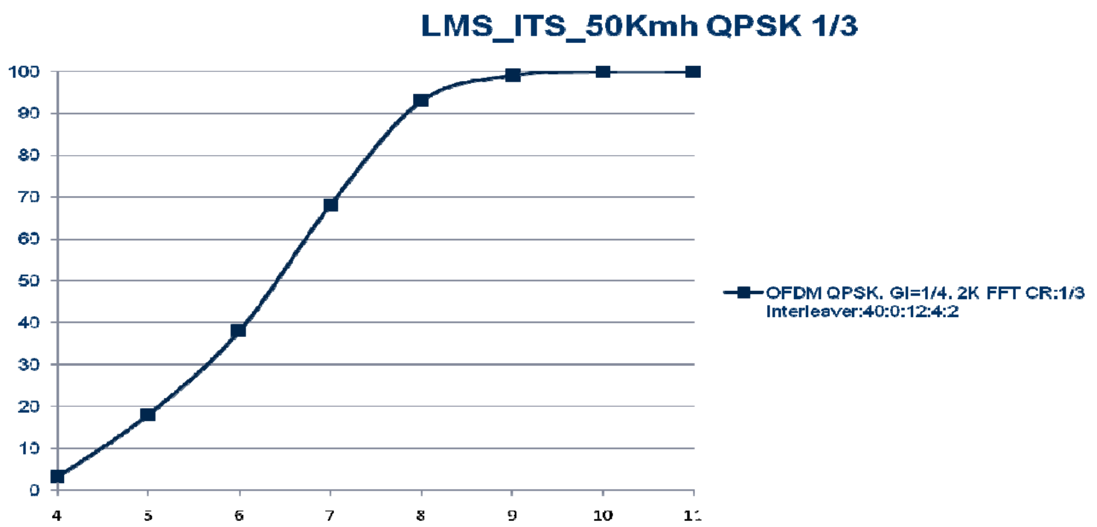


Figure A.13.22: OFDM QPSK 1/3 results in LMS-ITS 50 km/h

The ESR5(20) 90 % is obtained at C/N = 7,8 dB.

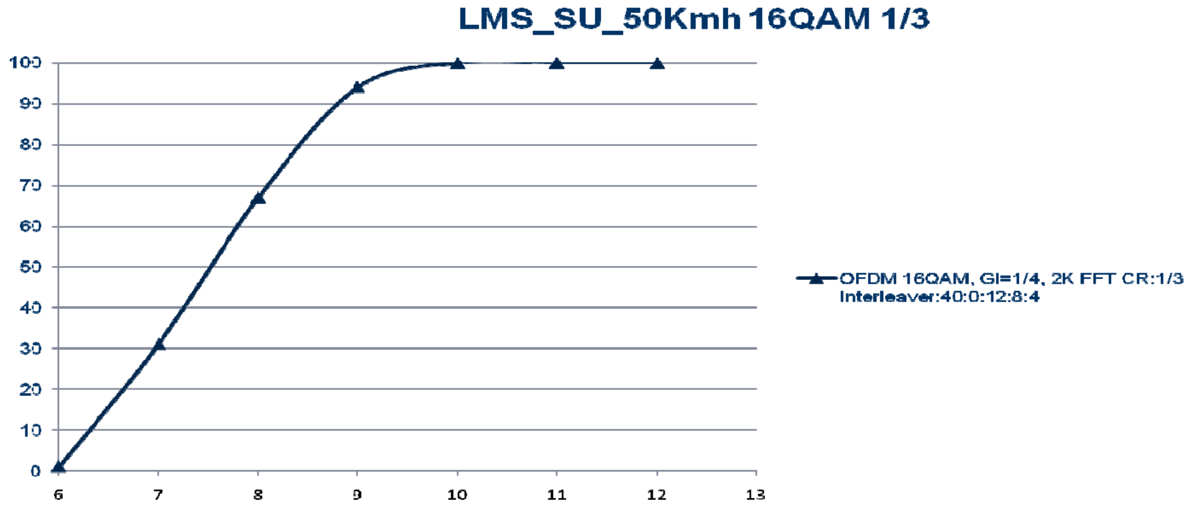


Figure A.13.23: OFDM 16QAM 1/3 results in LMS-SU 50 km/h

The ESR5(20) 90 % is obtained at C/N = 8,7 dB.

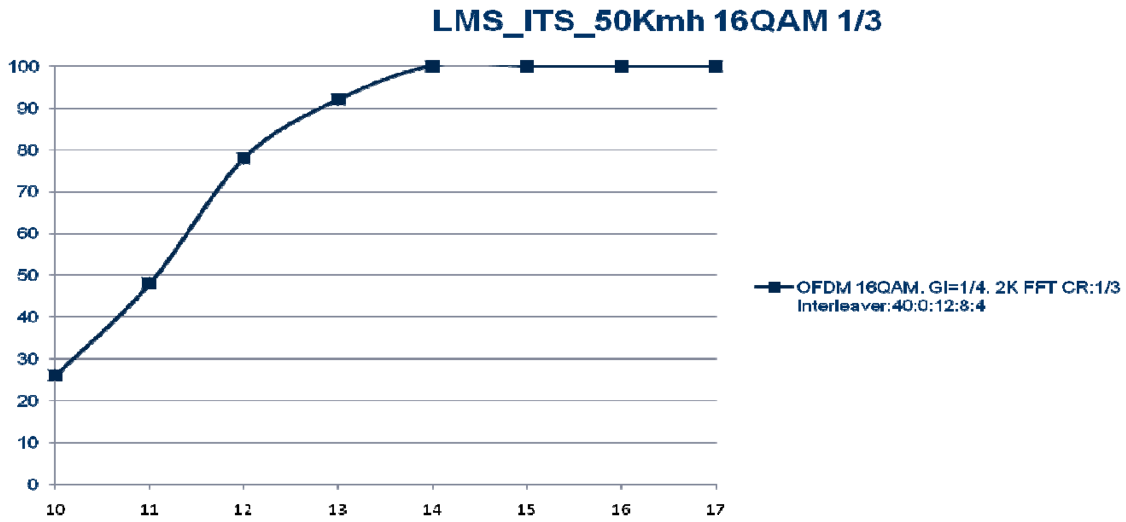


Figure A.13.24: OFDM 16QAM 1/3 results in LMS-ITS 50 km/h

The ESR5(20) 90 % is obtained at C/N = 13 dB.

A.13.1.2.2.3 Conclusions on receiver 2

The receiver 2 allows a comparison between TDM and OFDM, as the measurements have used the same channel emulator.

Table A.13.6: Summary of required C/N (dB) @ ESR5(20) 90 % with receiver 2

Receiver/Channel Emulator	TDM		OFDM	
Modulation/coding	**SU-50**	**ITS-50**	**SU-50**	**ITS-50**
QPSK 1/3	3,5	7,8	4	7,9
16QAM or 16 APSK 1/3	9,5	14,5	8,7	13

If we compare TDM and OFDM, tested with the same channel emulator:

- QPSK 1/3 in both TDM and OFDM waveforms have practically the same performances in SU and ITS.

- 16QAM 1/3 and 16 APSK show some differences: OFDM performs better, especially in ITS (1,5 dB) but the interleaver configurations are not the same.

A.13.1.2.3 Conclusions

Table A.13.7 gives a comparison in OFDM of the different receivers with different channel emulators.

Table A.13.7: Comparison between different receivers and channel emulators in LMS and OFDM waveform for ESR5(20) 90 % fulfilment

Receiver/Channel Emulator	Receiver 1/CE1		Receiver 1/CE2		Receiver 2/CE3	
Modulation/coding	**SU-50**	**ITS-50**	**SU-50**	**ITS-50**	**SU-50**	**ITS-50**
QPSK 1/3	6	9,5	4,3	8,7	4	7,9
16QAM 1/3	10,5	14	9,7	13,2	8,7	13
NOTE: CE is for Channel Emulator. 1,2,3 refers to the type of Channel Emulator.						

Observing the measurement results summarized in table A.13.7 we can derive the following conclusions:

- With QPSK 1/3 the differences between Receiver 1/CE1 and Receiver 2/CE3 are of 2 dB in SU and 1,6 dB in ITS.
- These differences reduces to 0,3 dB for SU and 1,2 dB in ITS when using Receiver 1/CE2 (which tends to prove that CE2 and CE3 are based on the same LMS channel model).
- With 16QAM 1/3, the differences between Receiver 1/CE1 and Receiver 2/CE3 are of 1,8 dB in SU and 1 dB in ITS.
- These differences are reduced to 1 dB in SU, and 0,3 dB in ITS when using Receiver 1/CE2 (which tends to prove that CE2 and CE3 are based on the same LMS channel model.

Considering that CE2 and CE3 are based on same LMS channel model, the two receivers perform approximately the same with OFDM in LMS channels (around 1 dB maximum difference). The difference is likely to be related to the different channel estimation algorithms implemented.

Receiver 2 with CE3 show quite comparable performances in TDM and OFDM for LMS channels when QPSK modulation is used. For 16APSK/16QAM modulation the results are slightly better for OFDM compared to TDM.

Concerning comparison with simulations results from clause A.12 with reference receiver channel estimation, we have table A.13.8.

Table A.13.8: Results comparison with simulations results

Receiver/Channel Emulator	Receiver 1 CE2		Receiver 2 CE3		Simulations with channel estimation	
Modulation/coding	**SU-50**	**ITS-50**	**SU-50**	**ITS-50**	**SU-50**	**ITS-50**
QPSK 1/3 OFDM	4,3	8,7	4	7,9	3,7	9,2
QPSK 1/3 TDM	N/A	N/A	3,5	7,8	3,7	8

Conclusions are as follows:

- OFDM Receiver 1 with CE2 is in line with the simulations.
- OFDM, Receiver 2 is in line for SU channel, but better with ITS channel simulations. The difference is likely to be related to the faster reacquisition time of the Receiver 2 then what modelled in the simulations.
- In TDM, receiver 2 is still slightly better than the simulations. The difference is likely to be related to the faster reacquisition time of the Receiver 2 then what modelled in the simulations.

A.13.1.3 Hybrid channels

A.13.1.3.1 Configurations

The Hybrid Channel tests aim to address hybrid situations when the on-air signal is a combination of terrestrial and satellite signals. This situation is particularly interesting when the satellite and the terrestrial signals are both using OFDM at the same frequency. This mode is called DVB-SHA SFN mode in the standard.

Hybrid SFN channel is defined by the combination at the receiver of:

- One satellite link under ITS or SU conditions.
- One or several terrestrial links under TU6 conditions.

In the case of Hybrid SFN, the two signals are at the same frequency, using the same modulation/coding/interleaving schemes, and carrying exactly the same data. In this section, only SHA-SFN configurations have been retained from simulations and laboratory tests:

- OFDM over satellite and terrestrial at same frequency bands.
- A single terrestrial repeater is emulated.

The objectives of the measurements are:

- Assess the improvement of system coverage beyond the edge of the terrestrial cell coverage where the received C/N is below the required threshold.
- Assess the possible existence of interference when satellite and terrestrial signals are not within the GI window and the impact of these interferences.
- Assess the influence of the GI on the interference reduction when the interferences are existing.

Taking into account the above objectives it will not be necessary to test the following "weak" hybrid situations:

a) A very high terrestrial C/N (for instance 20 dB): this terrestrial signal will be very "dominant" and no satellite component impact will be observed.

b) A very low terrestrial C/N as also no terrestrial component impact will be observed being the satellite signal dominant.

The different tests conditions are the following:

- Waveform baseline configurations:
  - 5 MHz/2k/GI 1/8 or 1/4.
  - Long Interleaving : class 2 and class 1 plus IFEC.
- Satellite channels configurations:
  - ITS 50 km/h.
  - SU 50 km/h.
- Terrestrial network configurations:
  - 2 kW EIRP single repeater.
  - Different repeater heights: 30 m, 45 m and 60 m.
  - TU 6 channel.
  - Use of Costa Hata model in SU and Rural conditions.

- The Receiver is a car receiver, single antenna with satellite oriented receiver (circular polarized antenna) as most of the satellite reception area will be on roads and highways, and is located at the point corresponding to the terrestrial C/N received value.

The operating mode for the simulations is the following:

- Based on the terminal location and repeater characteristics, terrestrial C/N is settled at the wanted value, which corresponds to a certain differential delay.
- The satellite received C/N is settled at the required value to comply with the ESR5(20) fulfilment ratio of 90 %.
- Then compute the ESR5 with the CGC repeater on.
- Observe if there is a performance improvement or degradation.

The basic principles of the laboratory tests are the following:

- Take the reference curve in LMS (ITS or SU) satellite reception with class 2 and/or IFEC.
- Set the terrestrial C/N to the wanted value, which corresponds to a certain differential delay.
- Perform the hybrid configuration. tests for different satellite received C/N.
- Compare the ESR5 fulfilment curve with the LMS reference curve mentioned before.
- Observe if there is a performance improvement or degradation.

As said above, the tests will be performed with class 2 and class 1 plus IFEC receiver configurations. In class 2, only QPSK 1/3 and 16QAM 1/3 configurations have been tested. For class 1 plus IFEC, the following configuration has been tested: 16QAM ½ at physical layer plus MPE-IFEC 70 % with GI = 1/8.

In a hybrid channel, there are three main parameters to consider:

- The received terrestrial C/N.
- The received satellite C/N.
- The difference of time of arrival of satellite and terrestrial signals, called here Delta T (ΔT). In the test, the satellite signal C/N is varying for each couple: C/N terrestrial/Delta T corresponding to mobile position.

The different couples are provided here below for each repeater configuration. To limit the number of cases, terrestrial C/N is comprised between -10 dB and + 10 dB (5 values).

Tables A.13.9 to A.13.11 have been used for class 2.

Table A.13.9: Configurations for 30 m

Repeater: 30 m	C/N (dB) terrestrial	Delta T (µs)	To GI=1/8 (44,8 µs)	To GI =1/4 (89,6 µs)	Distance/Rep (km)
Sub-Urban Terr/ LMS-SU 50	-10	57,07	127 %	64 %	9,4
	-5,1	40,82	91 %	46 %	6,8
	0	28,35	63 %	32 %	4,9
	4,3	19,79	44 %	22 %	3,7
	10,4	12,3	27 %	14 %	2,5
Rural-Terres/ LMS-ITS 50	-9,6	107	239 %	119 %	17,5
	-5,2	80,26	179 %	90 %	13,1
	0	57,07	127 %	64 %	9,4
	4,9	40,82	91 %	46 %	6,8
	10	6,99	16 %	8 %	4,9

Table A.13.10: Configurations for 45 m

Repeater: 45 m	C/N (dB) terrestrial	Delta T (µs)	To GI=1/8 (44,8 µs)	To GI =1/4 (89,6 µs)	Distance/Rep (km)
Sub-Urban Terr/ LMS-SU 50	-10,2	73,60	164 %	82 %	12
	-4,7	48,24	108 %	54 %	8,3
	0,6	34,49	77 %	39 %	5,8
	4,6	25,38	57 %	29 %	4,4
	9,7	17,23	38 %	19 %	3,4
Rural-Terres/ LMS-ITS 50	-9,8	140,57	317 %	157 %	17,5
	-4,8	100,30	224 %	112 %	15,9
	-0,2	73,60	164 %	82 %	12
	5,3	48,24	108 %	54 %	8,3
	9,4	25,38	57 %	29 %	4,4

Table A.13.11: Configurations for 60 m

Repeater: 60 m	C/N (dB) terrestrial	Delta T (µs)	To GI=1/8 (44,8 µs)	To GI =1/4 (89,6 µs)	Distance/Rep (km)
Sub-Urban Terr/ LMS-SU 50	-7,6	73,60	164 %	82 %	12
	-2,2	48,24	108 %	54 %	8,3
	3	34,49	77 %	39 %	5,8
	6,8	25,38	57 %	29 %	4,4
	10,8	17,23	38 %	19 %	3,4
Rural-Terres/ LMS-ITS 50	-6,9	140,57	314 %	157 %	17,5
	-2	100,30	224 %	112 %	15,9
	2,4	73,60	164 %	82 %	12
	7,8	48,24	108 %	54 %	8,3
	11,8	25,38	57 %	29 %	4,4

For class 1 plus IFEC, different configurations have been tested. They are based on the DVB-SSSP Validation Task Force. In this configuration, the delta T is varying from 0 % of the GT to 200 % by steps of 50 %, and the terrestrial C/N has the following values in dB: -10, - 6, -3, 0, 3, 6, 10.

The satellite received C/N in line of sight is assumed to be 12 dB and is fixed.

A.13.1.3.2 Results

A subset of the results contained in this clause have already been presented in the clause 7. Figures A.13.25 to A.13.31 represent the required C/N in satellite channel when in presence of terrestrial channel to get 90 % ESR5(20). All possible repeater heights have been considered. The horizontal line represents the C/N required in LMS channel only. There are two cases: one with GI 1/8 and the second one with GI 1/4. We consider first the cases with SHA-SFN class 2 terminals.

A.13.1.3.2.1 Suburban channels

Two cases are provided as examples:

a) QPSK 1/3 at 50 km/h.

b) 16QAM 1/3 at 50 km/h.

The satellite channel emulator is set in the SU configuration and the speed is 50 km/h. The terrestrial channel emulator is set at TU6 50 km/h, with the required C/N value.

For a 30 m terrestrial repeater height, the following results have been obtained.

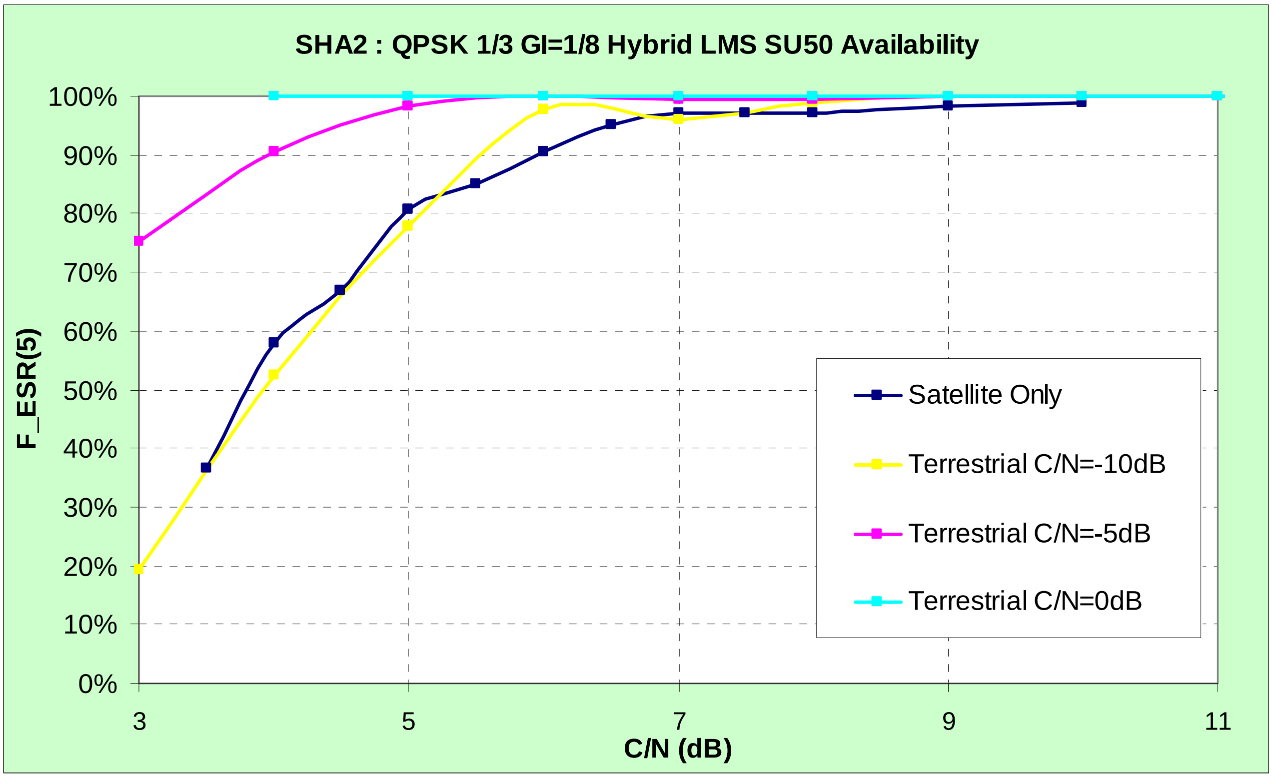


Figure A.13.25: Performances in sub Urban Areas with QPSK rate 1/3 @ 30 m

As the curve corresponding to C/N = 0 dB already shows a constant ESR5(20) of 100 %, the two other cases corresponding to C/N equal to 5 dB and 10 dB are not represented.

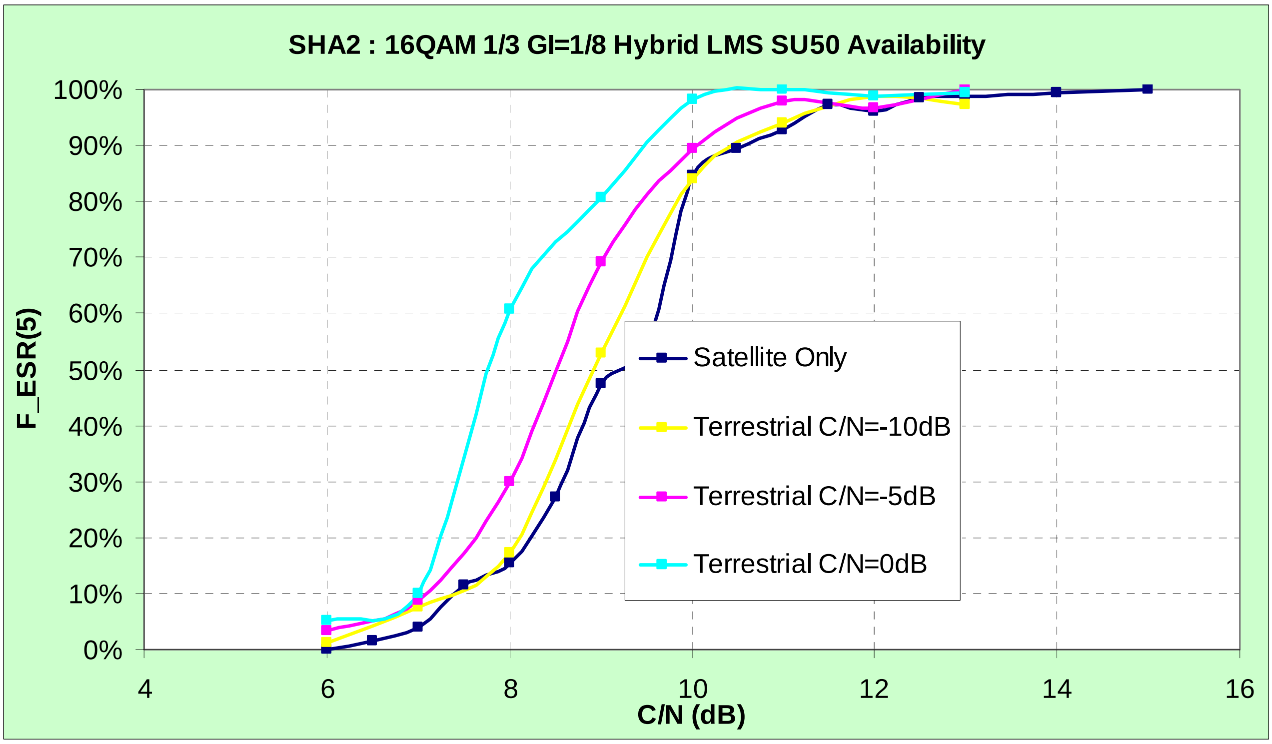


Figure A.13.26: Performances in sub Urban Areas with 16QAM rate 1/3 @ 30 m

From this example derive the following conclusions can be derived:

- Most of the hybrid reception cases provide improvement compared to the satellite only reception as the required C/N is decreasing.
- When the terrestrial signal is under the *C/N* threshold (for instance C/N = 0 dB in QPSK while 2 dB are required), the resulting quality of service with satellite on (and even with very low satellite C/N values) is above 99 % ESR5.
- With 16QAM, even the case C/N = 0 dB provides improvement to the satellite channel.

With 45 m terrestrial repeater height, the following results have been obtained.

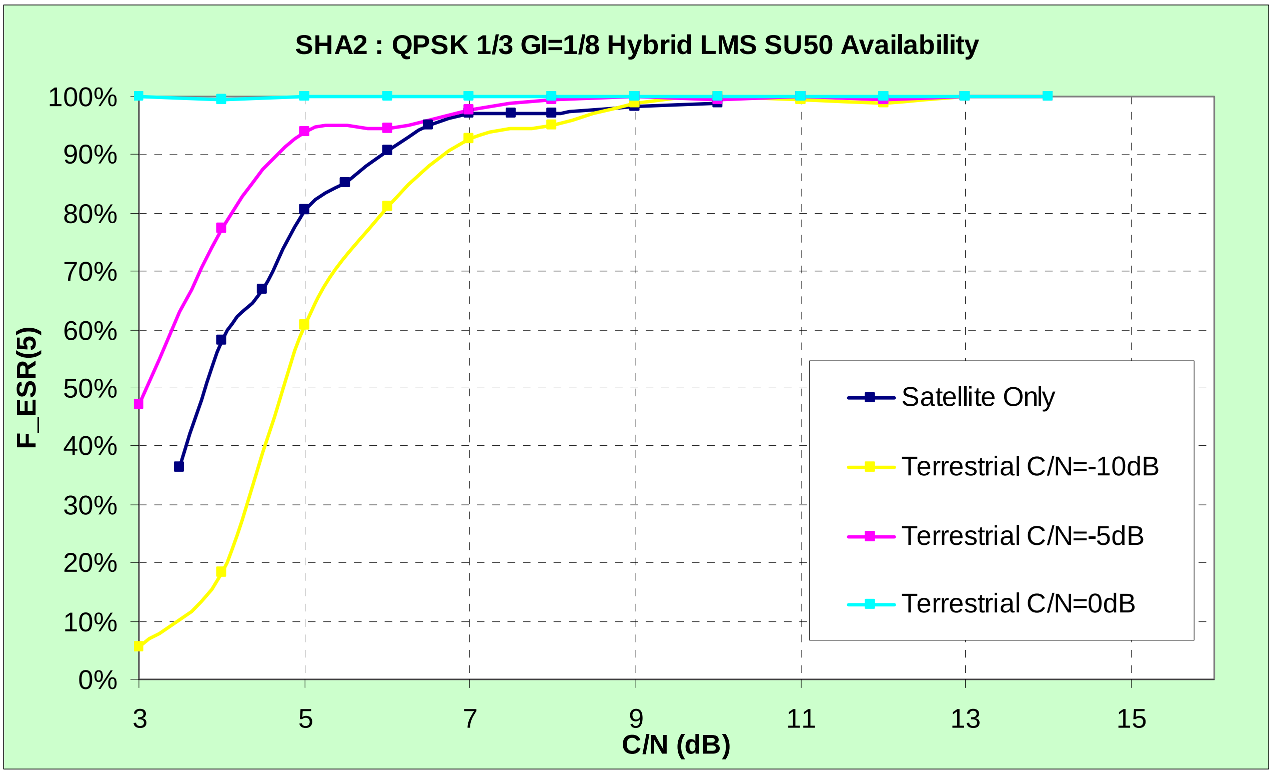


Figure A.13.27: Performances in sub Urban Areas with QPSK rate 1/3 @ 45 m

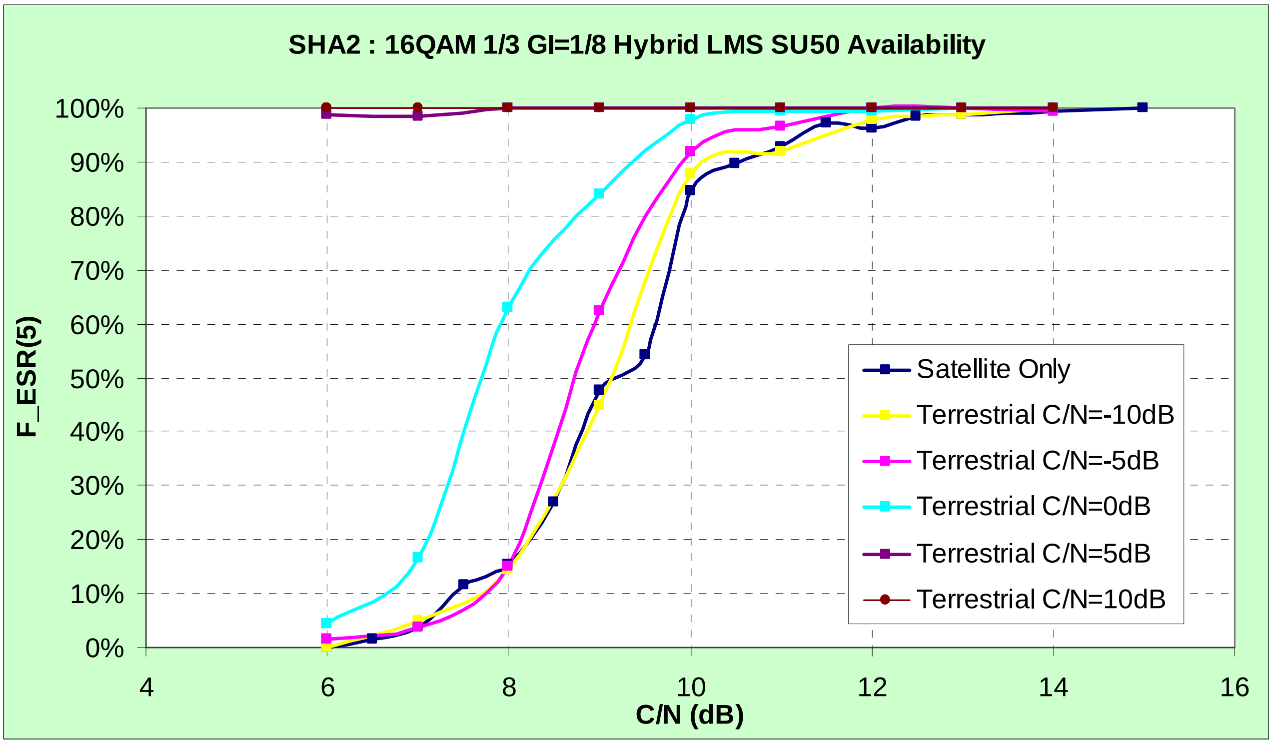


Figure A.13.28: Performances in sub Urban Areas with 16QAM rate 1/3 @ 45 m

From this example we can derive the following conclusions:

- Most of the hybrid reception cases provide improvement compared to the satellite only reception as the required *C/N* is decreasing.
- When the terrestrial signal is under the *C/N* threshold (for instance *C/N* = 0 dB in QPSK while 2 dB are required), the resulting quality of service with satellite on (and even with very low satellite *C/N* values) is above 99 % ESR5(20).
- In QPSK, the case with terrestrial C/N = -10 dB, and with differential delay equal to 164 % of the GI, shows some slight degradation in the satellite reception (< 0,5 dB).
- With GI of 1/4, there is no more degradation with QPSK 1/3.

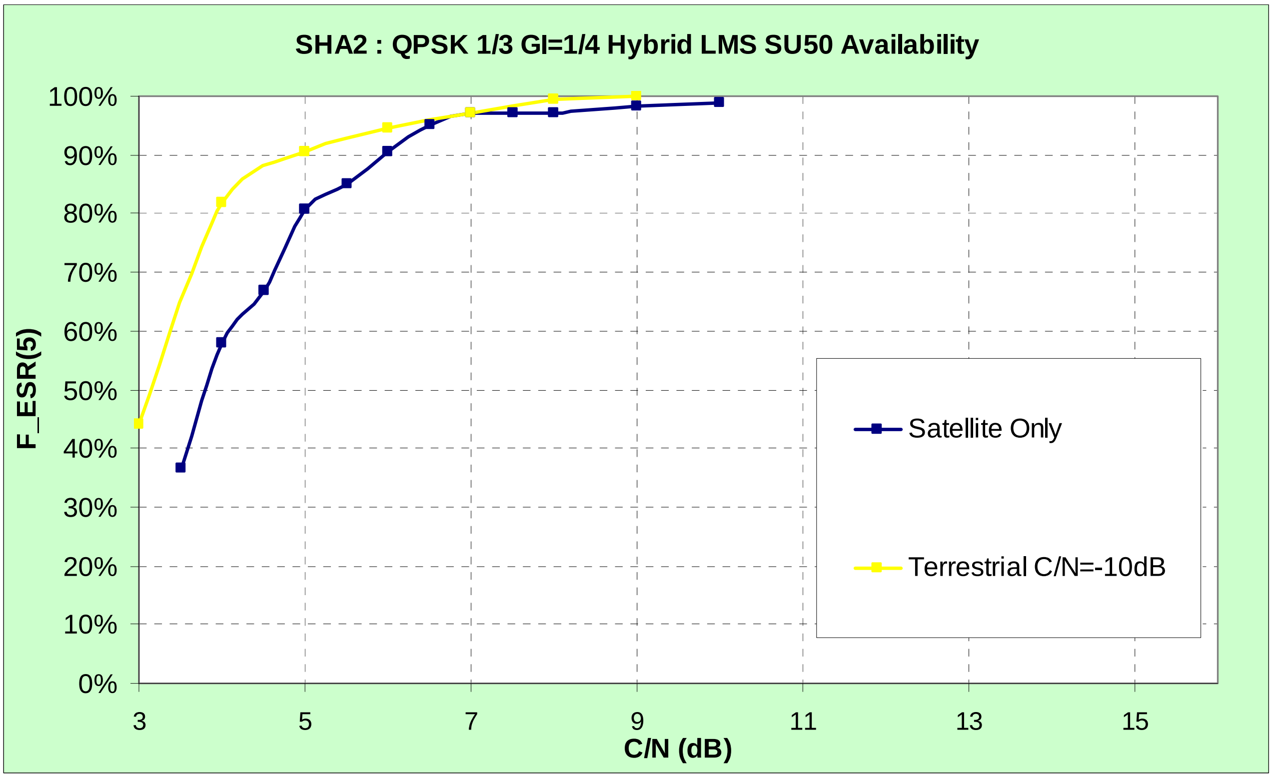


Figure A.13.29: Performances in sub Urban Areas with QPSK rate 1/3 @ 45 m and GI 1/4

Finally with 60 m height terrestrial repeater height, the following results have been obtained.

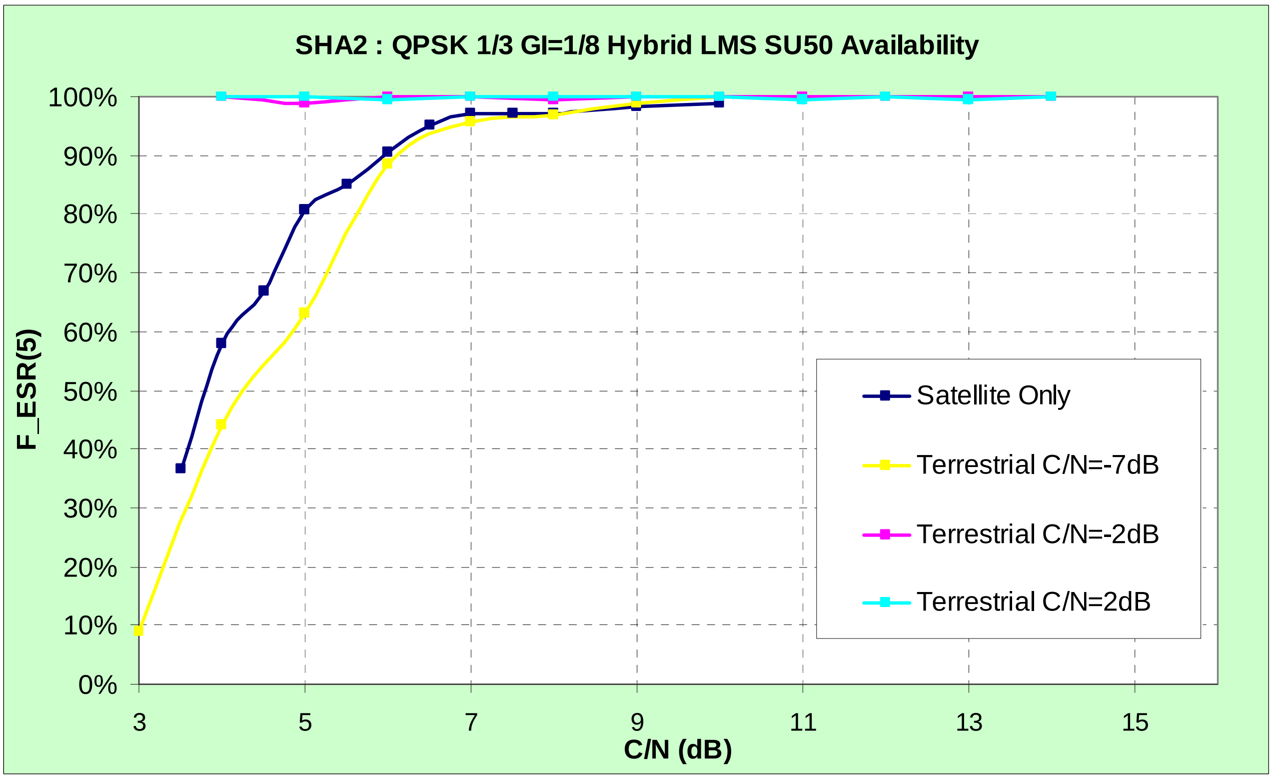


Figure A.13.30: Performances in sub Urban Areas with QPSK rate 1/3 @ 60 m

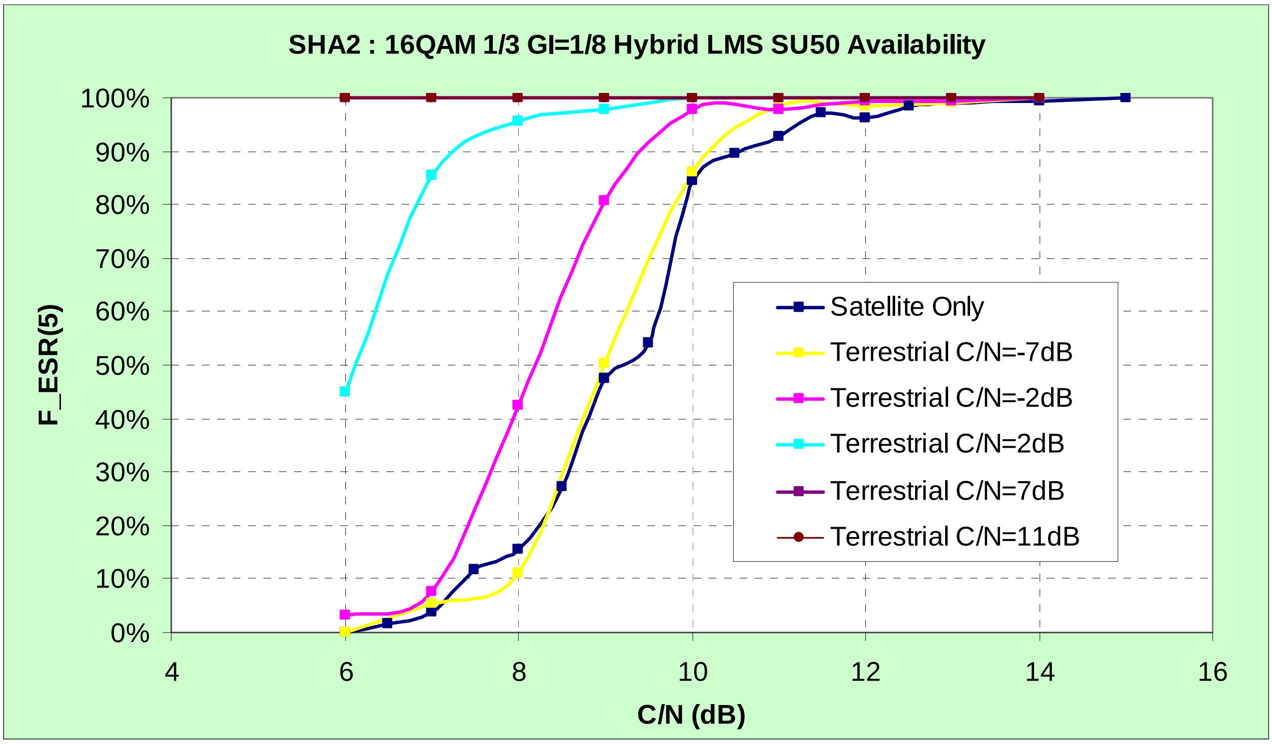


Figure A.13.31: Performances in Sub Urban Areas with 16QAM rate 1/3 @ 60 m

With 60 m height, there are some degradations for QPSK when differential delay is over 164 % of the GI (C/N = -7 dB).

A.13.1.3.2.2 Rural Channel

For the rural channel configuration two cases are provided:

a) QPSK 1/3 at 50 km/h.

b) 16QAM 1/3 at 50 km/h.

The satellite channel emulator is set in the ITS configuration and the speed is 50 km/h. The terrestrial channel emulator is set at in the TU6 configuration at 50 km/h. The same three cases for the different repeater antenna height are provided.

For a 30 m terrestrial repeater height, the following results have been obtained:

- There is noticeable improvements when terrestrial C/N is over 0 dB.
- With larger differential delays (more than 200 % of the GI), there are some degradations for both QPSK and 16QAM , below 0,5 dB.

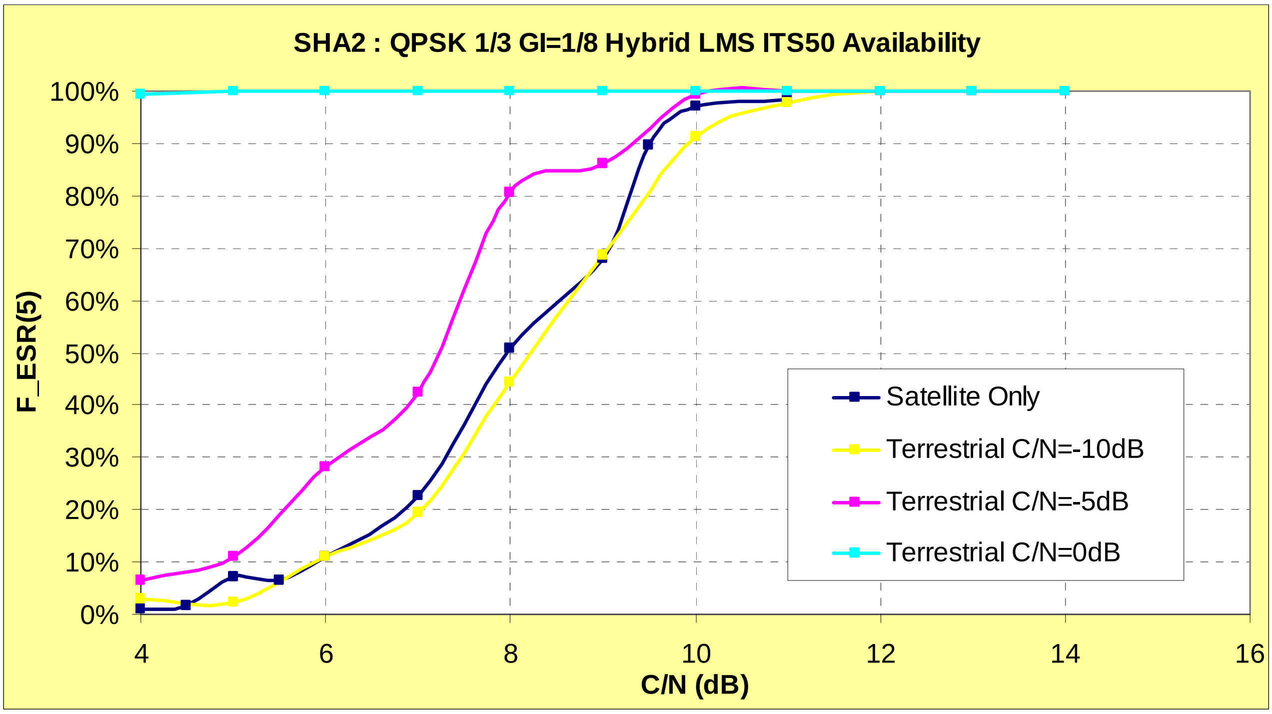


Figure A.13.32: Performances in Rural Areas with QPSK rate 1/3 @ 30 m

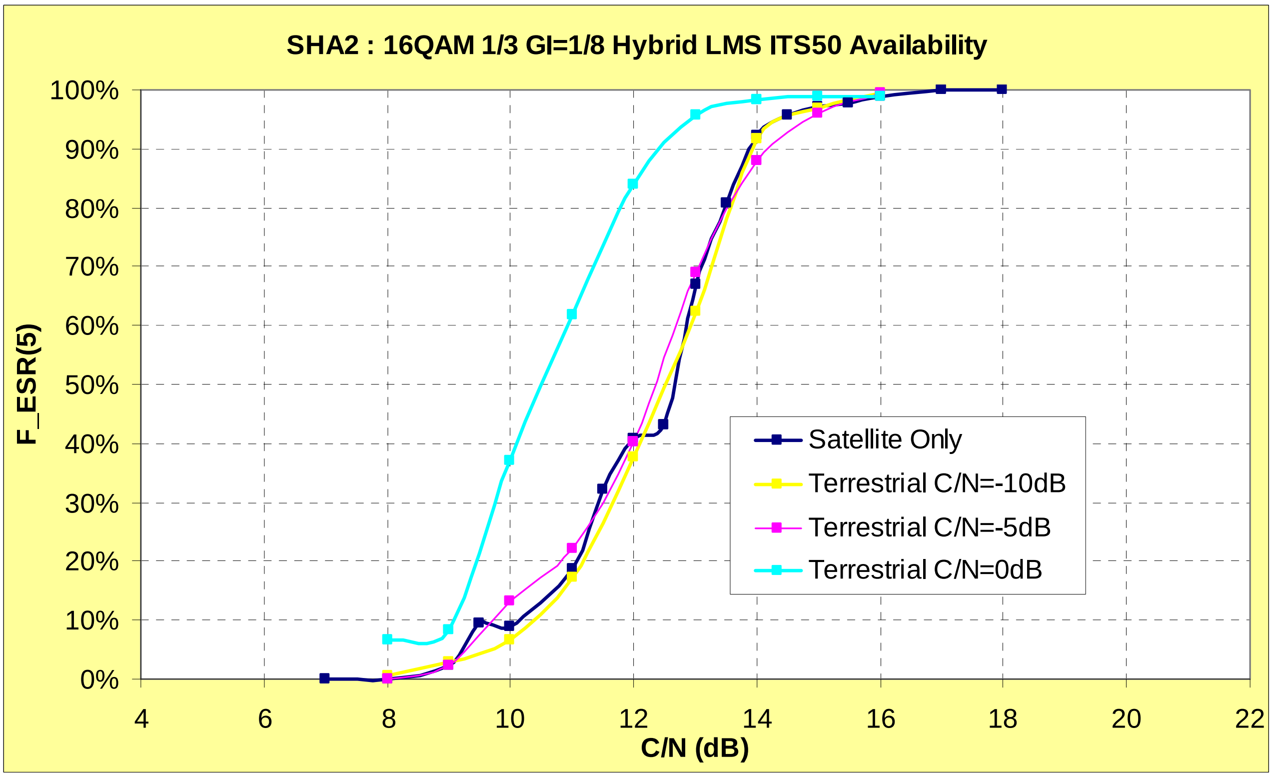


Figure A.13.33: Performances in Rural Areas with 16QAM rate 1/3 @ 30 m

With 45 m terrestrial repeater height, the following results have been obtained:

- In QPSK, there are some degradation when differential delay is over 200 % of the GI.
- For the other cases there is still high improvement of the coverage.
- In 16QAM, there is a slight degradation when differential delay are over 200 % of the GI, but below 0,5 dB.

- The use of GI 1/4 provides improvement with QPSK 1/3, reducing the interference.

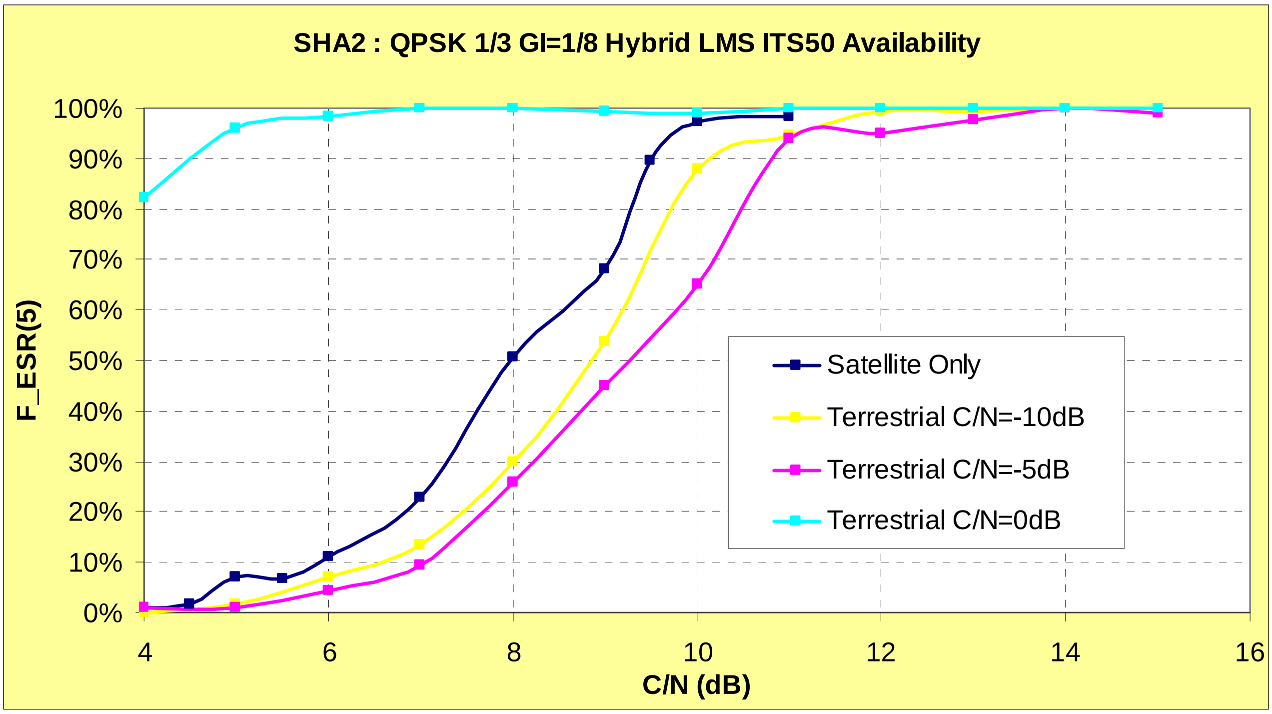


Figure A.13.34: Performances in Rural Areas with QPSK rate 1/3 @ 45 m

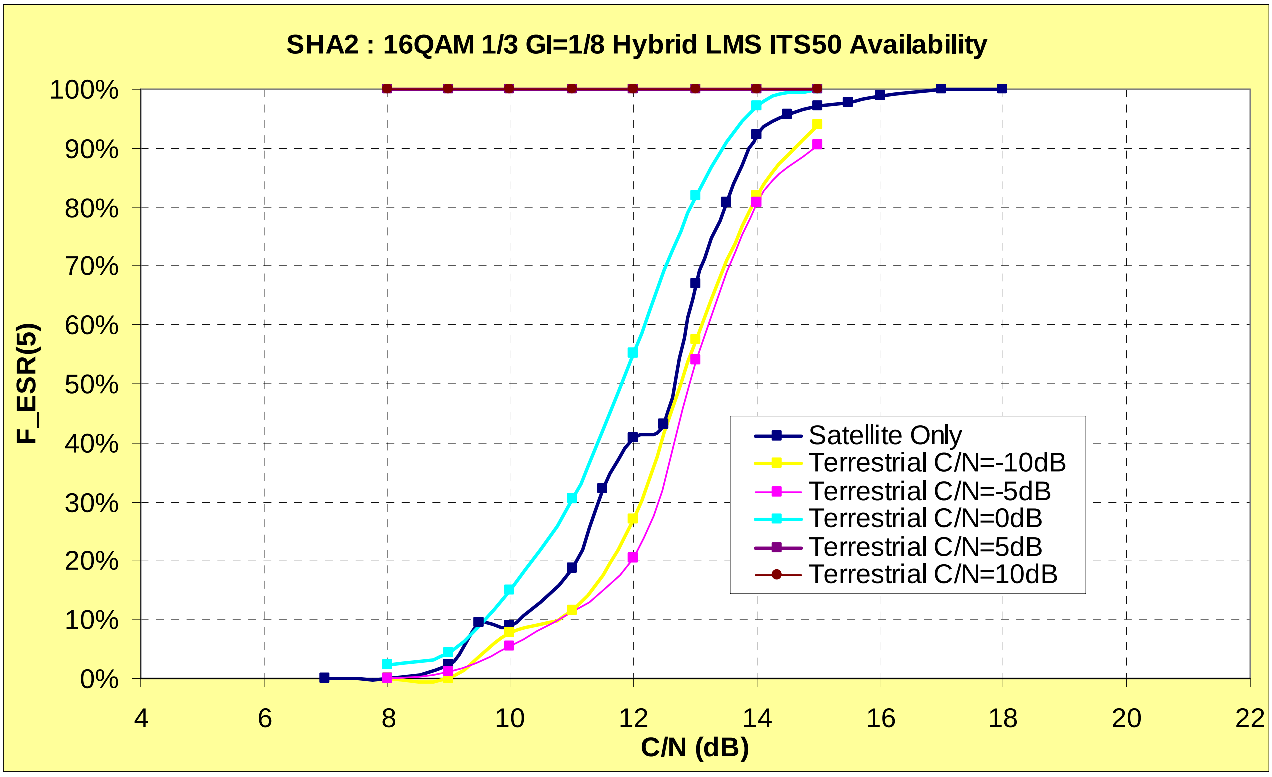


Figure A.13.35: Performances in Rural Areas with 16QAM rate 1/3 @ 45 m

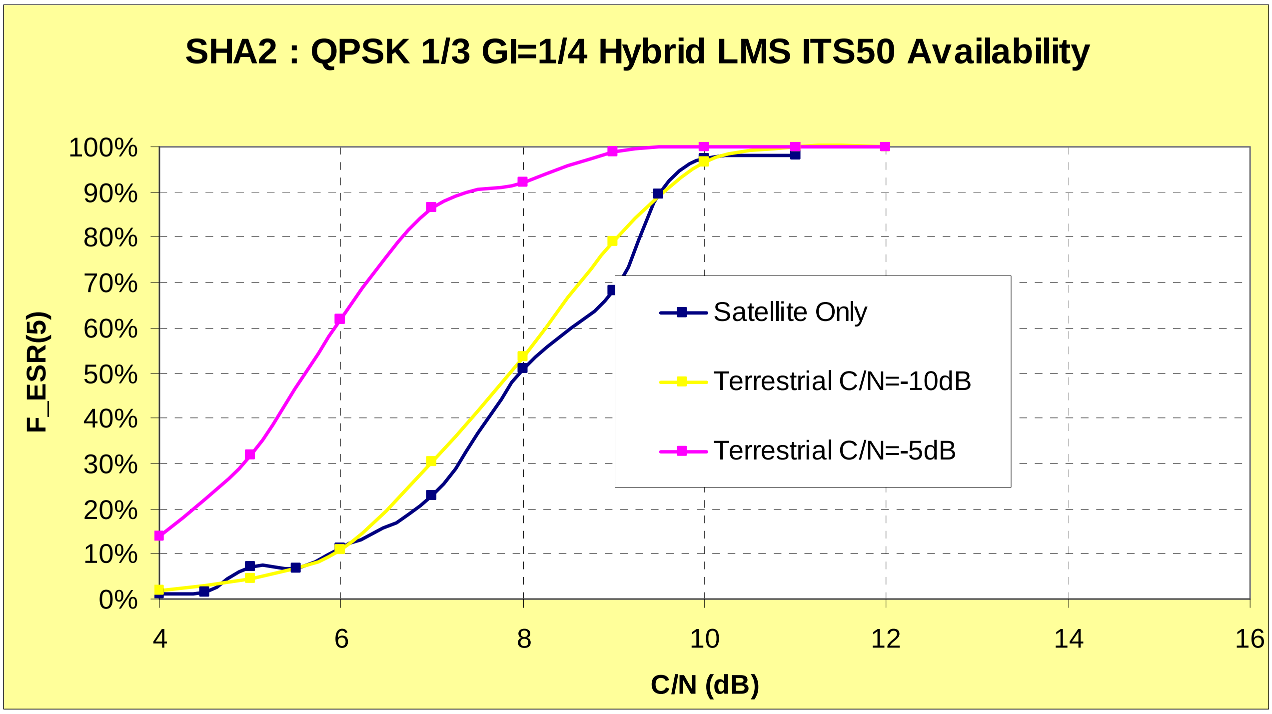


Figure A.13.36: Performances in Rural Areas with QPSK rate 1/3 @ 45 m and GI 1/4

Finally with 60 m terrestrial repeater height, the following results have been obtained:

- For both modulation schemes, when differential delay is above 200 % of the GI, there are noticeable degradations that can reach more than 2 dB.
- To mitigate these interference one can use GI 1/4.

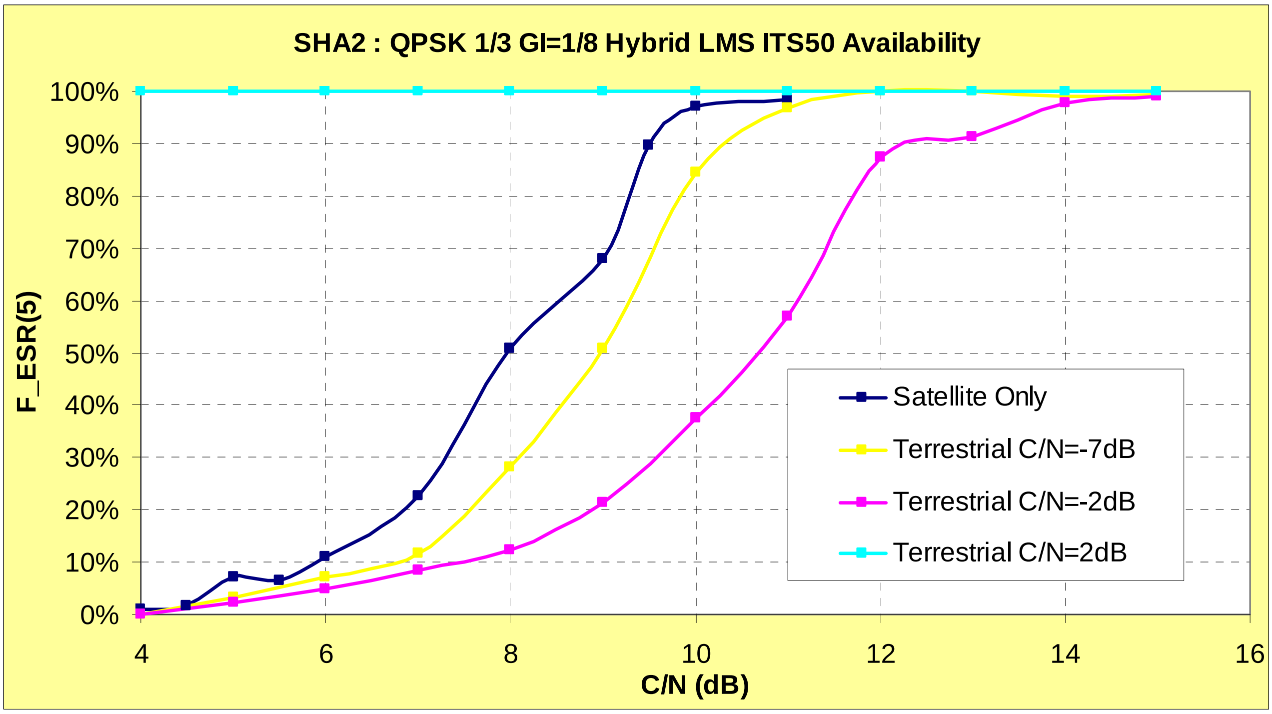


Figure A.13.37: Performances in Rural Areas with QPSK rate 1/3 @ 60 m

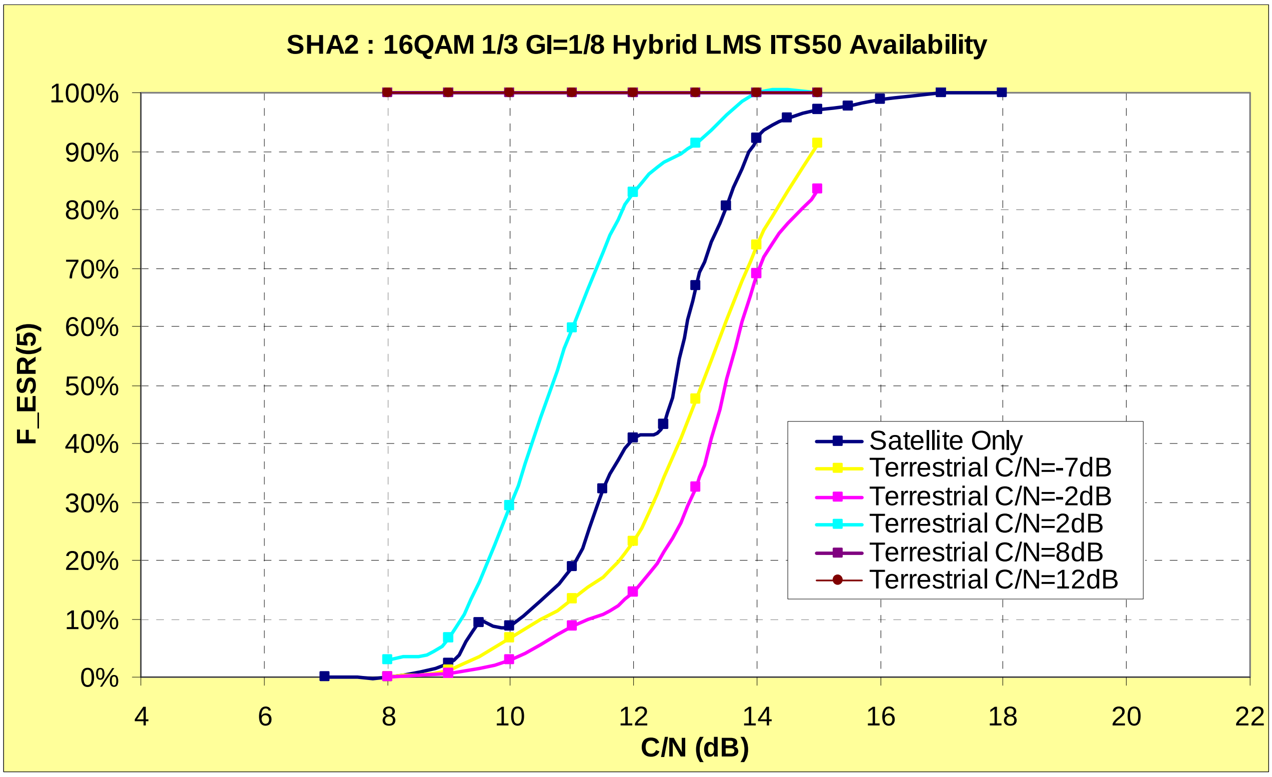


Figure A.13.38: Performances in Rural Areas with 16QAM rate 1/3 @ 60 m

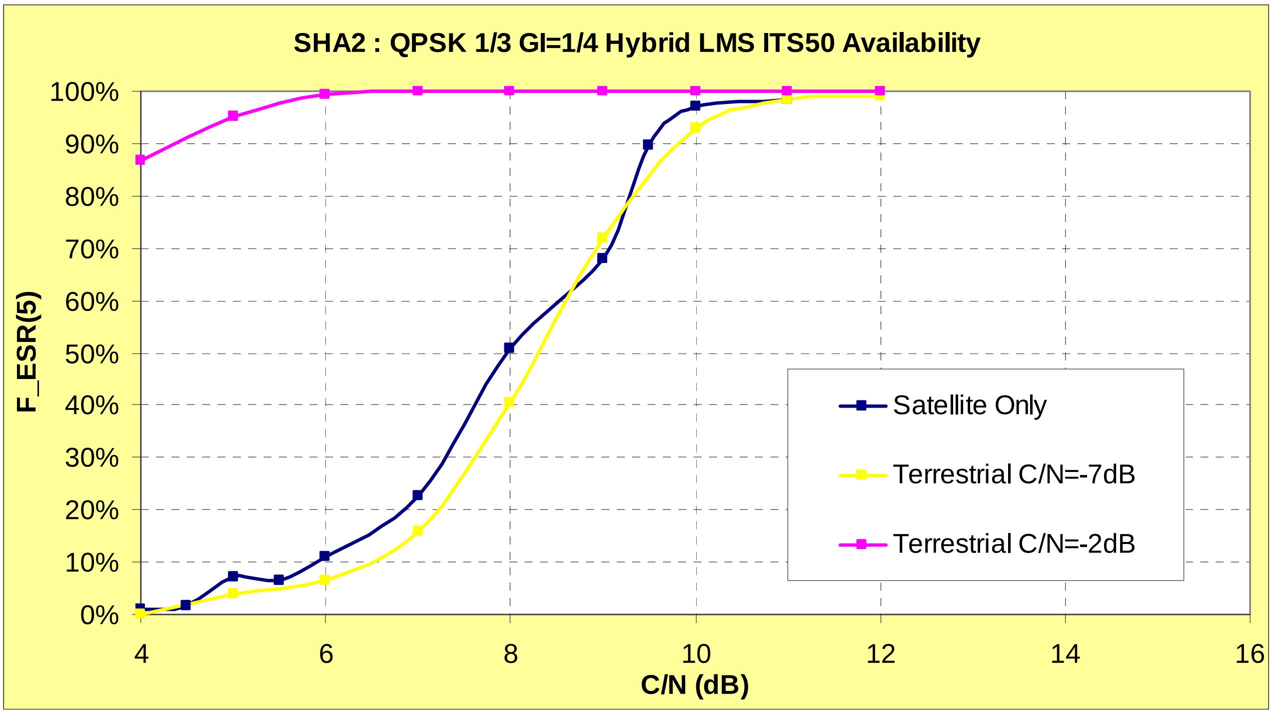


Figure A.13.39: Performances in Rural Areas with QPSK rate 1/3 @ 60 m and GI 1/4

A.13.1.3.2.3 Case of class 1 plus MPE-IFEC

In the case of MPE-IFEC, the following configuration has been tested: 16QAM ½ at physical layer plus MPE-IFEC 70 % with GI 1/8. The results show some degradation for 60 m repeater height but improvement in all other cases. As said above, the case of 1/4 with 16QAM 1/3 was not tested, but as degradation with 1/8 was lower for 16QAM than for QPSK, with GI 1/4, there is no doubt that there will be no more degradation. With MPE-IFEC,the same results are provided with GI 1/8. For the case of MPE-IFEC, not all the cases have been measured, due to lack of time. The representative cases are summarized in table A.13.12, which represents the ESR5(20) fulfilment ratio (last three lines). The satellite C/N is 12 dB which provides 90 % ESR5(20) in the sub urban 50 km/h configuration.

Table A.13.12: Summary of results for MPE-IFEC

MPE-IFEC	**GI ratio**				
C/N terrestrial	**0 %**	**50 %**	**100 %**	**150 %**	**200 %**
-6 dB	90 %	90 %	87 %	68 %	62 %
0 dB	92 %	92 %	91 %	51 %	8 %
6 dB	100 %	100 %	100 %	99 %	94 %

From the table above we can derive the following conclusions:

- When the terrestrial signal is strong (C/N = 6 dB for instance) and whatever the differential delay, the improvement is always excellent: 100 % ESR5 when differential delay is up to 100 % of the GI. When differential delay exceeds 100 % of the GI, some improvement is also noticeable.
- When terrestrial signal C/N is around 0 dB and with and differential delay below 100 % of the GI, the results are good. The performance is degraded when the differential delay increases, but we observe ESR5 fulfilment reduction to 8 % corresponding to only 2 dB C/N degradation due to the sharp roll-off of the ESR5 curve.
- When the terrestrial signal is weak, (C/N around -6 dB), there is only a slight degradation when differential delay is equal to 100 % of the GI (around 0,1 dB), but some degradation for larger delays, quite low in terms of C/N, even if the ESR (5) value varies quite a lot.

A.13.1.3.3 Interpretation and conclusions

Summarizing some of different laboratory measurements, figure A.13.40 illustrates the different hybrid network operational zones.

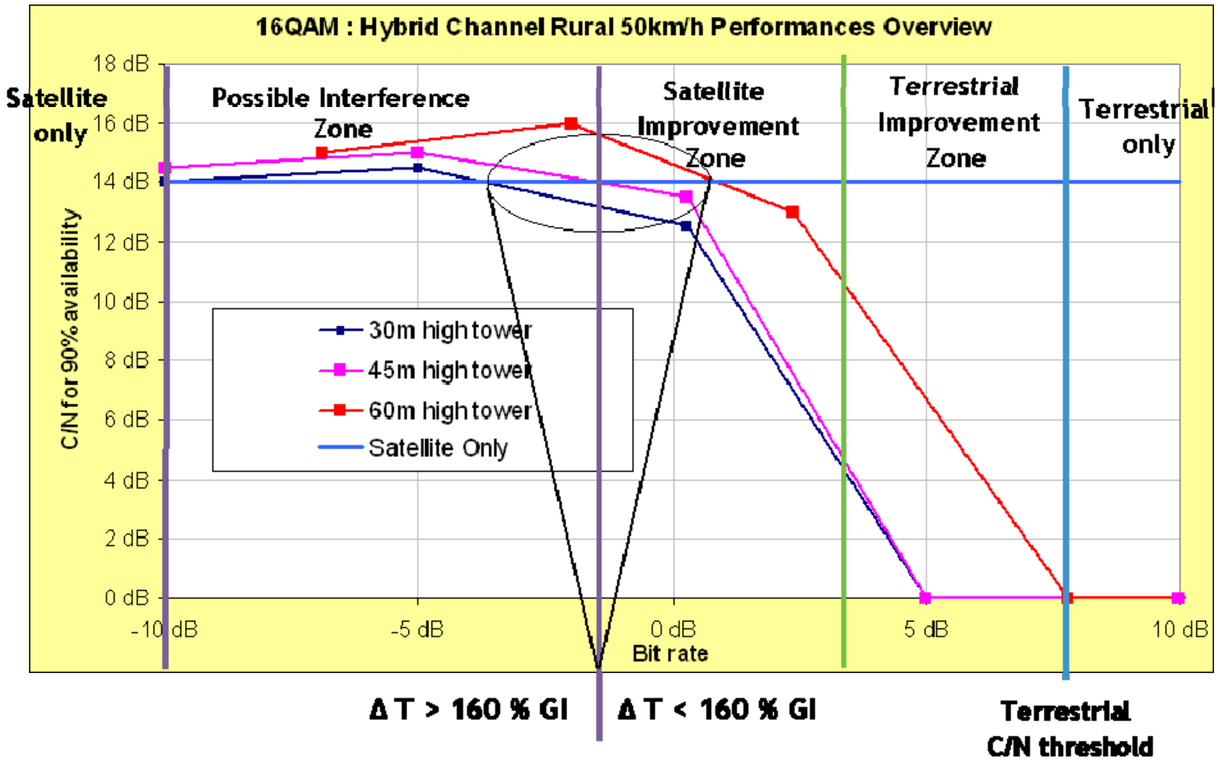


Figure A.13.40: Illustration of the different zones in rural areas with 16QAM rate 1/3

It is possible to distinguish the different zones (starting from right in the diagram):

- When received terrestrial signal C/N is above required threshold, we can consider that we are under terrestrial coverage, and the satellite signals provides some improvement, but is not essential.
- The second zone corresponds to a zone where satellite signal is improved by the terrestrial one. Even if the satellite signal is quite weak (for instance if received by a handset, the combination of the two signals result in a good quality signal).
- The third zone corresponds to an improvement of the satellite signal with a "residual" edge of terrestrial coverage signal.
- The fourth zone corresponds to possible interference zone between the satellite and terrestrial signals. The border between the two previous zones corresponds to a differential delay of around 160 % of the GI.
- The last zone corresponds to satellite only signal as terrestrial signal is falling below -10 dB, but it can have some interference impact.

In the case of suburban area, the interference zone is reduced as shown in figure A.13.41.

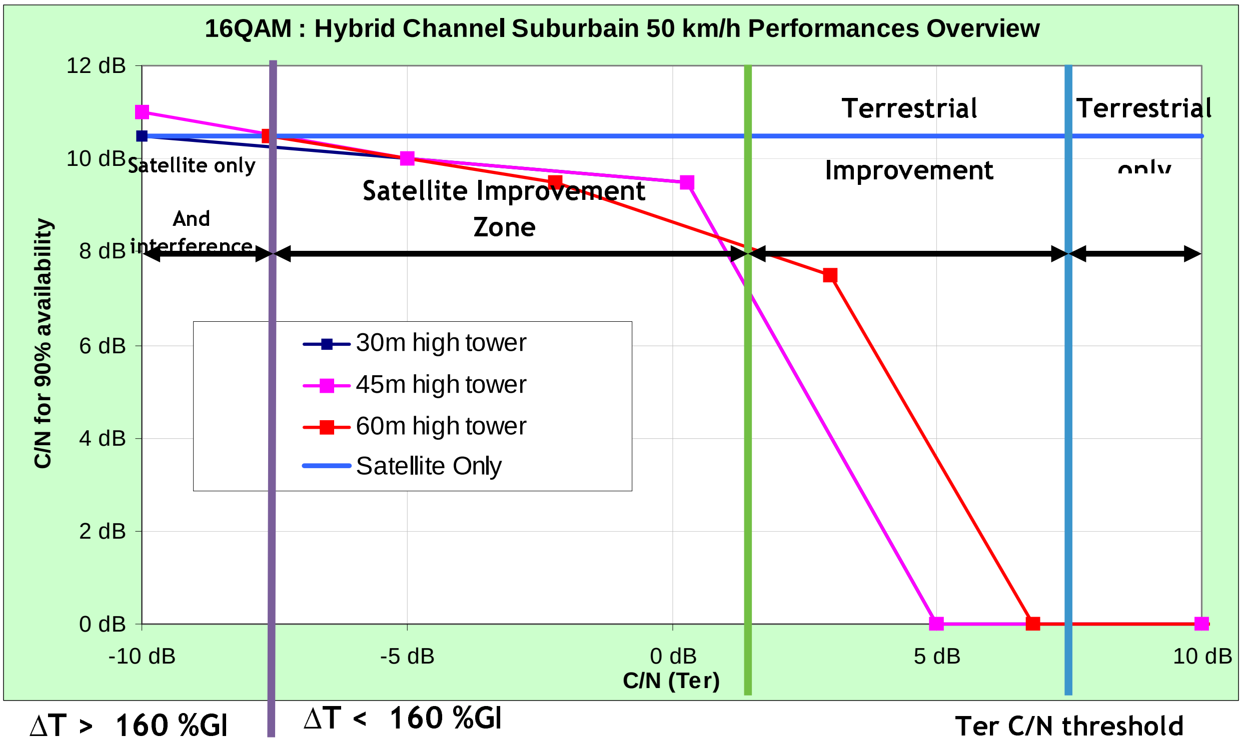


Figure A.13.41: Illustration of the different zones in Sub Urban areas with 16QAM rate 1/3

Some conclusions and rules can be derived from the previous results:

- The hybrid channel architecture gives an evident gain of quality of services, as it increases in most of the cases the ESR (5) ratio.
- In the northern direction, the differential delay is below 100 % of the GI on along distance, even for GI 1/8, and there is no risk of interference, as the terrestrial signal becomes very low at the edges.
- On the southern direction (in sight of the satellite), the gain is noticeable, even at more than 100 % of the GI.
- When the delay is more than 150 % of the GI approximately , there can be some interference at te edges of the contour, but they can be mitigated by different means: using a large GI when possible, implement timing advance at repeaters level, or reducing radiated power towards southern direction. The repeater could use two sector antenna at the edge of the zone, and oriented towards northern directions.

Hence a hybrid network planning tool can be designed according to the following principles:

1) Compute the geometrical hybrid SFN zones taking into account the possible timing advance: roughly the zones where differential delay is below 150 % (TBC) of the GI. In this region we will assume that satellite and terrestrial signal components will sum up constructively.

2) Compute terrestrial only coverage deriving the region within the SFN where considering the TU6 corresponding DVB-SH C/N threshold $(C/N)_{Ter}$, $C/N > (C/N)_{Ter}$.

3) Compute inside the geometrical SFN zone the equivalent C/N taking into consideration the satellite LOS by using the following simplified formula:

$$\left(\frac{C}{N}\right)_{eq}[\text{dB}] = 10\cdot\log\left[10^{\frac{\left(\frac{C}{N}\right)_{Sat}[\text{dB}]}{10}} + 10^{\frac{\left(\frac{C}{N}\right)_{Ter}[\text{dB}]}{10}}\right].$$

- In this case the QoS. could be determined by using maps like in clause 11.7.4. The calculation will give inputs for deriving possible reduction of number of sites or their transmitted power.

4) Outside the geometrical SFN zone we assume that the terrestrial signal component will now interfere with the satellite one. The resulting C/N is now computed by:

$$\left(\frac{C}{N}\right)_{eq}[\mathrm{dB}] = -10 \cdot \log\left[10^{-\left\{\frac{\left(\frac{C}{N}\right)_{Sat}[\mathrm{dB}]}{10}\right\}} + 10^{-\left\{\frac{\left(\frac{C}{N}\right)_{Sat}[\mathrm{dB}]}{10} - \frac{\left(\frac{C}{N}\right)_{Ter}[\mathrm{dB}]}{10}\right\}}\right].$$

A.13.2 Field trials results

A.13.2.1 Introduction

Several organizations undertook a set of trial campaigns to validate the operation of DVB-SH. The aim of these trials was to confirm on the field and under realistic conditions the performances predicted via simulations and laboratory tests.

The first two examples of trials (Pau and Torino) were terrestrial oriented (as deployed before the launch of any ad hoc satellite). The main objectives were the following:

- Waveform performances evaluation.
- SFN gain characterisation.
- Receive antenna diversity characterisation.
- Validation of network engineering rules (integration on UMTS sites for instance).
- Validation of the Radio Planning strategy.

The third example analyses the satellite coverage (using Eutelsat W2A S-band payload) with a SH-A configuration on a vehicular environment in a long route from Paris to Barcelona.

The fourth campaign (Pisa) was focused on satellite and hybrid reception and it used the S-band signal from W2A satellite payload. The main objectives were:

- Evaluation of Satellite coverage in different environments (urban, sub-urban and rural).
- Assessment of hybrid gain: code combining.
- Comparison of class 1 versus class 2 receivers.

A.13.2.2 Pau trial

A.13.2.2.1 Introduction

For over one year, an intensive field test campaign of the DVB-SH technology was performed over an extensive terrestrial zone covering approximately 15 Km² of urban, suburban and rural environments jointly set-up by the French mobile network operator SFR® and Alcatel-Lucent™ in the city of Pau (France), over a typical mobile cellular networking topology (sites disposition, engineering and heights), actually having a large sharing of site infrastructure between the mobile cellular UMTS service and the DVB-SH broadcast service (typically, in-building cabling and antennas, with limited modifications). Seven UMTS sites were selected and shared with DVB-SH transmitters.

The Pau trial has been carried out in the French city of Pau in 2007 (Phase 1) and 2008 (Phase 2), sharing 7 UMTS sites of the French operator. In the present document, we compare the performances on the field of the DVB-SH waveform in S-band (2 170 MHz to 2 200 MHz) under terrestrial conditions with the laboratory tests done with a TU6 channel.

The basic system architecture is represented here below.

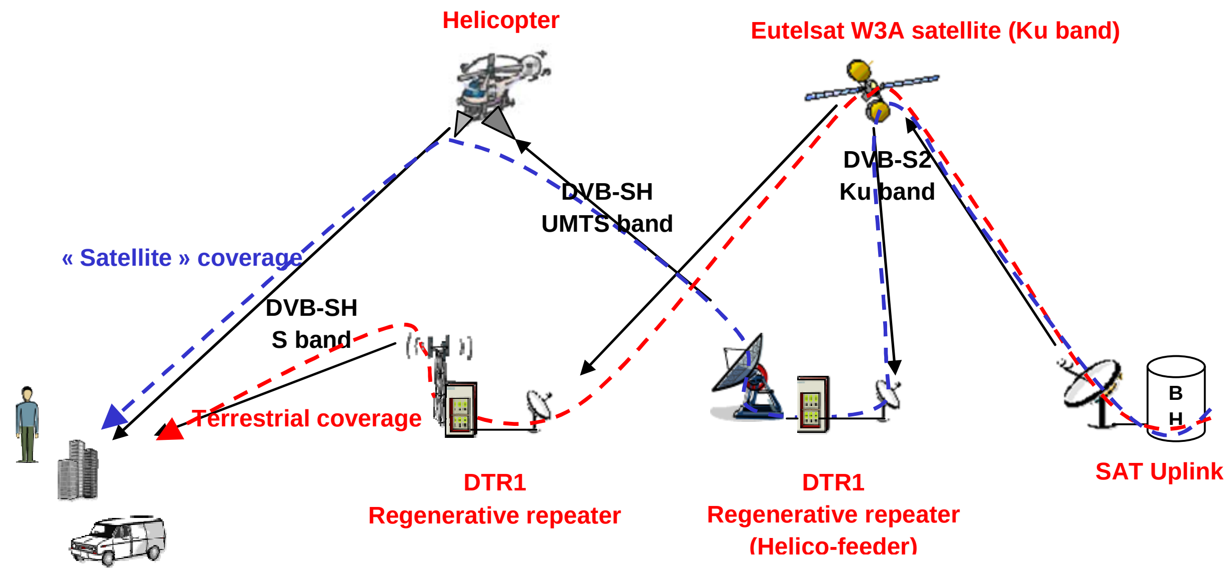


Figure A.13.42: Pau trial high level setup configuration

A.13.2.2.2 Experimental set up

Several phases of measurements were conducted overall, benefiting in steps of successive improvements brought by field tools and by measurements protocols. Alcatel-Lucent™ and SFR® were primarily interested in the end-to-end performances results, directly representative of actual service conditions delivered to end users while using SFN at network level and integrated reception antennas diversity at terminal level. SFN measurements benefited from the capabilities of the Alcatel-Lucent™ S-UMTS band emitter to broadcast several carriers (up to three in totally SFN conditions). The antenna diversity of order two was evaluated thanks to an innovative pre-industrial development by SAGEM® Wireless (called the "T2 program").

Precise quantitative measurements were carried out using terrestrial-only coverage (in SFN configuration and with antenna diversity at the receiver). A simulation of a hybrid coverage (combining signals from terrestrial repeaters and a satellite "emulator" provided by helicopter stationary flight) was used for qualitative results not presented here. Several reception conditions were evaluated (Vehicular, In-car, Indoor and Outdoor pedestrian).

Figure A.13.43 gives an example of radio network planning including the sites location inside Pau.

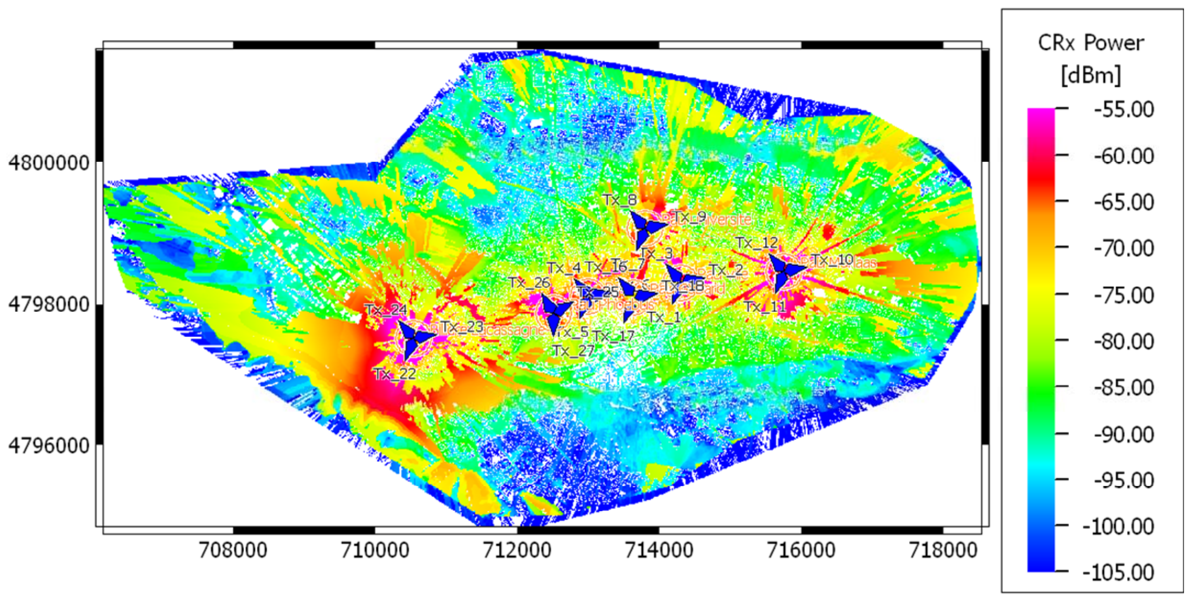


Figure A.13.43: Example of Radio Network Planning in Pau

A.13.2.2.2.1 Base Station Configuration

The 7 test sites, spread throughout the city, were reusing existing UMTS base station antennas and feeder. A diplexer, close to the UMTS antenna feeder, allowed injecting the DVB-SH signals (see figure A.13.44). Sites equipped with a TMA (the Tower Mounted Amplifier) had to be upgraded as the one in place was filtering the DVB-SH signals out. Each site was equipped with a satellite dish receiving signal from the head-end located in SES-Astra premises. On each site, up to 3 carriers were broadcasted simultaneously. The first tests aimed at validating the impact of antenna sharing on the UMTS service. No impact of DVB-SH on the UMTS service was found by SFR® using its key performance indicators (e.g. call setup and completion rate).

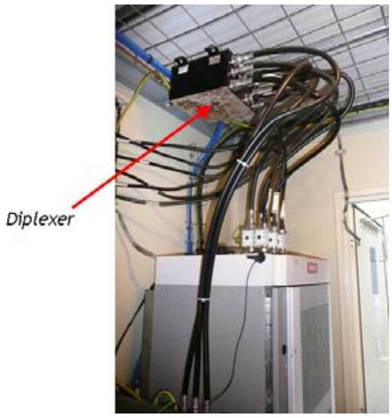


Figure A.13.44: Diplexer multiplexing UMTS and DVB-SH signals

A.13.2.2.2.2 Helicopter Configuration

A helicopter flying in stationary position with a constant elevation angle emulated the satellite. The helicopter embedded a transparent repeater (i.e. a simple frequency transponder with amplifier) equipped with a circular polarization antenna as on an actual satellite (see figure A.13.45). The radiated power was adjusted in order to deliver on the ground a power equivalent to the one of the satellite.

Figure A.13.45: The helicopter transmit antenna system

A.13.2.2.2.3 Receiver Configurations

Three types of receiving situations were evaluated using various antennas setup (see figures A.13.46 to A.13.48):

- Vehicular reception using 2 test receiver and 2 outdoor car-mounted antennas.
- In-car reception (see figures A.13.46 to A.13.48) using 2 test receivers and 2 handheld antenna systems.
- Indoor and outdoor pedestrian reception using up to 3 test receivers and 3 handheld antenna systems installed on a cart. For outdoor pedestrian the cart was pushed by a person along the test road, each SH receiver was connected to an independent antenna system. The indoor pedestrian conditions were emulated with the cart pulled by a rope and a motor, the SH receivers were connected via splitters (1 in diversity, 2 without diversity).

Figure A.13.46: Different kinds of roof top antennas

Figure A.13.47: In-car receivers

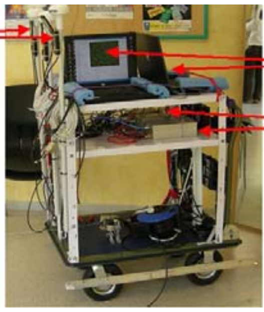


Figure A.13.48: Indoor receivers and processing unit installed on a cart

A.13.2.2.3 Waveform performance results

We consider the two cases in outdoor and indoor.

A.13.2.2.3.1 Waveform configurations

In most of the cases, unless specified, the characteristics of the waveform are the following, using a frequency slot in: inside 2 170 MHz to 2 185 MHz.

Table A.13.13: Pau waveform configurations

BW (MHZ)	Modulation/ coding	FFT	GI	Capacity (Mbps)	Short Interleaving	Short Interleaving duration (s)	Uniform Interleaving structure	Uniform duration (s)	MPE-IFEC
5	QPSK 1/3	2k	1/8	2,468	8	0,265			
5	QPSK 1/3	2k	1/8	2,468			4/0/1/0/59	10	
5	QPSK 1/3	2k	1/8	2,468	8	0,265			B=17 S=9 B+S = 26
5	16QAM 1/3	2k	1/8	4,494	8	0,132			

A.13.2.2.3.2 Outdoor pedestrian

The outdoor vehicle route and the corresponding network planning results are shown in figure A.13.49.

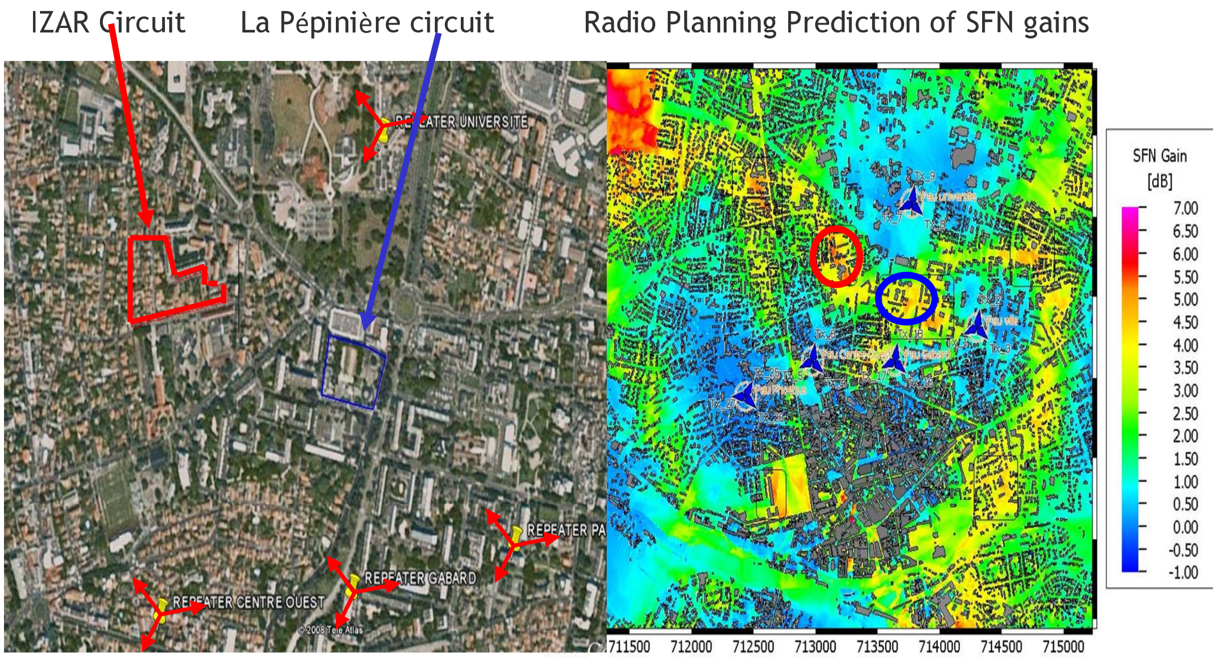


Figure A.13.49: Reference Circuits in Pau

For the measurements, the results for 16QAM 1/3 and QPSK 1/3 are presented in figure A.13.50. 5 000 points were measured, filtered and declared "good".

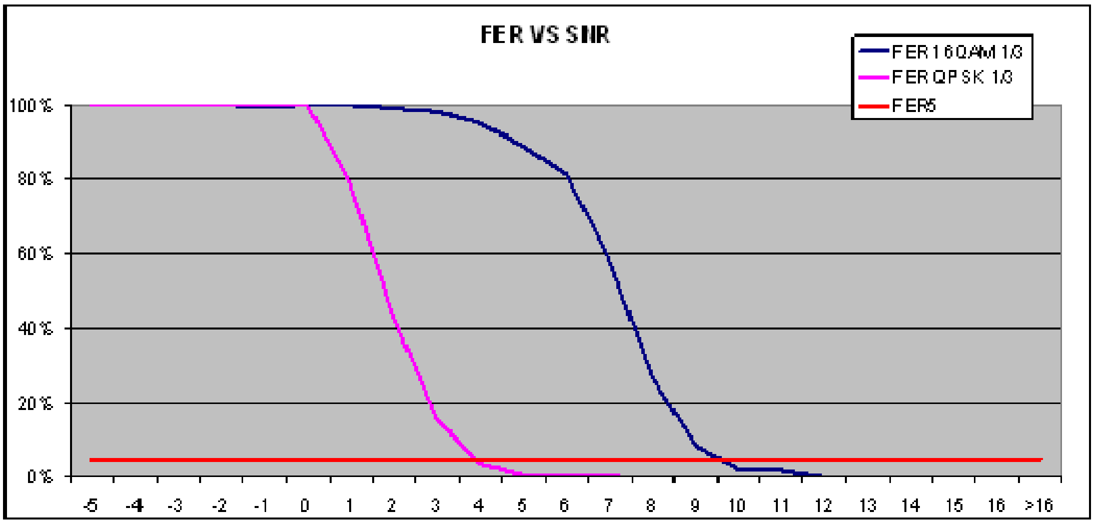


Figure A.13.50: Results of outdoor measurements: FER versus SNR

In conclusion, we have the following C/N values:

- For QPSK 1/3: C/N = 4 dB @ Low Speed.
- For 16QAM 1/3: C/N = 9,5 dB @ Low Speed.

The above results are slightly above to the TU6 channel laboratory test results reported in clause A.13. It is recalled that at low speed and using a TU6 channel emulator, the following laboratory measurements were observed:

- QPSK 1/3: 2,8 - 3,5 dB.
- 16QAM 1/3: 8,5 - 9,1 dB.

A.13.2.2.3.3 Indoor pedestrian

Two places were selected for indoor measurements:

- Leclerc commercial center.
- La Pepinière for its SFN gain.

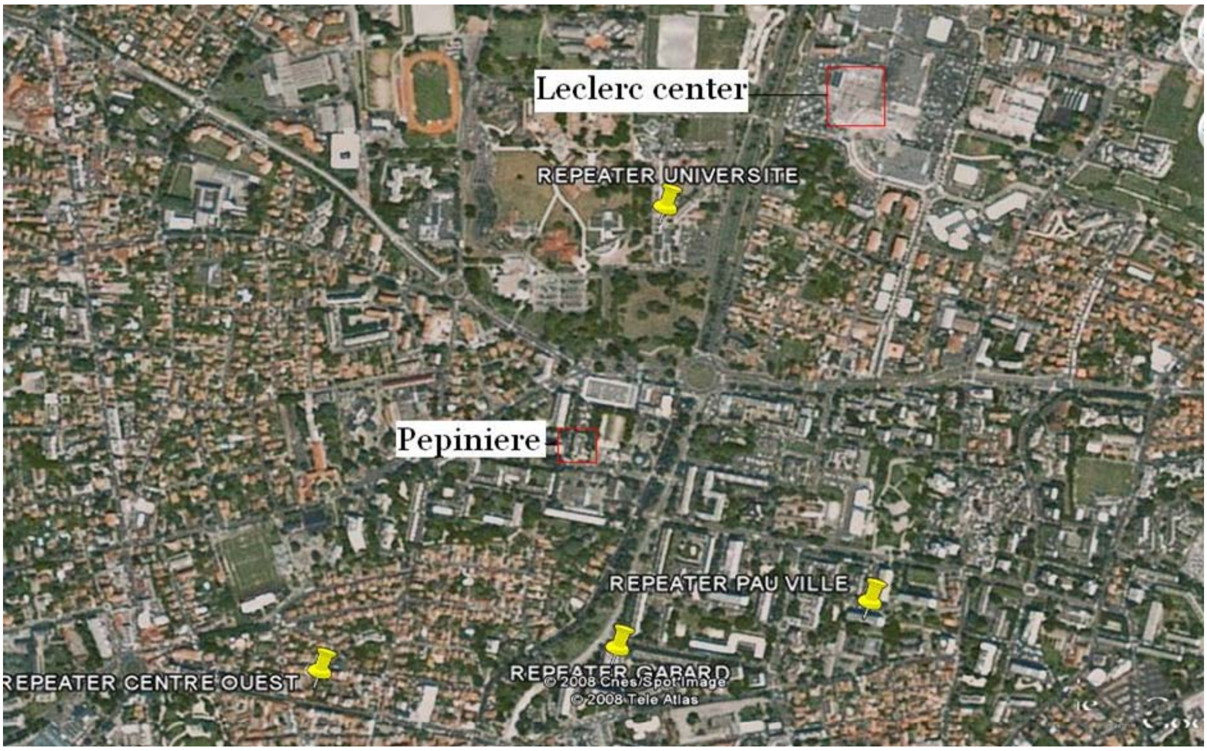


Figure A.13.51: Location of indoor sites

In this case only 16QAM 1/3 results are available and summarized in figure A.13.52.

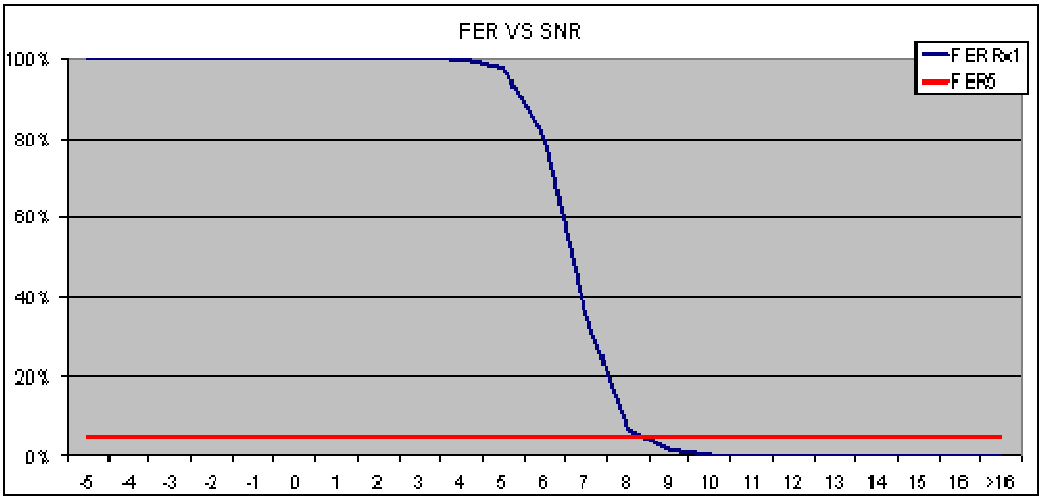


Figure A.13.52: Results of indoor measurements: FER versus C/N with 16QAM rate 1/3

The obtained C/N value for 16QAM 1/3 @ FER 5 % is 8,5 dB, which is exactly the value obtained with the same equipment using TU6 channel emulator at 5 km/h (19 Hz).

A.13.2.2.3.4 Waveform performances synthesis

The different waveform performances at low speed are summarized in table A.13.14 and compared with the results obtained with the same receiver under laboratory conditions.

Table A.13.14: Summary of waveform performances pedestrian

Modulation/coding	Terminal Type	Terminal Class	C/N @ FER 5 % Indoor Low speed	C/N @FER 5 % Outdoor Low speed
QPSK 1/3	Field trials	Class 1	Not available	4
	Lab test	Class 1	3,2	3,2
16QAM 1/3	Field trials	Class 1	8,5	9,5
	Lab test	Class 1	8,5	8,5

The differences between laboratory and field trial results is less than 1 dB.

A.13.2.2.4 SFN measurements

SFN gain is generally defined as the difference between the MFN case (i.e. each repeater uses a given frequency; the receiver selects along the time the best repeater) and the SFN case (i.e. all repeaters use the same frequency and the receiver combines the signal received from the different repeaters). Clearly, this is a purely "local" (point by point) definition. Local signals fluctuate widely within very short spatial ranges, especially in indoor conditions. When the various SFN-combining signals are very power unbalanced, the effect on "real end-user" service conditions may not be measurable under the usual "macroscopic" service availability criteria such as FER. Therefore, SFN measurements should be done in locations where SFN conditions are optimally met. This requires a specific radio network planning and network set-up optimization that, at the same time, respects propagation conditions in real networks, namely:

- Cell edge with low power signal, so that SFN creates an extended coverage (e.g. one individual signal does not ensure service while cooperative SFN signals do). Sources should be roughly of the same power level to guarantee non-biased conditions.
- The receiver antenna configuration should be representative of a real scenario; this was achieved using the RF part of a complete SAGEM® T2 mobile terminal.

The purpose of the test was to compare MFN and SFN in a real quasi-static environment using two repeaters. This was evaluated in two ways:

1) Field strength measurements (in dB):
    - Signal level measured in SFN (i.e. receiving simultaneously both repeaters on the same frequency).
    - Best signal level measured in MFN (i.e. receiving the repeater with the best signal. Both repeaters using different frequencies).
2) Transmission quality (in %):
    - Frame Error Rate (FER) in SFN.
    - Best FER measured in MFN.

A.13.2.2.4.1 Methodology

Two methods were used to assess the SFN gain.

Method 1: Simultaneous runs

This method benefits from the capability of the repeater used in the trials to broadcast simultaneously up to three DVB-SH 5 MHz signals:

- Repeater 1 is configured to broadcast two signals, one at frequency F1, the other one at frequency F3.
- Repeater 2 is configured to broadcast two signals, one at frequency F2, the other one at frequency F3.

Three receivers are set up to receive one of the three signals. Since frequencies F1, F2 and F3 are contiguous, we assumed similar propagation channel. This methodology allows simultaneous measurement of all signals, through a single RF chain, thus eliminating any propagation hazard, and it requires three different receiver tools, duly calibrated for simultaneous reception of the three carriers, and equipped with ad hoc filters.

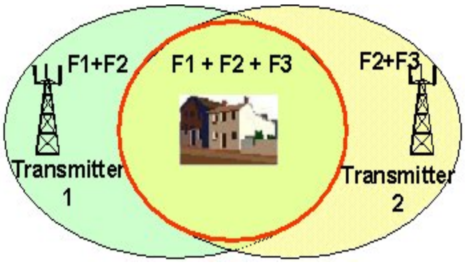


Figure A.13.53: Repeater configuration for simultaneous run tests

Tests were done on a single run on the test path; the three receivers measured parameters continuously over the test path.

Method 2: Successive runs

In this method, a single frequency F2 was used, from one repeater, then from the other one, then from both repeaters in SFN. A single receiver was set-up and received frequency F2.

Independently of the method used, the following conditions were also respected:

- For outdoor, about 100 measurement sites were selected along the roads and streets located in the test area.
- Each measurement site was a few meters long and separated by about 50 meters from the next measurement site.
- One minute measurements were performed in each site with the cart moving around.

For indoor tests were done in a room with continuous measurements at a speed of about 1m/mn.

A.13.2.2.4.2 The results

Field strength measurements results

SFN gain evaluation is done comparing the distribution of received signal levels. For each repeater, a CDF of the measured field strength was derived.

Figure A.13.54 represents the percentage of time the received level is above abscise. 77 % of the time, the received signal level was higher than -100 dBm for "Centre", -99 dBm for "Gabard" and -96,5 dBm in SFN. Measured SFN gain versus the single "Gabard" signal is 2,5 dB.

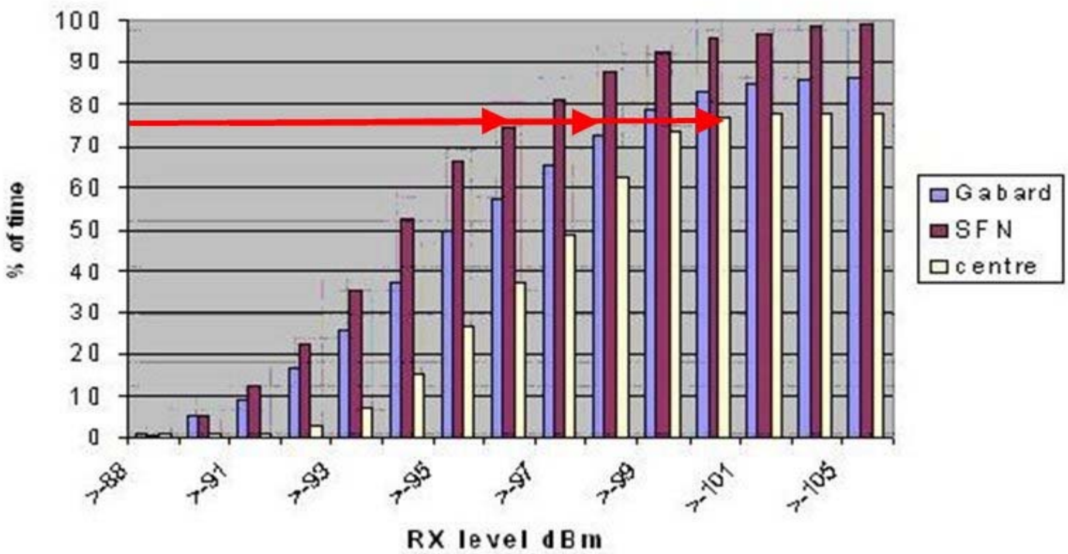


Figure A.13.54: Example of Rx level compared distribution

Figure A.13.55 represents the percentage of time C/N is below abscise, 70 % of the time (complementary to 30 %), the C/N is higher than 2,0 dB for "Centre", 3,5 dB for "Gabard" and 6,5 dB in SFN case, giving 3,0 dB of SFN gain.

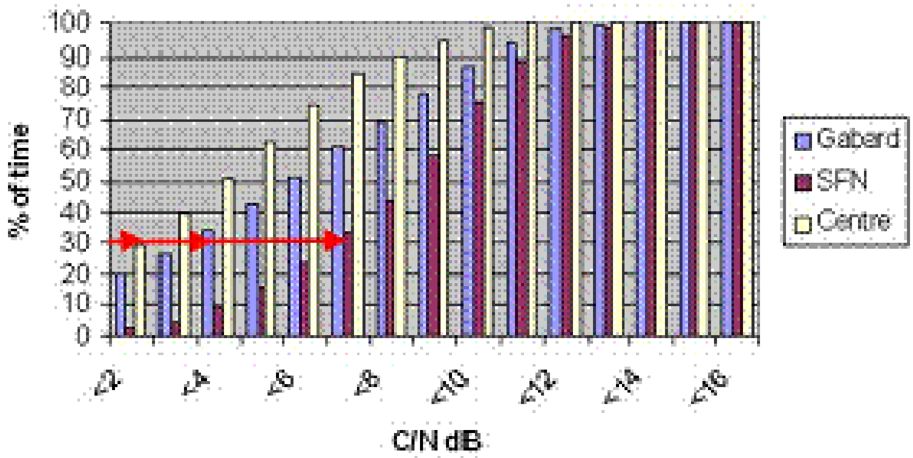


Figure A.13.55: C/N compared distribution

Transmission Quality Measurement Results

The three curves of FER versus time are plotted over the measurement path. The simultaneous measurements method is used.

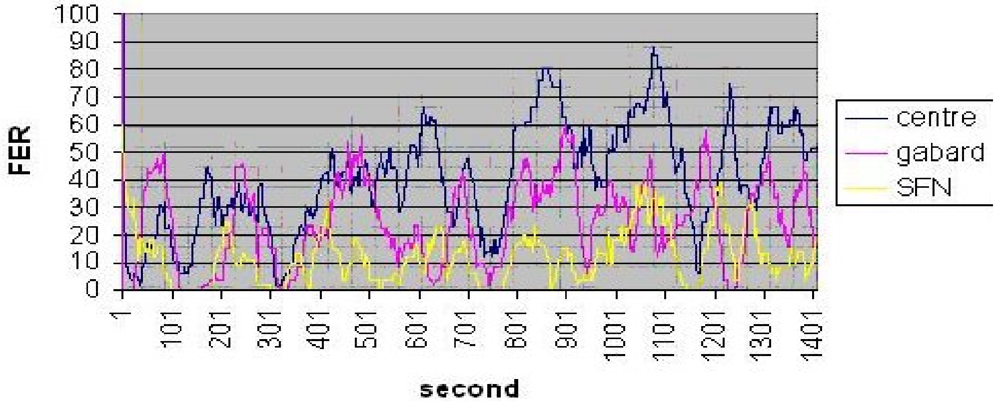


Figure A.13.56: FER measurements

A.13.2.2.5 Indoor diversity measurements

Space diversity works on the principle that when considering two points separated by half a wavelength, the probability that these two points correspond simultaneously to a minimum is very low. Using two receive antennas allows taking advantage of the space diversity. In S-band, antennas need to be separated by only a few centimetres which makes antenna diversity a feasible and attractive solution. Several methods exist to combine the signal from both antennas. The method used in the trial is the Maximum Ratio Combining (MRC) which weights the signal amplitude according to the square-root of their SNR, before coherently adding them.

A.13.2.2.5.1 Methodology

As for SFN measurements, it is preferable to do measurements in the area of the network where diversity clearly creates the conditions of a measurable improved reception. To characterize the diversity gain, 3 receivers were used, all connected to the same single handheld antenna system provided by the SAGEM® Mobiles T2 terminal. Figure A.13.57 details how the single antenna signals are fed to the several receivers: "Receiver 1" received only signal from "Antenna 1", "Receiver 2" received only signal from "Antenna 2", "Receiver 3" received signal from both antennas and combined these signals using the MRC method. Receiver 3 signals inputs could, in addition, be attenuated selectively, using wideband attenuators.

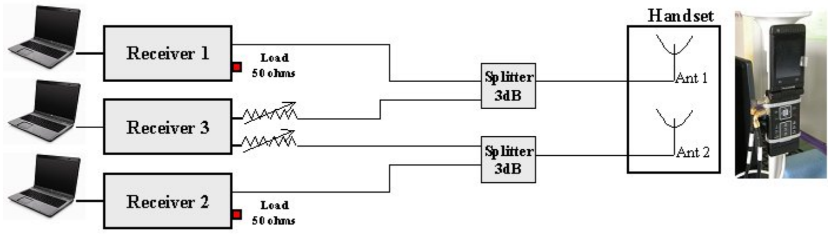


Figure A.13.57: Receivers configuration

These receivers, installed on a cart, were moved around the room and several thousand measurements were performed. At each run, the signal sent to "Receiver 3" was progressively attenuated until the quality of the video (FER) was equivalent to the one of the best receiver without diversity.

A.13.2.2.5.2 Diversity Measurements

The comparison is based on the global distribution of FER5 values (a FER below 5 % allows to display video without image freeze). This comparison is made between the receiver with diversity ("Receiver 3") and the best receiver without diversity ("Receiver 1" or "Receiver 2"). One run was performed per attenuation, with 3 600 to 6 000 measurements per run.

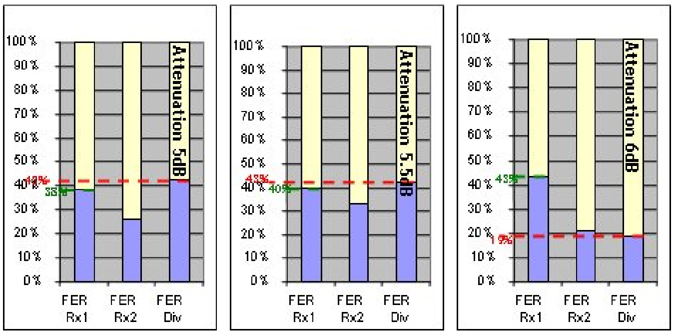


Figure A.13.58: Percentage of time with a FER below 5 %

The results show that the signal sent to the "Receiver 3" needs to be attenuated by 5,5 dB to get a performance similar to the best receiver without diversity ("Receiver 1" in the example of figure A.13.58). Therefore the diversity gain from this experiment can be estimated to be 5,5 dB. The diversity gain obtained in outdoor urban has been measured to be in the range of 4 dB to 6 dB. Diversity gain appears to be highly dependant on the urban environment.

A.13.2.3 Torino trial

A.13.2.3.1 Introduction

The objective of this trial was to demonstrate the quality of the DVB-SH technology on the field. The focus of this campaign was to assess the antenna diversity and the SFN gains. The capacity of DVB-SH waveform to counteract channel impairments thanks to efficient coding and interleaving schemes was also explored.

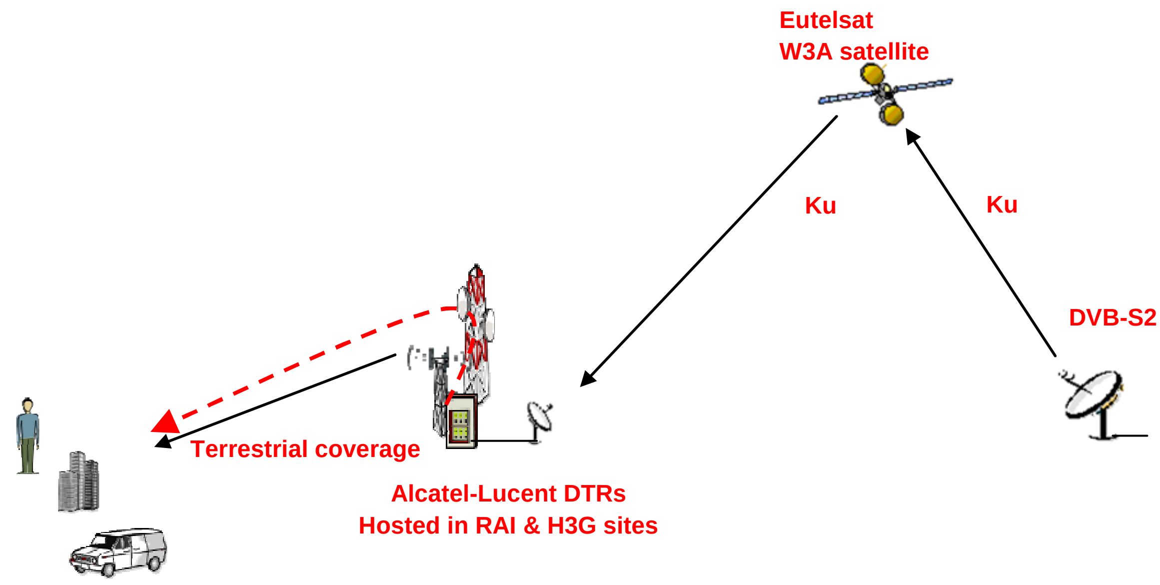


Figure A.13.59: Overview of the Torino field trial set up

The terrestrial network is composed of a combination of high power / high height sites (up to 480 m for Eremo) and lower power (30 w) regular heights collocated with 3G sites (provided by H3G operator).

A.13.2.3.2 Experimental set up

The experiment is based on the use of a combination of five Low Power sites collocated with 3G sites, and two high power sites (Eremo and Cernaia) located as shown in figure A.13.60.

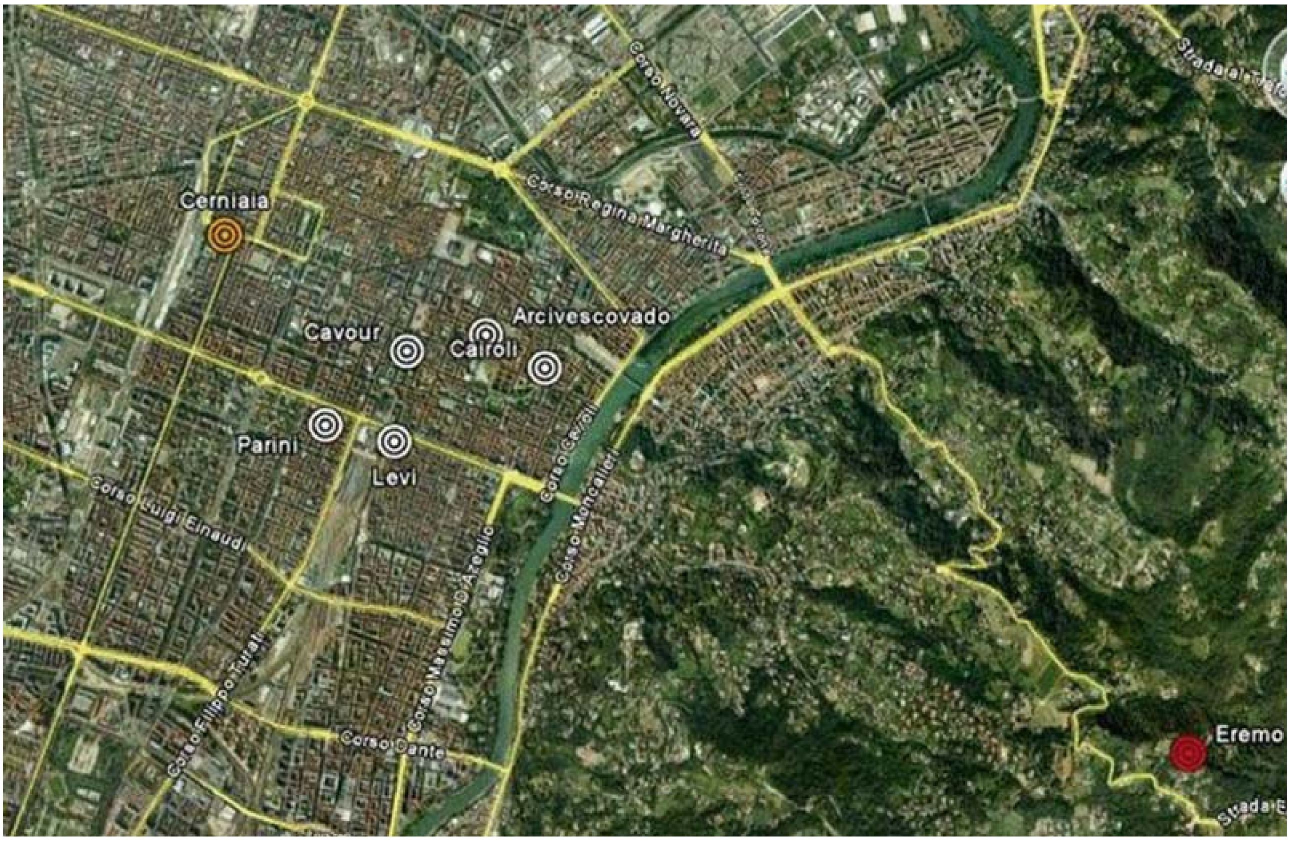


Figure A.13.60: Torino sites location

Figures A.13.61 and A.13.62 describe the block diagram of a low power site collocated with 3G site.

- The antennas are shared, thanks a diplexer, as already described in the Pau trial.
- The satellite dish is used for receiving the different services that will be demodulated, multiplexed and DVB-SH modulated through the tri sectors of the site.
- The TMA (Tower Mounted Amplifiers) are changed to increase bandwidth.

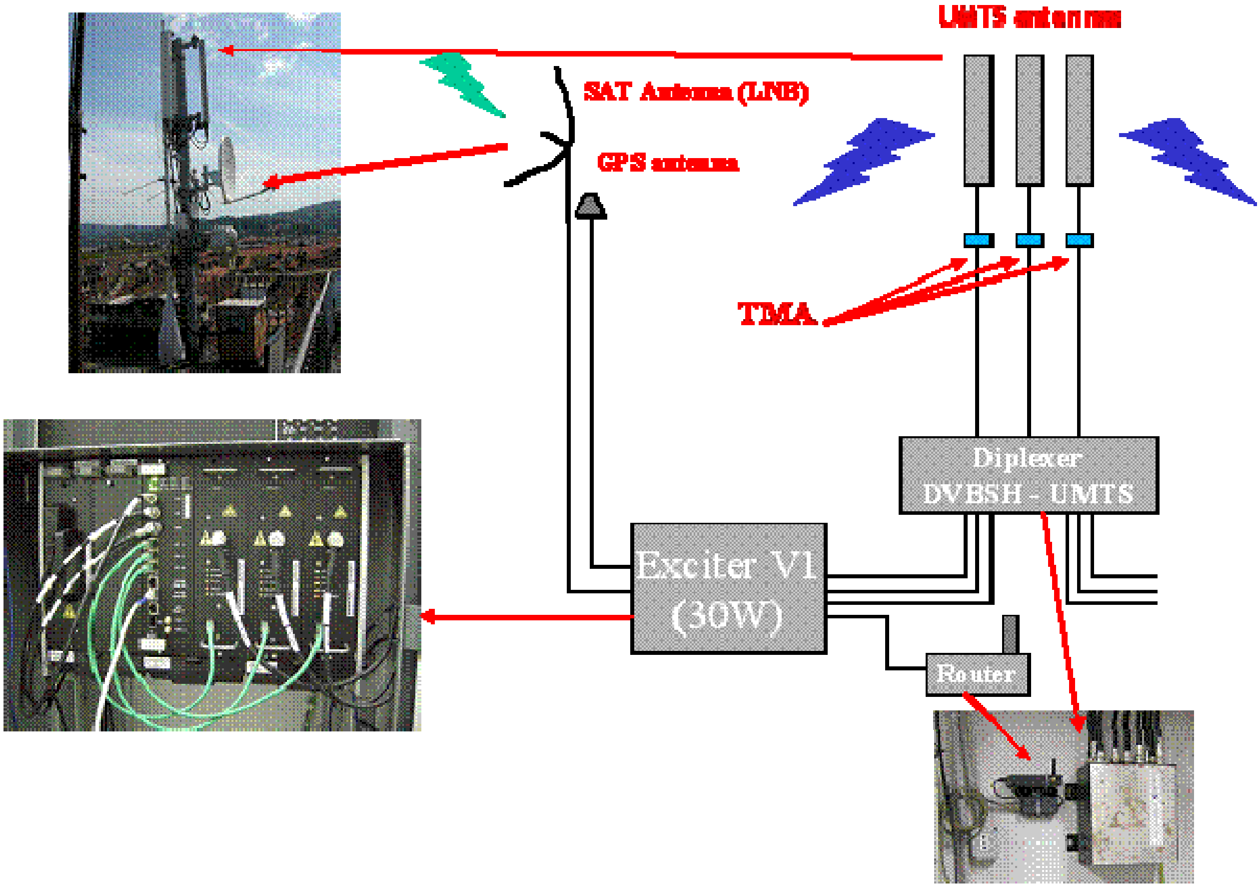


Figure A.13.61: Typical Low Power site configuration

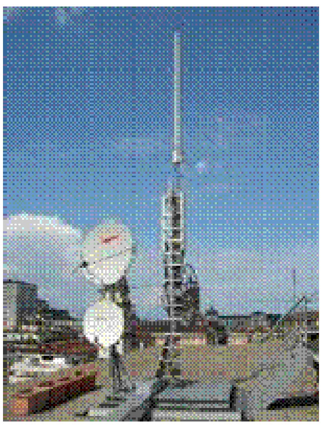


Figure A.13.62: Low Power site: Cavour site

The next site is a high power high altitude site: Eremo.

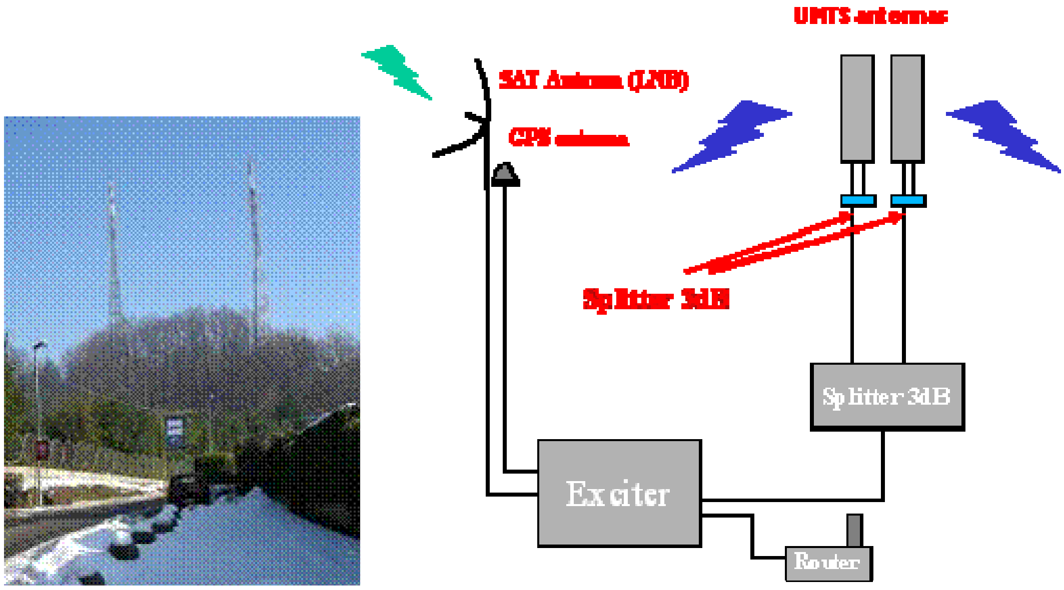


Figure A.13.63: Eremo site and Eremo configuration

The different receiver configurations are depicted hereafter:

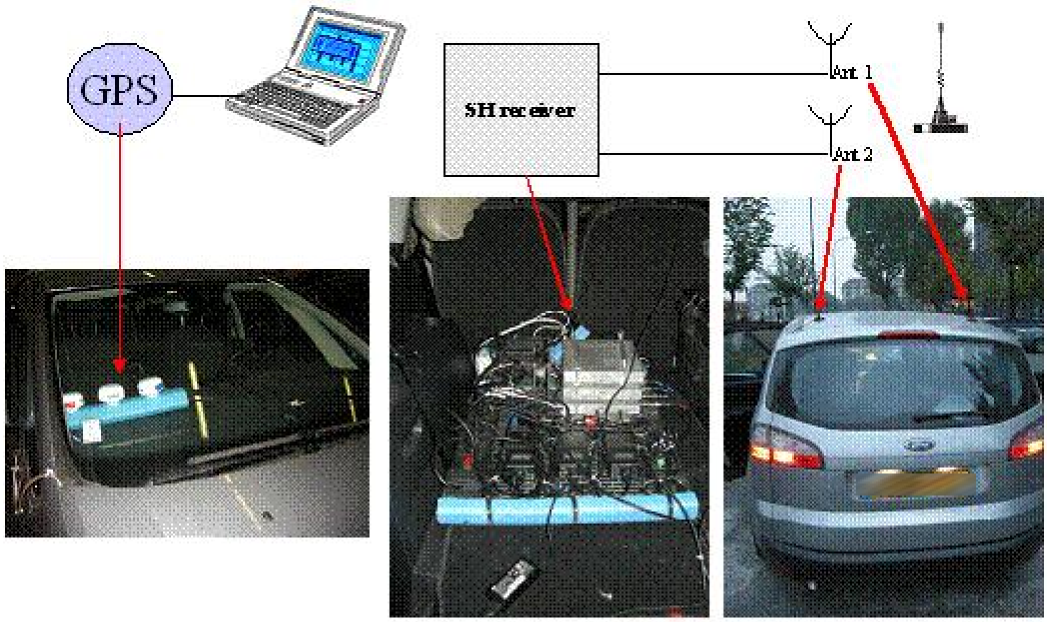


Figure A.13.64: Configuration of car receiver with roof top antennas

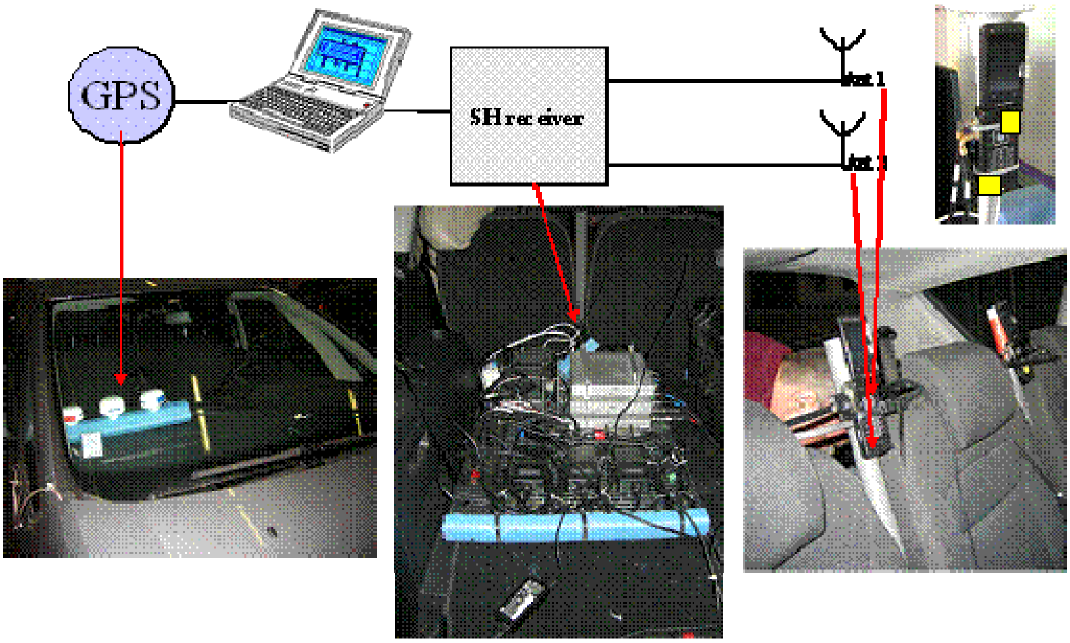


Figure A.13.65: In car reception configuration

A.13.2.3.3 Performances of DVB-SH waveform and diversity gain analysis in reception condition C

A.13.2.3.3.1 Waveform configurations

In most of the cases, unless specified, the characteristics of the waveform are the following, using a frequency slot inside the 2 170 MHz to 2 185 MHz band.

Table A.13.15: Torino waveform configurations

BW (MHZ)	Modulation/ coding	FFT	GI	Capacity (Mbps)	Short Interleaving	Short Interleaving duration (s)	Uniform Interleaving structure	Uniform duration (s)	MPE-IFEC
5	QPSK 1/3	2k	1/8	2,468	8	0,265			
5	QPSK 1/3	2k	1/8	2,468			2/0/1/0/59	5	
5	QPSK 1/3	2k	1/8	2,468			4/0/1/0/59	10	
5	QPSK 1/3	2k	1/8	2,468	8	0,265			B=3 S=2 B+S = 5
5	QPSK 1/3	2k	1/8	2,468	8	0,265			B=17 S=9 B+S = 26
5	16QAM 1/3	2k	1/8	4,494	8	0,132			
5	16QAM 1/3	2k	1/8	4,494			8/0/1/0/59	10	
5	16QAM 1/3	2k	1/8	4,494	8	0,132			B=3 S=2 B+S = 5
5	16QAM 1/3	2k	1/8	4,494	8	0,132			B=17 S=9 B+S = 26

A.13.2.3.3.2 Diversity gain evaluation with the SNR @ FER 5 method

In order to evaluate the diversity gain in outdoor environment, and possibly compare directly none-diversity values with the theory and laboratory tests, additional measurements have been performed using only one single rooftop antenna.This test was done with SH-A receivers from DiBCom® (SHA2).

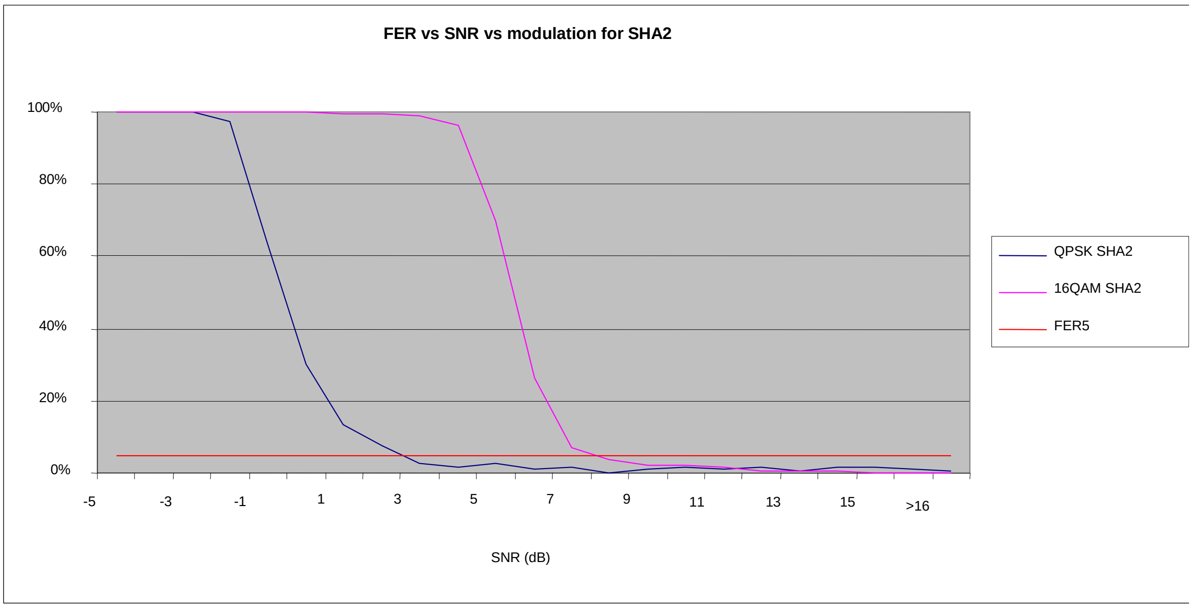


Figure A.13.66: FER vs SNR vs modulations for SHA2 (average) without diversity

The summary of the results from figure A.13.66 is in table A.13.16. For the comparison with lab test, a SHA2 receiver has been used with a TU6 channel. Since the tests in lab have been made only for 5 and 90 km/h the C/N value extrapolated is an average of those 3 values.

Table A.13.16: Summary field tests vs lab tests vs receivers for QPSK 1/3 and 16QAM 1/3 without diversity for SHA2

Receivers	Modulation	Field SNR @ FER=5%	Lab SNR @ FER=5% (50 km/h)
SHA2 average	QPSK 1/3	3 dB	2 dB
SHA2 average	16QAM 1/3	8 dB	7,2 dB

Table A.13.17: Modulation C/N difference without diversity in reception condition C for SHA2

Receivers	Modulation 1	Modulation 2	Field difference	Lab difference (50 km/h)
SHA2	QPSK 1/3 short	16QAM 1/3 short	+5 dB	+3,6 dB

The results were conservative whatever the antenna used (antenna 1 or antenna 2). The trade-off is about the same as when diversity 2 is used. In the following A.13.67, the results with diversity are presented.

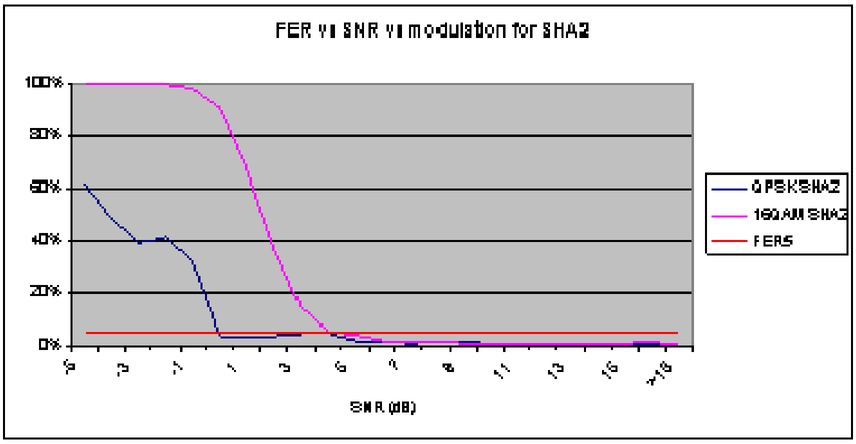


Figure A.13.67: FER vs SNR vs modulations for SHA2 (average) with diversity

The diversity gain can be calculated from the previous results.

Table A.13.18: Diversity gain vs modulations in reception condition C for SHA2

Receivers	Modulation	C/N with diversity	Diversity gain (SNR @ FER=5%)
SHA2 average	QPSK 1/3	0 dB	**3 dB**
SHA2 average	16QAM 1/3	4,5 dB	**3,5 dB**

With this method, the diversity gain is **3 dB for QPSK 1/3** and **3,5 dB for 16QAM 1/3**.

We cannotice also that the performances in lab trials using a TU6 channel emulator are in line (less than 1 dB difference) with on field performances, confirming the TU 6 channel model validity.

A.13.2.3.3.3 Conclusion on diversity gain in outdoor environment (reception condition C)

In outdoor environment, reception diversity of second order brings a significant advantage (more than 3 dB of gain in the tested configurations) on service quality and availability and above all on coverage extension.

A.13.2.3.4 Performances of DVB-SH waveform and diversity gain analysis in reception condition D

A.13.2.3.4.1 Diversity gain evaluation with the SNR @ FER 5 method

In order to evaluate the diversity gain in in-car environment, and compare directly none-diversity values with the theory and laboratory tests, additional measurements have been performed using only one single antenna. Both antennas have been tested in order to ensure the results are conservative and that the diversity gain is equal whatever the reference antenna. As before, that the test was done with SHA2 receivers.

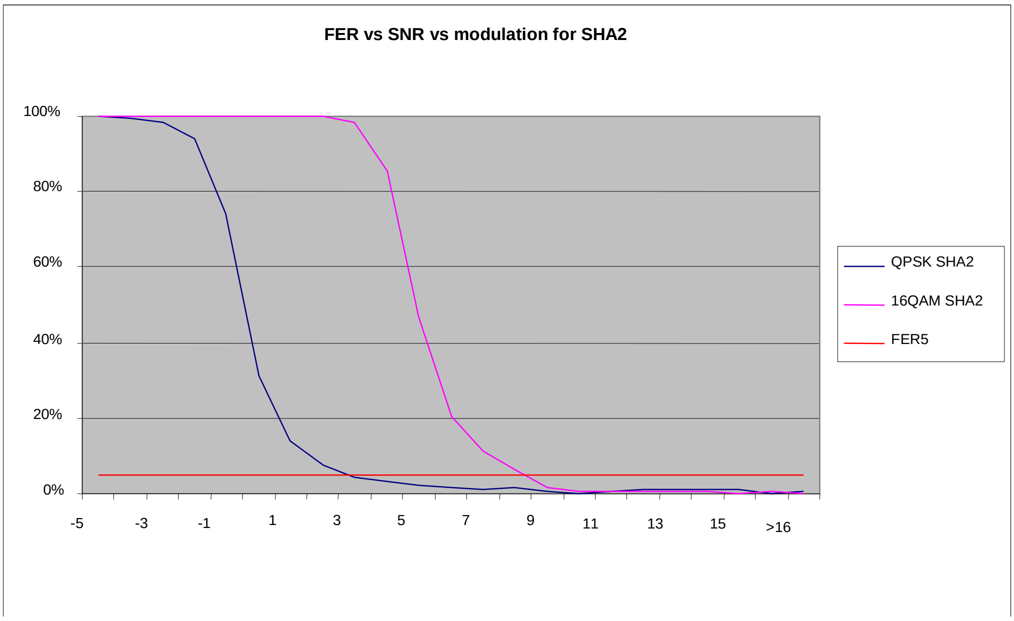


Figure A.13.68: FER vs SNR vs modulations for SHA2 (average) without diversity

The summary of the results is presented in table A.13.19. Regarding the comparison with lab test; the results have been taken from measurements for TU6 channels. As for the outdoor configuration, since the tests in lab have been made only for 5 km/h and 90 km/h, the C/N value extrapolated from those 3 values.

Table A.13.19: Summary field tests vs lab tests for QPSK 1/3 and 16QAM 1/3 without diversity for SHA2

Receivers	Modulation	Field SNR @ FER=5%	Lab SNR @ FER=5% (50km/h)
SHA2 average	QPSK 1/3	3,5 dB	3 dB
SHA2 average	16QAM 1/3	8,5 dB	6,6 dB

Figure A.13.69 and table A.13.20 the results with antenna diversity.

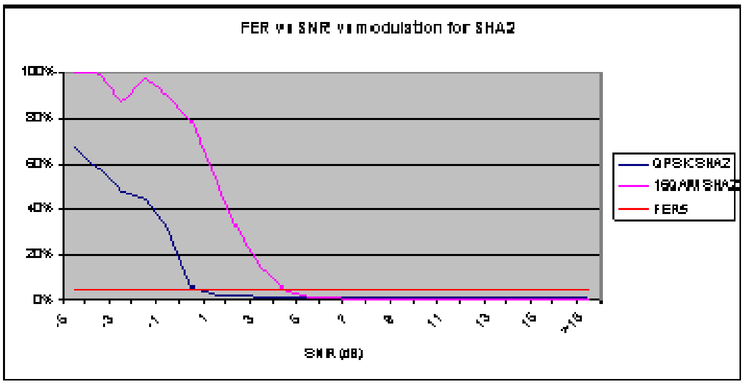


Figure A.13.69: Results with diversity

Table A.13.20 shows the diversity gain calculated from the trial results.

Table A.13.20: Diversity gain vs modulations for in-car environment for SHA2

Receivers	Modulation	C/N with diversity	Diversity gain (SNR @ FER=5%)
SHA2 average	QPSK 1/3	0,5	**2,9 dB**
SHA2 average	16QAM 1/3	4	**4,5 dB**

A.13.2.3.4.2 Conclusion on Diversity gain in reception condition D

In reception condition D, reception diversity of second order brings a significant advantage (more than 3 dB of gain in the tested configurations).

A.13.2.3.5 Impact of the interleaving scheme

The purpose of this set of trials is to assess the impact of the long interleaving scheme. In the following, a class 2 interleaver is compared with and class 1 plus IFEC configuration.

A.13.2.3.5.1 MPE-IFEC interleaving gain in Reception Conditions C and D

The following figures compare DVB-SH C/N performance for Class I without IFEC (named in the legend class 1) and with i-FEC of 5 s duration (named IFEC). Results are presented for the two selected modulations.

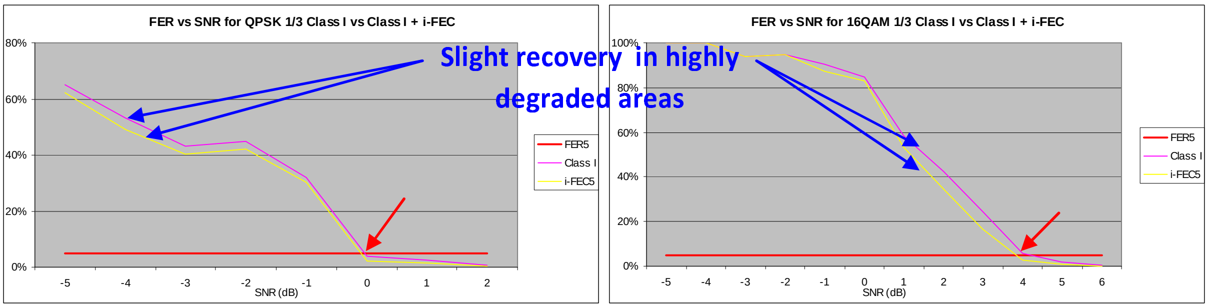


Figure A.13.70: Impact of IFEC 5 s

Figure A.13.70 shows that the 5 s IFEC lightly improves performances in the high PER area while the advantage at FER 5 % goes from marginal (in QPSK) to 0,2 dB (in 16QAM).

There is a sensible improvement when the IFEC length is extended to 26 s as seen in figure A.13.71.

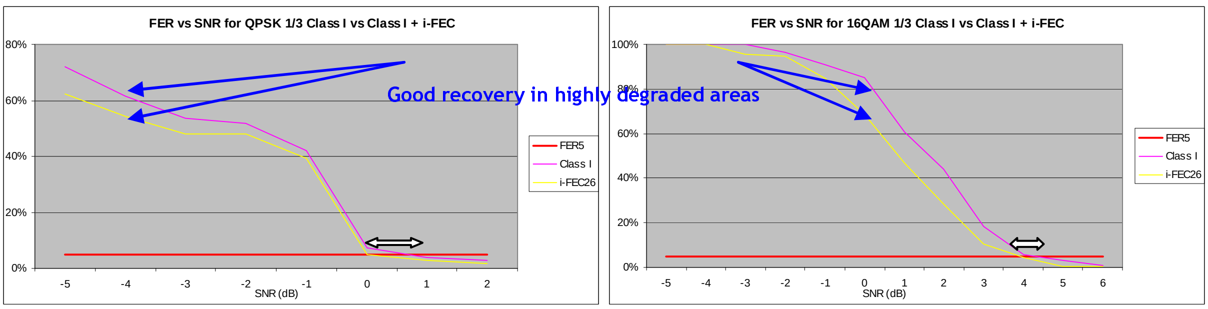


Figure A.13.71: Impact of IFEC 26 s

An IFEC of 26 s provides between 0,5 dB and 0,75 dB gain under pure terrestrial coverage. This is in line with TU6 lab measurements reported in clause A.13.1.

A.13.2.3.5.2 Class 2 interleaving gain in Reception Conditions C and D

This clause tackles the comparison between class 1 (short interleaver) and class 2 (long interleaver). For class 2, two interleaver durations are analysed i.e. 5 s and 10 s class 1 results do not include IFEC.

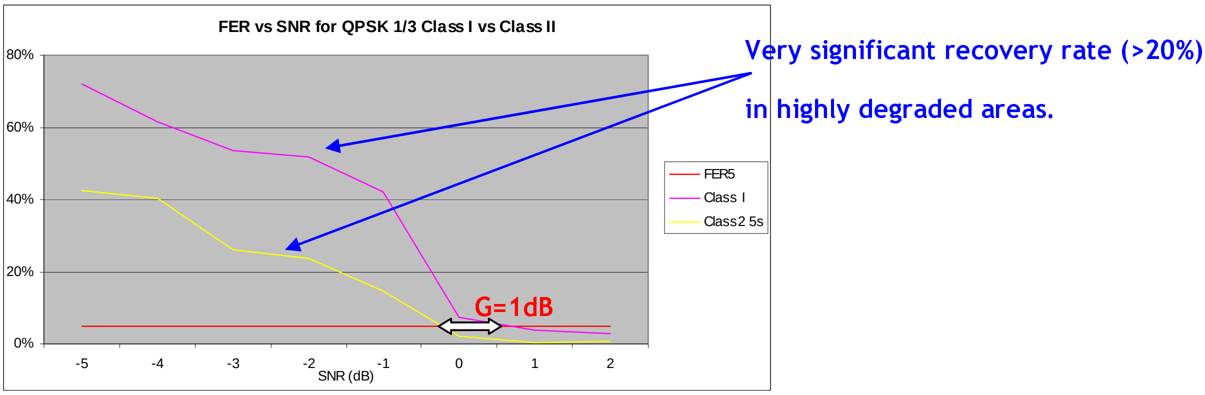


Figure A.13.72: DVB-SH C/N performance for Class I and class 2,5 s

An interleaver of 2,5 s allows recovering up to 20 % of the FER in highly degraded areas with 1 dB of improvement at FER 5 % (see figure A.13.72).

Figure A.13.73 shows the trials results for a 10 s interleaving.

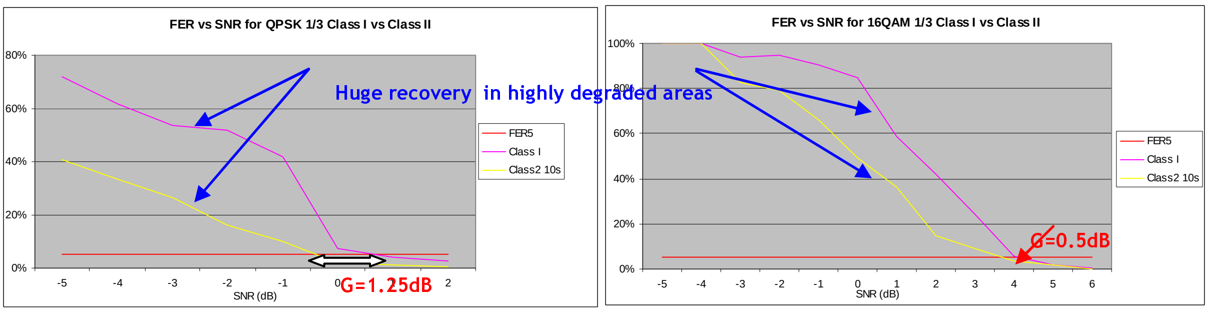


Figure A.13.73: DVB-SH C/N performance for Class I and class 2 10 s

Increasing the interleaving length improves slightly the performance both in highly degraded areas and at PER5% Gains range from 0,5 dB (16QAM) to 1,25 dB (QPSK) under pure terrestrial coverage (slightly better than TU6 laboratory measurements).

A.13.2.3.5.3 Conclusion on the use of long interleaving

Figure A.13.74 compares the performance of IFEC and class 2 configurations.

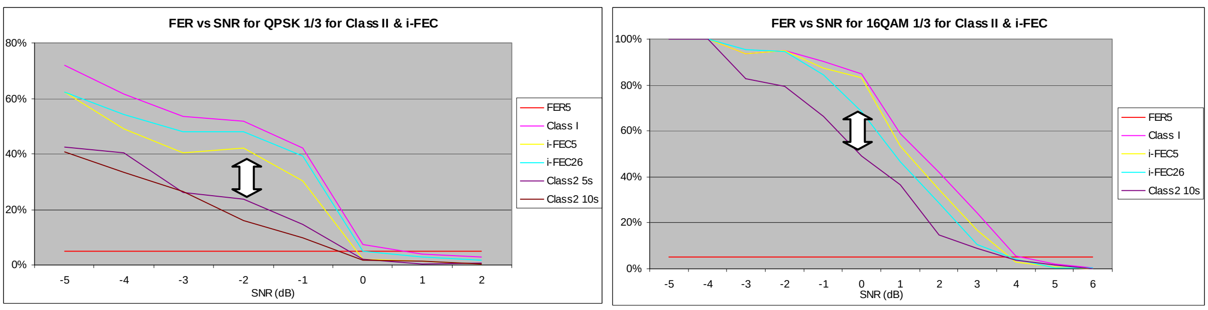


Figure A.13.74: DVB-SH C/N performance for i-FEC and class 2 interleavers

The following conclusions can be drawn:

- 5 s IFEC brings light improvements even in pure terrestrial coverage mainly in degraded areas.
- 26 s IFEC brings more important improvements and a significant gain of C/N in terrestrial coverage.

- class 2 is very efficient in highly degraded areas (20 % to 30 % recovery rate) and with an important gain of C/N for FER5 criteria.
- 5 s and 10 s class 2 show close performances.

A.13.2.4 Synthesis of results

Measurements without antenna diversity

Table A.13.21: Torino results without diversity at medium speed

Modulation/coding	Terminal Type	Terminal Class	C/N (dB) @ FER=5 % vehicular medium speed	C/N (dB) @FER=5 % In-car medium speed
QPSK 1/3	SHA2 Field	class 1	3	3,5
	SHA2 Lab	class 1	2	2
16QAM 1/3	SHA2 Field	class 1	8	8,5
	SHA2 Lab	class 1	7,2	7,2

Measurements of diversity gain outdoor and in car

During the Torino Field Trial diversity measurements have been made with vehicular terminal and with terminal in-car at medium average speed.

Outdoor:

Table A.13.22: Diversity gain in C conditions

Modulation	No diversity C/N (dB)	Diversity C/N (dB)	Diversity gain (dB)
QPSK 1/3 class 1	3	0	3
16QAM 1/3 class 1	8	4,5	3,5

In car diversity gain.

Table A.13.23: Diversity gain in D conditions

Modulation	No diversity C/N (dB)	Diversity C/N (dB)	Diversity gain (dB)
QPSK 1/3 class 1	3,5	0,5	3
16QAM 1/3 class 1	8,5	4	4,5

Improvement with Long Interleaver

Table A.13.24: Improvement with Long Interleaving

Modulation/coding	Terminal Type	Terminal Class	Improvement in C/N (dB) @ FER = 5 % vehicular medium speed
QPSK 1/3	SHA2 Field	Class 1 + IFEC / 5 s	0,1 (Diversity 2)
	SHA2 Lab	Class 1 + IFEC / 5 s	0,2
	SHA2 Field	Class 1 + IFEC / 25 s	0,75 (Diversity 2)
	SHA2 Lab	Class 1 + IFEC / 25 s	Bellow 0,5
	SHA2 Field	Class 2 / 9 s	1,25 (Diversity 2)
	SHA2 Lab	Class 2 / 9 s	No impact
16QAM 1/3	SHA2 Field	Class 1 + IFEC / 5 s	0,2 (Diversity v 2)
	SHA2 Lab	Class 1 + IFEC / 5 s	0,2
	SHA2 Field	Class 1 + IFEC / 25 s	0,5 (Diversity 2)
	SHA2 Lab	Class 1 + IFEC / 25 s	0,5
	SHA2 Field	Class 2 / 9 s	0,5 (Diversity 2)
	SHA2 Lab	Class 2 / 9 s	No impact

A.13.2.5 Measurements results of a DVB-SHA satellite signal from Paris to Barcelona

During February 2010, the QoS of the DVB-SHA signal transmitted in S-Band by the W2A satellite was measured in a car diversity receiver via two antennas mounted on the roof.

A.13.2.5.1 Receiver platform description

The modulation parameters were as follows:

- DVB-SHA class 1.
- Frequency carrier : 2 197,5 MHz.
- Right Hand Circular Polarization (RHCP).
- C/N threshold measured in Laboratory:
  - AWGN: C/N = -1 dB.
  - TU6: C/N = 1,3 dB.

The detail of the antennas mounted on the car roof is shown on figure A.13.75.

Table A.13.25: Waveform configuration

BW (MHZ)	Modulation/ coding	FFT	GI	Capacity (Mbps)	Short Interleaving	Short Interleaving duration (s)	Uniform Interleaving structure	Uniform duration (s)	MPE-IFEC
5	QPSK 1/2	2k	1/4	2,222	8	0,265	N/A	N/A	N/A

The configuration corresponds to table A.10.8, ID n° 7 in clause A.10.

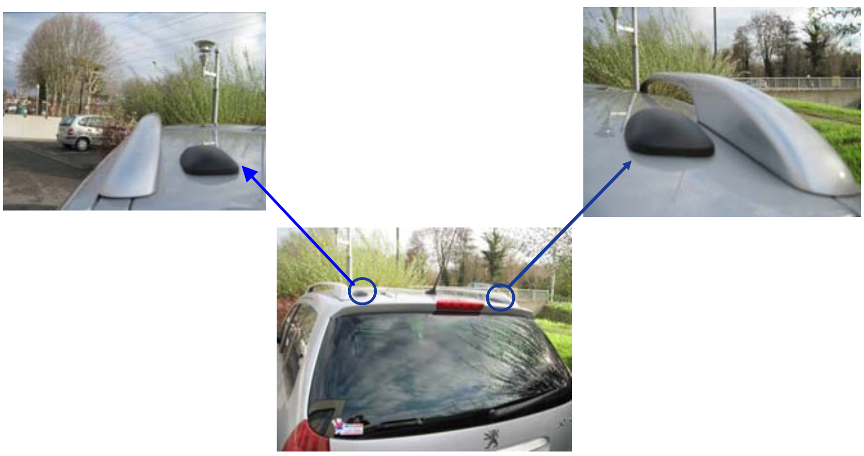


Figure A.13.75: Vehicle equipment overview

The vehicle roof rails were removed to avoid masking the satellite signal. The chipset used for the test integrates two tuner-demodulator working in diversity mode (MRC). The received signal was sampled every second and a monitoring software delivered all the needed information to monitor and post-process the trial information (RF level, C/N, Frame_Lock, GPS parameters, etc.).

A.13.2.5.2 Main measurements results

A.13.2.5.2.1 Erroneous seconds on over 1 000 km of highways

Considering the Frame_Lock criteria, the erroneous second rate (~MFER) is depicted on figure A.13.76 map for route segments of about one hour of duration.

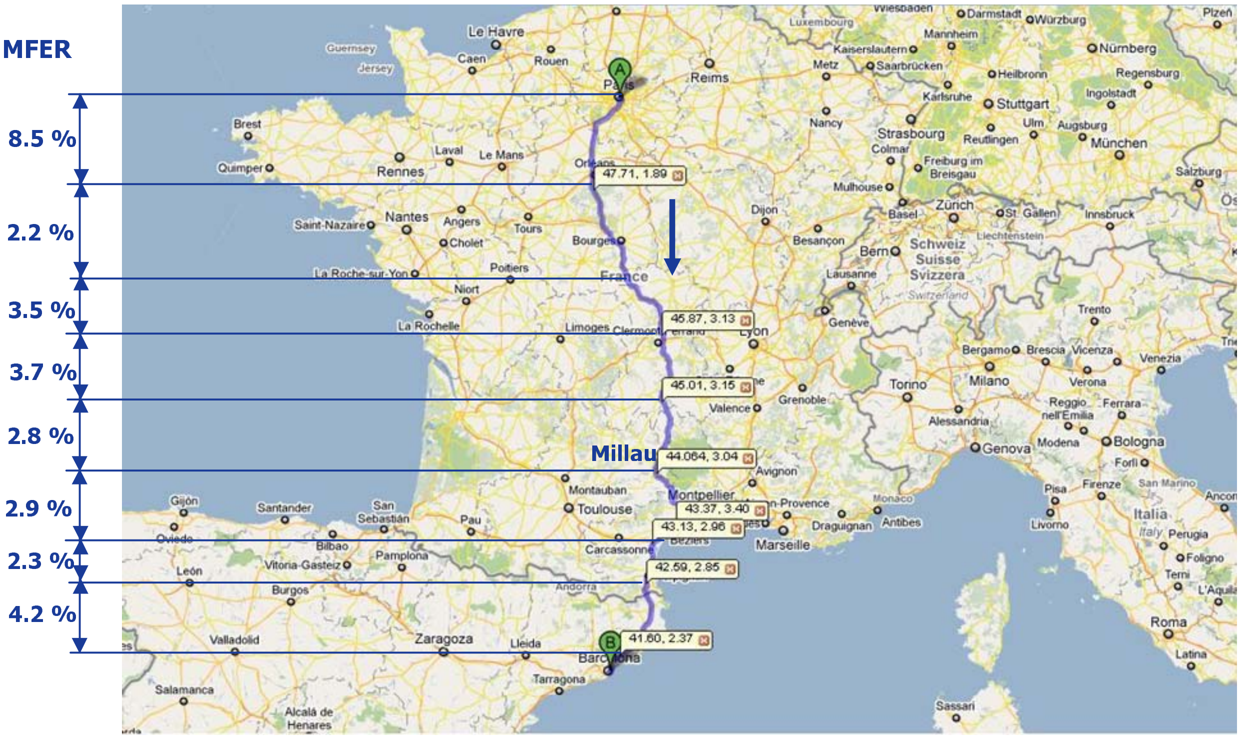


Figure A.13.76: MFER results all over the travel

The global MFER calculated along the whole route is of only 4,2 % (without MPE_IFEC), ESR5(20) fulfilment is about 85 %, in line with the results in clause A.12.

A.13.2.5.2.2 "Frame_lock" loss events distribution analysis

The probability density function of the loss of frame lock (masks) duration (figure A.13.77) shows that on the Highway:

- up to 62,6 % of the frame lock loss events have a duration less or equal to 1 s.
- up to 24,7 % of the frame lock loss events have a duration between 1 s and 2 s.

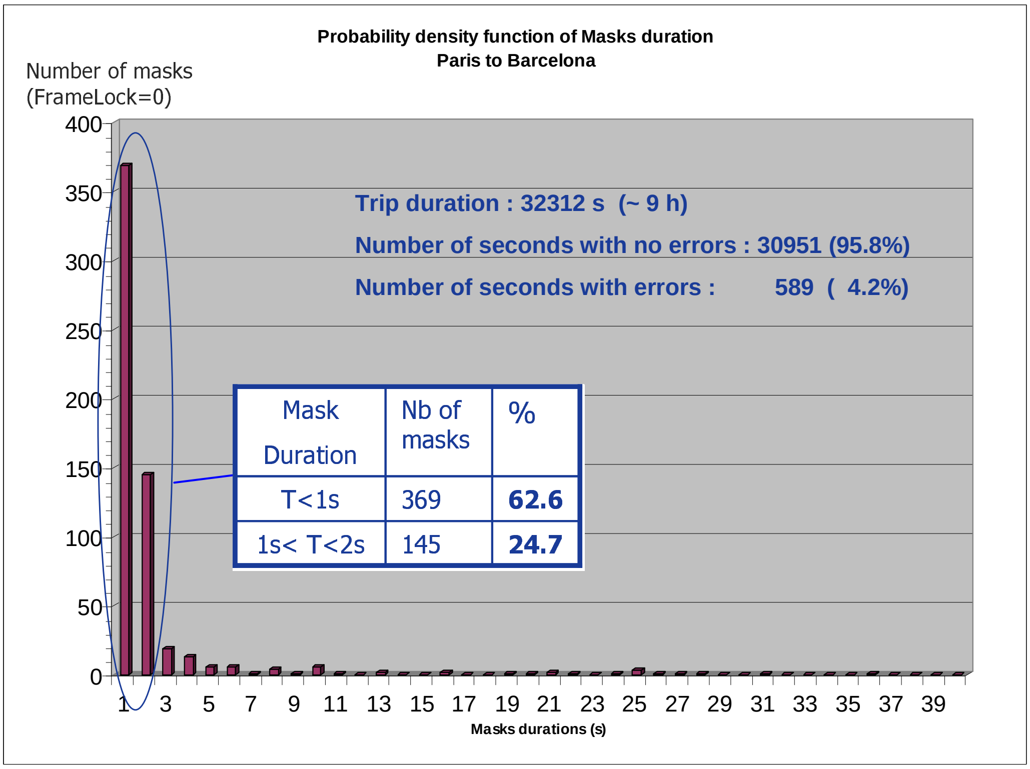


Figure A.13.77: Probability density function of masks duration

During the trip from Paris to Barcelona it was observed, that most of these short masks were due to short bridges crossing over the highway and to road signs masking the satellite. The associated cumulative distribution is depicted on figure A.13.78.

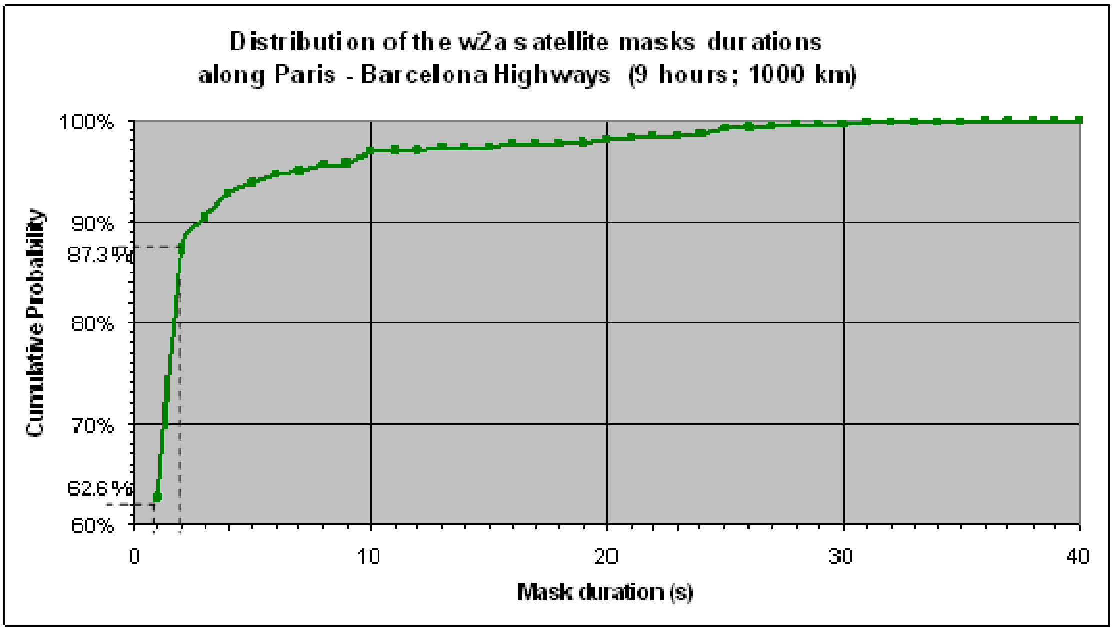


Figure A.13.78: Distribution of the W2A satellite masks duration

A.13.2.5.2.3 Erroneous seconds versus interleaving depth

The tests described above were made without class 2 interleaver or MPE-IFEC protection (only the 300 ms interleaver was active). However, from the traces recorded in the field, the performance of a class 2 interleaver can be emulated. Figure A.13.79 shows the remaining erroneous seconds versus the interleaver depth, calculated along the whole route.

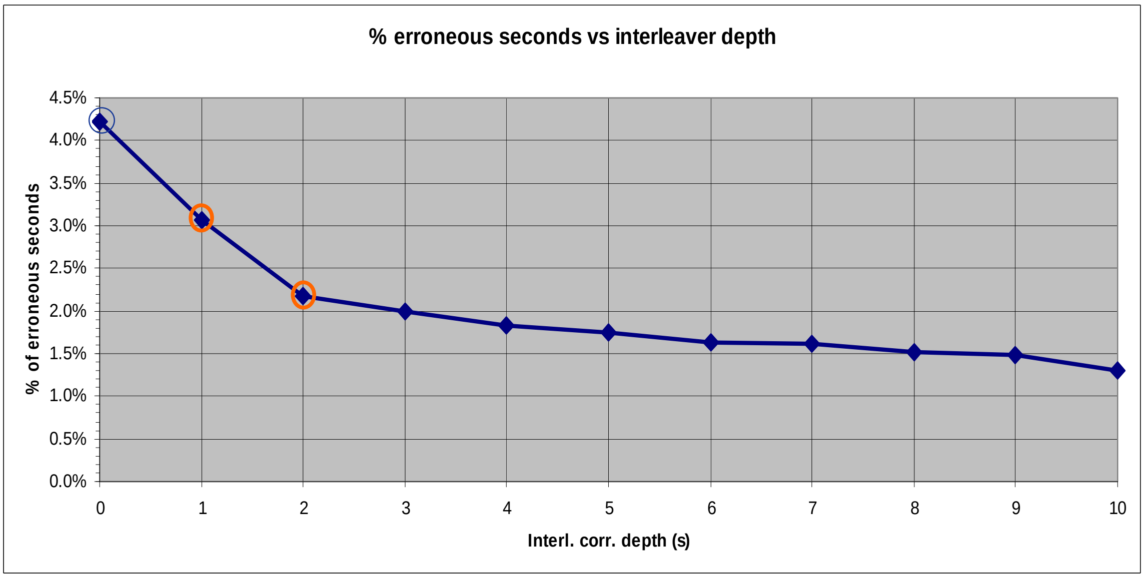


Figure A.13.79: Percentage of erroneous seconds versus interleaving depth

In conclusion:

- If the interleaver is able to compensate for interruption up to **1 s** the erroneous second ratio is reduced to **3,1 %** instead of **4,2 %**.
- If the interleaver is able to compensate for interruption mask of up to **2 s the erroneous second ratio is divided by nearly two: 2,2 % instead of 4,2 %**.

A.13.2.5.2.4 Percentage of interruptions versus interleaving depths

In addition to the decreasing of erroneous seconds ratio the interleaver decreases also dramatically the number of interruptions as seen in figure A.13.80.

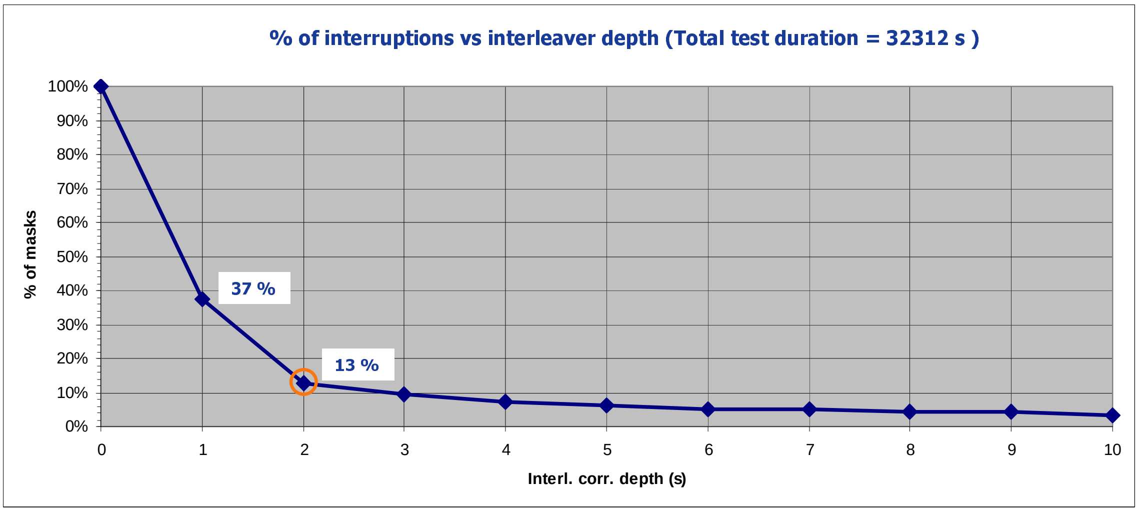


Figure A.13.80: Percentage of interruptions versus interleaving depth

Only 13 % of the initial number of interruptions remains after correction by an interleaver of 2 s depth.

A.13.2.5.3 Conclusions

- The vehicular outdoor quality of coverage of the W2A satellite tested between Paris and Amsterdam and Paris to Barcelona confirms the good performance of DVB-SH in satellite only coverage in a highway environment with ICO and Devas satellites trials.

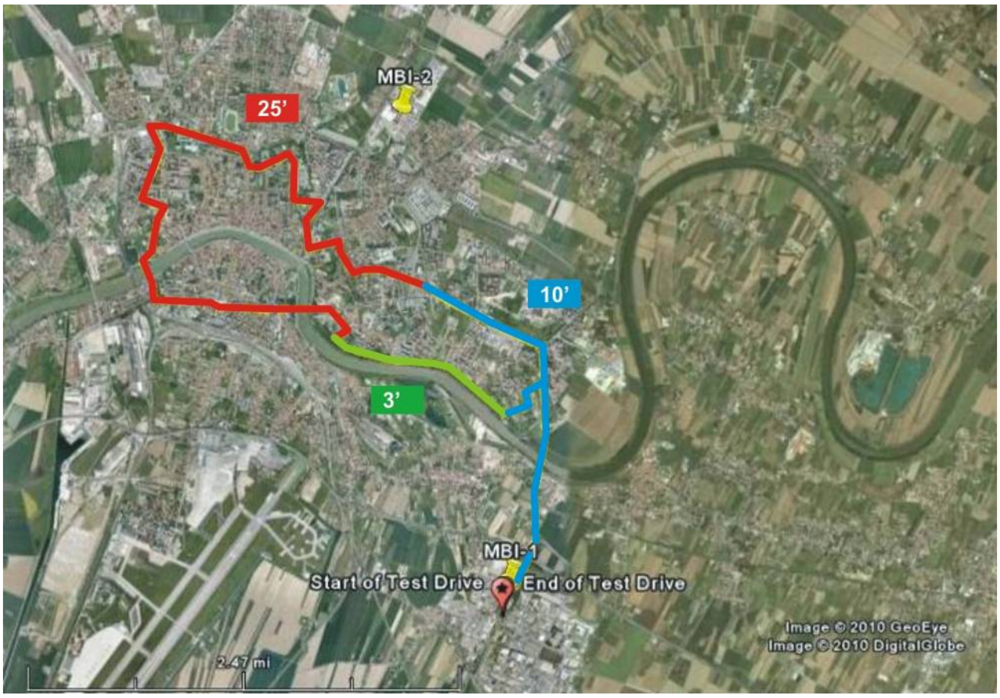
- Between Paris and Barcelona the measured MFER (in DVB-SHA and without any long interleaving process) was of around 4 % along 1 000 km and during 9 hours, corresponding to ESR5(20) fulfilment ratio of 85 %.
- An interleaving process correction simulation based on "frameLock" burst modelisation over the Paris-Barcelona highway shows that up to 87 % of the burst have a duration less than 2 s.
- For Mobile TV reception from satellite, in a car travelling on **highway**, the results indicate a small interleaving with a capacity correction of about 2 s is sufficient for correcting most of the interruptions (road bridges, signalling panels, etc.).
- Along the Paris-Barcelona route the resulting MFER, after a 2 s Interleaving/de-interleaving process, is approximately divided by two compared to the original one, corresponding to ESR5(20) fulfilment ratio of 95 %.

A.13.2.6 Pisa trial

These trials were carried out by a consortium of companies and institutions: Eutelsat, Fraunhofer IIS, DLR, UBS, SIDSA, MBI and Calearo co-funded under ESA ARTES34 program. The objective of these trials was to assess the performance of DVB-SH in the satellite and hybrid coverage provided in a variety of environments. Although the trials covered an extended set of configurations, this section will focus of DVB-SHB results.

A.13.2.6.1 Experimental Setup

The position of the two CGC repeaters (named MBI-1 and MBI-2) with respect to the test area is indicated in figure A.13.81. The two repeaters, both located at a rough height of 15 meters, aim to cover the city centre of Pisa, equipped with a 50 W amplifier over a unique sector, providing an EIRP of ~47 dBm+18 dB (antenna gain) - 3 dB (losses) = 62 dBm. Both repeaters were directly pointing the center of Pisa. The CGC intends to complement the satellite coverage provided by the W2A satellite. The overall duration of the route does indeed depend on the traffic conditions. Nevertheless, the estimated time for the completion of each session is in the order of ~38 minutes.



NOTE: The route used for the measurements is tracked (red-urban, blue-suburban, green-tree shadowing, below). Pisa, Italy. These pictures have been obtained via GoogleMaps (maps.google.com).

Figure A.13.81: Position of the CGC repeaters over the test area (above)

The measured received signal power (terrestrial only) along the route is depicted in figure A.13.82, while the predicted C/N is displayed in figure A.13.83.

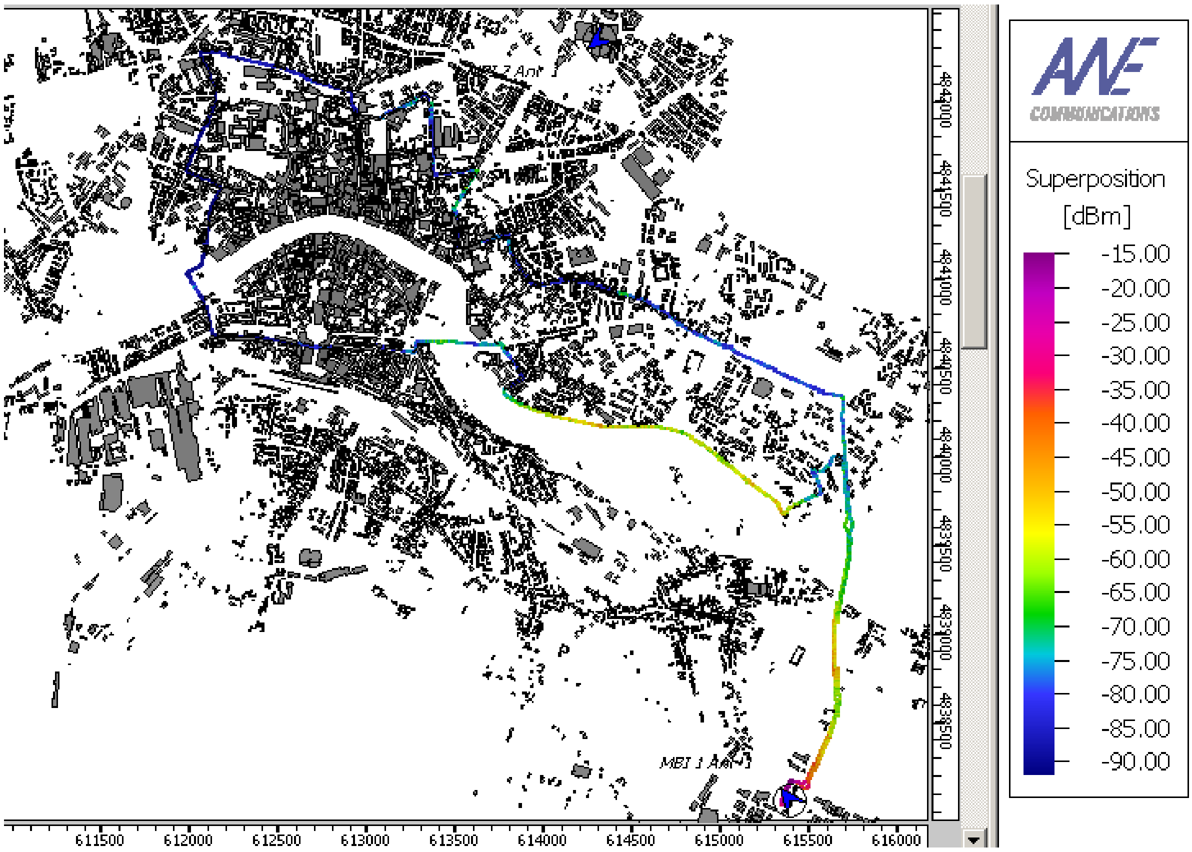


Figure A.13.82: Measured received power on Route 1 (terrestrial signal only)

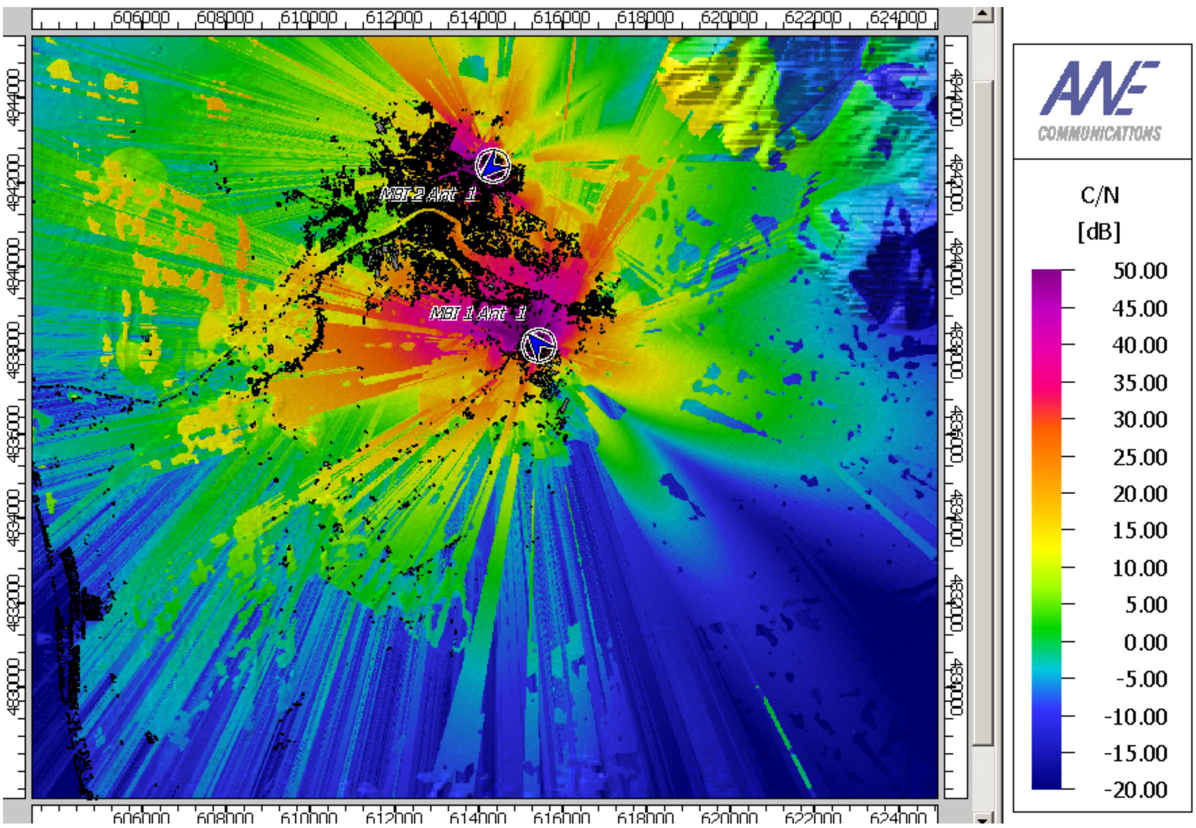


Figure A.13.83: Predicted C/N for the terrestrial signal

Figure A.13.84: Route 1

Figure A.13.85: Route 1, point A (left) and B (right)

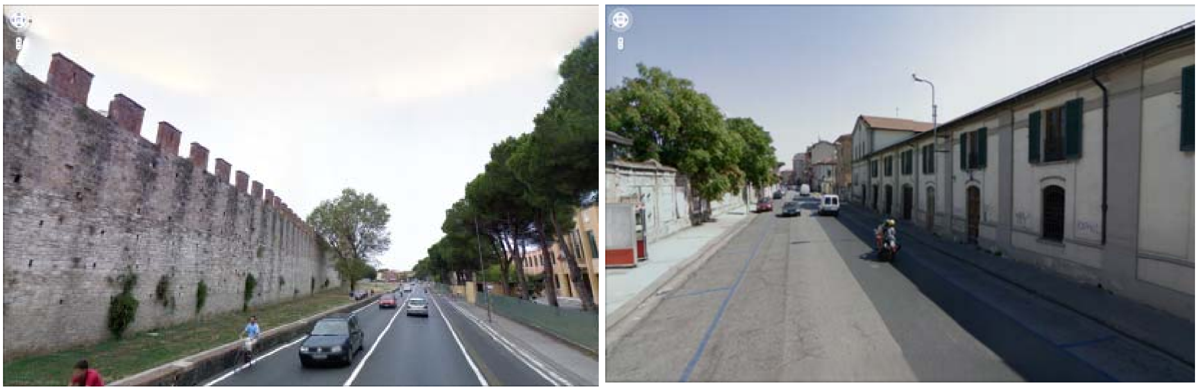


Figure A.13.86: Route 1, point C (left) and D (right)

Figure A.13.87: Route 1, point E

A.13.2.6.2 Description of the trials conditions

Trials were carried out on the above-described route in different configurations of the network: in satellite-only coverage, in terrestrial-only coverage, and with code combining. Whenever possible, for the tests, two receivers provided by two distinct manufacturers have been simultaneously used. The measurement setup has been organized in a way that the two receivers had access to the same antenna signal. The receivers were operated with a G/T of -21,5 dB/ °K. The LOS C/N measured on the satellite component is around 9 dB. The output of the two receives was input to dumping servers, collecting measures of quality of service at different layers, e.g.:

- C/N values.
- Physical layer / Transport Stream level error flags.
- MPE-level (burst) error flags.

A common dumping architecture was foreseen for all the receivers used during the Pisa trials. As showed in figure A.13.88 each receiver has to provide the output data to a centralized server, that will handle them in parallel with the data extracted from a GPS receiver, from a gyroscope system and from a fish-eye camera.

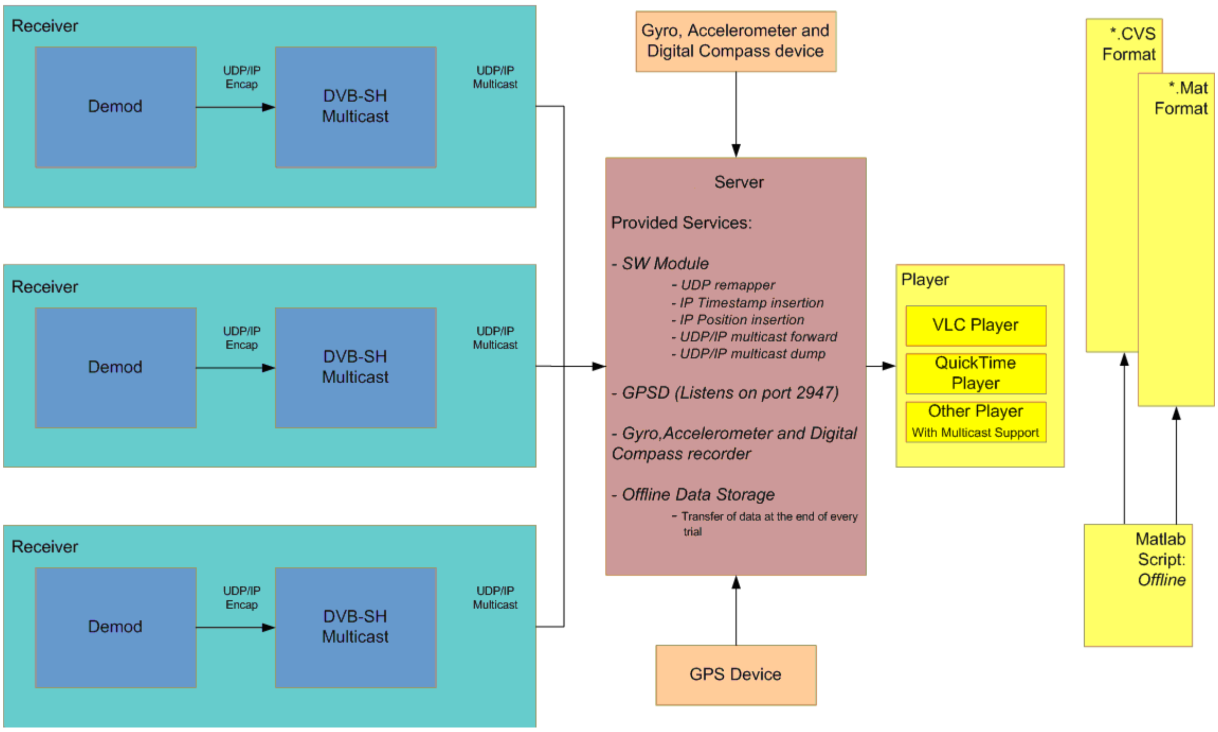


Figure A.13.88: Dumping architecture

The first receiver was equipped with the following features:

- S-Band Tuner.
- Antenna diversity with up to 4 antennas.
- DVB-SH OFDM and TDM demodulator.
- Maximum-Ratio-Combining (MRC).
- Code-Combining (CC) between OFDM and TDM signals.
- Class 1 and class 2 decoding capabilities.

The log-likelihood ratio data streams coming from identical demodulators are combined using the MRC method. The combined LLR are de-interleaved and then handed over to the turbo decoder. MPE-IFEC decoding (for class 1 configurations) is performed on a separate platform with a software (ANSI-C) implementation. The second receiver is a modular, multi-standard (DVB-T/H/SH) FPGA-based real-time prototyping. The real-time prototype is divided in two main parts:

- The main board. It is built around a Virtex-5 LX330 FPGA. This board contains the digital part of the demodulator. It also contains the external memories needed for Class-2 deinterleaver, along with the ARM7 main processor and the interfaces with the host.
- The RF board. This board contains a multi-band (UHF, S-Band) NXP tuner.

For these field trials, the demodulator contains two RF boards and two main (digital) boards where the whole receiver is mapped. The ARM7 processor subsystem adds flexibility to the receiver and controls the reception process.

The receiver supports the entire DVB-SH standard:

- Class 1 and class 2 deinterleaver profiles.
- SH-A, either MFN or SFN configurations. In SFN configuration, MRC antenna diversity is supported.
- SH-B, with Code-Combining between TDM and OFDM signals.
- MPE-IFEC processing is also performed by the demodulator.
- SH-B with code-combining and MRC antenna diversity of 2 could be supported by interconnecting several receivers for a total of 4 receiving antennas.

A.13.2.6.3 Selected configurations

The two configurations (denoted in the following of the document as SHB#2 and SHB#8) selected for the trials and are summarized in tables A.13.26 and A.13.27.

Table A.13.26: Configurations used for the on-field trials

	Configuration Setting		Set of values	SH-B #2		SH-B#8	
				SAT	CGC	SAT	CGC
	Class			Class 1	Class 1	Class 2	Class 2
	CIF (352*288) 25 fps		Session 4 - Italy	QPSK_1/2_S	16QAM_1/3_S	8PSK_1/3_U	16QAM_1/3_UL
	Frequency	Bandwidth [MHz]	5	5	5	5	5
		Channel Frequency [MHz]	2187.5 - 2192.5 - 2197.5	2192.5	2197.5	2192.5	2197.5
		Polarization		RHCP	V	RHCP	V
	Transmission Power (Satellite/Terrestrial)	Transmitted power [dB] (Sat or LP-HP)					
		OBO [dB]		2.5	-	2.5	-
	Performance	Theoretical C/N required in AGWN @BER 10^-5		1.1	3.7	1.6	3.7
	Spectral Efficiency (including LL protection)	Sat/Terr (bit/sec/Hz)		0.67	0.89	1	1.25
	Max. Network Delay	(ms)		600		600	
	W2A feeder link Symbol Rate	(Symbols/sec)			4821428		4821428
OFDM	Physical Configuration	FFT mode			2K		2K
		GI			1/4		1/4
		Modulation			16 QAM		16 QAM
		PHY FEC rate			1/3		1/3
		Interleaver configuration (com_mult/nof_late_taps/no_slices/ slice_dist/ not_late_incr)			5/48/1/0/0		20/24/9/5/12
		Early / late interleaver duration (ms)			0 / 122		0 / 5118
		MPEG TS total bit rate (Mbps)			4.443277		4.443277
	TS Layer Configuration	Services			2		2
		Repetition interval (ms)			975		975
		Burst duration (ms)			122		488
		TP per SH frame			360		360
TDM	Physical Configuration	TDM modulation		QPSK		8PSK	
		Roll-Off		0.15		0.15	
		PHY FEC rate		1/2		1/3 compl.	
		Interleaver configuration (com_mult/nof_late_taps/no_slices/ slice_dist/ not_late_incr)		5/48/1/0/0		60/0/12/4/2	
		Early / late interleaver duration (ms)		0/244		0/ 5605	
		Ext.. delay offset - TMC modulator only (ms)		60.14687179		56.24123077	
		MPEG TS total bit rate (bps)		3,850		3,850	
	TS Layer Configuration	Services		2		2	
		Number of SH frames		8		8	
		Synchronization LINK / PHY		No		No	
		Repetition interval (ms)		975		975	
		Burst duration (ms)		488		488	
		CW per SH frame		39		39	
		TP per service (total)		1248		1248	
Service	Reference Stream Parameters	PID [HEX]		101		101	
		UDP/IP Address		225.2.4.1		225.2.4.1	
		A/V port		4002/4000		4002/4000	
	Matrix dimension (FEC or IFEC)	ROWS Number (T)		768	768	768	768
		Applicative Number (C)		128	128	128	128
	Link Layer Configuration MPE-IFEC Parameters	LL-FEC Rate		0.66 (2/3)	0.66 (2/3)	1	1
		IFEC Columns (R)		64	64		
		B		17	17		
		S		9	9		
		D (D=0 or D=B+S)		0	0		
		Service bitrate at TS level (kbps)		910	910	485	485

Clearly the configuration SH B#2 deals with a Class-1 type, with satellite component configured with QPSK modulation and coding rate 1/2 (refer to table A.10.32, ID 7 in clause A.10, with a slightly longer interleaver, 244 ms vs. 181 ms), while the CGC part is configured with 16QAM and code rate 1/3 (refer to table A.10.6, ID 5 clause A.10, with a slightly longer interleaver, 122 ms vs. 105 ms). The scheme is complemented by an MPE-IFEC with B=17 and S=9, with nominal coding rate 2/3 (although after post-processing, it resulted to be actually closer to ½ due to padding).

The configuration SH B#8 is a class 2 type, with an interleaver of 5,6 s. The satellite component is protected by a rate 1/3 code, with 8-PSK modulation (close to refer to table A.10.53, ID 2 in clause A.10), while the CGC is still based on a 16QAM signalling and on coding rate 1/3 (refer to table A.10.25, ID 2 in clause A.10, with a different interleaver profile).

Table A.13.27: Summary of the configurations used for the trials, with reference to clause A.10

ID	Waveform configuration	Additional Parameters	Spectral Efficiency	SAT C/N	C/I	Comments
SHB2	SAT:T-QPSK_1/2_S	Table A.10.32 ID 7	0,67 (with IFEC)			AWGN C/N req=1,1 dB
	TER:16QAM_1/3_S	Table A.10.6 ID 5	0,89 (with IFEC)			AWGN C/N req=3,7 dB
SHB8	SAT:T-8PSK_1/3_U	Table A.10.53 ID 2	1			AWGN C/N req=1,6 dB
	TER:16QAM_1/3_UL	Table A.10.25 ID 2	1,25			AWGN C/N req=3,7 dB

A.13.2.6.4 Trial results

The trials results are summarized in table A.13.28. The overall test duration is indicated, and it is an approximated value given by the average dump lengths for the two receivers. Note that the slight differences in length of the dumps is due to different starting/stopping instants, and are usually less than 1 minute, which in any case may result in a bias of the measured service availabilities in the order of ±1 %.

We recall that the two configurations tested on the field:

- A class-1 configuration (SHB2), with satellite component configured with modulation QPSK and coding rate 1/2, while the CGC part is configured with 16QAM and code rate 1/3. The scheme is complemented by an MPE-IFEC with B=17 and S=9, with nominal coding rate 2/3. The MPE-IFEC duration is 25 s. The actual measured code rate for MPE-IFEC is actually lower, due to padding in the ADT tables, and it is close to 1/2. Hence, the overall efficiency of the class-1 configuration for the TDM part is approximately 2×(1/2)×(1/2)=0,5 b/s/Hz.
- A class-2 configuration (SHB8), with an interleaver of 5,4 s. The satellite component is protected by a rate 1/3 code, with 8-PSK modulation, while the CGC is still based on a 16QAM signalling and on coding rate 1/3. Hence, the overall efficiency of the class-2 configuration for the TDM part is approximately 3×(1/3)=1 b/s/Hz.

Table A.13.28: Summary of the field trial results.

Configuration	Receiver Class	Test duration [s]	Network status		ESR5(20) Service Availability	
			Satellite	CGC	RX 1	RX 2
SHB#2	1	1900	ON	ON	94%	-
SHB#2	1	2050	ON	ON	95%	-
SHB#2	1	2100	ON	ON	92%	93%
SHB#2	1	1700	ON	-	55%	54%
SHB#2	1	2050	ON	-	48%	52%
SHB#2	1	1750	OFF	ON	72%	70%
SHB#8	2	2100	ON	ON	89%	-
SHB#8	2	2120	ON	ON	91%	-
SHB#8	2	2050	ON	ON	-	90%
SHB#8	2	2150	ON	ON	90%	-
SHB#8	2	1850	ON	-	39%	40%
SHB#8	2	2600	ON	OFF*	50%	-
SHB#8	2	2200	OFF	ON	68%	65%

* - CGC turned off to avoid interference of the sat signal

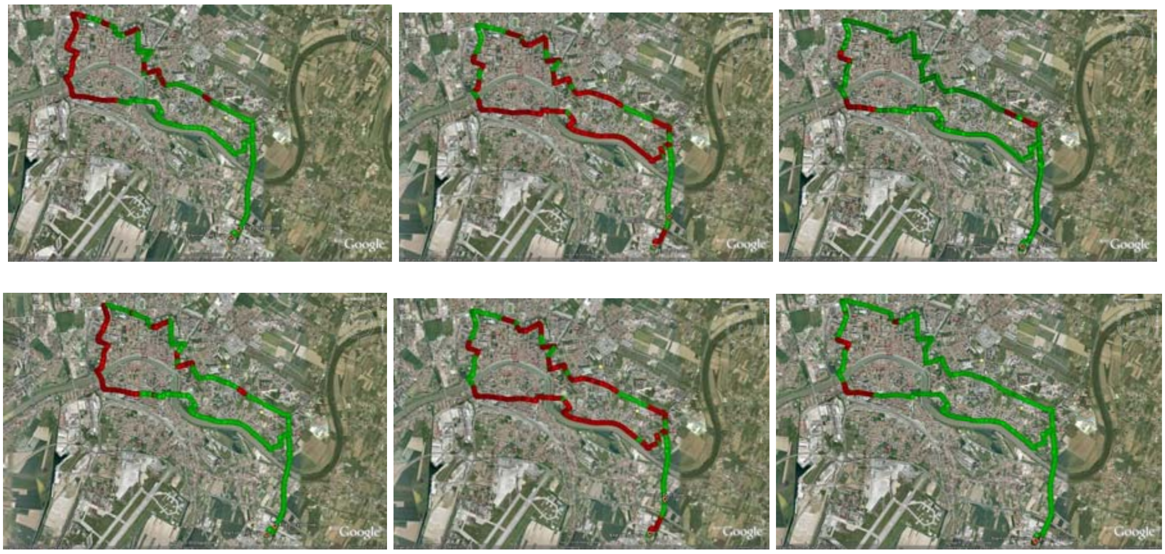
Tests have been carried out with 3 different network topologies: satellite-only (with CGC switched on) trials, terrestrial-only trials, and code-combining trials. A summary of test results is terms of service availability (ESR5(20) criterion) is provided in table A.13.28. The Class-1 configuration is clearly advantaged by its lower spectral efficiency with respect to the Class-2 configuration. However, in both cases, the code combining feature has ensured a seamless handover between satellite and terrestrial signals that is visible in the good service availability percentages achieved in hybrid reception. For the Class-1 configuration, service availability close to 95 % is achieved in hybrid reception, while a terrestrial-only reception would provide roughly 73 %. Similarly, for the class 2 configuration, a service availability of 90 % can be achieved thanks to code combining, while in the terrestrial-only case it would not exceed 70 %. In the table provided next, the performance of the two receivers in different sessions is depicted as well, confirming the stability of the obtained results.

More insights on the performance of the system can be obtained by evaluating the ESR5(20) criterion over location. The results for the class-1 case are provided in figure A.13.89, while those for the class-2 configuration are presented in figure A.13.90.

NOTE: Terrestrial only (left), Satellite only (mid), code combining (right). Red spots identify areas in which the ESR5(20) criterion is not fulfilled. Receiver 1 (top) and 2 (bottom).

Figure A.13.89: Service availability vs. location, class 1 configuration

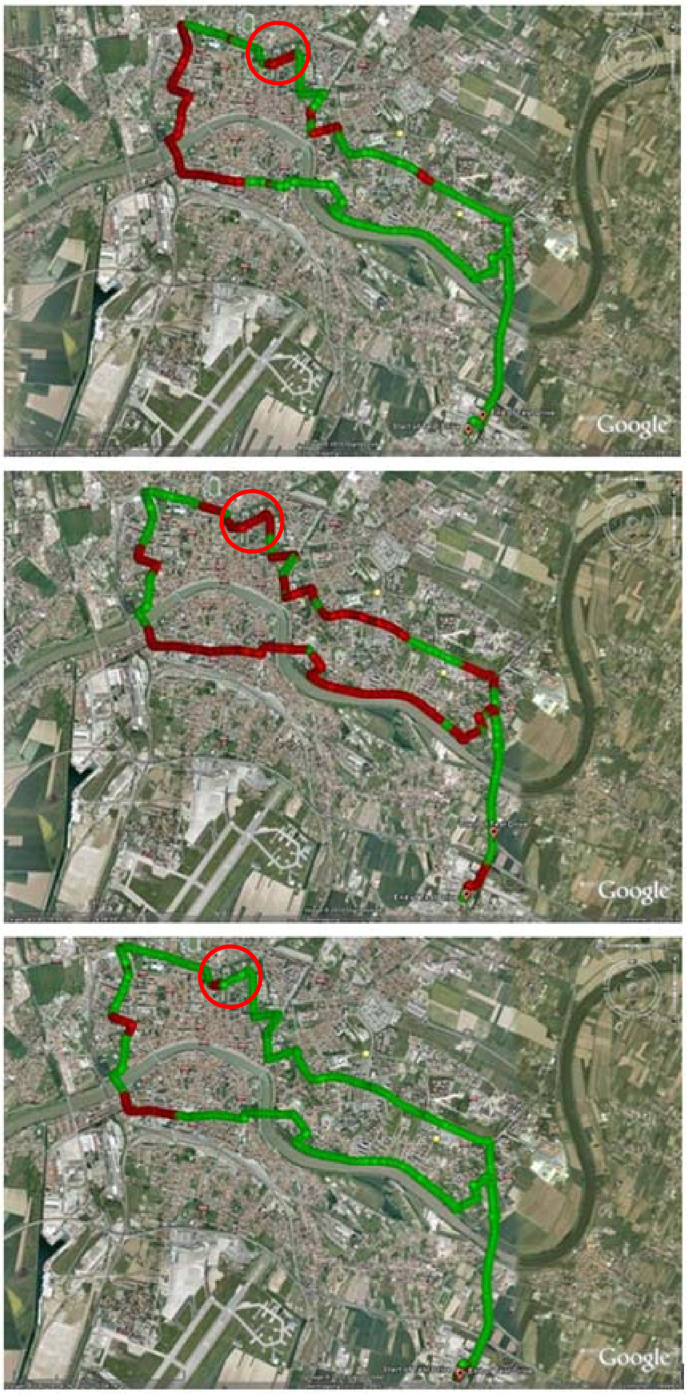
In both cases, the satellite component seems to suffer mostly in the parts of the route which are developed along the East-West axis. Interestingly, in areas where the neither the satellite component nor the CGC alone can provide coverage, their combination is successful. This is especially evident in figure A.13.90, on the western and in the northern parts of the route.



NOTE: Terrestrial only (left), Satellite only (mid), code combining (right). Red spots identify areas in which the ESR5(20) criterion is not fulfilled. Receiver 1 (top) and 2 (bottom).

Figure A.13.90: Service availability vs. location, class 2 configuration

The role of code combining is especially visible in the class 2 configuration case. A larger detail on that is presented in figure A.13.91: a part of the route (urban environment) where both the terrestrial only and the satellite only coverage are not sufficient to fulfil the ESR5(20) criterion while the code combining allows recovering the errors.



NOTE: Class 2 configuration. Terrestrial only (top), Satellite only (mid), Code combining (bottom).

Figure A.13.91: Role of code combining

An environment-based analysis is presented hereafter. The route has been split in 5 parts (according to figure A.13.92). The initial/final part of the route has been divided in two sub-parts (to take into account the different travelling direction), see figure A.13.92, parts A1/A2.

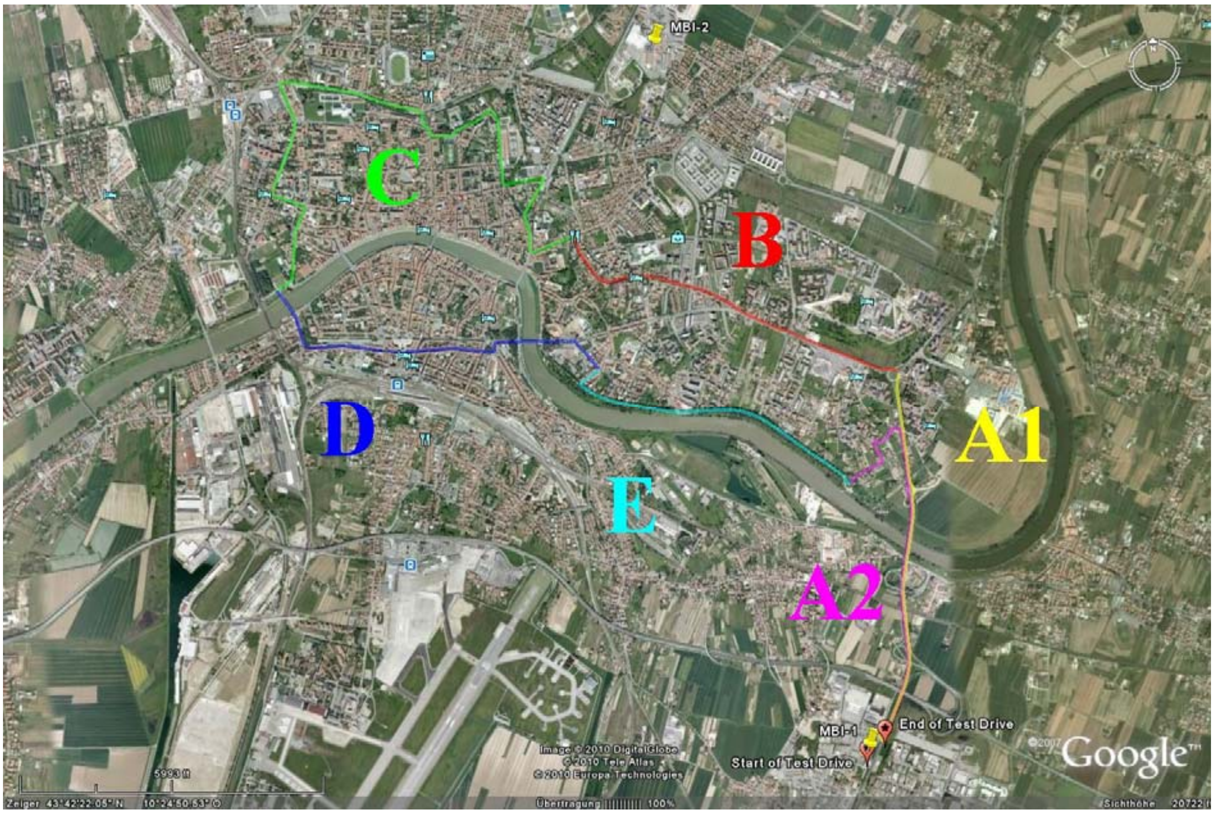


Figure A.13.92: Route segmentation by environment

For the class-1 configuration, the analysis is provided in the next figures.

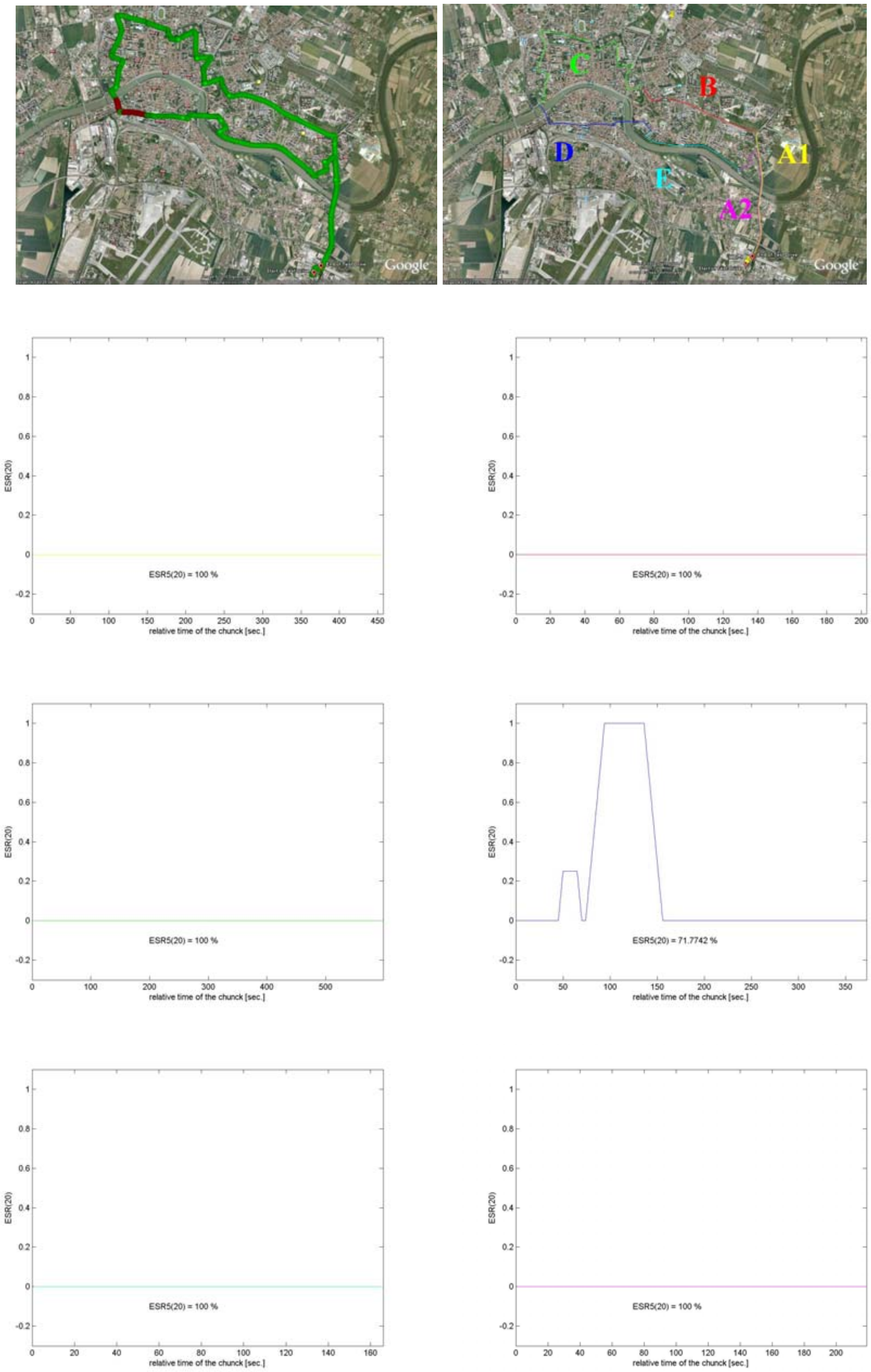


Figure A.13.93: Window status and ESR5(20) calculated for the different environments: A1 (top left), B (top right), C (centre left), D (centre right), E (bottom left), A2 (bottom right) Code combining

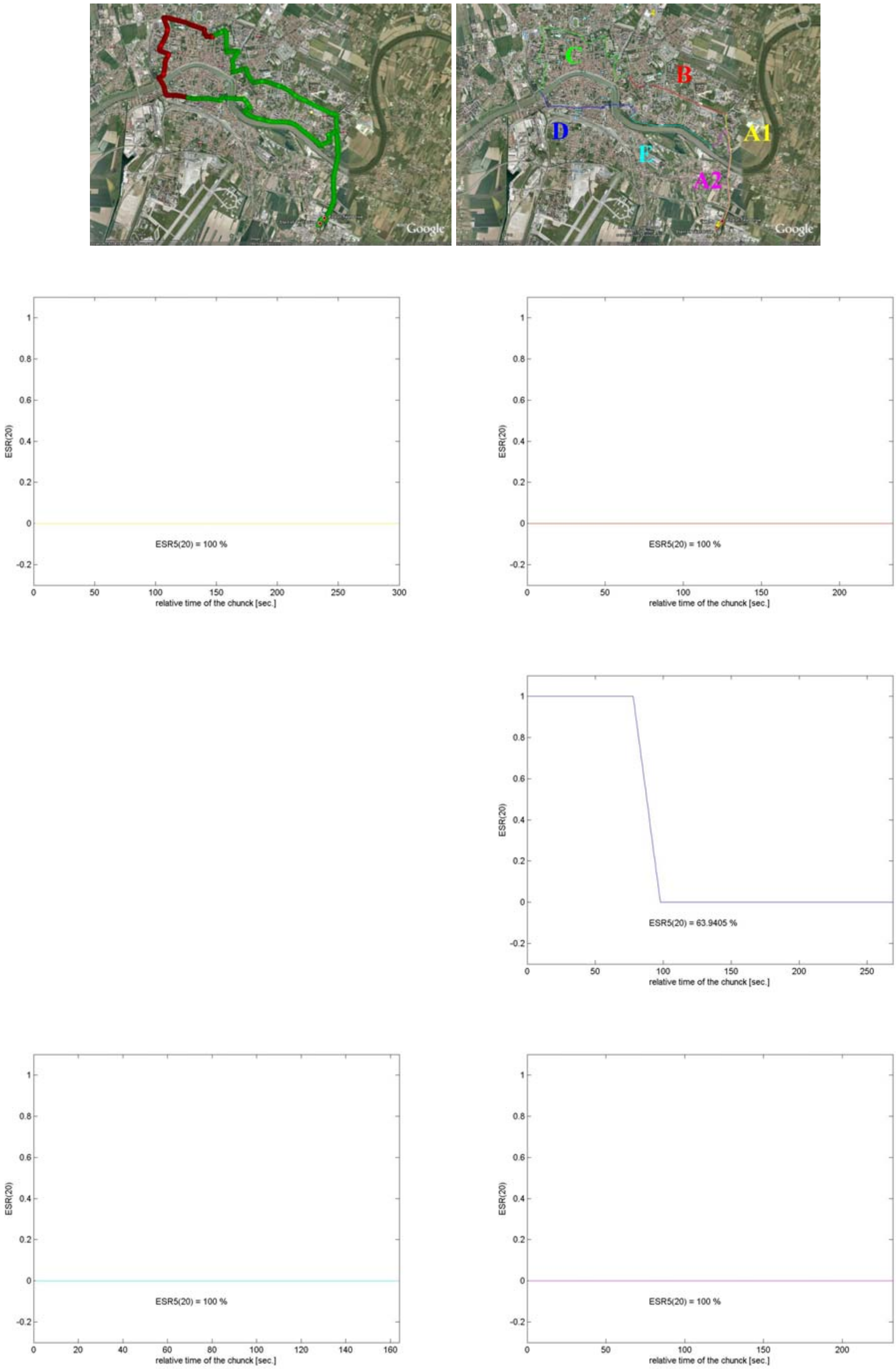


Figure A.13.94: Window status and ESR5(20) calculated for the different environments: A1 (top left), B (top right), C (centre left), D (centre right), E (bottom left), A2 (bottom right) Terrestrial only

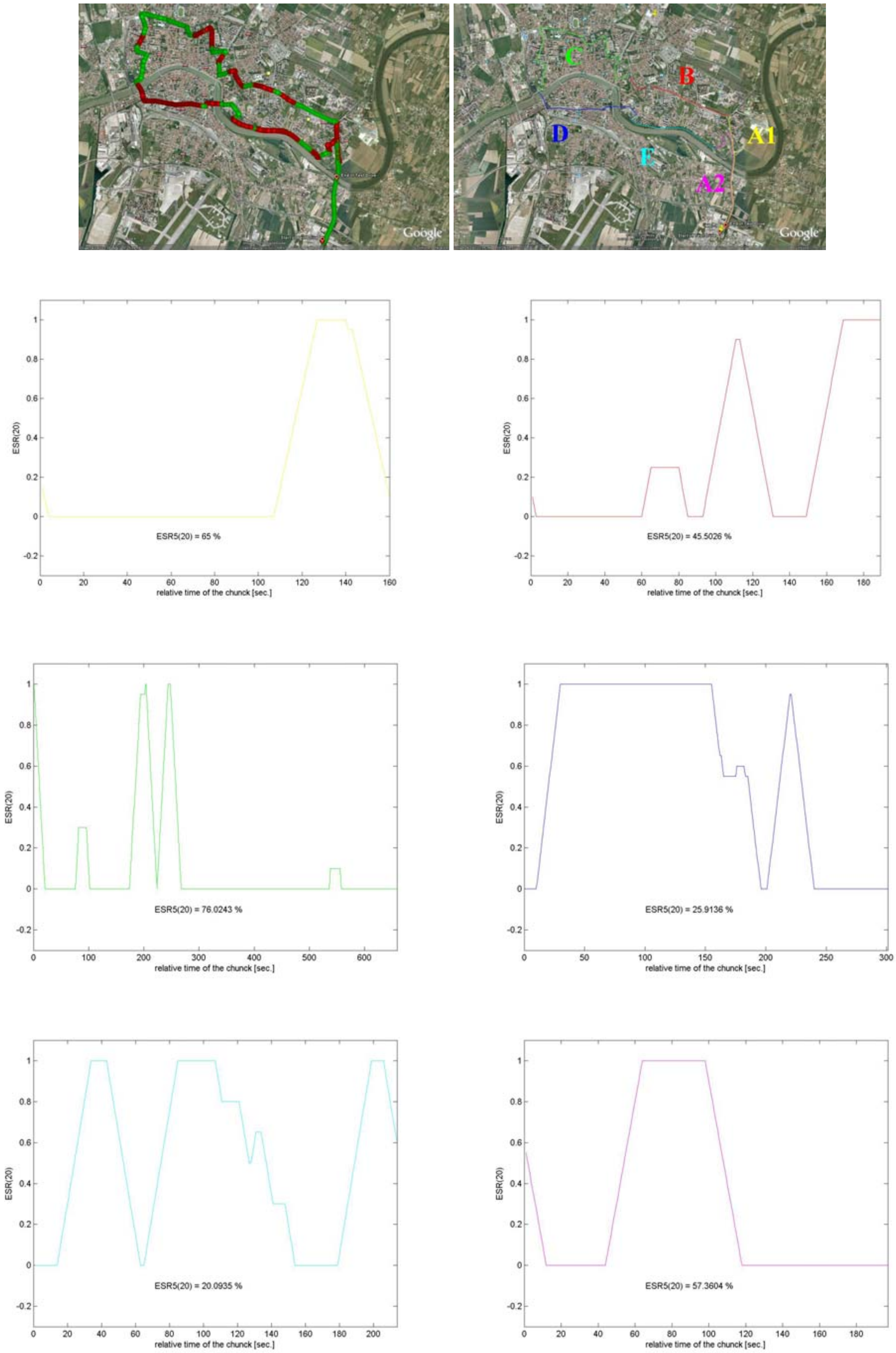


Figure A.13.95: Window status and ESR5(20) for the complete session (top) and for the different environments: A1 (top left), B (top right), C (centre left), D (centre right), E (bottom left), A2 (bottom right). Satellite only

The results for the class-2 configuration are depicted next.

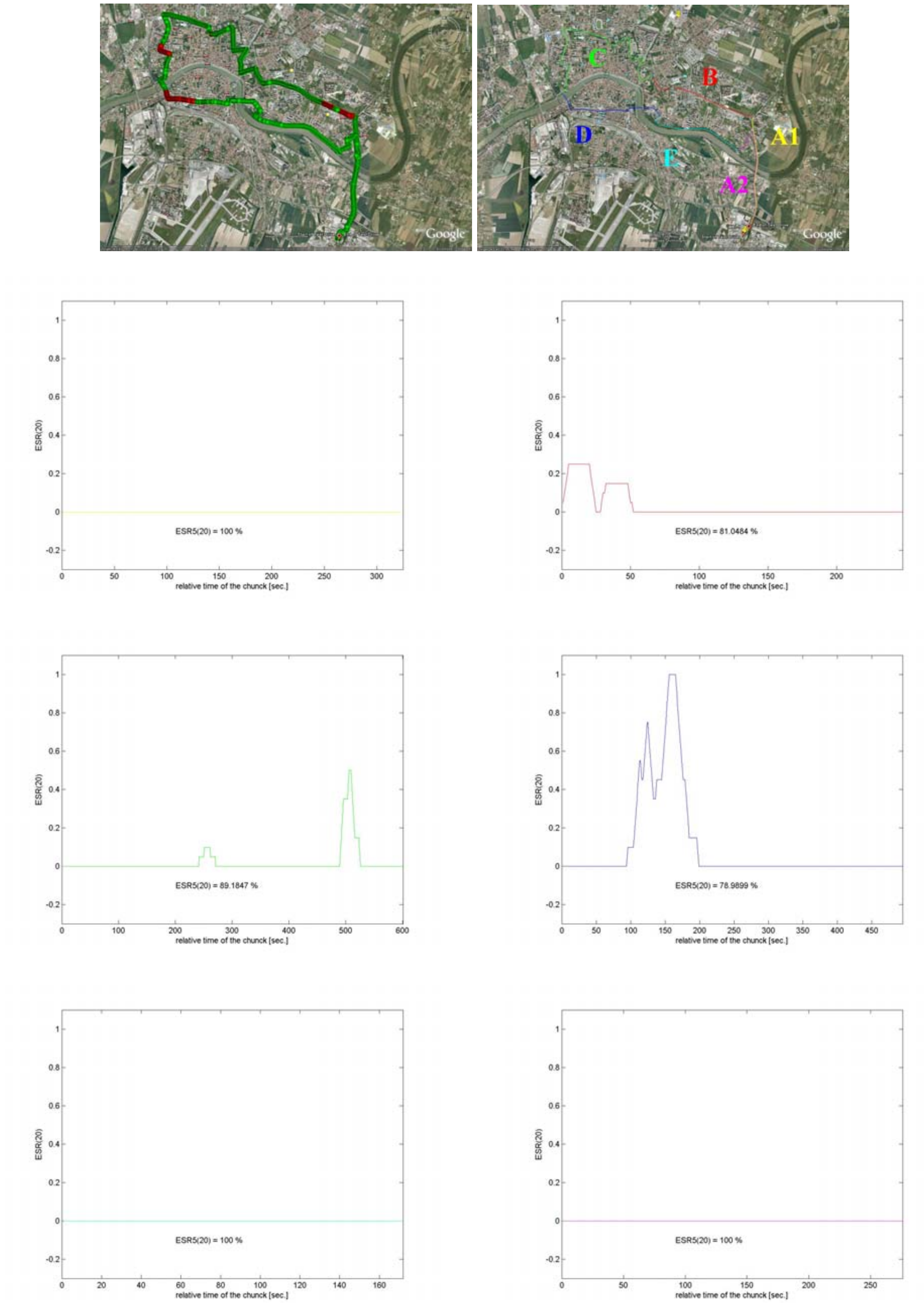


Figure A.13.96: Window status and ESR5(20) for the complete session (top) and for the different environments: A1 (top left), B (top right), C (centre left), D (centre right), E (bottom left), A2 (bottom right). Code combining

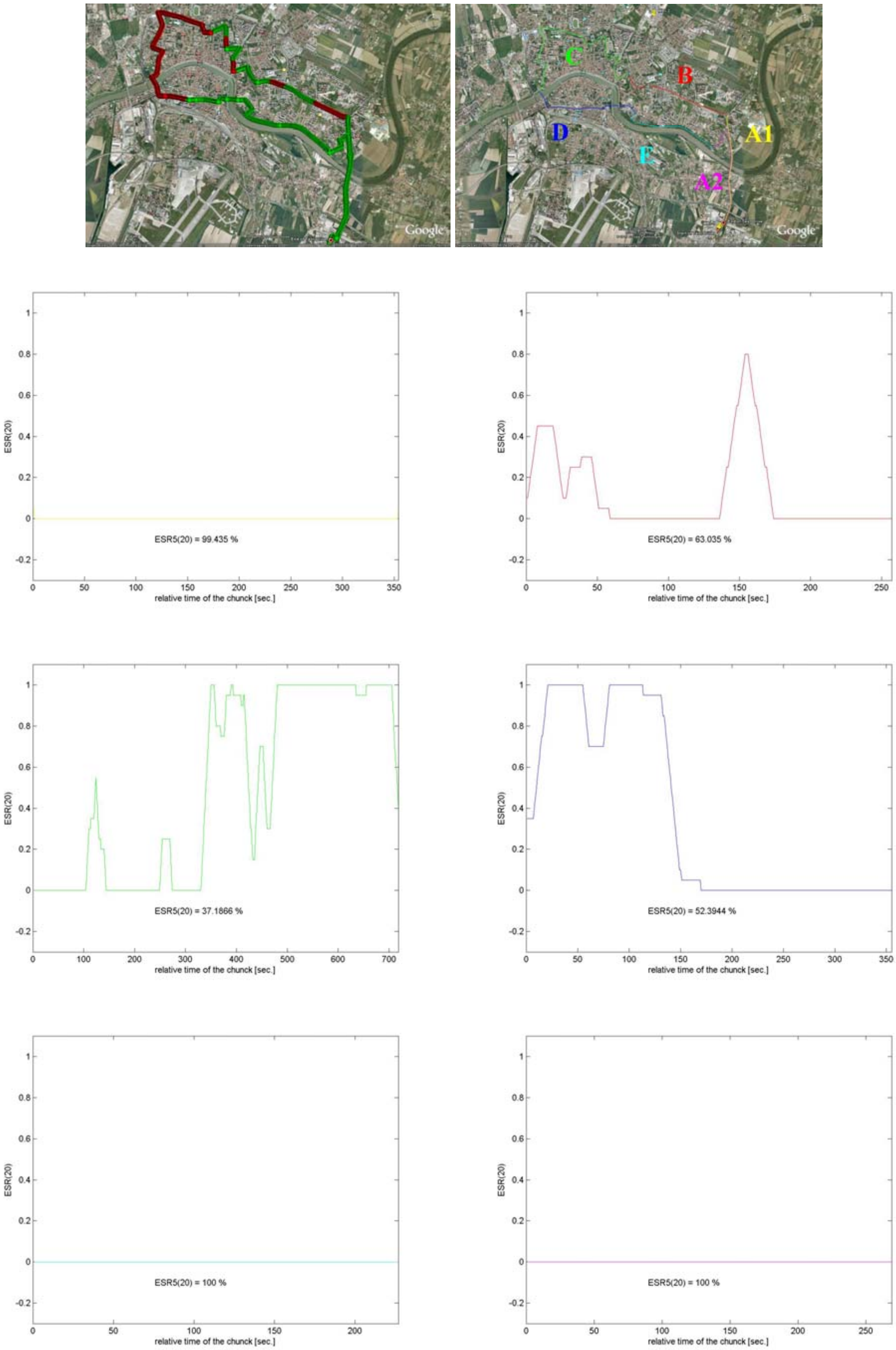


Figure A.13.97: Window status and ESR5(20) for the complete session (top) and for the different environments: A1 (top left), B (top right), C (centre left), D (centre right), E (bottom left), A2 (bottom right). Terrestrial only

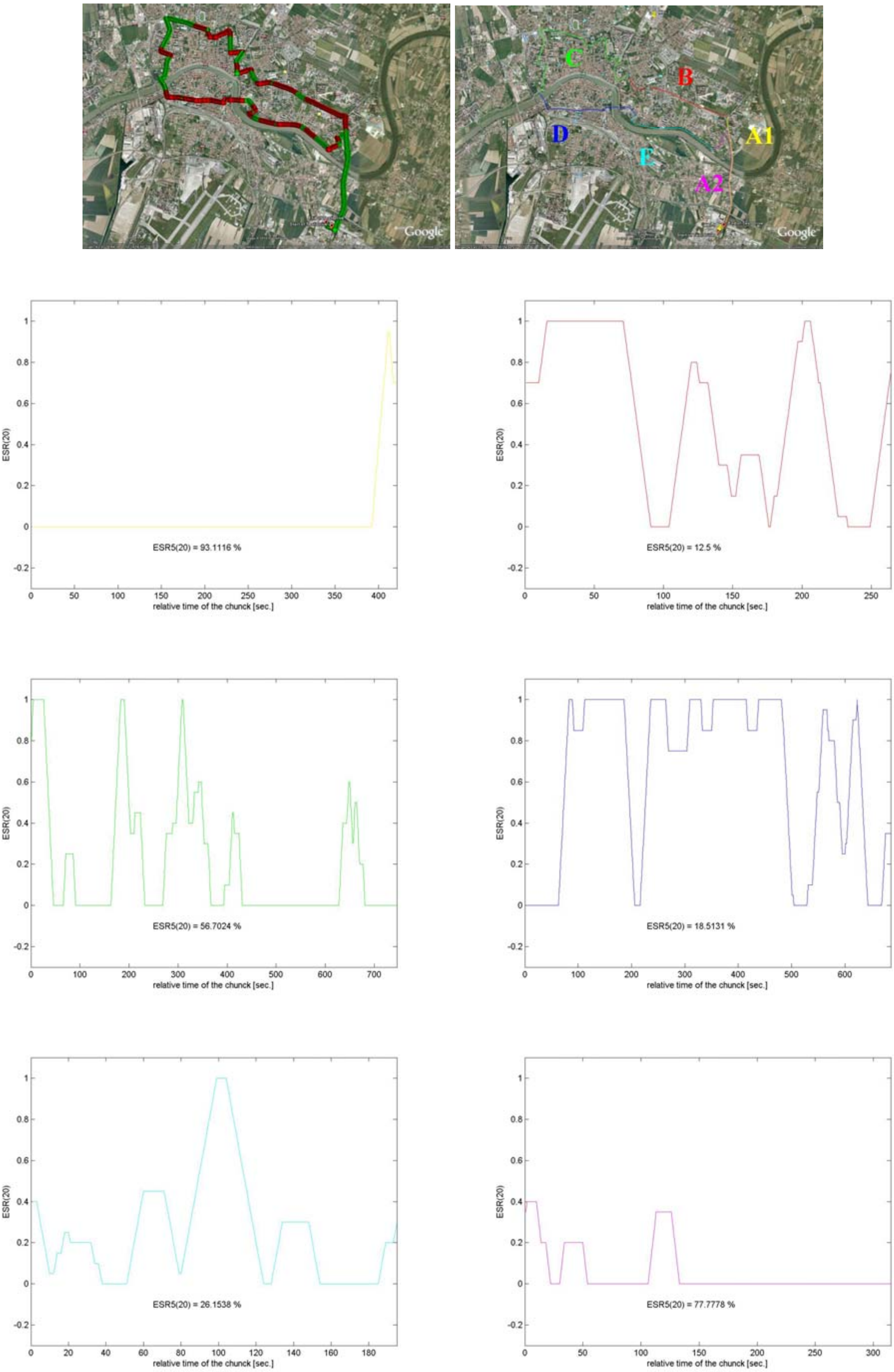


Figure A.13.98: Window status and ESR5(20) for the complete session (top) and for the different environments: A1 (top left), B (top right), C (centre left), D (centre right), E (bottom left), A2 (bottom right). Satellite only

A.13.2.6.4 Hybrid SFN trials

Trials in Pisa also covered hybrid SFN configurations where signals from satellite and terrestrial were combined in SHA SFN mode.

A.13.2.6.5 Conclusion on Pisa trials

These trials have proved that the code combining mechanism allows improving the coverage of DVB-SH systems in areas where neither the satellite nor the terrestrial signal could provide a good QoS. The capacity of the satellite to improve the CGC coverage in urban and sub-urban areas has also been confirmed.

By comparing two different waveforms with different spectral efficiencies, it has been shown that the long interleaver could be used to match the performance of a more robust (i.e. less spectral efficient) modulation and coding without trading-off the system capacity. This comes at the expense of the need of a longer memory and additional complexity in the receiver.

A.13.2.7 Conclusion on trial results

This clause has provided a compilation of the different trial campaigns carried out to assess the performances of DVB-SH in real scenarios.

All trials results confirm the performances expected and they do not deviate in more than few tens of dB from the results obtained in the laboratory tests.

The performance of the main features of DVB-SH has been assessed, namely: SFN gain, Code Combining gain, long Physical layer interleaver and MPE IFEC protection.

Satellite coverage has been evaluated in a highway scenario and in urban, sub-urban and rural environments. DVB-SH performance in satellite-only coverage is very good for the highway scenario and it can be further boosted thanks to the long interleaver protection. Satellite does also improve the coverage of the CGC in rural and sub-urban areas.

Even though long protection mechanisms (both long interleaver and MPEiFEC) where introduced to improve the performance on satellite channels, results show that they are also beneficial for terrestrial-only channels.

Antenna diversity has been proved to be an interesting technique to improve the coverage with a gain between 3 dB to 5,5 dB in the configurations analysed.

Annex B (informative): Interoperability with cellular telephony networks

B.1 Introduction

Mobile terminals can contain several radios and therefore co-existence of the DVB-SH receiver working in L-band (1 452 MHz to 1 492 MHz) and S-band (2 170 MHz to 2 200 MHz) with other radios, either cellular or connectivity, is required. The cellular radios can be GSM/EDGE (including GSM850, GSM900, DCS1800 and PCS1900) or UMTS while the connectivity radios can be wireless LAN or Bluetooth®. Simultaneous operation of the DVB-SH receiver in combination with one of the cellular or connectivity radios in a small sized terminal is very challenging.

B.1.1 General coexistence issues

The co-existence issues described in this clause consider worst case conditions:

Two main co-existence issues for the DVB-SH receiver can be distinguished:

- interference of the uplink signal of a simultaneous (cellular or connectivity) transmission with the DVB-SH receiver:
  - wanted transmitted signal: desensitization;
  - unwanted transmitted signal: power amplifier noise and spurious responses;
- interference of the downlink signal of a simultaneous (cellular or connectivity) reception with the DVB-SH receiver:
  - wanted transmitted signal from a base station: Adjacent Channel Selectivity (ACS);
  - unwanted transmitted signal from a base station: downlink Adjacent Channel Leakage Ratio (ACLR).

Next to this, undisturbed operation of the cellular or connectivity radio in presence of the DVB-SH receiver should also be maintained. Because DVB-SH does not have a uplink signal, the possible impairments caused by DVB-SH receiver are limited to:

- out of band unwanted signals in cellular or connectivity downlink (RX) band;
- effects to the cellular antenna pattern.

These problems are pure implementation issues and can be solved by proper terminal design.

B.1.2 Terminal Architectures

The architecture (relevant parts) of a typical terminal category 3 containing a DVB-SH receiver and a GSM, UMTS or WLAN radio is presented in figure B.1.1.

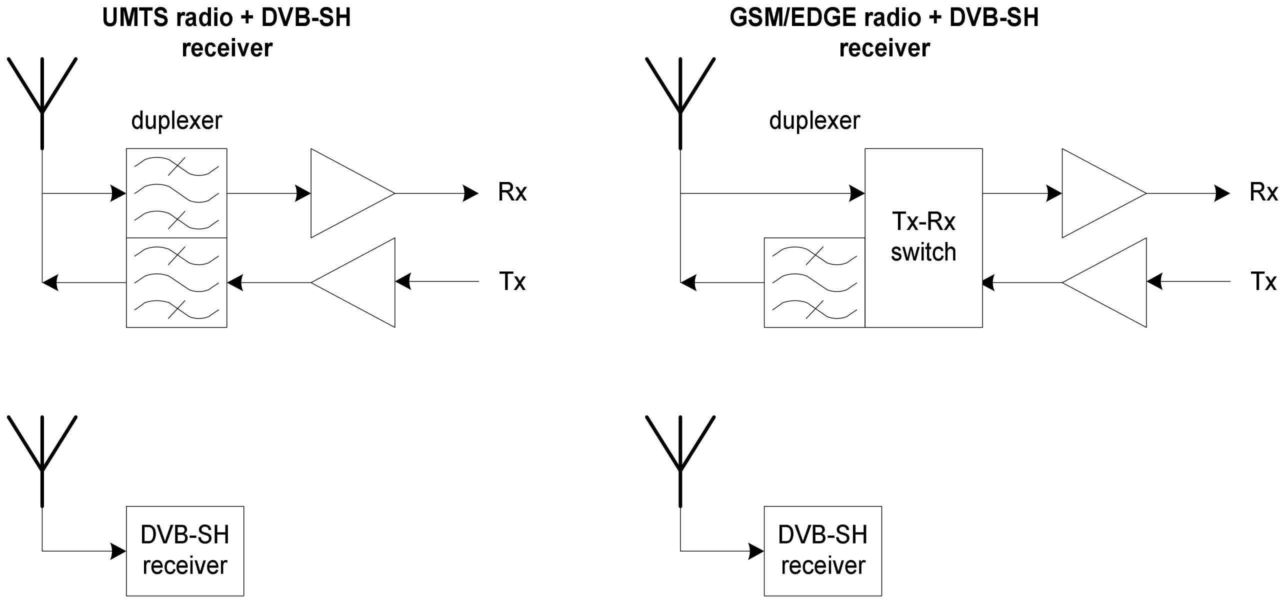


Figure B.1.1: Terminal architectures with cellular radios

Most probably the DVB-SH receiver and the cellular radio will have physically separate antennas. The antenna isolation between the antennas is frequency dependent and will be detailed in the next clause.

An important difference between UMTS and GSM radios is the duplex filter. UMTS will use duplex filters, but the majority of modern GSM radios uses a Tx/Rx switch. This has a major implication on the co-existence. The cellular radio uplink unwanted signal interference to the DVB-SH receiver will not be a problem in UMTS terminal if a duplexer is used. However the problem will be severe in a GSM terminal with Tx/Rx switch.

It is worth noting that in case of terminal category 2 the cellular radio transmitters are not considered. However, the WLAN radio transmitter might be present causing also signal interferences.

B.2 Frequency bands and power levels

Table B.2.1 gives an overview of the most relevant interferers for DVB-SH including frequency bands and power. For uplink signals, the transmit power and the assumed antenna coupling values are given for S-band receiver. Together they give the received power at the DVB-SH antenna.

Table B.2.1: Overview of interferers for DVB-SH receiver

Name	Frequency band	Transmit power	Antenna coupling	Power at DVB-SH antenna
GSM850 Tx	824 MHz to 849 MHz	33 dBm	-18 dB	15 dBm
GSM900 Tx	880 MHz to 915 MHz	33 dBm	-18 dB	15 dBm
DCS1800 TX	1 710 MHz to 1 785 MHz	30 dBm	-15 dB	15 dBm
PCS1900 TX	1 850 MHz to 1 910 MHz	30 dBm	-15 dB	15 dBm
FDD 3G band 1 Tx	1 920 MHz to 1 980 MHz	24 dBm	-9 dB	15 dBm
DD 3G band 1 Rx	2 110 MHz to 2 170 MHz	n.a.	n.a.	
ISM2400 (BT-WLAN)	2 400 MHz to 2 484 MHz	20 dBm	-9 dB	11 dBm

The most problematic cases are the wanted uplink and downlink signals from the FDD 3G band 1 (UMTS) due to its close proximity to the DVB-SH signal band in S-band.

B.3 Interference due to uplink signal

B.3.1 Desensitization

The transmitted cellular or connectivity signal has a very high power compared to the received DVB-SH signals. FDD 3G band 1 is the closest one with high power and will be considered as the worst-case situation.

FDD 3G band 1 transmitted power is +24 dBm. Part of this power is coupled to the DVB-SH antenna. The coupling loss is between the FDD 3G band 1 antenna and the DVB-SH antenna is assumed to be 9 dB. The received power at the DVB-SH antenna is therefore +15 dBm. Without any filtering the cellular TX signal present in the DVB-SH receiver input would be also +15 dBm. This very high interference signal level would cause severe blocking effects and reciprocal mixing.

The practical solution for co-existence is to insert a rejection filter in front of the DVB-SH receiver. The filter has to attenuate the FDD 3G band 1 Tx-signal to the allowed out of band unwanted signal level. The maximum level for hand portable terminal is -25 dBm. Stop band attenuation of the filter thus becomes:

$$\boxed{A_{filter} = P_{TX} - A_A - P_{max} = +24\ dBm - 9dB - (-25\ dBm) = 40\ dB} \quad \text{(B.1)}$$

where:

- A_{filter} = stop band attenuation of the reject filter;
- P_{TX} = Tx output power;
- A_A = coupling between the antennas (-9 dB);
- P_{max} = maximum allowed power at the DVB-SH receiver input.

Typically the insertion loss of the filter is in the order of 1,5 dB as described in clause 10. The sensitivity degradation is affected by the attenuation of the RF filter, by the receiver desensitization and phase noise performances and by antenna coupling between the two antennas at the blocker frequency.

B.3.2 Spurious and transmitted PA noise

Besides the wanted part of the FDD 3G band 1 uplink signal, the signal also contains unwanted parts like spurious responses and the out of band PA noise for terminal architectures without duplexer.

The GSM specification (TS 100 910 [30]) defines that within 100 kHz measurement bandwidth the power should not be greater than -79 dBm within GSM900 Rx frequency band (-71 dBm in case of DCS1800). This noise floor corresponds to the noise contribution of the transmitter power amplifier. That value can be extended for a noise floor value within the DVB-SH bands as a maximum value.

Assuming a 18 dB coupling loss between GSM900 and DVB-SH S-band antennas, the interference power entering DVB-SH receiver would be -80 dBm for 5 MHz bandwidth. With a 4,5 dB receiver noise figure, the total receiver input noise floor is -102,7 dBm. From this it is obvious that the transmitter output noise reduces the DVB-SH receiver sensitivity considerably. In order to degrade DVB-SH receiver sensitivity "only" by 3 dB the transmitter output noise would need to be lower or equal to the receiver thermal noise floor (-102,7 dBm).

Fortunately, noise in S band from GSM quadriband PA is lower than ETSI specification (20 dB typically). Nevertheless, if DVB-SH high sensitivity performance is targeted, an additional filtering will be required on TX power amplifier outputs.

Interoperability requirements

To guarantee interoperability between the radio systems the noise power at the DVB-SH receiver input (@ 5 MHz band) should fulfil the mask shown in figure B.3.1.

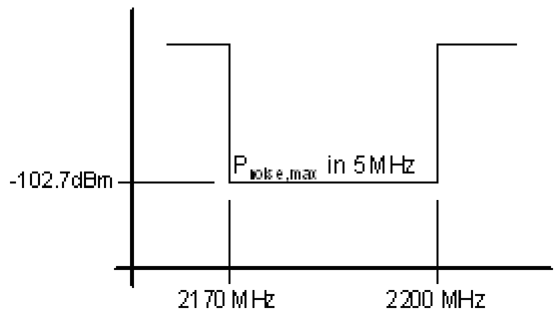


Figure B.3.1: Tx PA-noise Mask in S-band DVB-SH Receiver Input

The noise level is affected by the noise power of the power amplifier, by the attenuation of the possible low pass filter at the output of the transmitter PA as shown in figure B.3.2 and by antenna coupling between the two antennas at the DVB-SH reception frequency band.

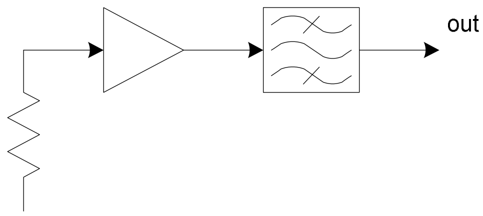


Figure B.3.2: GSM Tx Block Diagram

B.4 Cellular Radio Downlink Signal Interference to DVB-SH Receiver

The location of the DVB-SH received signal in S-band is next to the UMTS FDD downlink frequency range. This represents an additional interferer. Consequently, this is the most critical case for downlink signal interference.

B.4.1 Adjacent channel

Adjacent channel denomination will characterize hereafter 5 MHz adjacent channel (UMTS downlink channel next to DVB-SH channel) and per extension 10 MHz adjacent channel (one unoccupied UMTS downlink channel between UMTS downlink interferer and DVB-SH channel).

Adjacent interference is summarized through Adjacent Channel Interference Ratio (ACIR). This is defined by:

$$\frac{1}{ACIR} = \frac{1}{ACS} + \frac{1}{ACLR}$$

ACLR: Adjacent Channel Leakage Ratio which determines the power ratio transmitted by the UMTS downlink Base Station in its adjacent channel.

ACS: Adjacent Channel Selectivity which defines the ability of the DVB-SH receiver to demodulate the wanted signal in presence of interferer in its adjacent channel.

The UMTS specification TS 125 104 [15] gives a minimum of 45 dB ACLR for 5 MHz adjacent channel and a minimum of 50 dB ACLR for 10 MHz adjacent channel.

B.4.2 ACS requirements

A minimum receiver ACS should be achieved in order to avoid high DVB-SH receiver ACS contribution in ACIR value considering implementation margins in UMTS downlink ACLR.

B.4.2.1 Example of ACS value measurement

Different measurements have been made on DVB-SH receivers.

Figures B.4.1 and B.4.2 summarises the measurements methods, which is also explained in detail in EN 302 574-2 [i.41].

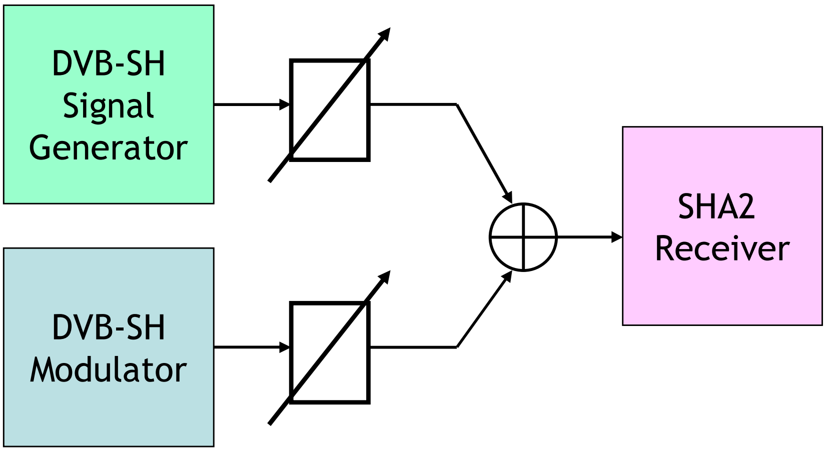


Figure B.4.1: Adjacent channel test bench

In figure B.4.2, the signal generator plays the role of the interferer, and generates a signal in excess power of M dB above the useful signal.

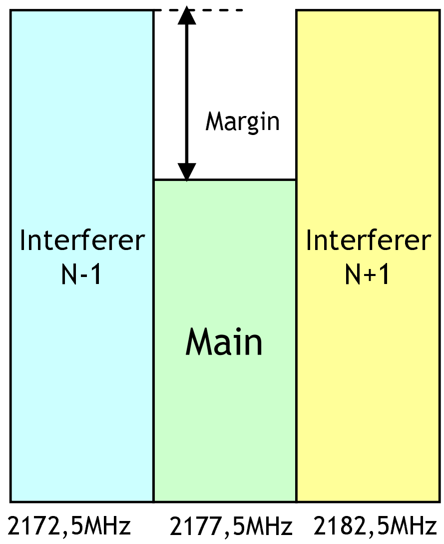


Figure B.4.2: Tx Adjacent channel spectrum

Figure B.4.2 shows the position of the main signal between two potential interferers at N-1 and N+1 frequency slots, where N is the main. The margin, called M represents the excess power of the interferer. The useful signal input power is set at -70 dBm, which corresponds to at least 25 dB above the required power to get the FER 5 % in AWGN.

Then the interferer power is increased until the useful signal is at the limit of required QoS. At that time we can write:

$$ACS = M + C/N_0.$$

The different measurements show that ACS is above 50 dB for N±1 and above 60 dB for N±2.

In the Harmonized Standard EN 302 574-2 [i.41] for User equipment in the MSS bands, the requirement for the ACS is the following:

- 48 dB for N±1.
- 55 dB for N±2.

B.4.2.2 Recommendations

The ACS performances requirement should be at minimum 48 dB for the N±1 channels and 55 dB for N±2 channels.

It should be possible to reach 50 dB and 60 dB as in the studied chipset.

B.4.3 Co channel Interference

The co-channel test consists of adding an interfering DVB-SH spectrum on the considered spectrum. This situation can be encountered in MFN networks on cell borders in area where signals from both cells can be received with equivalent level.

The same test bench as adjacent channel tests is used except that the frequency of the interfering signal is the same as the useful signal. The interfering signal uses the same DVB-SH modulation parameters with PRBS mode.

The method of the measurement consists of:

- Set the power of the useful signal to a given fixed value.
- Set the power of the interferer to find the FER5% criterion on the receiver.

Two scenarios are provided: one with QPSK 1/3 and one with 16QAM 1/3. The results are presented in table B.4.1.

Table B.4.1: Co-channels test results

	Max interferer C/I @ FER5	AWGN C/N	Co-channel limit compared to AWGN
Scenario 1	0,4 dB	-0,7 dB	1,1 dB
Scenario 2	6,1 dB	4,6 dB	1,5 dB

So

$$C/I_{co-channel}\ [\mathrm{dB}] = 1.5 + \left[C/N_{AWGN}\right]_{\mathrm{dB}}.$$

Annex C (informative): Spectrum efficiency and system throughput

C.1 Spectrum efficiency analysis

The following spectral efficiency calculations should be considered as illustrative and by no means representative of a specific real system. In this analysis, the frequency plan is assumed to be homogenous over the whole coverage region. It is considered that when the common content retransmission takes place in another frequency sub-band (MFN configuration) we assumed that the terrestrial physical and upper layer parameters can be different from the satellite ones. This allows inclusion of local content on top of the common content in the terrestrial retransmission.

The following configurations can be identified.

Table C.1.1

	Waveform	SFN/MFN
Config 1	SH-A	MFN
Config 2	SH-A	SFN
Config 3	SH-B	MFN

Examples for a 3-color reuse pattern of a 3 x 5 MHz satellite system are depicted below. The drawings show the frequency plan in a particular beam, with "Slot A" designating the frequency assigned to the satellite beam in question (this is the "satellite-protected" frequency, in which no adjacent beam interference occurs). Not the drawings are not to scale and that the frequency slots can be of different width in the different cases.

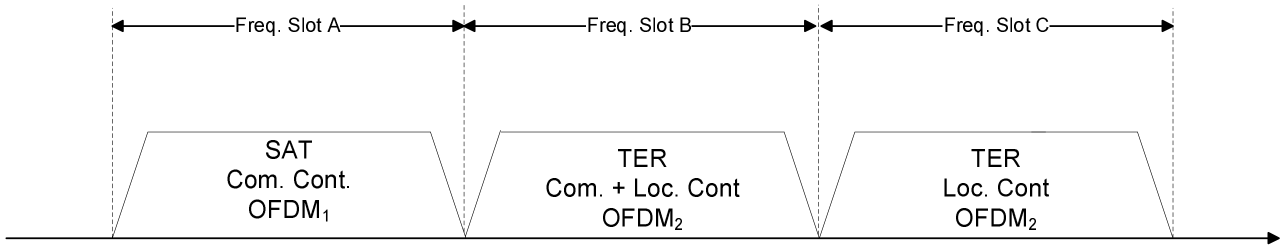


Figure C.1.1: Configuration 2 (SH-A, MFN, No split band)

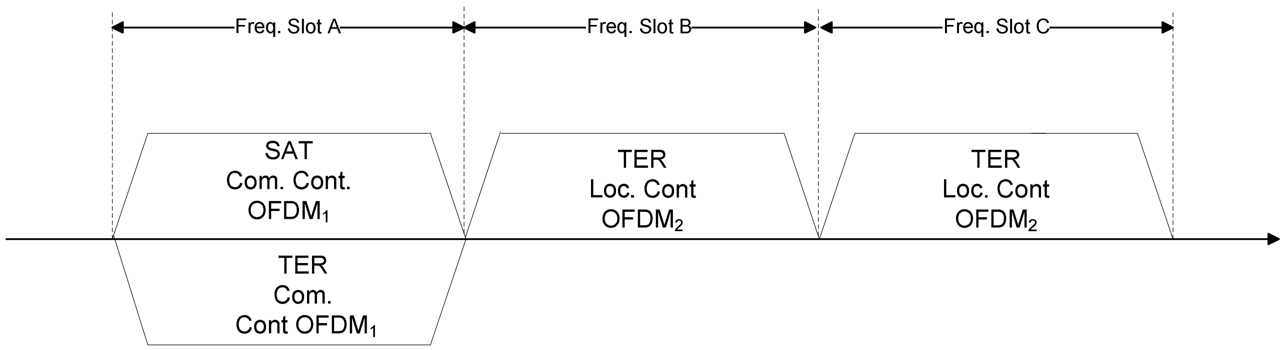


Figure C.1.2: Configuration 3 (SHA, SFN, No Split band)

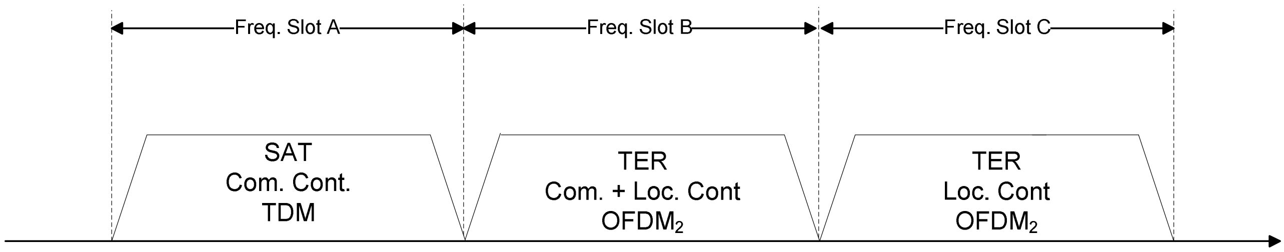


Figure C.1.3: Configuration 5 (SH-B, MFN, No split band)

C.1.1 Spectral Efficiency Calculations

We define the *waveform* spectral efficiency as the net bit rate of each waveform divided by the channelbandwidth. The following equations give the spectral efficiencies and the net bit rate as a function of modulation and coding parameters:

For TDM:

The net bit rate supported by the TDM carrier $Cap\_TP[TDM]$ is computed in clause 7. The TDM spectral efficiency is then:

$$\eta(\mathrm{TDM}) = Cap\_TP[TDM] / B_W^{TDM},$$

being B_W^{TDM} the TDM channelization bandwidth;

For OFDM:

The net bit rate supported by the OFDM carrier $Cap\_TP[OFDM]$ is computed in clause 7. The OFDM spectral efficiency is then:

$$\eta(\mathrm{OFDM}) = Cap\_TP[OFDM] / B_W^{OFDM},$$

being B_W^{OFDM} the OFDM channelization bandwidth.

Table C.1.2 contains numerical examples for the bit rate and spectral efficiencies calculations for TDM and OFDM waveforms having assumed:

- channelization bandwidth = B_W^{OFDM} = B_W^{TDM} = 5 MHz;
- TDM roll-off factor α_{TDM} = 0,15;
- OFDM guard interval GI = 1/8.

Table C.1.2

TDM, QPSK, FEC_rate 1/3	Net bit rate in 5 MHz= 2,633 Mbps	η(TDM) = 0,53 b/s/Hz
TDM, 16APSK, FEC_rate 1/4	Net bit rate in 5 MHz= 3,950 Mbps	η(TDM) = 0,79 b/s/Hz
OFDM, QPSK, FEC_rate 1/3	Net bit rate in 5 MHz= 2,468 Mbps	η(OFDM) = 0,49 b/s/Hz
OFDM, 16QAM, FEC_rate 1/4	Net bit rate in 5 MHz= 3,730 Mbps	η(OFDM) = 0,75 b/s/Hz

We are interested in two spectral efficiencies at beam level:

- the *Total spectral efficiency per beam*, η(TOTAL), defined as the net bit rate used for all content (Common+Local) available in the CGC coverage of a beam divided by the *satellite-protected* bandwidth allocated to that beam;
- the *Common content spectral efficiency per beam*, η(COM), defined as the net bit rate of the Common content available in the beam divided by the same bandwidth as above.

For the clarity of the formulas below, we use the following notations to further distinguish between different use cases of the OFDM waveform:

- η(OFDM): Efficiency of the OFDM waveform when used in CGC only;
- η(OFDM,Sat): Efficiency of the OFDM waveform when used in satellite, without SFN;
- η(OFDM,Hyb): Efficiency of the OFDM waveform when used jointly in satellite and CGC (SFN).

Let f_R be the number of colours in which we divide the overall bandwidth available in our system (i.e. the total number of sub-bands in a beam). Table C.1.3 gives the formulas for computing these efficiencies, once the waveform spectral efficiencies and the satellite frequency reuse have been selected.

Table C.1.3

Configuration	Total spectral efficiency per beam, η(TOTAL)	Common content spectral efficiency per beam, η(COM)
1) SH-A, MFN	(f_R -1) x η(OFDM)	η(OFDM,Sat)
2) SH-A, SFN	(f_R -1) x η(OFDM) + η(OFDM,Hyb)	η(OFDM,Hyb)
3) SH-B, MFN	(f_R -1) x η(OFDM)	η(TDM)

Comments:

In deriving the above formulas for MFN, it has been implicitly assumed that the terrestrial efficiency is always higher than or equal to the satellite efficiency so that local content can be included on top of the common content in the terrestrial retransmission.

In configuration 2), if the Hybrid frequency sub-band uses the same modulation and coding as the other terrestrial sub-bands then η(OFDM) = η(OFDM, Hyb) and the Total spectral efficiency per beam is f_R x η(OFDM). The terrestrial frequency reuse in a beam is exactly the satellite frequency reuse between beams.

If configuration 2 is to be compared with configuration 3, their Total spectral efficiency per beam differs by the η(OFDM,Hyb), in favour of configuration 2. As concerns the Common content spectral efficiency per beam, η(TDM) is usually higher than η(OFDM,Hyb), in favour of configuration 3.

If the satellite system is composed of N_{beams}, the *system spectral efficiency* can be defined as the total bitrates summed over all beams divided by the total frequency allocated to that satellite system. The following relationship can be easily shown:

System spectral efficiency = (N_{beams}/f_R) x (Total spectral efficiency per beam)

C.1.2 Numerical examples

In the following example of efficiency comparison between the different 5 possible configurations in a 6-beam, 3-color reuse system. For sake of simplicity the waveform used in all cases has a 5 MHz bandwidth. For this reason the split configurations 1) and 4) require a 10 MHz beam sub-slot bandwidth instead of 5 MHz as for cases 2), 3) and 5).

Table C.1.4

TDM, QPSK, FEC 1/3	η(TDM) = 0,53 b/s/Hz	Satellite in Config 3
OFDM, QPSK, GI=1/8, FEC 1/3	η(OFDM) = 0,49 b/s/Hz	Sat/Hybrid in Config 1, 2
OFDM, QPSK, GI=1/8, FEC ½	η(OFDM) = 0,75 b/s/Hz	CGC only

Table C.1.5

Configuration	η(TDM)	η(OFDM)	η(OFDM) Hybrid or Sat	η(COM)	η(TOTAL) with f_R =3	System (6 beams)
2) SH-A, MFN	NA	0,75	0,49	0,49	2 x 0,75 = 1,5	6/3 x 1,5 = 3
3) SH-A, SFN	NA	0,75	0,49	0,49	2 x 0,75 + 0,49 = 1,99	6/3 x1,99 = 3,98
5) SH-B, MFN	0,53	0,75	NA	0,53	2 x 0,75 = 1,5	6/3 x1,5 = 3

The efficiencies above can be converted into net bitrates as follows, assuming a 5 MHz waveform channelization.

Table C.1.6

Configuration	Total system bandwidth/Beam sub-slot bandwidth	Common bit rate in a beam	Total bit rate in a beam	System (6 beams)
2) SH-A, MFN	15/5 MHz	2,47 Mbps	7,5 Mbps	45 Mbps
3) SH-A, SFNt	15/5 MHz	2,47 Mbps	10 Mbps	60 Mbps
5) SH-B, MFN	15/5 MHz	2,63 Mbps	7,5 Mbps	45 Mbps

C.1.3 Conclusions on System Spectrum Efficiency

It is difficult to provide general recommendations on the best system frequency planning configuration as decision depends not only on spectral efficiency. In fact when deciding the optimal frequency planning one should also consider other system parameters such satellite antenna beam patters and gap fillers location which may impact the location and extent of exclusion zones (see clause 11). Based on the examples illustrated before from a pure spectral efficiency perspective the following conclusions can be derived:

- when the content to be broadcasted is different for different regions (e.g. linguistic regions) multibeam satellite configuration increase the common content spectral efficiency proportionally to the ratio between the number of beams and the number of colours used $\left(N_{beams} / f_R\right)$;
- SH-A/SFN configuration provides the highest overall spectral efficiency thanks to SFN operation between satellite and terrestrial gap fillers;
- SH-B achieves a slightly higher spectral efficiency than SH-A for common content delivery but has a lower spectral efficiency for local content distribution than SH-A. Power efficiency issues (e.g. HPA nonlinearity effects) are not part of this comparison.

As stated above the above discussion is only looking at the spectral efficiency aspects. More extensive discussion about SH-A and SH-B waveform pro and contra can be found in clause 7.

C.2 Throughput calculations

DVB-SH capacity at MPEG TS interface level may be calculated from the parameters defined in the waveform definition [1]. Specifically, it is found by first calculating the duration of the SH frame, and the number of MPEG TS packets in an SH Frame, and then using these results to determine the capacity.

To support seamless hand-over between OFDM and TDM, the SH frame length of the TDM part has been aligned to the SH frame length in OFDM. Therefore, the TDM parameters can not be calculated independently but imply a selection of OFDM parameters which is represented in the following equations.

C.2.1 Reference DVB-SH parameters

The throughput calculation is based on some reference DVB-SH parameters which are recalled in this clause.

C.2.1.1 OFDM Frame

The time related parameters of the OFDM frame are defined in multiple of the elementary period *T*. From EN 302 583 [1], clause 5.7.4.1,

$$T = \frac{7}{8 \cdot BW\_OFDM} \quad \text{(C.1)}$$

with BW_OFDM being the OFDM signal bandwidth, in MHz; BW_OFDM=1,6 MHz (for a "1,7 MHz" channel), 5 MHz, 6 MHz, 7 MHz or 8 MHz.

NOTE 1: *T* is in µS in the above equation; alternatively, *T* is in seconds if *BW_OFDM* is in Hz.

NOTE 2: The elementary period of the 1,7 MHz Channel, defined in EN 302 583 [1], clause 5.7.4, is equal to 7/12,8 µs; this corresponds to a *BW_OFDM* of 1,6 MHz.

The duration *Ts* of an OFDM symbol, defined in EN 302 583 [1], clause 5.7.4.1, is:

$$T_S = 1024 \cdot Mod\_FFT \cdot (1 + GI) \cdot T \quad \text{(C.2)}$$

with *Mod_FFT* being the FFT length in k; Mod_FFT = 1, 2, 4 or 8 and GI being the Guard Interval ratio; GI=1/4,1/8,1/16 or 1/32.

The number *Nb_Symb_OFDM_Frame* of OFDM symbol per OFDM frame, defined in EN 302 583 [1], clause 5.7.4.1, is:

Nb_Symb_OFDM_Frame = 68 (C.3)

The number of data subcarriers Nb_Data_Carrier per OFDM symbol, defined in EN 302 583 [1], clause 5.7.2, is:

Nb_Data_Carrier = 756 Mod_FFT (C.4)

with Mod_FFT being the FFT length in k; Mod_FFT = 1, 2, 4 or 8.

The number of bits per modulated data subcarrier in OFDM BpS_OFDM is:

BpS_OFDM = 2 or 4 (C.5)

with BpS_OFDM= 2 for QPSK and 16QAM hierarchical modulation, and 4 for 16QAM modulation.

The SH Frame is, as defined in EN 302 583 [1], clause 5.5.2.3, composed of 816 Capacity Units (CU) of 2 016 bit each, thus:

Nb_CU_SHF_OFDM = 816 (C.6)

Nb_Bit_CU = 2 016 (C.7)

C.2.1.2 TDM Frame

Prior to supporting unequal bandwidth for TDM and OFDM the TDM signal symbol rate had been calculated with the following formula:

$$Sr\_TDM = \operatorname{int}\left\{32\left(\frac{1+GI}{1+\alpha}\right)\right\}\frac{BW\_TDM}{32(1+GI)}$$

$$Sr_{TDM} \cdot T = \operatorname{int}\left(32\,\frac{1+GI}{1+\alpha}\right) \cdot \frac{BW\_TDM}{BW\_OFDM} \cdot \frac{7}{8 \cdot 32\,(1+GI)}$$

with α being the TDM Square-Root Raised-Cosine (SRRC) filter roll off factor; α =0,15; 0,25; or 0,35, GI being the OFDM guard interval, *BW_TDM* being the TDM signal bandwidth that may differ from the OFDM signal bandwidth BW_OFDM. This forumula, always correct when BW_TDM equals BW_OFDM, does not always respect the constraint of having an integer number of TDM PLSLOT per SH-frame as defined in EN 302 583[1], clause 5.6.4.1 for differing BW_OFDM and BW_TDM. Therefore the formula can be modified as follows (see note 2):

$$\mathrm{Sr_{TDM}} \cdot \mathrm{T} = \mathrm{int}\left(\mathrm{int}\left(32 \frac{1+GI}{1+\alpha} \right) \cdot \frac{BW\_TDM}{BW\_OFDM} \cdot \frac{28}{\mathrm{BpS\_OFDM}} \right) \cdot \frac{1}{1024} \cdot \frac{BpS\_OFDM}{1+GI} \tag{C.8}$$

where BpS_OFDM represents the OFDM modulation order (BpS_OFDM = 2, for QPSK and 16QAM hierarchical; BpS_OFDM = 4, for 16QAM). This equation (C.8) now serves as basic definition in EN 302 583 [1], clause 5.6.3.

NOTE 1: The rationale of this equation is that the TDM symbol rate may be simply derived from the OFDM elementary period *T, in order to* optimize the receiver design. Using this definition as well as (C.1), $\mathrm{Sr_{TDM}}$.T results is a rational number, whatever the GI, α , BW_TDM, BW_OFDM and BpS_OFDM values. Thus $\mathrm{Sr_{TDM}}$ may be easily generated in any case from the 1/*T* clock reference.

As defined in EN 302 583 [1], clause 5.6.4, the TDM SH Frame is composed of Physical Layer Slots (PL SLOTS) of *Nb_Symb_TDM_SLOT* symbols:

$$\mathrm{Nb\_Symb\_TDM\_SLOT} = 2\,176 \tag{C.9}$$

Each PL SLOT carries 2, 3, or 4 CU, equal to the order of modulation *BpS_TDM* (see EN 302 583 [1], clause 5.6.4.1):

$$\mathrm{Nb\_CU\_PLSLOT{=}BpS\_TDM} = 2, 3 \text{ or } 4 \tag{C.10}$$

NOTE 2: The number of TDM PLSLOT per SH-frame is given by the formula:

$\mathrm{Nb\_TDM\_PLSLOT} = \frac{\mathrm{Sr_{TDM}} \cdot \mathrm{T_{SHF}}}{\mathrm{Nb\_Symb\_TDM\_SLOT}} = \frac{\mathrm{Sr_{TDM}} \cdot \mathrm{T}}{\mathrm{Nb\_Symb\_TDM\_SLOT}} \cdot \frac{\mathrm{T_{SHF}}}{\mathrm{T}}$; $\mathrm{T_{SHF}} = \frac{\mathrm{Nb\_CU\_SHF\_OFDM} \cdot \mathrm{Nb\_Bit\_CU}}{\mathrm{BpS\_OFDM} \cdot \mathrm{Nb\_Data\_Carrier}} \cdot \mathrm{T_S}$ and with (C.6), (C.7), (C.2), (C.4) $\frac{\mathrm{T_{SHF}}}{\mathrm{T}} = 2228224 \cdot \frac{(1+\mathrm{GI})}{\mathrm{BpS\_OFDM}}$ so that $\mathrm{Nb\_TDM\_PLSLOT} = (\mathrm{Sr_{TDM}} \cdot \mathrm{T}) \cdot \frac{1024 \cdot (1+\mathrm{GI})}{\mathrm{BpS\_OFDM}}$; $\mathrm{Nb\_TDM\_PLSLOT} = \mathrm{int}\left(32 \frac{1+GI}{1+\alpha} \right) \cdot \frac{BW\_TDM}{BW\_OFDM} \cdot \frac{28}{\mathrm{BpS\_OFDM}}$. But, as defined in EN 302 583, clause 5.6.4.1 this value has to be integer: $\mathrm{Nb\_TDM\_PLSLOT} = \mathrm{int}\left(\mathrm{int}\left(32 \frac{1+GI}{1+\alpha} \right) \cdot \frac{BW\_TDM}{BW\_OFDM} \cdot \frac{28}{\mathrm{BpS\_OFDM}} \right)$; The TDM symbol rate can then be derived as follows: $\mathrm{Sr_{TDM}} = \mathrm{Nb\_TDM\_PLSLOT} \cdot \frac{2176}{\mathrm{T_{SHF}}}$, $\mathrm{Sr_{TDM}} = \mathrm{int}\left(\mathrm{int}\left(32 \frac{1+GI}{1+\alpha} \right) \cdot \frac{BW\_TDM}{BW\_OFDM} \cdot \frac{28}{\mathrm{BpS\_OFDM}} \right) \cdot \frac{1}{896} \cdot \frac{BpS\_OFDM \cdot BW\_OFDM}{1+GI}$.

C.2.2 Calculation of the SH Frame duration

The SH frame length is defined as a function of the length of OFDM frames. Therefore, the frame length durations of both an SH-frame in OFDM and TDM modes are identical. The number of OFDM Frames per SH frame is defined in the table 5.10 of the waveform definition [1]. It may be calculated from the equations (C.3) to (C.7)from above. The related equation is the following:

$$Nb\_OFF\_SHF = \frac{Nb\_CU\_SHF\_OFDM \cdot Nb\_Bit\_CU}{BpS\_OFDM \cdot Nb\_Data\_Carrier \cdot Nb\_Symb\_OFDM\_Frame}$$

where the numerator is the number of data bits per SH Frame, and the denominator is the number of bits per OFDM Frame.

Using (C.3) to (C.7), the above equation reduces to:

$$Nb\_OFF\_SHF = \frac{32}{Mod\_FFT \cdot BpS\_OFDM} \tag{C.11}$$

So, the equation of the SH Frame duration is the following:

$$T_{SHF} = Nb\_OFF\_SHF \cdot Nb\_Symb\_OFDM\_Frame \cdot T_S \tag{C.12}$$

Using (C.2), (C.3) and (C.8), it reduces to:

$$T_{SHF} = 2176 \cdot 896 \cdot \frac{1+GI}{BpS\_OFDM \cdot BW\_OFDM}, \qquad \text{(C.13)}$$

with T_{SHF} in µS.

C.2.3 Calculation of the number of MPEG TS packets per SH Frame

Each turbo code word contains exactly 8 MPEG TS Packets. Thus, the number of MPEG TS packets per SH Frame is:

$$Nb\_TP\_SHF = Nb\_TP\_CW \cdot Nb\_CW\_SHF, \qquad \text{(C.14)}$$

with *Nb_CW_SHF* being the number of FEC code words per SH frame, and *Nb_TP_CW* being equal to 8.

Following FEC coding and rate adaptation, the FEC codeword length in bits is:

$$Nb\_Bit\_CW = \frac{Nb\_Bit\_CW\_0}{cr},$$

with Nb_Bit_CW_0 = 12 096, that is the codeword length assuming a code rate of 1, and cr being the nominal FEC code rate, as defined in EN 302 583 [1], clause 5.3.1.

The number of code word per SH Frame is related to the number of CU per SH Frame:

$$Nb\_CW\_SHF = \begin{cases} \mathrm{int}\left(\dfrac{Nb\_CU\_SHF}{Nb\_Bit\_CW \;/\; Nb\_Bit\_CU}\right), & \text{for OFDM} \\ \mathrm{int}\left(\dfrac{(Nb\_CU\_SHF-3)}{Nb\_Bit\_CW \;/\; Nb\_Bit\_CU}\right), & \text{for TDM} \end{cases}$$

with *Nb_CU_SHF* being the number of CU per SH Frame, int*(x)* being a function which returns the largest integer less than the value *x*. The TDM case is different from OFDM, because the TDM signalling field (3 CUs) needs to be substracted.

$$Nb\_CW\_SHF = \begin{cases} \mathrm{int}\left(\dfrac{Nb\_CU\_SHF \cdot Nb\_Bit\_CU \cdot cr}{Nb\_Bit\_CW\_0}\right), & \text{for OFDM} \\ \mathrm{int}\left(\dfrac{(Nb\_CU\_SHF-3)\cdot Nb\_Bit\_CU \cdot cr}{Nb\_Bit\_CW\_0}\right), & \text{for TDM} \end{cases} \qquad \text{(C.15)}$$

NOTE: $\dfrac{Nb\_Bit\_CW\_0}{Nb\_Bit\_CU}$ simplifies to 6 (a codeword occupies 6/cr CUs).

For OFDM, as defined in (C.6), the number of CU per SH frame is:

$$Nb\_CU\_SHF\_OFDM = 816 \qquad \text{(C.16)}$$

For TDM, the number of CU per SH frame is given in EN 302 583 [1], clause 5.6.1. It is obtained from the following calculation:

$$Nb\_CU\_SHF\_TDM = Sr\_TDM \cdot T_{SHF} \cdot \frac{Nb\_CU\_PLSLOT}{Nb\_Symb\_TDM\_SLOT} \qquad \text{(C.17)}$$

Using (C.8), (C.1), (C.9), (C.10) and (C.13), it reduces to the following expression:

$$Nb\_CU\_SHF\_TDM = \mathrm{int}\left(\mathrm{int}\left(32\frac{1+GI}{1+\alpha}\right)\cdot\frac{BW\_TDM}{BW\_OFDM}\cdot\frac{28}{\mathrm{BpS\_OFDM}}\right)\cdot\frac{1}{1024}\cdot\frac{BpS\_OFDM}{1+GI}\cdot 2228224\cdot\frac{(1+\mathrm{GI})}{\mathrm{Bps\_OFDM}}\cdot\frac{Bps\_TDM}{2176}$$

$$Nb\_CU\_SHF\_TDM = \mathrm{int}\left(\mathrm{int}\left(32\frac{1+GI}{1+\alpha}\right)\cdot\frac{BW\_TDM}{BW\_OFDM}\cdot\frac{28}{\mathrm{BpS\_OFDM}}\right)\cdot Bps\_TDM \quad \text{(C.18)}$$

So, the number of MPEG TS packets per SH Frame is:

for OFDM, using (C.14), (C.15) and (C.16):

$$Nb\_TP\_SHF = 8\cdot\mathrm{int}(136\cdot cr) \quad \text{(C.19)}$$

for TDM, using (C.14), (C.15) and (C.18):

$$Nb\_TP\_SHF = 8\cdot\mathrm{int}\left(\left(\mathrm{int}\left(\mathrm{int}\left(32\frac{1+GI}{1+\alpha}\right)\cdot\frac{BW\_TDM}{BW\_OFDM}\cdot\frac{28}{\mathrm{BpS\_OFDM}}\right)\cdot Bps\_TDM - 3\right)\cdot\frac{cr}{6}\right) \quad \text{(C.20)}$$

C.2.4 Calculation of the capacity

An MPEG TS packet is 188 bytes long, that is *Nb_Bit_TP* bits.

$$\mathrm{Nb\_Bit\_TP}=1\,504 \quad \text{(C.21)}$$

So the bit rate capacity at the MPEG-TS interface is:

$$Cap\_TP = Nb\_Bit\_TP\cdot\frac{Nb\_TP\_SHF}{T_{SHF}}\ \text{[bps]}, \quad \text{(C.22)}$$

for OFDM, using (C.13), (C.19) and (C.22):

$$Cap\_TP\_OFDM = \frac{47}{272\cdot 28}\cdot\frac{\mathrm{int}(136\cdot cr)}{1+GI}\cdot BpS\_OFDM\cdot BW\_OFDM\ ;$$

for TDM, using (C.13), (C.20) and (C.22):

$$Cap\_TP\_TDM = \frac{47}{272\cdot 28}\frac{\mathrm{int}\left(\left(\mathrm{int}\left(\mathrm{int}\left(32\frac{1+GI}{1+\alpha}\right)\cdot\frac{BW\_TDM}{BW\_OFDM}\cdot\frac{28}{\mathrm{BpS\_OFDM}}\right)\cdot Bps\_TDM - 3\right)\cdot\frac{cr}{6}\right)}{1+GI}\cdot BpS\_OFDM\cdot BW\_OFDM$$

Annex D (informative): Low Latency Interleaver extension

D.1 Introduction/Concept/Idea

The idea of the low latency extension to the DVB-SH waveform is to embed a low latency (LL) service into regular services using a long time interleaver. Thus it is possible to transmit services either with a long interleaver with the known benefits or with a low latency interleaver with reduced latency. As the error robustness is reduced through shortened interleaver, different code rates for both services are possible. Both service modes can be used in parallel.

By the chosen way to embed the LL service in the stream, a receiver that does not have this low latency option, is able to decode all services transmitted over the long time interleaver (in the following called regular latency services, in short RL services).

Inserting the low-latency services is re-using the concept of SH services also used for SH time slicing, or local content insertion.

Key facts on low-latency extension:

- low and regular latency services available in parallel;
- backwards compatibility:
  - RL content:
    receivers not aware of low-latency extension or content can safely decode regular latency content without any limitations.
  - LL content:
    regular receivers decode the low-latency content with errors and thus discard it (if eventually decoding is possible, content is removed by PID filter).
  - no performance impact on regular-latency content.
- signalling is based on concepts of SH-services;
- PSI/SI signalling is based on local content insertion;
- low-latency content uses the same FEC scheme and the same time interleaver concept.

D.1.1 Concept

An interleaver delays the interleaver units (IUs) by a certain time. The delay depends on the chosen interleaver. The IUs are fed into tapped delay lines, so that their delay is given by the length of the delay line. The IUs are fed sequentially to all 48 delay lines. The placing of 48 IUs to the interleaver (one IU into every tapped delay line) is called an interleaver cycle (see Figure D.1.1, from [1]).

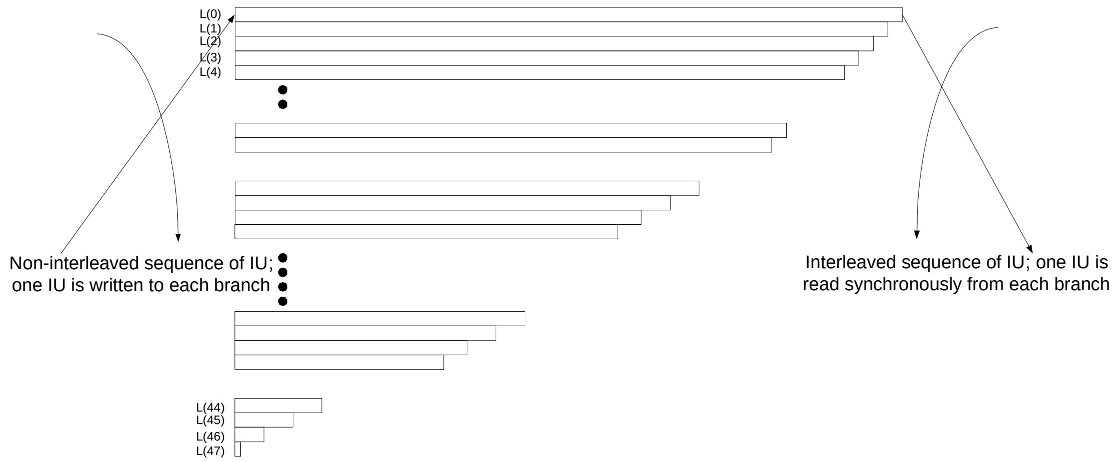


Figure D.1.1: Regular Latency Interleaver (from [1], Figure 5.9)

The DVB-SH time interleaver is characterized by the following facts:

- due to tap-delay lines delay is different for each tap, each IU of one interleaver cycle has a different delay;
- every IU has a delay according to the length of the tap it is passed through;
- due to integer number of ICs of one codeword, the interleaver is synchronous to the SH frame, so e.g. the first IU of each SH-Frame has the same delay;
- IUs which are separated by one (or N) SH frame(s) at the input of the interleaver, are also separated by one (or N) SH frame(s) at the output of the interleaver.

In DVB-SH every SH-frame and even each codeword consists of an integer number of interleaver cycles. So the pattern (source to destination, placement of IUs inside the interleaver output stream) of IUs is static for each interleaved SH-Frame. The difference in source SH-Frames of the regarded IUs is the also the difference in SH-Frames of the IUs in the interleaved codeword stream. The position inside the encoded SH-Frames is the same for IUs with the same source position (compare in Figure D.1.2 IU 0 from the source SH-Frames always maps to the IU 3 in the next interleaved SH-Frame). The IU placement at the interleaver output differs for other modulations and interleaver configurations.

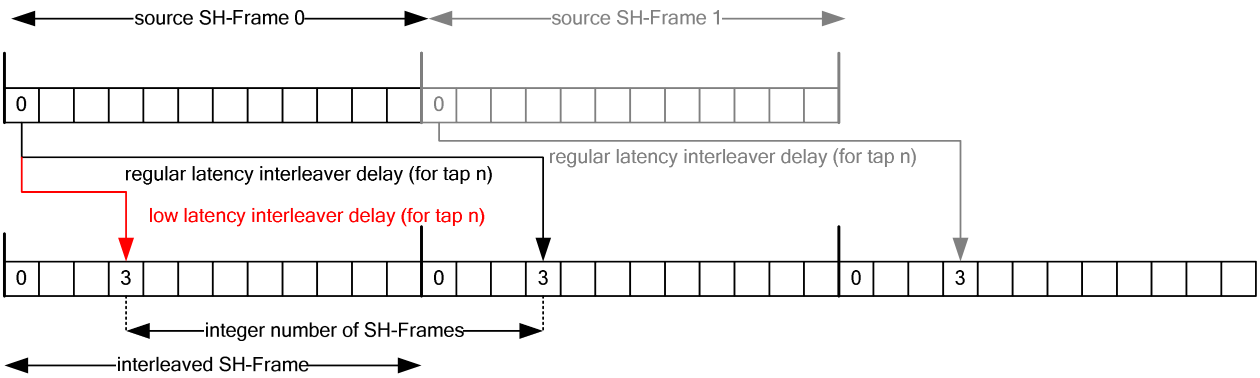


Figure D.1.2: Relation between regular latency and low latency interleaver delay (on the example of one interleaver tap)

So, from the IU point of view, each IU n (in Figure D.1.2: e.g. IU 0) is placed at a position X_n (in the figure e.g. position 3) in the SH-Frame Y_n (in the figure e.g. next interleaved SH-Frames) SH frames later than the current one. So the IU position X_n in each SH-Frame belongs always to the same source IU and so to the same codeword position in the non-interleaved SH-Frames (as the SH time interleaver is SH-Frame periodic). In other words, every IU of a source SH-Frame is placed at a location inside an interleaved SH-Frame, input IUs of another SH-Frame, are mapped to the same *relative* output location, only shifted by the difference in source SH-Frames.

The idea of the low latency interleaver is that for certain codewords the interleaver is shortened by using a modulo SH-Frame function. So, for the IUs of these codewords, the mapping is calculated to place at the position X_n in the *current* interleaved SH-Frame (or the following one, if IU is one of last IUs of the source SH-Frame) (in Figure D.1.2, see the red arrow). This position would normally be occupied by the IU 0 of the previous source SH-Frame, but if the low latency interleaver is applied to all SH-Frames, the concerning position is not occupied by the IU from the previous SH-Frame.

So, if the assignment to low latency data and regular latency data is done on complete codewords, the regular latency stream will not be corrupted. If a stream is deinterleaved by a regular latency interleaver, the IUs of the low latency data are mapped to complete codewords, but the IUs are from different SH-Frames. So a regular receiver will not be able to decode the codewords of the low latency data, while the codewords of the regular latency data will not be affected at all. A codeword belonging to the LL multiplex will result in a decoding error. If a regular receiver may decode by chance the LL multiplex, it will be able to distinguish the packets by the different PIDs. Details in clause D.4.2.

The low latency profile is qualified by the following points:

- FEC code rate for the low latency code words (CR_{LL}) can be assigned independently from the regular latency code rate (CR_{RL}), its value is included in the signalling functions (details, see clause D.2);
- The physical layer time interleaver profile for the low latency code words is directly derived from the RL physical layer time interleaver profile by a simple "modulo" operation, no signalling is required (refer to clause D.2.4 for the interleaver profile);
- The assignment of codewords between RL and LL is periodic within each SH-Frame, while changing the relative shares is possible through a reconfiguration called "re-multiplexing" (non-disruptive for RL and LL, refer to clause D.4). The assignment of RL and LL code words is signalled accordingly;
- Code combining between TDM and OFDM in SH-B remains completely untouched as the same code words are used for RL transmission. Additional code words on one of the two branches (one branch possibly having higher throughput than the other, local content) can be assigned either to RL or LL;
- LL code words are never code-combined between TDM and OFDM and allow therefore independent handling of LL traffic on satellite and terrestrial component;
- Backwards compatibility: The regular latency data can be decoded completely with a regular receiver without modifications. The low latency data can be decoded with a slightly modified receiver. The interleaver is set to the modified values of the low latency interleaver. The decoding control is adapted if different code rates are used (see also clause D.1.3).

D.1.2 Architecture

Figure D.1.3 gives an overview on a possible modulator architecture including the low latency extension. There are two streams that are transferred to the modulator, one for the regular latency data and the optional one for the low latency data. The data rate of the regular latency stream will match exactly the expected data rate for a regular latency only system. To regular modulators the regular latency stream can be fed without any compatibility issues as the regular latency stream remains completely backward compatible, thus regular modulators with a single input will continue to work in such environment, nevertheless in an SFN environment all modulators need to follow the same rule.

In the low latency branch many blocks from the regular latency branch are reused unmodified, as Figure D.1.4 shows. The low latency stream is adapted on the base of IUs, before it is written synchronously with the regular latency stream to their respective time interleavers. After the block called "Post-interleaver Multiplexer", the processing is exactly the same as in a regular modulator. For regular modulators the "Post-interleaver Multiplexer" reduces to static mapping of the RL stream to the output (can be considered as transparent in terms of the regular stream).

The details are described in the clauses below.

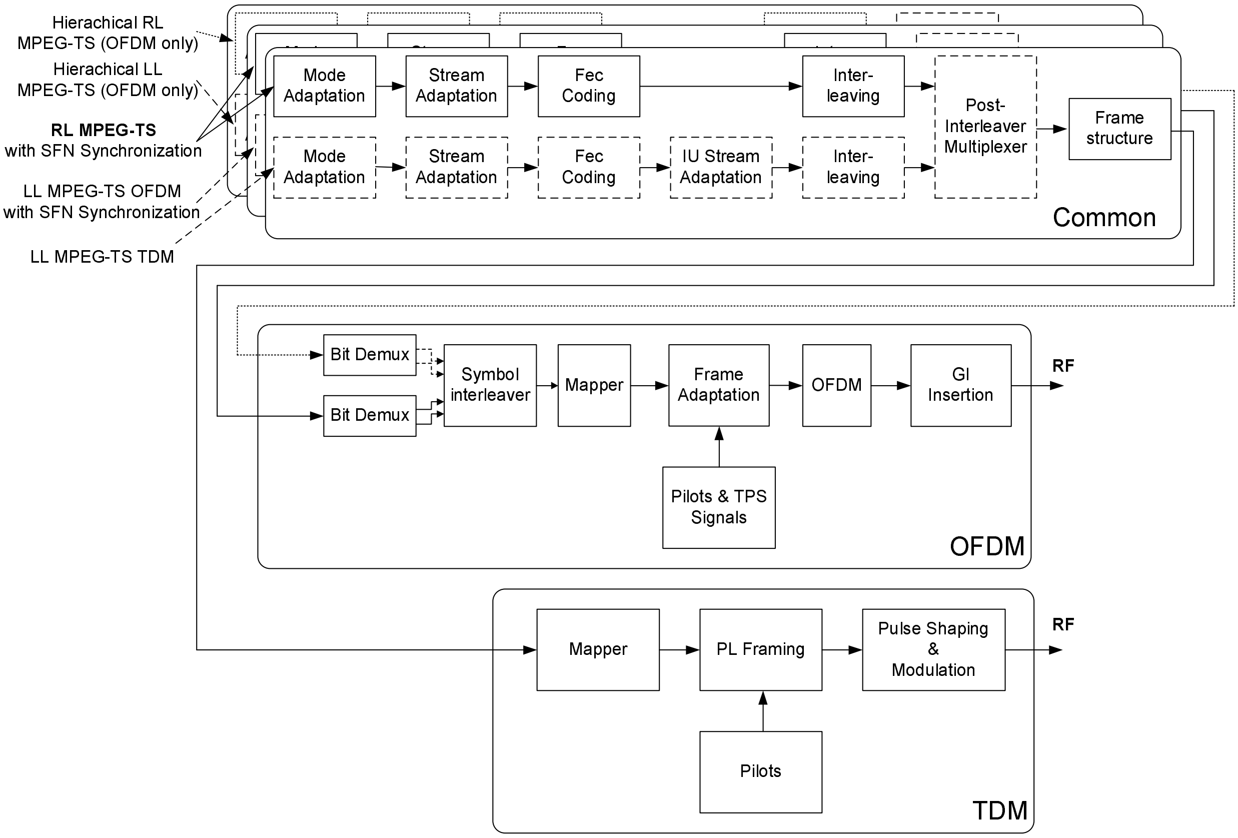


Figure D.1.3: Functional block diagram of the DVB-SH modulator with additional LL MPEG TS input

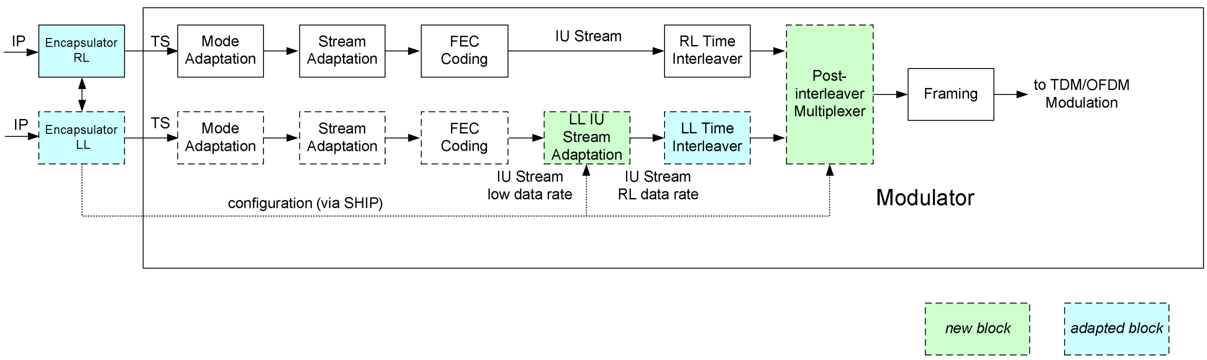


Figure D.1.4: Architectural Overview of a modulator including LL extension

D.1.3 Backwards Compatibility

This clause summarizes the backwards compatibility of the low-latency extension regarding equipment at different processing steps.

Table D.1.1 summarizes the behaviour of a regular or extended modulator in case of regular TS only or regular and low-latency TS input.

Table D.1.1: Backwards Compatibility Matrix (Transmitter Side)

	Regular modulator	Modulator with low latency option
Regular TS only	modulator operates on one stream according to regular processing.	modulator operates on one stream according to regular processing, low-latency content is Null.
Regular TS AND low latency TS	modulator operates on one stream according to regular processing, low-latency content is grounded before (not used by) modulator.	modulator operates on two streams according to low-latency extension.

Table D.1.2 summarizes the behaviour of regular or extended receiver in case the received signal is generated by a regular or extended modulator assuming regular and low-latency TS has been generated by the encapsulator and transferred to the modulator.

Table D.1.2: Backwards Compatibility Matrix (Receiver Side)

	Regular receiver	Receiver with low latency option
Regular modulator	standard equipment behaviour, receiver will decode Null TS packets at the positions intended for the LL service.	receiver can decode the complete RL service. LL service is signalled, but not present (due to regular modulator). LL service positions will be Null TS packets, thus no LL SHIP and no LL service decoding.
Modulator with low latency option	receiver will be able to decode the complete RL service, LL service will be regarded as non-decodable EFRAMES (see clause D.4.2).	receiver will be able to decode compete RL and LL service.

D.1.4 Latency

D.1.4.1 Latency through transmission network

The LL multiplex is transmitted over the transmission network in the same way as the regular latency or RL multiplex. Its latency depends on the kind of transmission. If the stream is transmitted to a satellite or via a satellite feed (e.g. to CGCs), the trip delay is quite large. A latency of 240 ms should be calculated for the network. On the other hand, if the feed is via a cable network e.g. to CGCs, the latency can be in the range of some ms.

Nevertheless the longest distribution path defines the necessary delay for the on-air timing. This delay is given in the SHIP packet as maximum_delay. It adds directly to the LL latency, so this delay should be kept as short as possible, which increases the requirement to the transmission network.

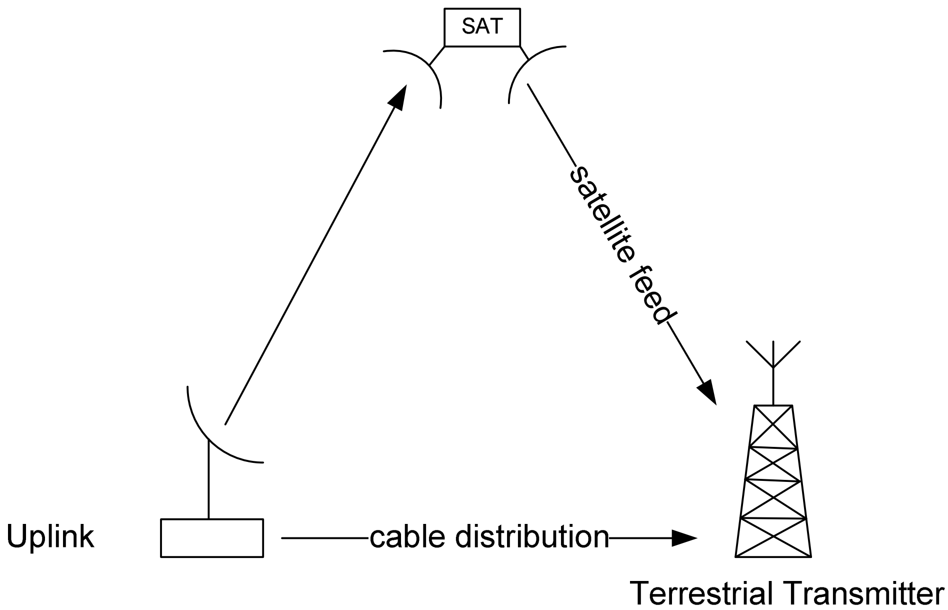


Figure D.1.5: Different possibilities to distribute the stream to terrestrial transmitters

D.1.4.2 Latency in the modulator

The modulator receives the TS stream(s) and transmits them as RF signals. The modulators calculate the transmit time of the interleaver, align the SH-Frames and compensate the interleaver delay of different transmitting branches.

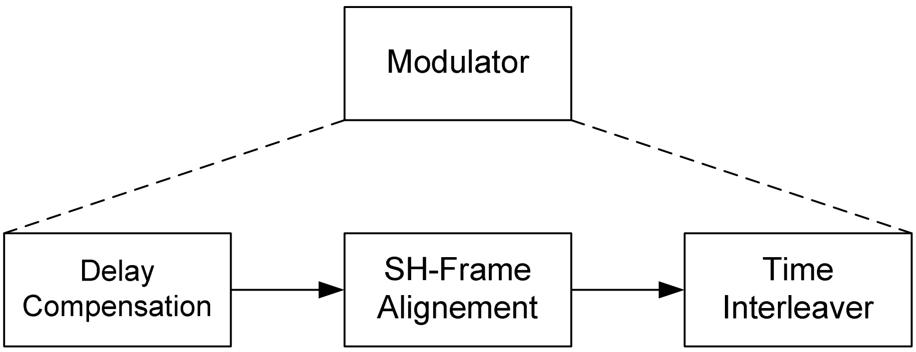


Figure D.1.6: Latencies in the modulator

D.1.4.2.1 Interleaver Delay Compensation

The same SH-Frames should arrive at the same time at the receiver, even if the time interleaver delay of two transmitting branches are different (e.g. satellite, terrestrial repeaters). To achieve this, the modulator with the shorter time interleaver introduces the interleaver delay compensation (refer to clauses 7.2.3.4 and 7.5). The signal is delayed by the difference of the interleaver lengths in time. This delay may be introduced at the input or the output of the modulator (according to clause 7.2.3.4). The delay can be in the range of some SH-Frames and it may have fractional parts of an SH-Frame.

Clause D.3.1.4 describes the requirement for the modulators and the handling of the delay compensation so that there results no additional delay for the LL multiplex.

D.1.4.2.2 SH-Frame Alignment

The RL and LL multiplex is aligned in the modulator. The processing is uniquely defined, so that each modulator of one branch does exactly the same processing. This is a prerequisite for an SFN. So also the MPEG-TS of the LL multiplex is transmitted with a Framing which is defined in the SHIP packets (see clause D.3.1.4).

At the input of the transmitting time interleavers the LL SH-Framing is aligned to the RL SH-Framing. This is necessary for a synchronous behaviour of each modulator and for the correct processing of the post-interleaver multiplexer.

As the LL SH-Frame is aligned with the RL SH-Frame in the modulator, this may introduce a fixed latency up to the length of a SH-Frame. The SH-Framing is settled by the Encapsulator, as it inserts the gaps for the LL content. In the modulator the processing of LL and RL is synchronized at the interleavers, so that the LL content is placed in the RL gaps left in the RL stream (see Figure D.1.2). Therefore the SH-Frames of RL and LL have to be synchronized, in order to have the SH-Frame starts aligned at the inputs of their respective interleavers. This may result in delaying the LL stream until the next RL SH-Frame start, which may last up to one complete SH-Frame. This additional delay for the low-latency content is not recommended, the following clause describes how this can be accomplished.

The delay of the LL content can be reduced if the LL mux is split into the SH-framing (with dedicated SH-frame start) and the LL payload content. The framing needs to be tight to the RL framing, whereas the content can be shifted. In other words the framing can be considered as containers, which will be filled with LL content to obtain the minimum delay. This principle will become clear in the following:
For achieving the best results, the encapsulator should align the SH-Framing of LL and RL. If the LL stream is generated for a modulator which does not need to apply interleaver delay compensation, the SH-Frames should be synchronous to the RL SH-Frames (see also Figure D.1.7). If the modulator hast to apply delay compensation, the LL SH-Frame is delayed (regarded as shifted) by the fractional part based on SH-Frames of the delay compensation (see also Figure D.1.8). In both cases the position for the LL burst data are calculated with respect to the SH frame start of the respective stream. The incoming LL payload is then encapsulated and assigned to the next possible LL burst. If this rule is not used, the LL service latency is increased. In other words: the actual position of the SH frame start in LL is shifted with respect to the SH frame start of the RL stream. Nevertheless for encapsulation LL payload into the LL stream the closest gap in terms of time is used.

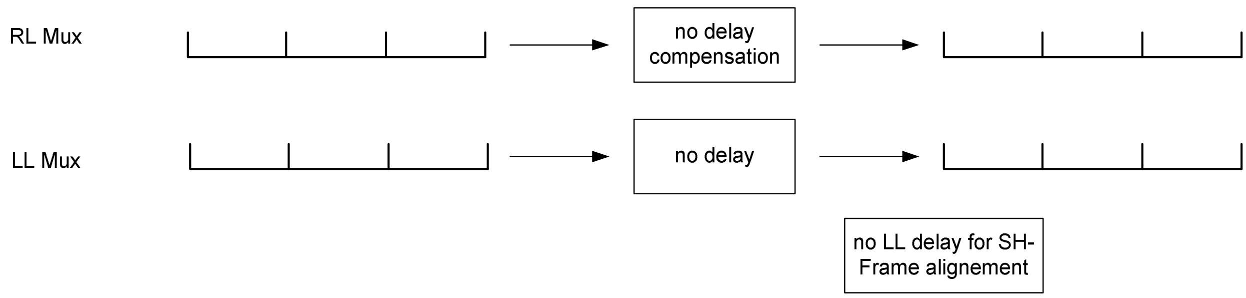


Figure D.1.7: Method of avoiding additional latency through SH-Frame alignment in case of *no* interleaver delay compensation

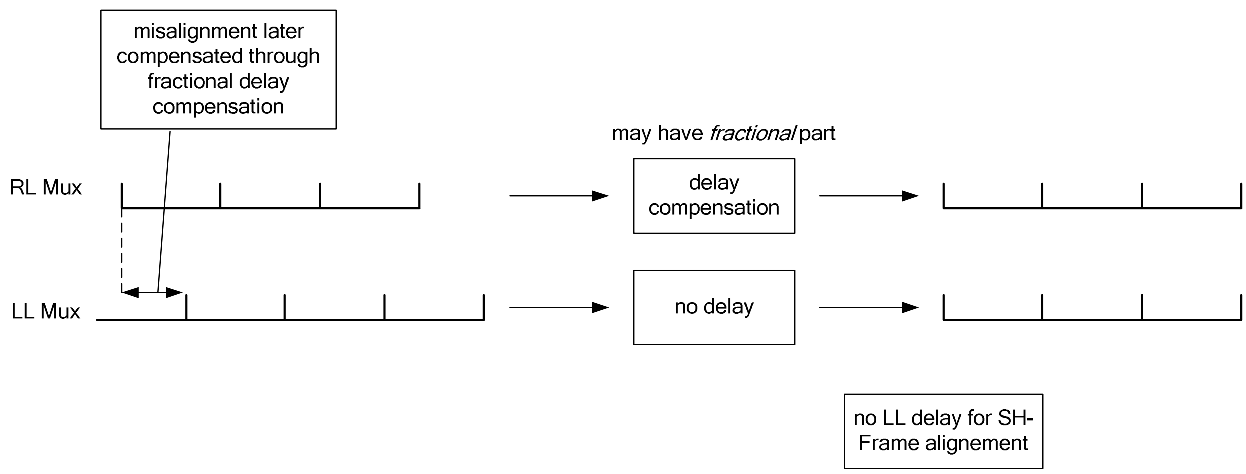


Figure D.1.8: Method of avoiding additional latency through SH-Frame alignment in case of interleaver delay compensation

D.1.4.3 Latency through the interleaver

The time interleaver introduces the interleaver delay. The low latency interleaver shortens that delay, but, depending on the interleaver parameters, this delay still will be up to one SH-Frame. The parameters for the RL time interleaver also define the LL time interleaver and so the interleaver delay for RL and LL.

The LL time interleaver delay can be optimized by the choice of the RL interleaver parameters. If all RL delay lines (transmitter side) are multiples of one SH-Frame, even a zero LL interleaver is possible.

The interleaver delay is introduced by the transmitter and receiver interleaver together, it is $D_{ILV,\,LL}$.

D.2 Physical Layer Processing

The Low Latency Interleaver implies changes in two parts of the processing. First the Common part of the Encoder has to be extended, then the signalling is modified to cope with the introduced LL service. The signalling is described in the clause D.3.

According to [1] the regular input TS is of regular SH frame capacity, thus the RL processing is the same as without low-latency extension. The LL input only carries the used LL payload, i.e. only EFRAMES carrying LL information. The LL stream is brought to the full SH Frame capacity by module LL IU stream adaptation.

The first processing steps until the FEC Coding (including the rate adaptation and bit-wise interleaving) of the LL MPEG TS are the same and follow the regular processing of DVB-SH without the low latency option. The RL and LL streams are processed synchronously, but independently. Up to this stage the processing is operating on a codeword by codeword basis. The first new block is the LL IU Stream Adaptation (as shown in Figure D.2.1).

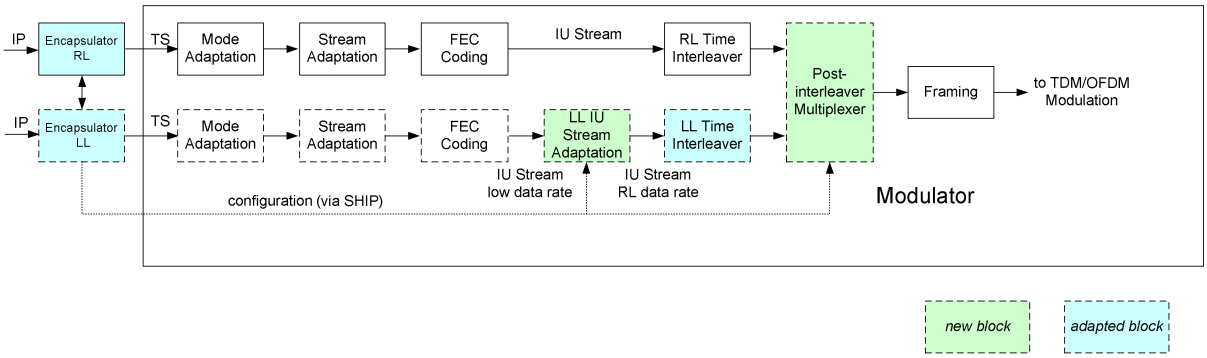


Figure D.2.1: Functional block diagram of the common data processing of a DVB-SH transmitter (for TDM and OFDM)

The processing, described in the following sections is only done for the LL multiplex. It enlarges the LL multiplex to the SH-Frame capacity by inserting dummy IUs at RL service positions and distributing the LL payload to the LL burst. It eventually adds padding IUs at the end of LL bursts.

According to Figure D.2.1 the modulator receives two input transport streams. The processing aligns the adapted and encoded LL stream with the RL stream and generates the streams according to Figure D.2.2. If the low latency option is used, the RL multiplex contains payload and gaps for the LL data. The positions of the gaps are provided by the mux_assoc-vector, this shows whether a codeword belongs to RL or LL, thus a given RL codeword position can either belong to RL or LL.

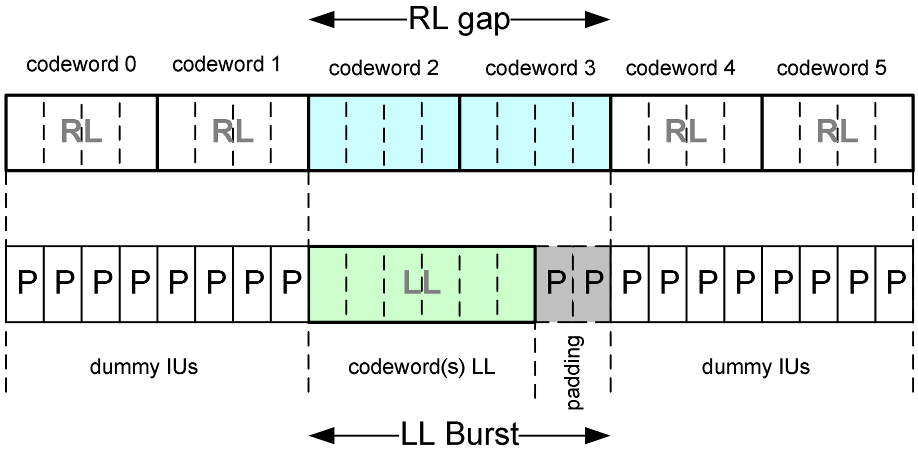


Figure D.2.2: Exemplary relation of RL and LL IU stream before time interleavers, relationship between RL gap and LL burst, time granularity is one IC (= 48 IUs)

The encapsulator prepares the RL and LL streams. On the basis of the mux_assoc-vector, the code rate and puncturing pattern for the RL and LL stream the payload capacity of both streams is calculated. The detailed processing of the Encapsulator is described in the following clauses.

D.2.1 Capacity of LL and RL stream

D.2.1.1 Capacity of the RL stream

The RL stream contains the RL payload data and the gaps, later replaced by the LL data. The capacity for the RL payload is directly derived form the mux_assoc-vector. For each codeword marked as RL (see Table B.1 in [1]) one EFRAME of RL payload can be inserted. So:

$$\text{num\_RL\_payload\_MPEG-TS} = \sum \text{mux\_assoc}[0\ldots N_{CW,RL} -1] \times 8$$

where $N_{CW,RL}$ is the applicable number of codewords for the regarded modulation. The mux_assoc-vector may be larger than applicable e.g. if local content is used for the OFDM modulation, but the RL payload is calculated for the satellite path.

D.2.1.2 Capacity of the LL stream

D.2.1.2.1 Capacity of an LL burst

As the LL code rate (CR_{LL}) may differ from the RL code rate (CR_{RL}), the number of LL codewords (for burst i: $N_{CW, LL,i}$) inserted in an RL gap may also differ from the number RL codewords (for gap i: $N_{CW, RL, i}$) of (i^{th}) RL gap, therefore fractional numbers of LL codewords per burst would be possible.

Derived from mux_assoc-vector, a defined number (later called b) of bursts of LL data can be inserted in the RL multiplex. A LL burst is the correspondence in the LL multiplex to a RL gap in the RL multiplex.

To keep the implementation of the receiver close to the regular receiver only an integer number of LL codewords is inserted into a RL gap/LL burst while the fractional part is padded.

The integer number of LL codeword $N_{CW, LL, i}$ for burst i is calculated by from the number of RL codewords in the corresponding RL gap i ($N_{CW, RL, i}$):

$$N_{CW, LL, i} = \text{floor}\ (N_{CW, RL, i} / CR_{RL} \times CR_{LL})$$

The additional IUs of the LL burst, which cannot host a complete LL codeword, are filled by padding as described in [1], clause B.1.4.2).

NOTE: Each codeword consists of an integer multiple of ICs (by definition).

An OFDM Frame in a regular system is padded with 0 to 18 CUs of PRBS IUs (see [1], 5.5.2.3). If a LL multiplex is used, these interleaver units are automatically assigned to the LL multiplex and included in the calculation of LL bursts.

Please note, that the OFDM SH-Frame padding may form an LL burst of its own, if the last codeword of the SH-Frame is assigned to the RL multiplex, or the padding extends the last LL burst, if the last codeword of the SH-Frame is assigned to the LL multiplex. This enlarges the LL data rate without loss for the RL data rate. This is only applicable for OFDM as the TDM padding is not interleaved.

The number of RL gaps is set to b.

In the case of OFDM SH-Frame padding and last codeword of the SH-Frame is assigned to the LL multiplex, the formula of the last LL burst (b-1) changes to:

$$N_{CW, LL, b-1} = \text{floor}\left(\left(2 \times N_{CW, RL, b-1} / CR_{RL} + N_{padding\_CUs\_OFDM} / 3\right) \times CR_{LL}/2\right)$$

where $N_{padding\_CUs\_OFDM}$ is the number of padding CU at the end of an OFDM SH-Frame (as stated in [1] 5.5.2.3) The number of LL bursts is b.

If the last codeword is assigned to the RL multiplex, the OFDM SH-Frame padding forms a LL burst of its own with the number b:

$$N_{CW, LL, b} = \text{floor}\left(\left(N_{padding\_CUs\_OFDM} / 3\right) \times CR_{LL}/2\right)$$

The number of LL burst is b+1.

D.2.1.2.2 Capacity of the LL stream

So the amount of payload ($N_{CW, LL}$) the LL multiplex can bear in one SH-Frame is calculated as the sum of the LL codewords of all LL bursts per SH - Frame.

$$N_{CW, LL} = \sum^{i}(N_{CW, LL, i}),$$

where i = 0… b-1 (or i = 0… b, if OFDM SH-Frame padding forms an extra LL burst, see above)

NOTE 1: Padding can be minimized by adequate choice of the multiplex association vector with respect to the coderates on RL and LL (CR_{RL} and CR_{LL}). (e.g. there is no padding at the end of an LL burst, if two integer numbers r, s exist, so that $2/CR_{RL} \times r = 2/CR_{LL} \times s$ is fulfilled. The RL gap of r codewords then carries s LL codewords).

NOTE 2: The losses through padding can be minimized by using only a small number of LL bursts per SH-Frame.

NOTE 3: The time between two LL burst causes jitter for the LL service in the order of the burst distance.

D.2.1.2.3 Padding calculations of an LL burst

At the end of the LL burst i, it may be necessary to introduce padding to fill the RL gap i completely. As each codeword in DVB-SH consists of an integer number of interleaver cycles (ICs), the introduced padding is also an integer number of ICs. The introduced number of padding IC ($N_{IC, pad, i}$) is calculated for each LL burst i by:

$$N_{IC, pad, i} = N_{CW, RL, i} \times 2/CR_{RL} - N_{CW, LL, i} \times 2/CR_{LL}$$

NOTE 1: Each IC consists of 48 IUs of 126, so an IC consists of 48 × 126 bit = 6 048 bit.

NOTE 2: In the case of OFDM SH-Frame padding the number of burst may be b+1 instead of b (see above).

The padding is done via a PRBS (like in clause 5.5.2.2 – Padding part) for energy dispersal. The padding sequence is initialized with every LL burst. The number of bits extracted from the PRBS equals the number of bits needed for the padding at the end of the LL burst.

D.2.2 MPEG TS generation (Encapsulator)

D.2.2.1 Multiplex Association Vector (mux_assoc-vector)

D.2.2.1.1 Setting up Multiplex Association Vector (mux_assoc-vector)

According to the requirements in data rate, stream latency and jitter the Encapsulator calculates the number of needed LL codewords for one SH-Frame $N_{CW,LL}$.To limit the maximum jitter and latency the complete number ($N_{CW,LL}$) of LL codewords is split into an amount b of LL bursts, each containing a number ($N_{CW, LL, i}$, i being a number between 0 and b-1) of LL codewords. b should be small to reduce the padding losses, but a small number of bursts also increases the latency and jitter of the LL stream.

The number of codewords per LL burst does not need to be exactly the same for all bursts. The number of LL codewords per LL burst defines the needed number of RL codewords for the corresponding RL gap. The number of RL codewords ($N_{CW, ,RL, i}$) for the given number of LL codewords ($N_{CW, LL, i}$) is given by:

$$N_{CW, RL, i} = \text{floor}(N_{CW, LL, i} / CR_{LL} \times CR_{RL})$$

There is no padding, if $N_{CW, LL, i} / CR_{LL} \times CR_{RL}$ results in an integer number.

NOTE 1: If there is padding at the end of an OFDM SH-Frame, where the number of padding CUs is pad_CUs then the following two cases apply:

- Case 1: the last codeword of the SH-Frame is assigned to LL, then the formula for the last burst changes to:

$$N_{CW, RL, b-1} = \text{floor}((N_{CW, LL, b-1} + \text{pad\_CUs}/6) / CR_{LL} \times CR_{RL})$$

- Case 2: the last codeword of the SH-Frame is assigned to RL, then there is an additional burst, which may contribute to the LL codewords per SH-Frame ($N_{CW, LL}$) but which does not need any RL codeword (or RL gap).

The RL gaps are then distributed over the SH-Frame. To reduce the jitter an equidistant distribution is recommended. The number of the starting codeword of each burst i is defined (as burst_start_i). As the mapping of the burst is the same for all SH-Frames, the distance between the last burst of an SH-Frame and the next is also given.

If fast-access for OFDM systems is required, the first burst is placed at the beginning of the SH-Frame (and this codeword contains the LL SHIP packet, see later).

The multiplex association vector is created with N_{CW} elements, which are initialized to RL (entries = 1). Then, for every burst i (burst_$_i$) (being assigned to LL) with beginning codeword burst_start_$_i$, $N_{CW, RL, i}$ codewords are set to LL (entries = 0). This is further described below.

NOTE 2: In this description so far only the RL gaps / LL-burst are defined as bursts, capacity assigned to RL is not defined as burst (i.e. the burst-index i is related to LL content), this will change in the next clause.

D.2.2.1.2 Translating mux_assoc-vector to a burst-length representation

To be included in the DVB-SH signalling the mux_assoc-vector can not be inserted as is, but has to be mapped to a burst / burst_length description. In this description now the complete capacity is considered as bursts. As the mux_assoc-vector this burst description is given in terms of RL codewords. Each codeword is either assigned to RL or LL, regarding the mux_assoc-vector the consecutive assignment of codewords to the same transmission-mode (RL or LL) is now considered as burst.

Therefore the mux_assoc-vector is transformed into a sequence of nof_burst bursts (index k = 1, … nof_bursts-1), where the burst_length_$_k$ = $N_{CW, RL, i}$, for all bursts k corresponding to a LL-burst i.

NOTE: The mux_assoc-vector has a length of N_{CW} elements, as given in [1], B.1.4.4.

- Each consecutive assignment of codeword to one latency mode (either RL or LL) is one burst. Each burst k has a starting codeword (named burst_start(k)) and a burst length (named burst_length(k)).

- Each change in assignment in the mux_assoc-vector defines a new burst start, the following cases are possible
  1. mux_assoc[i] = 0 (LL), mux_assoc[i+1] = 1 (RL) defines a burst_start at i+1 and
  2. mux_assoc[i] = 1 (RL), mux_assoc[i+1] = 0 (LL) defines a burst_start at i+1 and
- The first burst gets the index 0 and starts at codeword 0, so burst_start(0) = 0.
- Given that the change in assignment between i and i+1 is the k-th (k starting at 1) change, the variable burst_start(k) = i+1
- The burst_length of burst k is defined by:
  - burst_length(k) = burst_start(k+1) – burst_start(k)
  - Given that there are n bursts, the last burst length is defined by burst_length(n-1) = N_{CW} – burst_start(n-1).
- The burst mode (RL or LL) of each burst k is given by mux_assoc[burst_start(k)].

The calculated burst lengths are used in the given order from index 0 to n-1 to be inserted in the service_synchronization_function in the LL SHIP for the "Length" parameter (see clause D.2.2.1.4).

D.2.2.1.3 Translating mux_assoc-vector to a SF representation

For the SF, the burst_lengths values are transferred to the SF burst_length – field in the correct order. Their RL respectively LL attributes are transferred to the latency_mode field.

This translation is defined in [21], clause 4.11.

D.2.2.1.4 Translating mux_assoc-vector to SHIP - representation

For the LL-SHIP, the burst_lengths are signalled via the service_synchronization_function in the correct order.

In the RL SHIP, the first_burst_latency_mode parameter has to be set either to LL if the first burst is a LL burst (case 1), otherwise to RL (case 2).

This translation is defined in [21], clause 4.10.2.5.0.

D.2.2.2 RL MPEG TS

The RL MPEG TS is generated according to special rules, so that the multiplexing with low latency data does not corrupt or erase any RL payload. This processing is called the low latency adaptation of a TS. So for each codeword, that is not assigned to the RL multiplex, 8 Null MPEG TS are inserted (8 TS = one turbo codeword = one EFRAME) at the correct position (e.g. if codeword 0 is assigned to RL, codeword 1 to LL, codeword 2 to RL, the encapsulator fills TS packets 0 to 7 and 16 to 23 with RL payload, TS 8 to 15 are null MPEG TS packets). The resulting low latency adapted RL TS is shown in figures D.2.3 and D.2.6.

This ensures that the RL multiplex has the same data rate as a regular latency DVB-SH stream with the same modulation parameters. So this stream can be fed to a regular modulator without constraints. The receiver would then receive some extra null MPEG TS packets, which do not affect its processing.

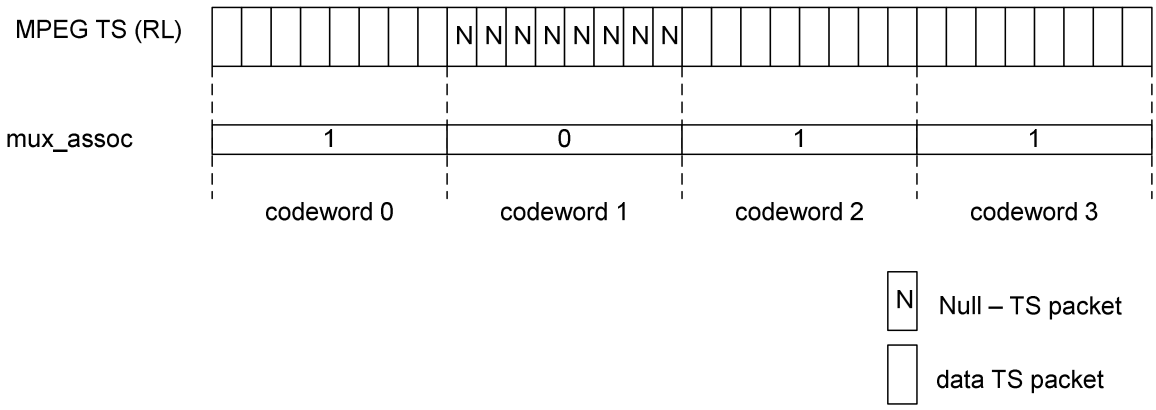


Figure D.2.3: Mapping of mux_assoc-vector to MPEG-TS packets

The assignment of codeword to the multiplexes is calculated by the encapsulator and given in the mux_assoc-vector. The defined positions, where the low latency (LL) multiplex is inserted is filled with Null TS-Packets. This can be achieved in two ways.

1) The encapsulator generates a stream with the data rate of the RL payload (given in clause D.2.1.1 as num_RL_payload_MPEG-TS). In the following the Null MPEG TS packets AND the SHIP are inserted at the positions, which are stated by the mux_assoc-vector. For each codeword assigned to RL, 8 RL payload MPEG TS packets are used from the previously generated stream. For each codeword that is **not** assigned to RL, 8 null MPEG-TS packets are inserted. No RL packets are discarded. The SHIP packet are inserted in the RL part. The SHIP states the SH-Framing, otherwise the codewords boundaries cannot be determined (see Figure D.2.4).

NOTE 1: The option with a low latency adapter does only work, if no SH-Services are used for SH time slicing, which would implies knowledge of the time interleaver by the Encapsulator. In the case of parallel uses of SH-Service for SH time slicing, the next method is used.

2) The encapsulator generates a stream with the full data rate of an SH-Frame. But it also inserts the SHIP packet and the null MPEG TS packets as described in 1. (see also Figure D.2.5).

The mux_assoc-vector is static over the time. After its initial generation it can only be changed by a re-multiplexing (see clause D.4)

NOTE 2: With the second method it is easier to set the delta-t timestamps of MPE time slice streams correctly.

NOTE 3: The difference of point 1 and 2 is the number of processing units. In point 1 the encapsulator may be a regular one, which does not need to know about low latency interleaver option, this is done by a separate processing unit, in Figure D.2.4 given as "low latency adapter". This adapter performs the missing steps for LL extension in accordance with the requirements in [21] (Null packet insertion, modification of (RL-) SHIP (insertion of ll_service_function, pointer to next SH frame start), update on MPE delta-t time-stamps (if applicable)).

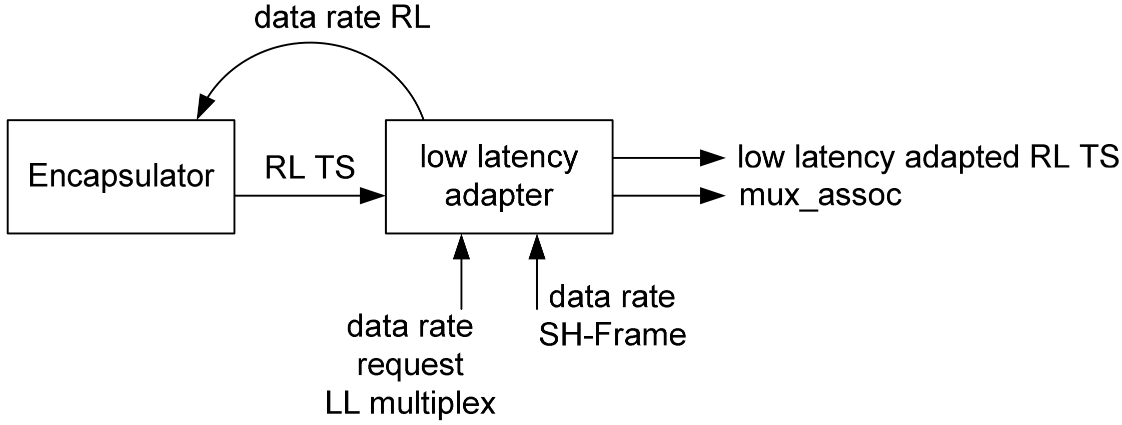


Figure D.2.4: RL stream generation with a regular encapsulator and an adaptation block

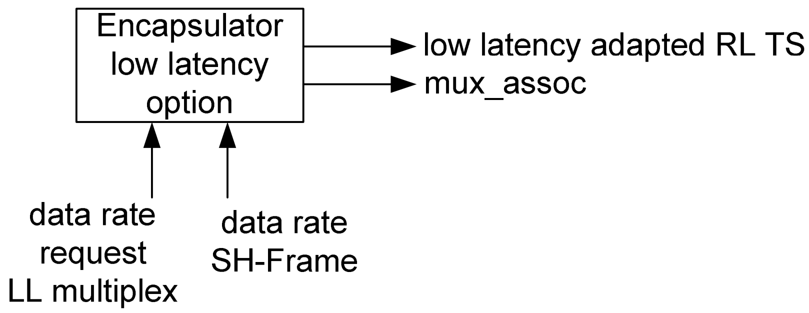


Figure D.2.5: RL stream generation with a modified encapsulator (direct generation of low latency adapted RL TS)

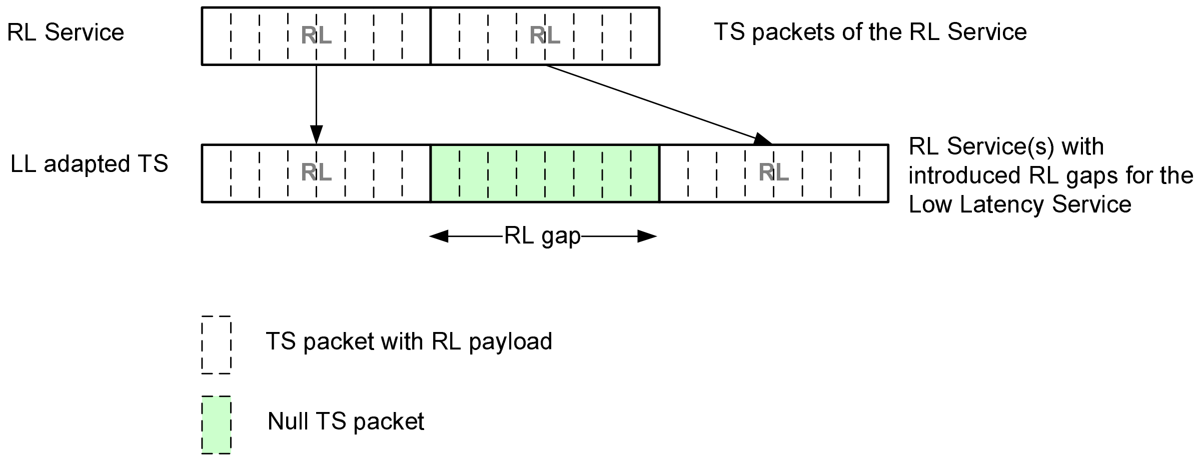


Figure D.2.6: Low Latency Adapted TS

D.2.2.3 LL MPEG TS

An LL MPEG TS SH-Frame is generated with the exact number of:

$$N_{CW,\,LL} \times 8$$

MPEG-TS packet by the encapsulator. A SHIP is inserted to define the SH-Frame boundaries. There are no gaps inserted for the RL multiplex. The distribution to the LL bursts is done by the LL IU Stream adaptation in the modulator

The LL (and RL) multiplex is generated according to the rules of the local content, which are described in [21], clause 4.2. (In short: The PAT of RL and LL is complete and the same for both. It is transmitted at least in the RL multiplex. The PMTs for the LL services are only contained in the LL multiplex, the PMTs for the RL services are only contained in the RL multiplex. NIT, INT, SDT should be mainly transmitted in the RL multiplex due to higher data rates)

D.2.3 LL IU Stream Adaptation

The incoming TS-packets of the LL service are grouped to EFRAMES. There are SHIP packets for the LL and for RL multiplex. These are necessary for the modulator to synchronize the two input stream. The receiver will get 2 different multiplexes, as they have different physical parameters. The receiver needs the LL SHIP for fast access strategies to the LL multiplex.

For each SH-Frame, the first $N_{CW,\,LL,}$ EFRAMES are used for the further processing. Additional EFRAMES are dropped. This should not happen, if the LL multiplex is generated as described in clause D.2.2.3. If EFRAMES are missing in the LL multiplex, the modulator does not transmit power.

The $N_{CW,LL}$ EFRAMES are encoded and processed according to the desired LL code rate CR_{LL}.

The resulting IU stream is divided to b (or b+1, see clause D.2.1.2) LL burst with $N_{CW,\,LL,i}$ codewords. Each LL burst is padded with $N_{IC,\,pad,\,i}$ padding interleaver cycles (Switch X in Figure D.2.7).

The resulting LL IU-Stream is build according to the following rules:

- For each codeword assigned to the RL multiplex, the according number of dummy IUs ($48 \times 2/CR_{RL}$) is inserted (Switch X in Figure D.2.7 to dummy IUs).
- Beginning at the first position of i-th LL burst/RL gap, the $N_{CW, LL,i} \times 48 \times 2/CR_{LL}$ LL payload IUs are inserted (Switch X to LL payload IUs).
- Then at the end of each LL burst, $N_{IC, pad, i}$ padding IC or $N_{IC, pad, i} \times 6\,048$ padding bits of are PRBS are inserted (refer also to clause D.2.1.2.3) (Switch X to PRBS).

The resulting LL IU stream has the same data rate as the complete RL IU stream. (see clause 7.6 for regular latency/RL data rate calculation)

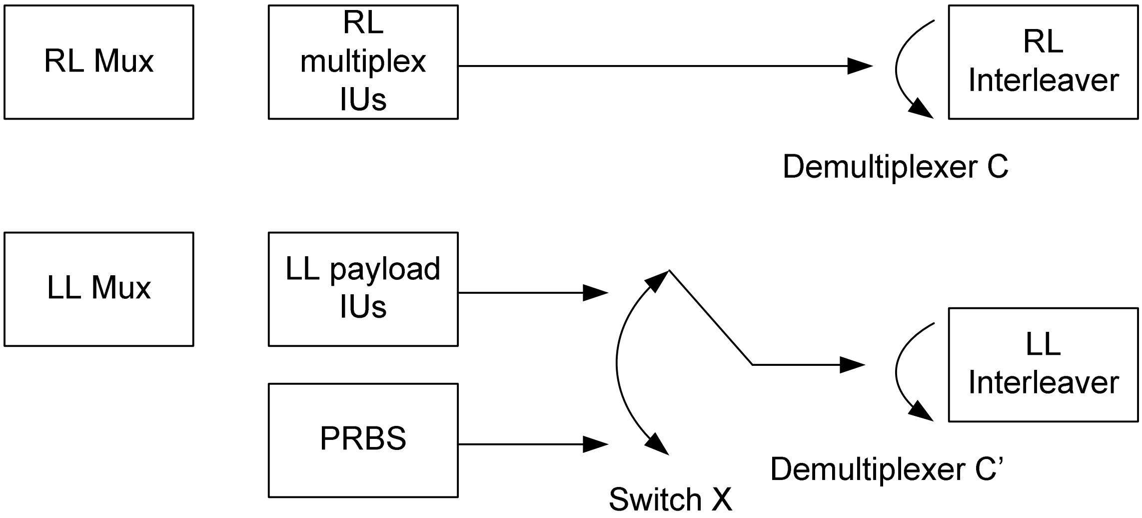


Figure D.2.7: Processing of RL and LL multiplex in the modulator

Both IU streams are multiplexed by the perfect synchronous Demultiplexers C/C' to the corresponding RL/LL interleavers.

Please note: The IU Stream of the RL multiplex is generated in the same way as in regular equipment. (refer to Figure D.2.1 and [1], up to 5.4)

D.2.4 Low latency time interleaver

The interleaver profile of the low latency interleaver is defined in the transmitter side (while the RL interleaver was defined at receiver side). This results in a straight-forward and unambiguous description.

D.2.4.1 Transmitter Side

The RL interleaver profile is defined for the Receiver. (see [1], clause 5.4).

The interleaver profile for the LL multiplex is derived from the RL interleaver profile of the RL Transmitter. $D_{TX, RL}[i]$ is the tap-length of the i-th tap of the RL transmitter interleaver.

NOTE 1: $D_{Tx, RL}[0]$ is the longest tap, $D_{Tx, RL}[47]$ is the shortest tap with a delay equal to zero.

The LL transmitter interleaver is derived by the following formula:

$$D_{Tx, LL}[i] = D_{Tx, RL}[i] \bmod L_{SH, IC}$$

NOTE 2: mod() is the modulo function.

D.2.4.1 Receiver Side

Further more is:

$$D_{ILV,\,LL} = \max(D_{Tx,\,LL}[0\ldots47])$$

NOTE: max() chooses the largest element.

The LL receiver interleaver is defined by:

$$D_{Rx,\,LL}[i] = D_{ILV,\,LL} - D_{Tx,\,LL}[i]$$

So the longest tap of the LL interleaver gets a delay assigned to the value of $D_{ILV,\,LL}$, which is the lowest possible for the chosen RL interleaver.

Both multiplexes are synchronized at the input of the interleavers. When the first IU of the RL stream is fed to the RL interleaver, also the first LL IU is fed to the LL interleaver.

D.2.5 Post-Interleaver Multiplexer

The multiplexer F selects IU-wise between the outputs of the two multiplex interleavers. The order of the selected outputs is derived from (a) the sequence of codewords of the different multiplexes (which is periodic) and (b) the disperser profile of RL.

Let be p[n] the sequence of the multiplex labels (if RL or LL IU carries useful data) for every interleaver cycle at the inputs of the channel interleavers. This vector is derived from the mux_assoc-vector (length N_{CW}).

p[n] has the length of:

$$L_{SH,\,IC} = N_{CW} \times 2/CR_{RL} \text{ (2 Interleaver cycles per unencoded codeword)}$$

For each codeword (with respect to CR_{RL}) that is assigned to the RL multiplex, $2/CR_{RL}$ "ones" are inserted in p[n], for each codeword that is not assigned to the RL multiplex, $2/CR_{RL}$ dummy IUs are inserted.

Moreover, let $D_{Tx,LL}[0..47]$ again represent the interleaver profile of LL in the transmitter, i.e. $D_{Tx,LL}[i]$ is the delay (in terms of interleaver cycles) of the tapped delay line i.

Let m be the index of the current interleaver cycle, which is written into the channel interleaver. m has the values of 0 to ($N_{CW,\,RL} \times 2/CR_{RL}$ -1), the first IC (number 0) starts at the start of an SH-Frame, which is indicated by the SHIP packets for the RL and LL multiplex.

The order of the multiplex selected by multiplexer F for this interleaver cycle m is:

$$p[m-D_{Tx,LL}[0]],\ p[m-D_{Tx,LL}[1]],\ p[m-D_{Tx,LL}[2]]\ \ldots,\ p[m-D_{Tx,LL}[47]].$$

The multiplexer at the output of the interleavers chooses one IU, either from the RL or the LL interleaver. The decision is made for each delay line separately. The not-chosen IU is dropped. At the i-th delay line it chooses:

$$p[(m- D_{Tx,LL}[i] + N_{CW} \times 2/CR_{RL}) \bmod (N_{CW} \times 2/CR_{RL}))]$$

If the calculated value is one, it chooses the RL IU, if zero, it chooses the LL IU. This pattern is periodic for each SH-Frame. The multiplexer always chooses either RL data, LL data or LL padding. The dropped IUs are either dummy IUs (if the LL part is dropped) or IUs that result from the inserted Null-TS packets in the RL stream.

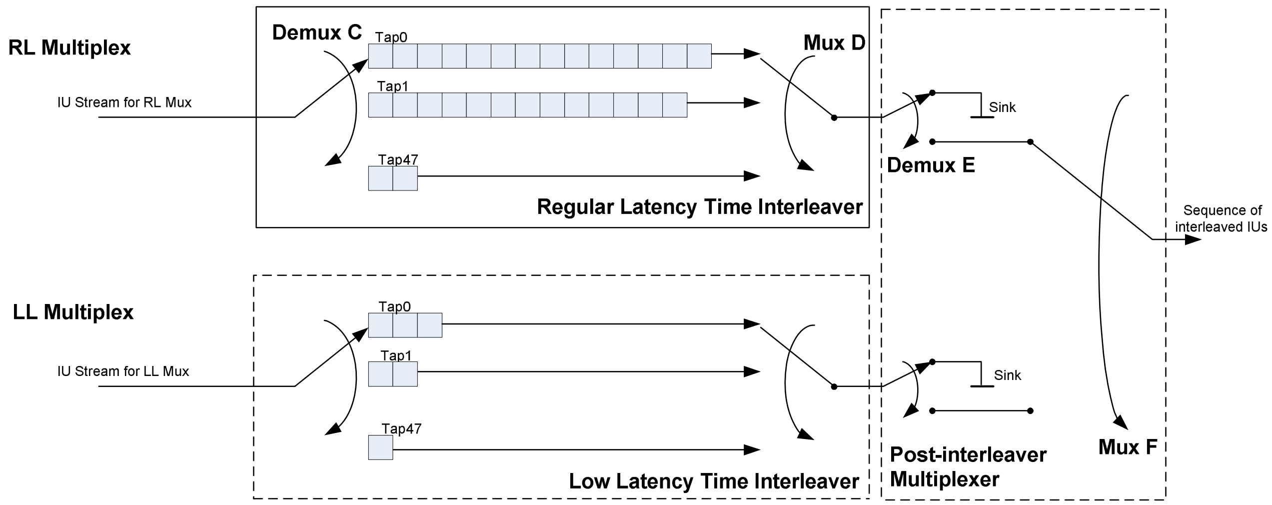


Figure D.2.8: The low latency channel interleaver consisting of the two channel interleavers for the multiplexes

D.3 Signalling

D.3.1 Physical Layer/MPEG TS

D.3.1.1 Signalling Field – TDM

The signalling field of the TDM SH-Frame has to be extended in the following way. Receivers with the low latency option decide by the bit 112 to 119 if the signalling for a LL service is present. If not, the low latency option is not used. Legacy receivers ignore the parameters behind bit 111.

Table D.3.1: Signalling field with LL multiplex extensions

Parameters for the DVB-SH frame with signalling field					
Start bit index	**Parameter**	**Description**	**Wordsize (bits)**	**Format**	**Comment**
0	Signalling_Version	Version number of the DVB-SH signalling format	8	U8	Fixed to 0 other values are RFU If values are ~= 0 the receiver ignores the signalling field.
8	RFU	RFU	8	U8	RFU bits
16	Frame_Width_CUs	DVB-SH frame width in Cus	12	U12	CUs are used as the unit in order to allow receivers to know the width of the DVB-SH frame.
28	Punct_Pat_ID	ID number of the Turbo code puncturing pattern	4	U4	See table 5.2.
32	Common_multiplier	Tap length common multiplier	6	U6	Values from [1…63], is by default the "late" part step; 0 is not allowed.
38	Nof_late_taps	Number of taps in the late category	6	U6	Values from [0…48], whereas "0" signals no late part available, and "48" signals only late part available.
44	Nof_slices	Number of slices over which the data is distributed	6	U6	Values from [1…63], if only late part is used, this value is set to 1.
50	Slice_distance	Distance between two slices	8	U8	Values from [0…255]; is multiplied with the SH frame capacity in IU and divided by 48 to get increment in IU. Value set to 0 if interleaver applies only to 1 slice.
58	Non_late_increment	Increment between taps inside the non-late slice(s)	6	U6	Values from [0…63]; is multiplied with common_multiplier to get increment in IU. Value set to 0 if interleaver applies only to 1 slice.
64	RFU	RFU	32	U32	RFU bits.
96	CRC_16	CRC-16 over the first 96 bits	16	U16	Polynomial as defined in clause 5.1.1.
112	LL_par_present	LL parameters present 0: no 1: yes	1	U1	DVB-SH signal has the extended format with 2 multiplexes
113	RFU	RFU	1	U1	RFU bits
114	LL_SAT_active	LL content over the satellite branch is active	1	U1	1: there is LL content distributed over the satellite path
115	Next_Conf	0: current configuration 1: future configuration	1	U1	Flag stating if the transmitted burst_description and LL_Punct_Pat_ID_TDM are valid for the current multiplex (0) of LL or the future multiplex (1)
116	LL_Punct_Pat_ID_TDM	Puncturing pattern ID of LL for the TDM	4	U4	
120	RFU	RFU	4	U4	RFU bits
124	nof_bursts	Number of RL/LL burst + 1, up to 16 are possible	4	U4	0: 1 burst, …, 15: 16 bursts
128	16x burst_description	[15]: latency_mode (1: RL, 0: LL) [14:11]: RFU [10:0]: burst_length	256	16xU16	Only the first nof_burst+1 burst_descriptions are used. All additional are written to zero. The sum of all used lengths summarizes to the length of the TDM SH-Frame length in EFRAMES.
384	CRC_16	CRC-16 over the previous 272 bits	16	U16	Polynomial as defined in clause 5.1.1
400		RFU	746	U1	Remaining bits are RFU bits.
Total length of Signalling field			**1 146**		

D.3.1.2 SHIP

In the SHIP packet, the LL multiplex is described for the modulators. For this purpose, a new function (ll_service_function) is defined and the existing function service_synchronization function is reused.

The ll_service_function is used to transmit the necessary physical layer parameter, while the service_synchronization_function give the LL service positions.

Table D.3.2: (Table A.6 of SH-WF ([1]): Tag value of functions

Function	function_tag value
tx_time_offset_function	0x00
tx_frequency_offset_function	0x01
tx_power_function	0x02
private_data_function	0x03
cell_id_function	0x04
Enable_function	0x05
bandwidth_function	0x06
service_loc_function	0x07
service_sync_function	0x08
tdm_function	0x09
group_membership_function	0x0A
ll_service_function	0x0B
tdm_auxiliary_function	0x0C
Future_use	0x0D to 0xFF

Table D.3.3: Function ll_service

Syntax	Number of bits	Identifier
ll_service_function() {		
function_tag	8	
function length	8	
for (i=0; i<N; i++) {		
ll_ship_cw_present	1	
first_service_latency_mode	1	
modulation_id	2	
ll_punct_pat_id	4	
if (ll_ship_cw_present==1) {		
ll_ship_cw	16	
}		
}		
reserved	7	
wait_for_enable_flag	1	
}		
Syntax	**Number of bits**	**Identifier**

ll_ship_cw_present: Indicates if the start_cw – field is present.

first_service_latency_mode: Indicates if the first service (service ID) belongs to the RL multiplex (1) or to the LL multiplex (0). The services alternate strictly between RL and LL.

modulation_id: Indicates the transmission path the given parameters are valid for (see table below).

modulation_id	Description
00	SAT (TDM/OFDM)
01	TERR HP (OFDM)
10	TERR LP (OFDM)
11	RFU

In case of OFDM without hierarchical modulation the ID TERR_HP is used.

ll_punct_pat_id: Puncturing pattern ID of LL service (see [1], table 5.3).

ll_ship_cw: This parameter gives the number of the EFRAME in the SH-Frame, which bears the SHIP packet of the LL multiplex (This value only changes during remultiplexing)

wait_for_enable_flag: If this flag is set to "0" then attach_detach function has to be activated immediately. If this flag is set to "1" then attach_detach function has to be activated immediately after having received the corresponding enable_function.

reserved: Set to 0.

D.3.1.3 Generating the TDM signalling field in a TDM modulator

By the TDM modulator some content of the RL/LL-SHIP packet will be inserted into the TDM SF (see Table D.3.1). Most of the contents are directly derived from the SHIP.

Attention has to be paid to the timing requirements in case of re-multiplexing (description see clause D.4). If a new configuration is enabled by the SHIP enable_function, this new configuration is put valid (next_conf = 0) in the next TDM-SF following this SHIP packet. I.e. The SHIP is contained in the data-part of the TDM frame, a configuration in the TDM-SF is put valid with the next SF following the SHIP with the enable function.

D.3.1.4 Synchronization of LL and RL in the modulator

The LL multiplex is synchronized with a SHIP packet and the same mechanism as the RL multiplex.

The RL multiplex is synchronized between TDM and OFDM so that the SH-Framing is aligned. To reach this goal, the shorter interleaver delays the stream by the (possibly fractional to SH-Frame base) difference of the longest interleaver delay taps (see clause 7.5).

The LL Multiplex should be inserted in the next possible SH-Frame. All delay, that may be avoided, should be avoided (see clause clause D.1.4 for a details). But the LL multiplex (the LL IU stream) is aligned with the RL multiplex (IU stream) on SH-Frame base at the input of the time interleaver. This ensures that the LL multiplex is placed at the RL gaps by the Demultiplexer E. From this point on, the processing of RL and LL is synchronous as it is the same IU stream.

For the exact description of the synchronization, some aspects are explained in advance:

Some characteristics of the LL interleaver differ from the RL Interleaver. For RL interleavers the tap 0 ($D_{Rx, RL}[0]$) in the receiver is fixed to zero, so in the transmitter $D_{Tx, RL}[0]$ is the largest delay tap. It gives the total interleaver delay for RL interleaver ($D_{ILV, RL} = D_{Tx, RL}[0]$). For the LL interleaver the $D_{Tx, LL}[0]$ in the transmitter is not necessarily the longest tap, as it only is the fractional part of SH-Frames of the RL interleaver tap ($D_{Tx, LL}[0] = D_{Tx, RL}[0] \bmod L_{SH}$). The LL interleaver delay is the length of the longest LL tap, so $D_{ILV, LL} = \max(D_{Tx, LL}[i])$. From the definition of the LL receiver interleaver, it is obvious that the tap 0 is not necessarily zero, as $D_{Rx, LL}[0] = D_{ILV, LL} - D_{Tx, LL}[0]$. (Refer to clause D.2.4).

The definition of the "start of an SH-Frame" is for the LL multiplex the same as for the regular latency system:

For OFDM this is defined in clause 7.2.4.1, for TDM in clause 7.2.4.2.

As the RL and LL part of the stream are aligned at the input of the interleaver only the "first data IU of LL" is clarified: The first IU always refers to the first IU of the LL IU stream of the SH-Frame before the time interleaver. Please note: this IU is a dummy IU if the first regular latency codeword is assigned to the RL multiplex (see Figure D.3.2 and Figure D.3.3, where codeword 0 corresponds to the start of an SH frame).

If both IU streams are synchronized at the input of the time interleaver, so that both IU 0 are written synchronously to the interleavers, also at the output the IU 0 are synchronous, as the tap 0 of the LL interleaver is the fractional part of the RL interleaver tap (fractional referred to SH-Frames), only there is an integer delta of SH-Frames between them.

Modulators have to compensate the time interleaver delay between OFDM and TDM, if they are the branch with the lower delay. The modulator hast to delay then the stream by the difference between the longer and the shorter interleaver delay. This ensures the synchronous SH-Frame reception in the receiver.

For the description of the synchronization of the LL to the RL multiplex, this leads to two different solutions, one without delay compensation (1.), the other with delay compensation (2).

1. Without delay compensation:
When the LL multiplex is generated for a modulator without delay compensation, the Encapsulator generates the same SH-Framing for LL and RL multiplex. That results for the SHIP in the LL multiplex in the same synchronization time stamp (STS) as for the RL multiplex. Then, the rule for the "on_air_SH_frame_start" is adapted from the rules given for the RL in the clause 7.5):

$$\text{on\_air\_SH\_frame\_start}_{LL} = \text{on\_air\_SH\_frame\_start}_{SHIP} + D_{Tx,\,LL}[0]$$

(Unlike the RL stream, not the LL interleaver length $D_{ILV,\,LL}$ is added, but the length of the tap 0!)

If the STS and the maximum delay in the SHIP are then set identical for LL and RL (and SH-Frames are aligned in the Encapsulator), the LL SH-Frame n is perfectly aligned before the interleaver with the RL SH-Frame n-k, where k = $D_{Tx,\,RL}[0]$ div $L_{SH,\,IC}$, where div() is the integer division.

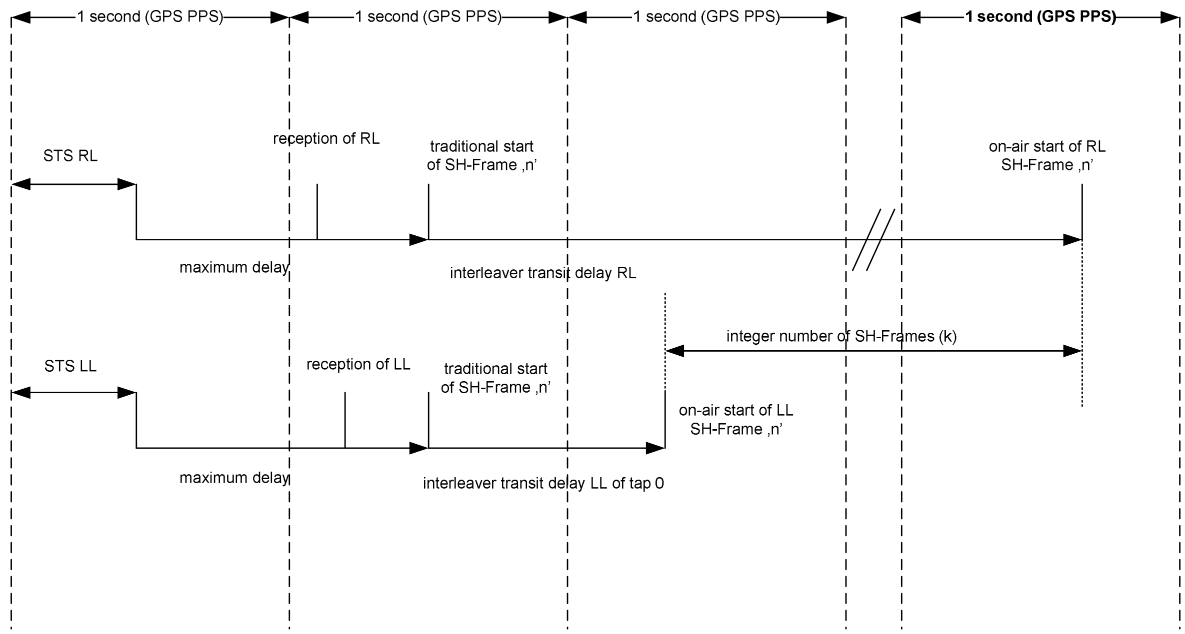


Figure D.3.1: Synchronization of RL and LL multiplex in the modulator without delay compensation

2. With delay compensation:

As the delay compensation is necessary to align the RL SH-Frames of OFDM and TDM in the receiver, it is not needed for the LL multiplex, because the content of ***SAT*** and ***TERR*** is not combinable and likely to be different. In the contrary, if the delay compensation is also added to the LL multiplex, this increases the latency of the LL. This latency is likely to be in the range of multiple SH-Frames, which is not acceptable for a low latency transmission.

So, in the following, if a modulator is processing also the LL multiplex, it adds the delay compensation only to the RL multiplex and it does this before the time interleaver. This implies that the solution 1 (clause 7.5.2.3.1) is not applicable for a LL modulator. Solution 2 (clause 7.5.2.3.2) is applicable, with the restriction, that the base for the LL interleaver calculation is still the non-extended time interleaver. The third solution (clause 7.5.2.3.3) is applicable without restrictions and allows the insertion of a LL multiplex with low latency.

The delay compensation d_{comp} is splitted to a fractional ($d_{comp,\,frac}$) and an integer part ($d_{comp,\,int}$) with respect to SH-Frames, so that:

$$d_{comp} = d_{comp,\,int} + d_{comp,\,frac}$$

where $d_{comp,\,int}$ is the equivalent to m (integer) SH-Frames.

The only difference to the processing without delay compensation is that the start of the LL SH-Frame is delayed in the encapsulator by $d_{comp,\,frac}$ with respect to the RL SH-Frame start. This leads to a synchronized situation of the SH-Frames of LL and RL after the delay compensation, only LL SH-Frame n is synchronous with RL SH-Frame n-m.

The actual maximum delay for LL and RL stays the same.

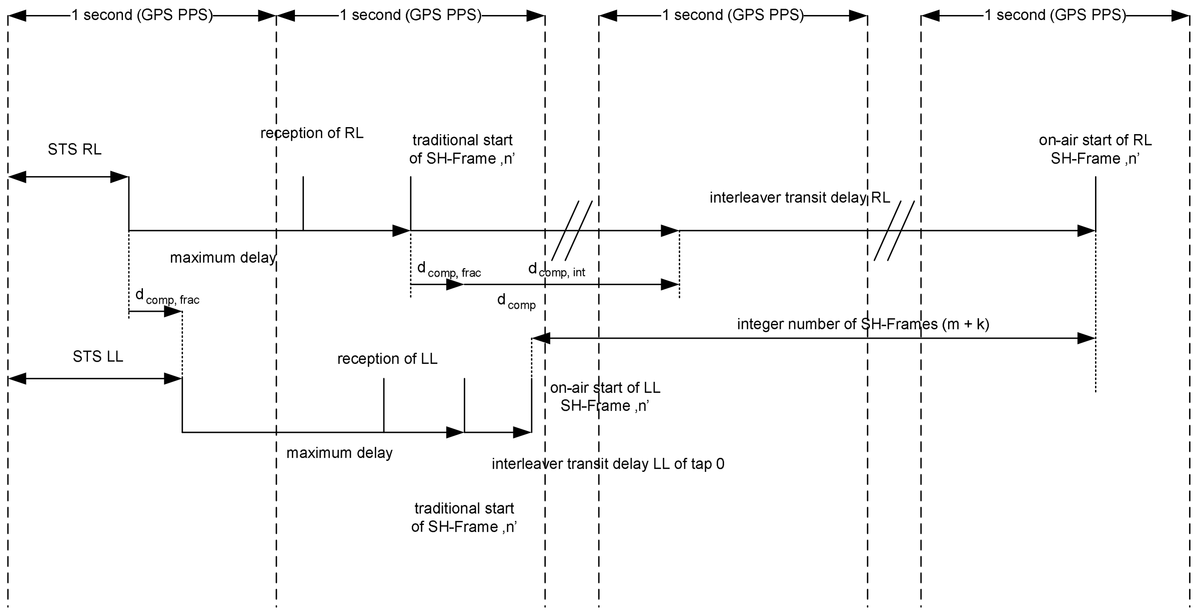


Figure D.3.2: Synchronization of LL and RL multiplex in the modulator with delay compensation

The two multiplexes, when processed in the above described way are synchronized at the interleaver input. The first IU of each frame is written synchronously in the concerning interleavers. The SH-Frame start is pointed out by the SHIP packet.

NOTE: The SH-Frames of RL and LL are not aligned in the Receiver after the interleavers, but that is not necessary. The delay of the LL – SH-Frame start behind the RL SH-Frame start at the output of the de-interleaver is $D_{ILV, LL} - D_{Tx, LL}[0] = D_{Rx, LL}[0]$

D.3.2 PSI/SI

For signalling the LL service the same mechanisms as for the local content are used. So only a small modification to the PSI/SI and their description is necessary.

One modification is the use of 4 reserved bits in the SH_delivery_system_descriptor to signal the presence and the puncturing pattern of the LL service.

Table D.3.4: Extended SH_delivery_system_descriptor for LL services

Syntax	Number of bits	Identifier
`SH_delivery_system_descriptor(){`		
`descriptor_tag`	8	uimsbf
`descriptor_length`	8	uimsbf
`descriptor_tag_extension`	8	uimsbf
`diversity_mode`	4	bslbf
`reserved`	4	bslbf
`for (i=0; i<N; i++){`		
`modulation_type`	1	bslbf
`interleaver_presence`	1	bslbf
`interleaver_type`	1	bslbf
`reserved`	1	bslbf
`ll_service_mode`	4	uimsbf
`if (modulation_type == 0) {`		
`Polarization`	2	bslbf
`roll_off`	2	bslbf
`modulation_mode`	2	bslbf
`code_rate`	4	bslbf
`symbol_rate`	5	bslbf
`scrambling_mode`	1	bslbf
`} else {`		
`bandwidth`	3	bslbf

Syntax	Number of bits	Identifier
priority	1	bslbf
constellation_and_hierarchy	3	bslbf
code_rate	4	bslbf
guard_interval	2	bslbf
transmission_mode	2	bslbf
common_frequency	1	bslbf
}		
if ((interleaver_presence == 1) {		
if (interleaver_type == 0) {		
common_multiplier	6	uimsbf
nof_late_taps	6	uimsbf
nof_slices	6	uimsbf
slice_distance	8	uimsbf
non_late_increments	6	uimsbf
} else {		
common_multiplier	6	uimsbf
reserved	2	uimsbf
}		
)		
}		
}		

ll_service_mode: If these bits are set to '0000' no LL service is present. In any other case this vector give the "code_rate + 1" (see existing tables). (If regular latency signaling is applied, the reserved bits are set to 0, so for receivers with low latency extension, no LL service would be signaled, which is a safe behavior).

NOTE: This descriptor does not contain any representation of the mux_assoc-vector (which means a description of the RL/LL burst structure). The vector can be acquired by the discovery algorithm described in clause D.3.3. If TDM modulation is used, the vector may be received via the Signalling Field (SF). If a fast access to the LL in OFDM mode is desired, the recommendations are given in the same clause.

D.3.3 LL Multiplex Description and Discovery Strategies

The LL multiplex is transmitted as separate MPEG-TS to the modulator. Each MPEG-TS (LL and RL) bears its own SHIP packet, which are referred as LL SHIP and RL SHIP in the following.

The ll_service_function is used in the modulator to control the LL multiplex path and in the receiver to allow the decoding of the LL multiplex. It gives the necessary parameters for an entry point to the LL multiplex. These parameters are available previous to decoding the LL multiplex. In TDM these parameters are also transmitted in the Signalling Field (SF). In OFDM they are NOT transmitted in the TPS. So this function is optional for the TDM RL SHIP and mandatory for the OFDM RL SHIP. It is also optional for the LL SHIP. It is RECOMMENDED to include the ll_service_function in the OFDM LL SHIP to enable fast access methods (the receiver needs to know, if the first service ID is LL or RL, this is described in the ll_service_function).

The first_cw parameter is an optional parameter. If not present, the LL SHIP is found in the first codeword of the SH-Frame.

If the service synchronization function is used in the LL SHIP, it gives the multiplex association for LL and RL service. The number of codewords is the number of EFRAMES in one SH-FRAME. If satellite and terrestrial transmission is used, the number of codewords either matches the number of EFRAMES of the shorter or the longer SH-FRAME. In the second case, the codewords up the length of the shorter SH-FRAME refer to the common content, the remaining to the local content. In the first case, all codewords refer to the common content.

For the SI the SH_delivery_system_descriptor is modified. 4 of the 5 reserved bits in the modulation loop are used to state the presence and the punct_pat_id. If the value is "0000", no LL is present in the described modulation. If unequal to zero, the value gives the punct_pat_id + 1 according to table 5.3 of [1].

LL discovery algorithm:

"cold start" – TDM: find the Low Latency parameter in the Signalling Field and use them to decode (they are complete, punct_pat_id and a representation of the LL multiplex association).

"cold start" - OFDM: decode the RL as normal, find the RL SHIP. Now the ll_punct_pat_id is known (ll_service_function is mandatory in OFDM RL SHIP). Either decode the first EFRAME or, if present, the start_cw EFRAME as a LL codeword and so find the LL SHIP with the complete description of the LL multiplex association).

"warm start" (PSI/SI known): For TDM the same as "cold start". For OFDM, use the given (SI) LL code rate and try to decode the first codeword of the SH-Frame to find the LL-SHIP. If successful, the LL multiplex association is known, if it fails, proceed like "cold start" - OFDM.

D.4 Re-multiplexing

There is also the possibility to change the LL data rate without stopping LL or RL service. During the data rate change, the data rates of both multiplexes may be decreased.

D.4.1 Processing

Data rate change in the multiplexes (in the sequel referred to as re-multiplexing) means a change of the multiplex association vector mux_assoc[n], n = 0,..,N_{CW} - 1.

A re-multiplexing event is announced by the wait_for_enable_flag either in the ll_service_function or the service_synchronization_function, whichever will change in the SHIP. If this bit is set to 1, the functions bears the now configuration. The new configuration is enabled by the enable_function in the SHIP. The new configuration gets valid at the start of the SH-Frame following the enabling of the function.

A new multiplex association vector can only become valid at the start of an SH frame. A re-multiplexing event at the start of SH frame k is actually a sequence of several phases, which are different for Encapsulator, Modulator and Receiver.

Modulator:

The modulator switches the multiplex association vector (and all its depended controls) at the start of the SH-Frame which follows the enabling of the new configuration. The SH-Frame start is considered at the input of the interleavers of the modulator. The processing in the modulator is according to [1], clause B.2.

Receiver:

For the receiver, this results in a switching at the SH-Frame boundary, which follows the enabling in the LL SHIP. The switching cannot be controlled by the RL SHIP, as this is fed through the regular interleaver, with possibly a delay of some SH-Frames. The receiver has to be aware, that the RL and LL SH-Frames starts may not be aligned any more in the Receiver.

Encapsulator:

The Encapsulator has to take care that no payload data is lost, while the modulator makes a hard switching. As the post-interleaver-multiplexer in the modulator is switched at the SH-Frame boundary, some of the data in the interleaver might be lost. To avoid loss of data, the encapsulator first needs to reduce the payload of the stream that is intended to be reduced, before, some time (SH-Frames) later, the other stream can increase its payload. The amount of guard time between payload reduction and increase depends on the interleaver length and on the kind of switching (RL to LL or vice versa).

The re-multiplexing processing of the encapsulator is defined in [21], clause 4.10.2.7.6.

The interleaver units (IUs) of one codeword are delayed between zero to the delay of the longest tap, the last being the interleaver delay. So the number of involved SH-Frames is given by the interleaver delay, rounded up to the next complete SH-Frame. For the last codeword of an SH-Frame the current plus the calculated number of SH-Frames are involved. So the interleaver influence relevant for the re-multiplexing is calculated by:

L = ceil(longest tap RL Interleaver in SH-frames)

The RL interleaver delay (L) is relevant for switching a codeword position from RL to LL, in this case the fading out of the interelaver has to be considered over L SH-frames.

For LL the interleaver delay is by design not larger than 1 SH-Frame, so the switching a codword from LL to RL is completed after 1 SH-Frame.

For RL, the parameter L is calculated separately for each transmission path, e.g. TDM and OFDM. The time relevant for the Encapsulator is the largest re-multiplexing length:

$$L = \max\{ L_{TDM}, L_{OFDM} \} \text{ (see note 2)},$$

where for TDM and OFDM the respective value is calculated by:

$$L_{TDM/OFDM} = \text{ceil}\,(D_{Rx,N,TDM/OFDM}[47] / L_{SH,\ IC,\ TDM/OFDM}),$$

and where ceil(x) denotes the smallest integer larger than or equal to x. Hence, $L_{TDM/OFDM}$ is the number of SH frames, over which the RL interleaver disperses the payload for TDM or OFDM, respectively, rounded upwards.

NOTE 1: Re-assigning a codeword from RL to LL (LL to RL) has to be considered as change of data-rate for this coderate, the appropriate rules above apply to this codeword.

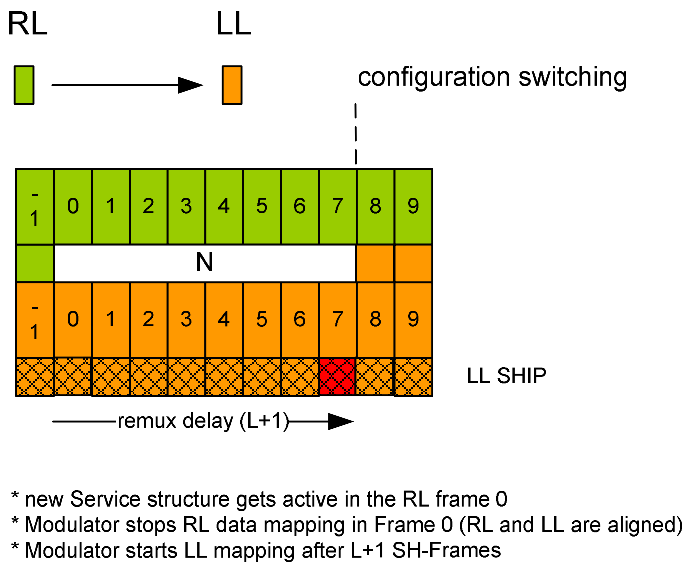
NOTE 2: The max() – operation ensures that also for systems with two different interleavers the longest interleaver delay is respected. So the interleaver delay compensation in the modulator with the shorter interleaver is automatically included.

D.4.2 Receiver Behaviour (informative)

NOTE: The modulator behaviour is described in [1], Annex B.

In the process of re-multiplexing the data rate, which can safely be used for payload is reduced.

If a codeword is switched from RL to LL, the MPEG TS will (for that codeword) contain only Null MPEG TS packets for a duration of L+1 SH-Frames before switching, as not all the interleaver units of these RL codewords will be transmitted. By chance, the receiver may be able to decode some of these codewords (depending on the used code rate). But then they will only contain Null MPEG-TS packets.



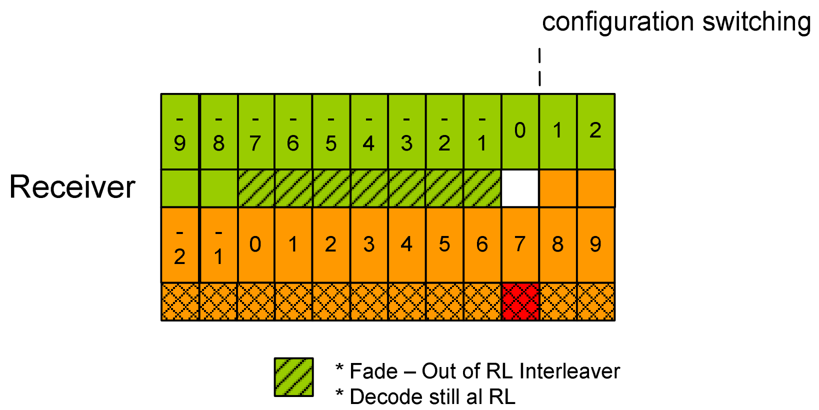


Figure D.4.1: Re-multiplexing a codeword from RL to LL

If two transmission paths (OFDM and TDM) are used, in the path with the shorter interleaver it may be possible to decode some RL codewords (containing Null MPEG TS packets). For some of these codewords, all interleaver units are transmitted, enabling correct decoding.

If a codeword is switched from LL to RL, it may be possible, that a receiver can decode some RL codewords before the re-multiplexing time L+1 is over, but these codewords contain only Null MPEG TS packets.

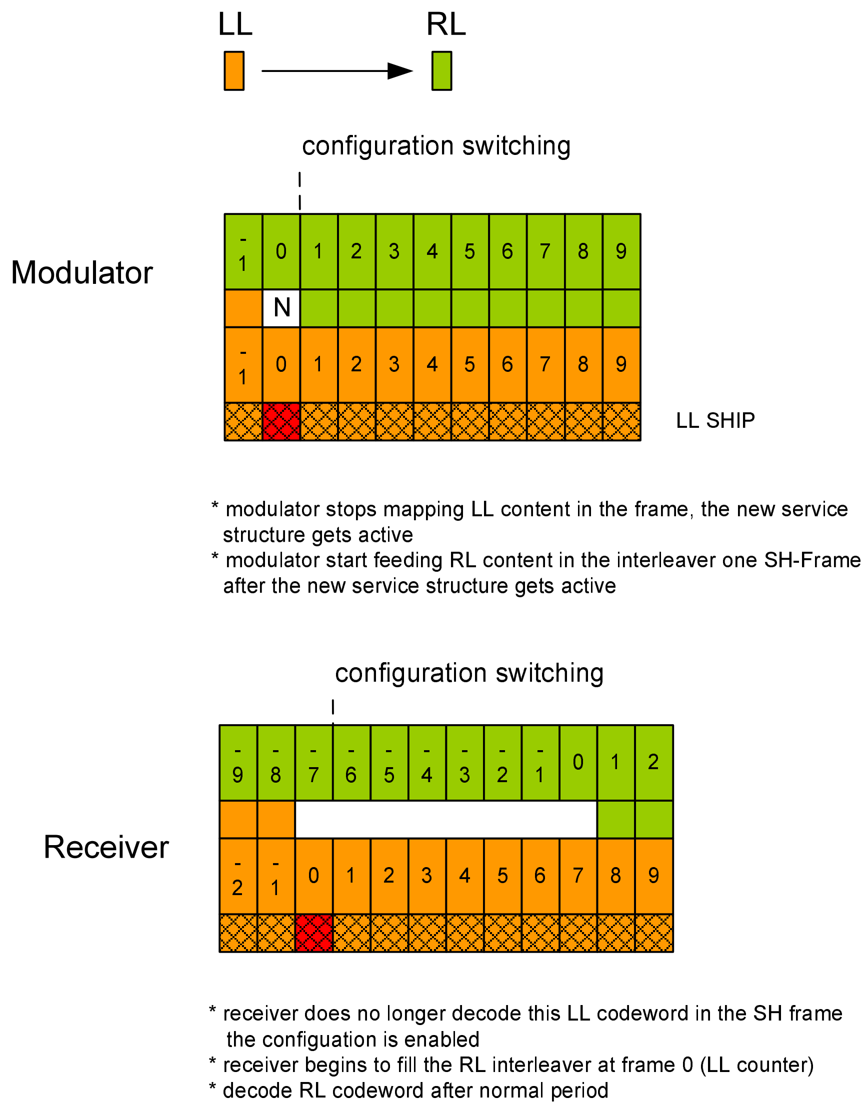


Figure D.4.2: Re-multiplexing a codeword from LL to RL

In principle this is the same for LL codewords in both switching directions.

It is important to check the CRC and PID of all incoming MPEG-TS packets for correct processing in the following receiver processing modules.

Summary:

As can be seen from Figure D.4.1 and Figure D.4.2, the receiver will decode RL/LL codewords according to the burst-description valid from the LL SHIP, or TDM-SF (valid one frame after enable_function). All codewords that are assigned to LL will be decoded as low-latency payload, all other codewords will be considered as regular-latency codewords. By this approach it is assured that all valid (filled) content (being regular- or low-latency) is decoded by the correct branch.

For clarification:

- Using the SHIP: the configuration is valid for the next SH frame start after the SH frame that contains the SHIP.
- Using the TDM SF: a current configuration (next_conf = 0) is valid for the current SH frame.

Annex E (informative): DVB-SH B Scrambling

E.1 Introduction

The revised DVB-SH standard [38], p. 33 is specifying that the physical layer frame of DVB-SH B is covered by a scrambling sequence similarly to DVB-S2 and based on the same type used in the W-CDMA 3GPP standard [15]. Such a programmable spreading sequence in DVB-S2 enables:

a) To ensure the randomization of the repeating I+jQ pilot symbols so that no discrete components appear in the transmitted signal spectrum.

b) To counteract co-channel and co-frequency interference, e.g. from other satellite beams (inter-beam interference) or from other antenna polarizations (intra-beam cross-polar interference).

c) To randomize narrowband co-channel interference, by spreading it over a wider band. The demodulator performance will be therefore independent from the central frequency of the narrow-band interference.

d) To allow a carrier unique signature to ensure easy separation between different carriers or operators.

The previous version of DVB- SH standard [1] assumed as for DVB-S2 when used in broadcasting mode that only the complex scrambling sequence number 0 needs to be implemented. This specified lack of the scrambling sequence programmability leads to the impossibility of de-correlate different carriers transmitted from different beam and/or polarizations, with an impact on points b) and d). As shown in the following, the former DVB-SH B lack of scrambling sequence programmability was leading to performance degradation in a multi-beam satellite with frequency/polarizations reuse.

Likewise DVB-S2 standard, DVB-SH has been recently extended [38] to allow generating a large set of complex scrambling sequences (although of smaller size than the DVB-S2 one to ease the signalling of this information). The use of different scrambling sequences in the different satellite beams/polarisations is avoiding the possible channel estimation degradation due to the cross-talk between the different carriers.

A more detailed explanation of this phenomenon is provided in clauses E.2 and E.3.

E.2 Multi-beam Channel Estimation Analysis

This clause analyze the effect of co-frequency and co-polar inter-beam interference (or alternatively intra-beam cross-polar interference) to exemplify the concept. The simple case of two distinct channels is used to simplify the notation. It is assumed the symbol timing of the different co-channel carriers to be aligned. This is a likely situation when the modulators for the different carriers are collocated from a common time reference.

E.2.1 DVB-SH B without Extension

This clause refers to the case where only one complex scrambling sequence is implemented [1]. Under the assumptions reported in clause E.2, in case of two co-frequency channels using the same scrambling sequences the received signal complex envelope is:

$$\tilde{r}(t) = A_1 \exp\left[j\left(\omega_1 - \omega_0\right)t + \varphi_1(t)\right] \sum_{k=-\infty}^{k=+\infty} \left[p_1(k) + jp_1(k)\right]\left[C_I(k) + jC_Q(k)\right] g_T(t - kT - \tau_1) +$$

$$+ A_2 \exp\left[j\left(\omega_2 - \omega_0\right)t + \varphi_2(t)\right] \sum_{k=-\infty}^{k=+\infty} \left[p_2(k) + jq_2(k)\right]\left[C_I(k) + jC_Q(k)\right] g_T(t - kT - \tau_2) + \tilde{n}(t),$$

where ω_0 is the nominal carrier frequency, A_m, ω_m, φ_m, τ_m are the amplitude, carrier frequency, carrier phase and group delay of the *m*-th modulated DVB-SH carrier related to different RF transmission chain, $p_m(k)$ and $q_m(k)$ are the m-th modulated DVB –SH carrier I-Q APSK symbols, $C_I(k)+jC_Q(k)$ is the DVB-SH "unique" unit power complex scrambling sequence [38] T is the symbol duration, $g_T(t)$ is the modulator square-root raised-cosine impulse response and $\tilde{n}(t)$ is the complex AWGN process. The difference in carrier frequency, phase and group delay of the two distinct DVB-SH signals considered is due to the fact that they are up-linked from the gateway to the satellite on different frequencies and going through different satellite transponders before being transmitted towards the user.

Assuming that the group delay difference between the two channels is negligible compared to the symbol duration i.e. $\left(\tau_1-\tau_2\right)/T \ll 1$ so that we can assume that $\tau_1 \approx \tau_2$, that the two carrier frequencies are close to the nominal value ω_0 so that $\omega_1 \approx \omega_2 \approx \omega_0$ (see note), ideal timing recovery after the symbol matched filter of the demodulator, then one gets for the *i*-th symbol matched filter sample at time $iT+\tau_1$:

$$\tilde{s}_{SMF}(i)=\tilde{r}(t)\otimes g_R(t)\Big|_{t=iT+\tau_1}=A_1\exp\left[j\varphi_1\left(iT+\tau_1\right)\right]\left[p_1(i)+jq_1(i)\right]\left[C_I(i)+jC_Q(i)\right]+$$
$$+A_2\exp\left[j\varphi_2\left(iT+\tau_1\right)\right]\left[p_2(i)+jq_2(i)\right]\left[C_I(i)+jC_Q(i)\right]+\tilde{n}(i),$$

where $g_T(t)$ is the demodulator square-root raised-cosine impulse response and $\tilde{n}(i)=\tilde{n}(t)\otimes g_R(t)\Big|_{t=iT+\tau_1}$ is the complex noise sample at the symbol matched filter output.

NOTE: Residual small frequency offset can be included in the time variant carrier phase.

Assuming that the start of the physical layer frames of the two carriers generated at the same gateway location is time aligned, it follows that also the pilot symbols locations will be aligned in time at the demodulator thus being the pilot sequence the same for the two carriers $p_1(i)=p_2(i),\ q_1(i)=q_2(i)$. Thus for the demodulated pilot symbols $\tilde{s}_{SMF-P}(i)$ we get:

$$\tilde{s}_{SMF-P}(i)=\left\{A_1\exp\left[j\varphi_1\left(iT+\tau_1\right)\right]+A_2\exp\left[j\varphi_2\left(iT+\tau_1\right)\right]\right\}$$
$$\cdot\left[p_1(i)+jq_1(i)\right]\left[C_I(i)+jC_Q(i)\right]+\tilde{n}(i).$$

The last step for pilot-aided channel estimation requires removing the known pilot and scrambling sequence. This is achieved as:

$$\tilde{p}(i)=\tilde{s}_{SMF-P}(i)\cdot\frac{1}{\sqrt{2}}\left\{\left[p_1(i)+jq_1(i)\right]\left[C_I(i)+jC_Q(i)\right]\right\}^*$$
$$=\left\{A_1\exp\left[j\varphi_1\left(iT+\tau_1\right)\right]+A_2\exp\left[j\varphi_2\left(iT+\tau_1\right)\right]\right\}+\tilde{n}_p(i),$$

where:

$$\tilde{n}_p(i)=\tilde{n}(i)\frac{1}{\sqrt{2}}\left\{\left[p_1(i)+jq_1(i)\right]\left[C_I(i)+jC_Q(i)\right]\right\}^*.$$

The estimated pilot phase will be given by:

$$\hat{\vartheta}_P(i)=\arg\left\{A_1\exp\left[j\varphi_1\left(iT+\tau_1\right)\right]+A_2\exp\left[j\varphi_2\left(iT+\tau_1\right)\right]+\tilde{n}_p(i)\right\}$$

Instead of the true phase value:

$$\vartheta_P(i) = \arg\left\{A_1 \exp\left[\, j\varphi_1\left(iT+\tau_1\,\right)\right]\right\}$$

The cross-talk in the channel estimation is clearly visible in the above equation. To be remarked that typically the pilot phase is estimated not over a single pilot symbol but over the DVB-SH field of 80 symbols to reduce the thermal noise effect thus:

$$\hat{\vartheta}_P = \arg\left\{\sum_{i=1}^{80} \underbrace{A_1 \exp\left[\, j\varphi_1\left(iT+\tau_1\,\right)\right]}_{\text{useful signal component}} + \underbrace{A_2 \exp\left[\, j\varphi_2\left(iT+\tau_1\,\right)\right]}_{\text{co-frequency interference}} + \underbrace{\tilde{n}_p(i)}_{\text{AWGN}}\right\}$$

It has to be remarked that the problem mentioned above is also affecting the DVB-SHA standard whereby there is no possibility to apply a scrambling sequence decorrelating the co-channel interfering carriers. Instead as discussed in the following section for DVB-SH B a simple extension of the current standard will allow to solve the problem.

E.2.2 DVB-SH B with Extension

This clause refers to the case where different complex scrambling sequences are used per carrier as from the most recent version of the DVB-SH standard [38]. Under the assumptions reported in clause E.2, in case of two co-frequency channels using different scrambling sequences the received signal complex envelope is:

$$\begin{aligned}\tilde{r}(t) = A_1 \exp\left[\, j\left(\omega_1-\omega_0\right)t+\varphi_1(t)\right]\sum_{k=-\infty}^{k=+\infty}\left[\,p_1(k)+jp_1(k)\right]\left[C_{I,1}(k)+jC_{Q,1}(k)\right]g_T(t-kT-\tau_1\,)+\\ +A_2 \exp\left[\, j\left(\omega_2-\omega_0\right)t+\varphi_2(t)\right]\sum_{k=-\infty}^{k=+\infty}\left[\,p_2(k)+jq_2(k)\right]\left[C_{I,2}(k)+jC_{Q,2}(k)\right]g_T(t-kT-\tau_2\,)+\tilde{n}(t),\end{aligned}$$

where ω_0 is the nominal carrier frequency, A_m, ω_m, φ_m, τ_m are the amplitude, carrier frequency, carrier phase and group delay of the *m*-th modulated DVB-SH carrier, $p_m(k)$ and $q_m(k)$ are the *m*-th modulated DVB –SH carrier I-Q APSK symbols, $C_{I,m}(k)+jC_{Q,m}(k)$ is the unit power complex DVB-SH *m*-th carrier specific scrambling sequence [1], p. 33, T is the symbol duration, $g_T(t)$ is the modulator square-root raised-cosine impulse response and $\tilde{n}(t)$ is the complex AWGN process.

Assuming that the group delay difference between the two channels is negligible compared to the symbol duration i.e. $\left(\tau_1-\tau_2\right)/T \quad 1$ so that we can assume that $\tau_1 \approx \tau_2$, that the two carrier frequencies are close to the nominal value ω_0 so that $\omega_1 \approx \omega_2 \approx \omega_0$ (seenote) ideal timing recovery after the symbol matched filter of the demodulator, then one gets for the *i*-th symbol matched filter sample at time $iT+\tau_1$:

$$\begin{aligned}\tilde{s}_{SMF}(i) = \tilde{r}(t)\otimes g_R(t)\Big|_{t=iT+\tau_1} = A_1 \exp\left[\, j\varphi_1\left(iT+\tau_1\,\right)\right]\left[\,p_1(i)+jq_1(i)\right]\left[C_{I,1}(i)+jC_{Q,1}(i)\right]+\\ +A_2 \exp\left[\, j\varphi_2\left(iT+\tau_1\,\right)\right]\left[\,p_2(i)+jq_2(i)\right]\left[C_{I,2}(i)+jC_{Q,2}(i)\right]+\tilde{n}(i),\end{aligned}$$

where $g_T(t)$ is the demodulator square-root raised-cosine impulse response, and $\tilde{n}(i) = \tilde{n}(t)\otimes g_R(t)\Big|_{t=iT+\tau_1}$ is the complex noise sample at the symbol matched filter output.

NOTE: Residual small frequency offset can be included in the time variant carrier phase.

Assuming that the start of the physical layer frames of the two carriers generated at the same gateway location is time aligned, it follows that also the pilot symbols locations will be aligned in time at the demodulator. Thus for the demodulated pilot symbols $\tilde{s}_{SMF-P}(i)$ we get:

$$\tilde{s}_{SMF-P}(i) = \left\{ A_1 \left[C_{I,1}(i) + jC_{Q,1}(i) \right] \exp\left[j\varphi_1 \left(iT + \tau_1 \right) \right] + A_2 \left[C_{I,2}(i) + jC_{Q,2}(i) \right] \exp\left[j\varphi_2 \left(iT + \tau_1 \right) \right] \right\} \cdot \left[p_1(i) + jq_1(i) \right] + \tilde{n}(i).$$

Compared to the current DVB-SH B approach now the two contributions from the two distinct carriers remain separate thanks to the adoption of different scrambling sequences.

The last step for pilot-aided channel estimation requires removing the known pilot and scrambling sequence. This is achieved as:

$$\tilde{p}(i) = \tilde{s}_{SMF-P}(i) \cdot \frac{1}{\sqrt{2}} \left\{ \left[p_1(i) + jq_1(i) \right] \left[C_{I,1}(i) + jC_{Q,1}(i) \right] \right\}^*$$
$$= \left\{ A_1 \exp\left[j\varphi_1 \left(iT + \tau_1 \right) \right] + A_2 \left[C_{I,1}(i) - jC_{Q,1}(i) \right] \left[C_{I,2}(i) + jC_{Q,2}(i) \right] \exp\left[j\varphi_2 \left(iT + \tau_1 \right) \right] \right\} + \tilde{n}_p(i),$$

where:

$$\tilde{n}_p(i) = \tilde{n}(i) \frac{1}{\sqrt{2}} \left\{ \left[p_1(i) + jq_1(i) \right] \left[C_{I,1}(i) + jC_{Q,1}(i) \right] \right\}^* .$$

The estimated pilot phase will be given by:

$$\hat{\vartheta}_P(i) = \arg\left\{ A_1 \exp\left[j\varphi_1 \left(iT + \tau_1 \right) \right] + A_2 \left[C_{I,1}(i) - jC_{Q,1}(i) \right] \left[C_{I,2}(i) + jC_{Q,2}(i) \right] \exp\left[j\varphi_2 \left(iT + \tau_1 \right) \right] + \tilde{n}_p(i) \right\}$$

Instead of the true phase value:

$$\vartheta_P(i) = \arg\left\{ A_1 \exp\left[j\varphi_1 \left(iT + \tau_1 \right) \right] \right\}$$

The cross-talk in the channel estimation is clearly visible in the above equation. To be remarked that typically the pilot phase is estimated not over a single pilot symbol but over the DVB-SH field of 80 symbols to reduce the thermal noise effect thus:

$$\hat{\vartheta}_P = \arg\left\{ \sum_{i=1}^{80} A_1 \exp\left[j\varphi_1 \left(iT + \tau_1 \right) \right] + A_2 \left[C_{I,1}(i) - jC_{Q,1}(i) \right] \left[C_{I,2}(i) + jC_{Q,2}(i) \right] \exp\left[j\varphi_2 \left(iT + \tau_1 \right) \right] + \tilde{n}_p(i) \right\}$$
$$\approx \arg\left\{ \sum_{i=1}^{80} A_1 \exp\left[j\varphi_1 \left(iT + \tau_1 \right) \right] + \tilde{n}_p(i) \right\}$$

having considered that thanks to the different scrambling sequences adopted for the different carriers:

$$\sum_{i=1}^{80} A_2 \left[C_{I,1}(i) - jC_{Q,1}(i) \right] \left[C_{I,2}(i) + jC_{Q,2}(i) \right] \exp\left[j\varphi_2 \left(iT + \tau_1 \right) \right] \ll A_1,$$

This corresponds to the wanted result for having an accurate channel estimation of each DVB-SH B carrier also in the presence of aggressive frequency reuse scheme.

E.3 Multi-beam Channel Estimation Performance Impact

In this clause the need for a programmable complex scrambling sequence is substantiated: two examples of interference source in multi-beam system are provided in the following, whereby (study case 1) frequency is reused in other beam with the same antenna polarization; or (study case 2) interference is coming from the opposite polarization from other beams or the same beam. In practice, real systems may adopt a combination of the two configurations described above.

E.3.1 Study Case 1 - Single Polarisation Multi-beam System

The first study case corresponds to a multi-beam system with a given colour scheme which is obtained reusing the frequency over the coverage region. The polarization is not reused within the same or other beams.

Simulations follow the scheme as shown in Figure E.1 different levels of C/I have been considered to emulate the user experience corresponding to different user locations within the beam and/or different frequency reuse schemes and/or satellite antenna patterns. Only one DVB-SH co-channel interfering signal has been considered. The useful and the interfering signals are both QPSK modulated with rate ½ FEC. The interfering signal start of frame is time aligned with the useful signal but with a small carrier frequency offset ($\Delta f \cdot T = 10^{-5}$, see clause E.2); this has also the effect of ensuring that the carrier phase is randomized over the simulation time. Being the interfering signal coming from the same satellite with the same antenna polarization it will experience the same fading as the useful signal. For the simulations we used the DVB-SH IG LMS Fontan's channel model [16] and [17]. The DVB-SH receiver includes the channel estimation described in clause A.11.

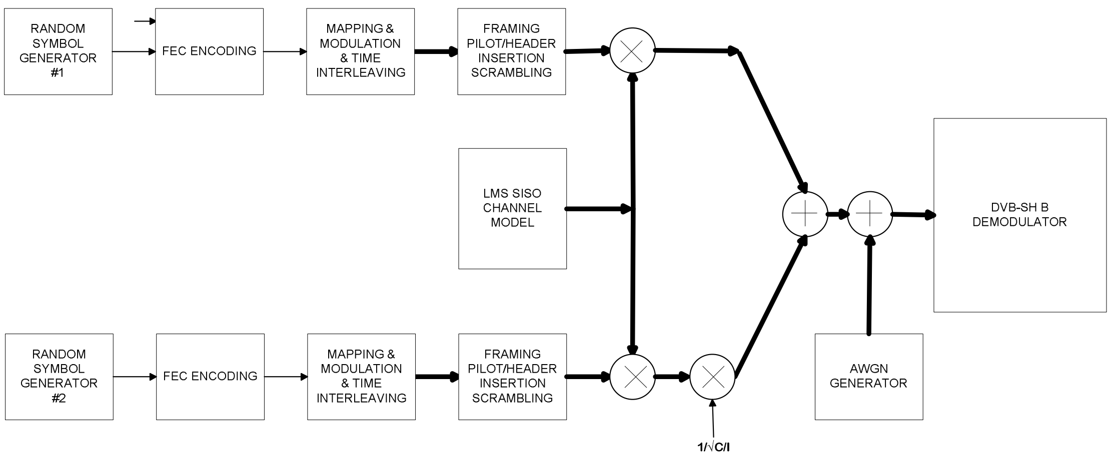


Figure E.1: DVB-SH B study case 1 simulator schematic

The simulation results are shown in Figure E.2. It can be seen that the impact of the co-channel interference on the useful channel BER/FER remains moderate i.e. less than 1 dB until C/I ≥ 10 dB.

In this case, the channel estimation error impact is also negligible i.e. less than 0,1 dB for the worst-case corresponding to C/I=12 dB. Although the pilots of the two carriers are not isolated because they use the same scrambling sequence, the interfering signal amplitude is fully correlated in terms of fading and shadowing with the useful signal. This may lead to a maximum (see note) useful signal phase estimation error caused by the inter-beam interference as reported in Table E.1.

Table E.1: Backwards Compatibility Matrix (Transmitter Side)

C/I (dB)	Maximum phase estimation error (degrees)
16	9,0
13	12,6
10	17,6
6	26,6

It is to note that a larger impact is expected as the spectral efficiency of the signal is increasing and in particular higher order modulations are considered.

NOTE: This situation occurs when the interferer carrier phase is orthogonal to the useful one.

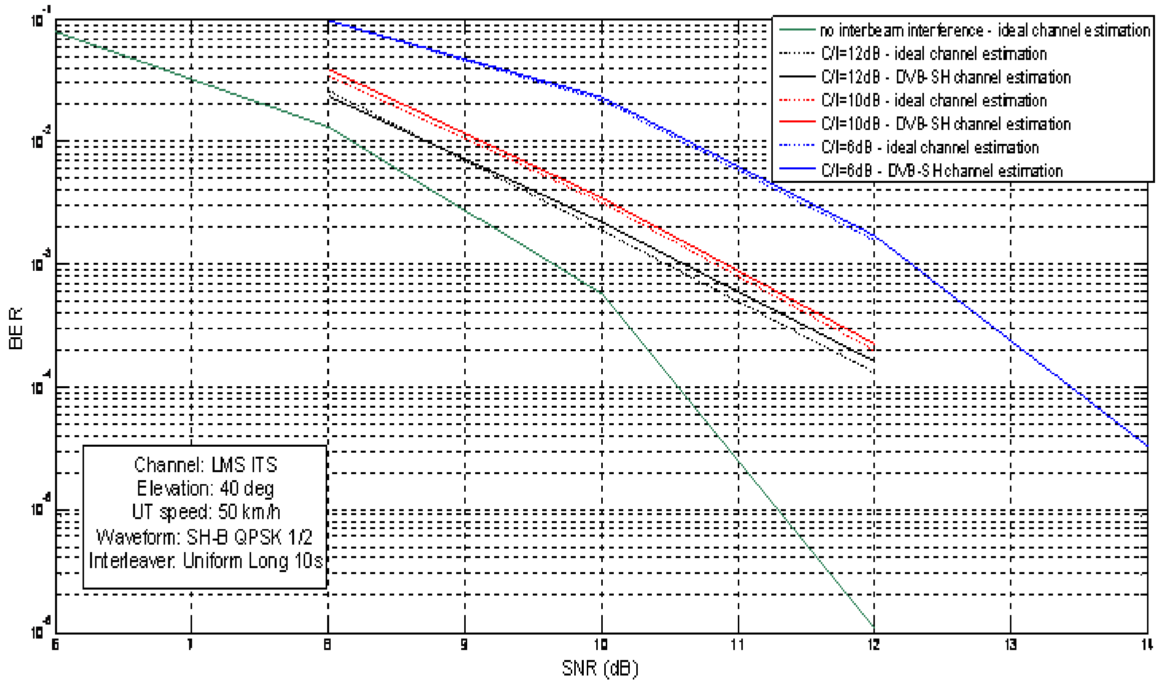


Figure E.2: DVB-SH B study case 1 simulation results

E.3.2 Study Case 2 - Dual Polarisation Multi-beam System

This second study case corresponds to a multi-beam system with a given colour scheme which is obtained reusing the polarization over the different beams or over the same beam. This means that near beams reusing the same frequency band are separated using a different antenna polarisation instead of a different frequency. In satellite multi-beam systems a combination of spatial (case 1) and polarisation isolation (case 2) is often used in practice.

The simulations shown in Figure E.3 we assume C/I =12 dB as reference level of cross-polar experienced by the users. As for study case 1, only one DVB-SH co-channel interferer has been considered. The useful and interfering signals are QPSK modulated with rate ½ FEC. The interferer start of frame is time aligned with the useful signal but has a small carrier frequency offset ($\Delta f \cdot T = 10^{-5}$, see clause E.2); this has also the effect of ensuring that the carrier phase is randomized over the simulation time. Being the interferer coming from the same satellite but with a different antenna polarization *it will not experience the same fading* as the useful signal. This is because based on experimental evidence it is assumed that there is a good degree of decorrelation between the fading (see note) experienced in the two different satellite polarizations. For the simulations we used a customised version of the satellite dual polarization MIMO channel. Being the system of interest a Multiple Input Single Output (2x1 MISO) only one output is considered. As for study case 1, the DVB-SH receiver includes the channel estimation described in in clause A.11.

NOTE: The shadowing processes will instead be highly correlated between the two polarizations.

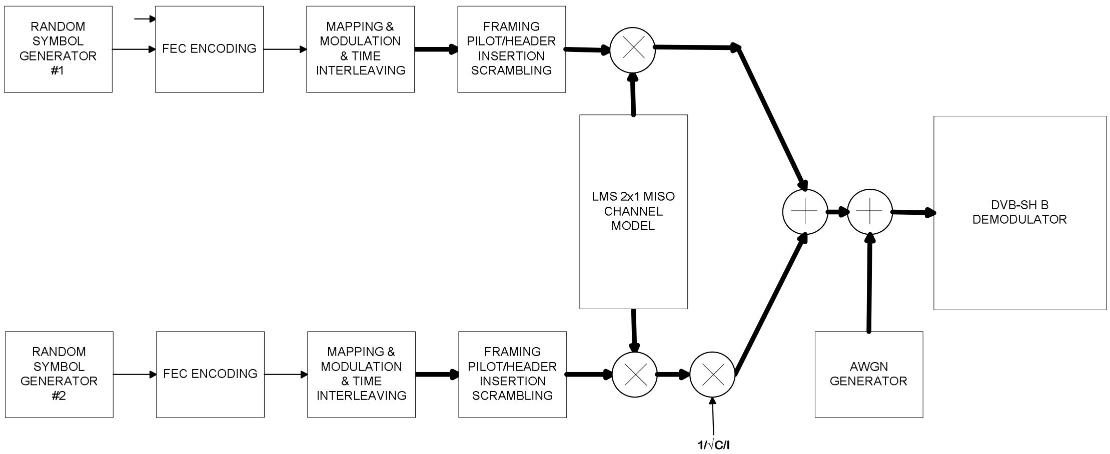


Figure E.3: DVB-SH B study case 2 simulator schematic

The simulation results are shown in Figure E.4. It can be seen that the impact of the co-channel interference on the useful channel for the assumed satellite cross-polar C/I=12 dB, the BER/FER degradation compared to the absence of cross-polar remains negligible if channel estimation is ideal. In case of channel estimation with a single scrambling sequence as currently specified in DVB-SH B the loss of performance is quite large (almost 2 dB at BER=1E-2). Instead using different scrambling sequences as recommended for multi-beam systems the performance loss with real channel estimation compared to the ideal channel estimation becomes negligible. To explain the increased performance loss compared to the study case 1, one may consider that in this case the useful and interfering signals are affected by *quasi independent* "locally" Rice fading processes which may increase the instantaneous phase and amplitude error of the useful signal pilot symbols. This is a much worst condition than the one experienced in the study case 1 whereby the useful and interfering signals undergo through the same fading process.

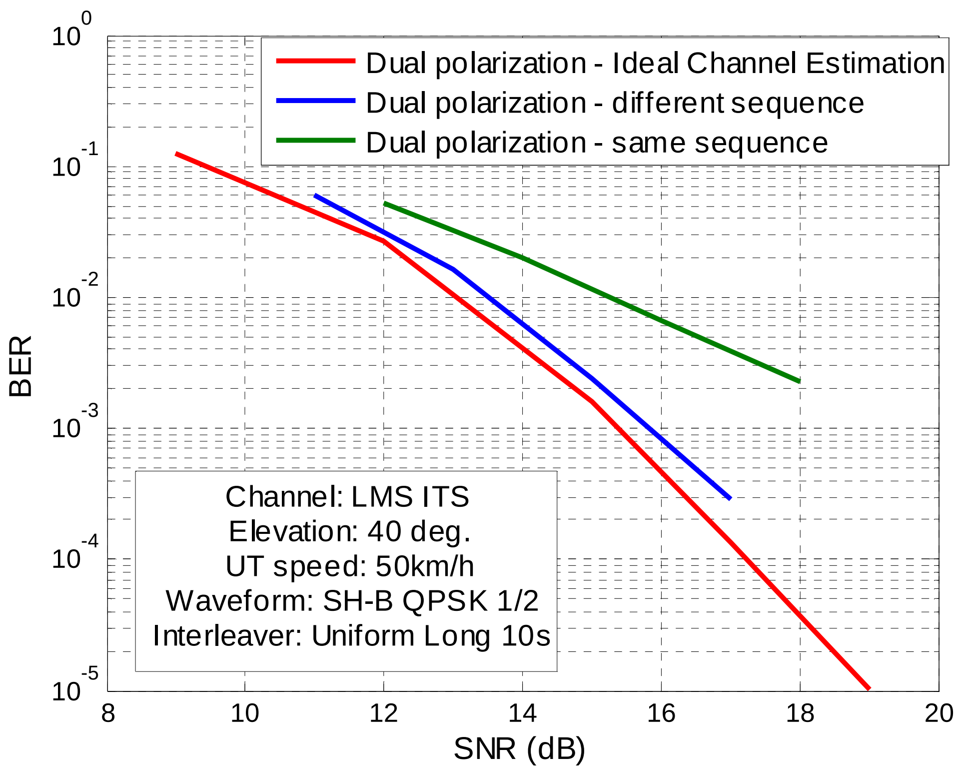


Figure E.4: DVB-SH B study case 2 simulation results

E.4 DVB-SH B Scrambling Sequences Practical Guide

E.4.1 Selection of the SH B Scrambling Sequence

Following the previous discussion and the DVB-SH B scrambling sequence programmability it is proposed to use different scrambling sequences for different co-frequency satellite beams and in particular when different polarisations are used to discriminate the beams. The use of a single scrambling sequence is recommended only in case of single beam satellite systems. In this case the scrambling sequence n = 0 is used.

E.4.2 Signalling of the SH B Scrambling Sequence

To help the demodulator initial acquisition it has been agreed to limit the number of possible scrambling sequences to 256. The SH B possible range of scrambling parameter n is 0 to 255 [38], p. 33. In this waveform specification the use of shifted sequences, as they could occur in DVB-S2, is forbidden.

In case of the presence of the OFDM terrestrial component in addition to the TDM SH-B satellite component the TDM parameters including the SH B scrambling sequence can be derived from the DVB-SH PSI/SI delivery *system_descriptor* signalling information [24]. More precisely there the parameter "scrambling_mode" states a scrambling parameter unequal to "0" and couples the exact value to the cell_id and polarization [24], Table 122.

In case of TDM only operations the TDM scrambling sequence parameter n is included in the SHIP function within the tdm_auxiliary_function [38], clause A.4.13, in order to let modulators know which value they should use (this value is derived form PSI/SI and inserted into the SHIP by the encapsulator.

History

Document history		
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