



**Digital Video Broadcasting (DVB);
Interaction channel for Satellite Distribution
Systems; Guidelines for the use of EN 301 790 in
mobile scenarios**

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Foreword

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

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

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Founded in September 1993, the DVB Project is a market led consortium of public and private sector organizations in the television industry. Its aim is to establish the framework for the introduction of MPEG 2 based digital television services. Now comprising over 200 organizations from more than 25 countries around the world, DVB fosters market led systems, which meet the real needs, and economic circumstances, of the consumer electronics and the broadcast industry

Introduction

The present document gives guidelines for the implementation of Digital Video Broadcasting (DVB) interaction channel for Satellite Distribution System (also known as DVB RCS: DVB Return Channel via Satellite) in mobile scenarios.

DVB RCS specification [i.1] describes several mandatory and optional features for geostationary satellite interactive system in mobile scenarios. Annex L of [i.3] already defines guidelines for some mobility scenarios characterised by Line-Of-Sight conditions with a channel that can be characterised as AWGN. This present document draws attention to the technical questions that need to be answered in setting up a DVB RCS network that supports mobility in Line of Sight and Non Line of Sight scenarios in different channel conditions and offers some guidance in finding answers to

them. This technical report considers the DVB-S2 standard [i.2] for the forward link transmission with the features to support mobility described in the DVB-RCS specification.

Outline of the present document

The present document provides some examples of implementation details related either with the physical layer (e.g. spreading, frame length, link budget), the link layer (MPE-FEC) or the medium access control layer (e.g. use of continuous carrier operation versus MF-TDMA, handover).

In order to ease the use of the present document, it shall be noted that from up to clause 8, the clause numbering is similar to the one used in [i.1]. Clause 9 maps with clause 10 in [i.1] and clause 11 presents performance results based on simulations

The present document only covers the extension to systems operating in mobile scenarios, guidelines for fixed scenarios are found in [i.3].

1 Scope

The present document should be read in conjunction with the normative document [i.1] in order to assist network operators, systems integrators, and equipment manufacturers in the realization of satellite based interactive services in mobile scenarios.

The present document should be interpreted as recommendations or good practices, but not as mandatory requirements. It is anticipated, however, that future procurement documents may reference elements of the present document as part of their system specification.

The present document is applicable to satellite systems as defined in [i.1] that target the provision of interactive services in mobile scenarios (maritime, aeronautical, railway and land-vehicular). In such a system, the RCSTs receive a Forward Link signal based on the DVB-S2 [i.2] specifications with the mobile extensions defined in [i.1].

The system as defined in [i.1] may be used in all frequency bands allocated to FSS or BSS services.

2 References

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

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

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

2.1 Normative references

Not applicable.

2.2 Informative references

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3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions in EN 301 790 [i.1] apply.

3.2 Symbols

For the purposes of the present document, the symbols given in EN 301 790 [i.1] and the following apply:

Δf	Differential Frequency Offset
E_s/N_0	Energy per symbol per Noise Spectral Density
P_{fa}	Probability of False Alarm
P_{md}	Probability of Missed Detection
r	Code rate
s	Seconds
sqrt	Square Root

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in EN 301 790 [i.1] and the following apply:

AES	Aeronautical Earth Station
AMSS	Aeronautical Mobile Satellite Service
BW	Bandwidth
CAC	Connection Admission Control
D-GPDI	Differential Generalised Post Detection Integration
DAMA	Demand Assigned Multiple Access
ECN	Explicit Congestion Notification
FEA	Functional Entity Actions
FIP	Forward Interaction Path
FL	Forward Link
FLSS	Forward Link Subsystem
FMT	Fade Mitigation Technique
FS	Fixed Service
GPDI	Generalised Post Detection Integration
GS	Generic Stream
GSE	Generic Stream Encapsulation
HO	Hand Over
IBR	In band Request
IMUX	Input Multiplexer
LOS	Line-of-Sight
MAT	Mean Acquisition Time
M&C	Management and Control
ML	Maximum Likelihood
MMT	Multicast Mapping Table
MODCOD	Modulation and Coding scheme
MPE	Multi Protocol Encapsulation
MSL	Minimum Scheduling Latency
MSS	Mobile Satellite System
MTBL	Maximum Tolerable Burst Length
MTU	Maximum Transmission Unit
NCF	Network Control Facility
NCPDI	Non Coherent Post Detection Integration
NMS	Network Management Subsystem
NOC	Network Operation Center

OBR	Out of Band request
OMUX	Output Multiplexer
PEP	Performance Enhancing Proxies
PLR	Packet Loss Rate
PDI	Post Detection Integration
PFD	Power Flux Density
PSD	Power Spectral Density
QoS	Quality of Service
RADIUS	Remote Authentication Dial-In User Service
RAS	Radio Astronomy Service
RLSS	Return Link Subsystem
ROC	Receiver Operating Characteristics
SACK	Selective Acknowledgement
SF	Spreading Factor
SL	Slope Ratio
SLA	Service Level Agreement
SYNC	Synchronization burst
TC	Threshold Crossing
TCP	Transmission Control Protocol
TS	Transport Stream
TWTA	Travelling Wave Tube Amplifier
VSAT	Very Small Aperture Terminal

4 Reference model

The Reference Model for an interactive satellite network for the mobile scenario, depicted in [i.1], includes all interconnections among Network Control Centre, Traffic Gateway(s), Feeder(s) and Terminals, which are possible from a functional viewpoint as well as all the possible mobile scenarios.

In practice, not all systems address all the mobile scenarios. This clause describes therefore the mobile scenarios that are more likely to be implemented for the service provision and their associated architecture.

The reference mobile DVB-RCS system architecture, envisaged in [i.1], is depicted in Figure 1 and described below.

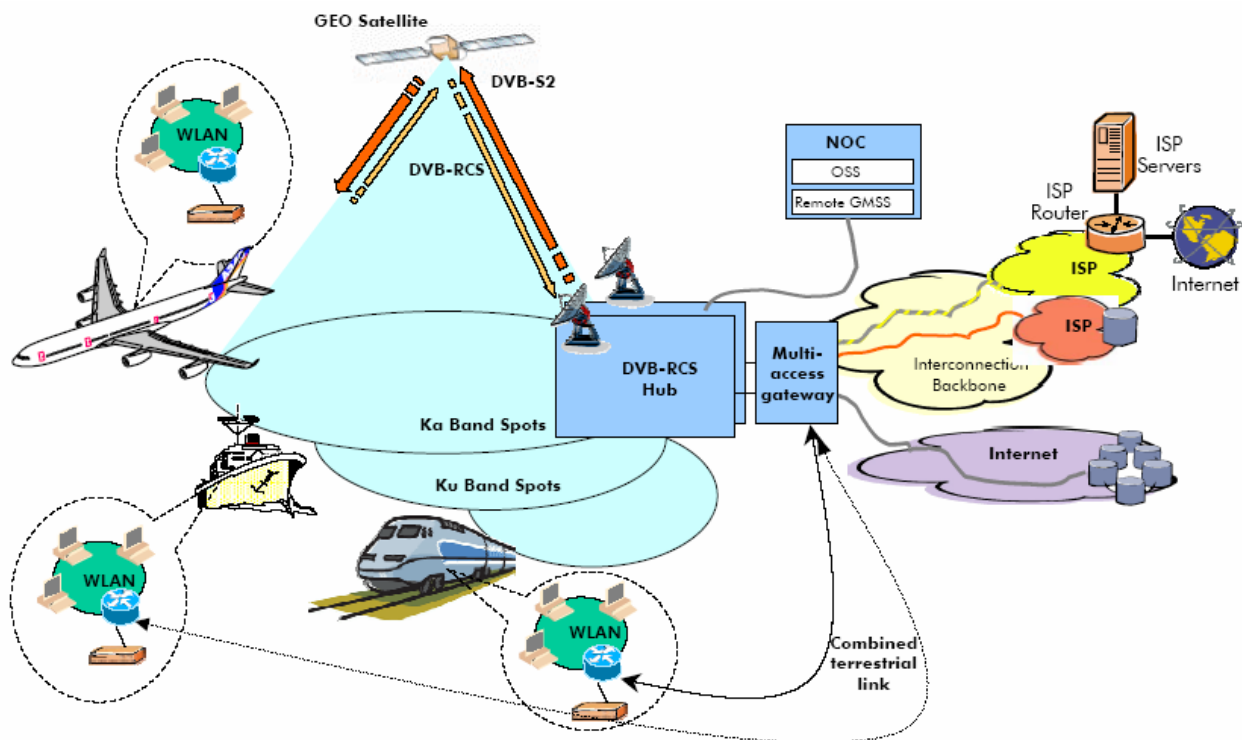


Figure 1: Overall system architecture for mobile interactive services via satellite

The space segment includes one or more GEO-satellites with a single or multi-beam configuration per satellite and with performances equivalent to those used for classical Ku-band or Ka-band FSS services. Two service coverage scenarios are identified: one for regional case (one country or part of continent covered); and one for global case (land coverage complemented by oceanic coverage, e.g. transatlantic).

On the terminal segment, the terminal is in most cases mounted on a mobile platform operating as an access point for multiple users. The RF characteristics (specifically, antenna minimum/maximum sizes) are adapted to the mobile service requirements which depend on the applications but also on the regulatory and accommodations constraints. Traffic aggregation at the RCST has to be considered due to the high number of users that can be attached to the same RCST in a mobile platform. Therefore, both the MF-TDMA mode and the continuous carrier operation described in [i.1] will need to be considered.

The telecommunications ground segment consists of the Network Operator Centre (NOC), the Network Control Centre (NCC) compatible with the definition of the DVB-RCS NCC and the gateways, providing access to the terrestrial networks. The NOC is in charge of sharing the satellite resources among network operators, allocating bandwidth to NCC's and centralised management of satellite handover if relevant. The RCST is managed by a single NCC within the satellite coverage area.

The different mobile scenarios that are envisaged in a mobile DVB-RCS system can be classified as follows:

- LOS (Line-Of-Sight) scenarios: these correspond to low-fading scenarios, which are almost always in LOS or close to LOS conditions. Aeronautical and maritime are the two main scenarios in this category.
- Non-LOS (Non-Line-Of-Sight) scenarios: these correspond to land-based where the LOS is frequently affected by frequent/deep/long signal blockages and shadowing. Railway and land-vehicular are the two main scenarios in this category.

4.1 Line-Of-Sight Scenarios

4.1.1 Maritime

The maritime scenario comprises mainly: passenger transportation ships (e.g., ferries and cruises), commercial ships (e.g., cargos and tankers), and private transportation ships (e.g., sailing boats).

Two coverage scenarios are particularly considered: a global one, corresponding to cruise routes (e.g., transatlantic cruises), and a regional one, corresponding to ferry routes (e.g., European coastal regions).

Regarding the modelling of the LOS channel conditions encountered in the maritime scenario, considering the usage of high directive antennas, presence of a strong LOS can be assumed, and the channel can be modelled as a pure Ricean channel with a very high Rice factor; i.e., very close to a AWGN channel.

4.1.2 Aeronautical

The aeronautical scenario comprises mainly: passenger aircrafts (including wide-body and single aisle aircrafts); and private aircrafts (e.g., executive jets).

Two coverage scenarios are particularly considered: a global one, corresponding to long-haul flights (e.g., transatlantic flights), and a regional one, corresponding to short and medium haul flights (e.g., over Europe)

Regarding the modelling of the LOS channel conditions encountered in the aeronautical scenario, experimental results at 18.6 GHz [i.4] show that the channel behaves like Ricean during normal flight situations and manoeuvres, with a Rice factor well above 20 dB. Light fades in the order of 3 dB were observed for manoeuvres with roll angles up to 20°, whereas only in case of extreme manoeuvres, with roll angles up to 45°, the influence of the aircraft structure resulted in deep fades in the order of 15 dB. In conclusion, in aeronautical scenarios, LOS conditions can be assumed and the channel can be fairly approximated by a AWGN channel for most of the time.

4.2 Non-LOS Scenarios

4.2.1 Railway

The railway scenario considered in the following comprises mainly high-speed long-distance trains.

Only the regional coverage scenario is considered since trains remain within one continent.

The mobility effects, such as multipath, shadowing and blockage, encountered due to the local environment in the vicinity of the mobile RCST, such as adjacent buildings, vegetation, bridges, and tunnels, result in sporadic severe fading.

Regarding the modelling of the railway channel conditions, different propagation measurements at Ku [i.5] and Ka bands [i.6],[i.7] were performed in the last decade, based on which, reference statistical channel models exist:

- For Ku-band, the behaviour of the land mobile satellite channel can be modelled using a 3-state (namely LOS, shadowed and blocked states) Markov chain model, where each state is further characterized by a Rice distribution. The transition matrix coefficients and the distribution parameters can be found in [i.5].
- For Ka-band, the behaviour of the land mobile satellite channel can be modelled using a 3-state (namely LOS, shadowed and blocked states) Markov chain model, where each state is further characterized by a Loo distribution. The corresponding parameters can be found in [i.6],[i.7].

For the specific case of the railway environment, dedicated measurements were performed at Ku-Band in Spring 2004 along the Italian railway in the framework of the EC co-funded project FIFTH [i.8]. As illustrated in Figure 2, the presence of several metallic obstacles along the railroad such as electrical trellises (power arch) and catenaries, i.e. electrical cables, must be taken into account in addition to the presence of other obstacles such as isolated buildings, tunnels or trees that may appear along railway lines. Following is a list of signal attenuations associated with different obstacles:

- Catenaries: typically attenuation of 2-3 dB
- electrical trellises”: up to 15-20 dB attenuation (depending on the geometry and layout of the obstacles and on the orientation of the railway with respect to the position of the satellite): length~0.5 m, distance between two consecutive trellises ~ 43 m
- Bridges: high signal loss for a length between a few meters up to tens of meters-50 m.
- Tunnels: long-term interruptions. .

The railway channel can be modelled by superimposing this kind of deterministic and (space) periodic fades on a statistical model accounting for unpredictable obstacles (such as the three-states Markov model with parameters similar to the land mobile highway channel) [i.9].

In conclusion, the railroad satellite channel is in LOS state most of the time. However, short blockages due to power arches as well as long blockages due to obstacles, such as buildings, vegetation, bridges, and tunnels, are also present leading to non-LOS effects.

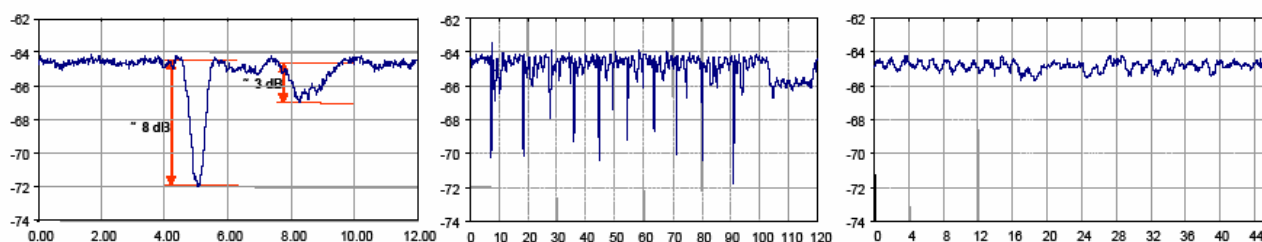


Figure 2: Examples of measured attenuation in dB caused by electrical trellises (left), electrical posts with brackets (mid) and catenaries (right) at Ku-band [i.8]

The interruption caused by short bridges can last from 0.6 sec (at 300km/h) to 9 seconds for (20 km/h). For the purposes of the analyses presented in this document, short blockages are considered up to 1 second of duration and the assumptions regarding trellis- induced fades are summarised in Table 1.

Table 1 Period Time and Obstacle Time for the Railway scenario

Speed	20 km/h	80 km/h	120 km/h	160 km/h	240 km/h
Period Time (s)	7.74	1.935	1.29	0.9675	0.645
Obstacle Time (s)	0.09	0.0225	0.015	0.01125	0.0075

4.2.2 Vehicular

The land-vehicular scenario comprises mainly: passenger vehicles (e.g., buses); commercial vehicles (e.g., trucks); and private vehicles (e.g., cars).

Only the regional coverage scenario is considered since land-vehicles remain within one continent.

Regarding the modelling of the non-LOS channel conditions encountered in the land vehicular scenario, the same statistical channel models described above for the railway scenario apply here as well. However, the effect of deterministic and (space) periodic fades due to power arches are not relevant here anymore, and hence, they shall not be taken into account.

5 Forward link

In case of mobile scenarios, the forward link signal received by the RCST can be adapted to the mobile conditions with the provisions adopted in [i.1].

The block diagram in Figure 3 depicts the main functionalities of a DVB-S2 forward link, including the optional mobile features. Although Figure 3 depicts the case for MPEG transport stream, GSE is also supported by both DVB-S2 [i.2] and the mobile features introduced in [i.1].

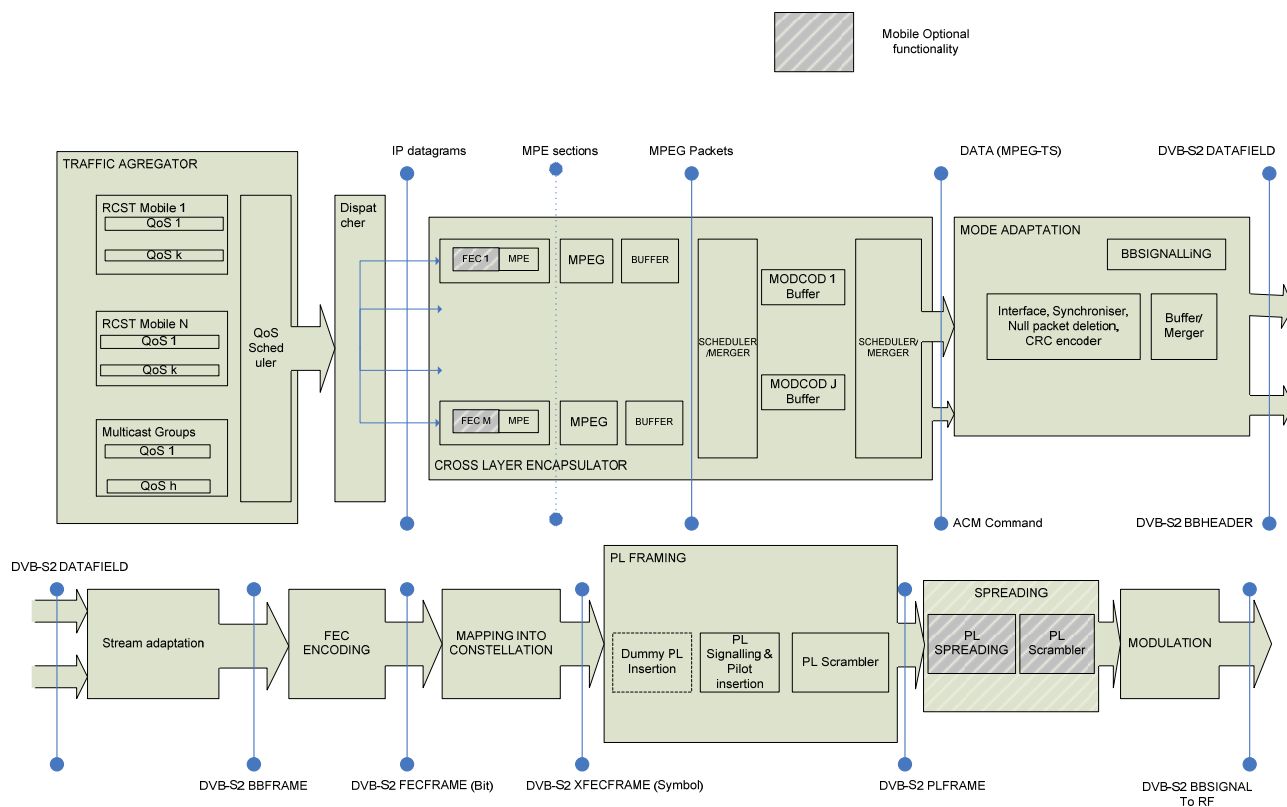


Figure 3: DVB-S2 high level block diagram (MPEG-TS case)

5.1 Spectrum Spreading in the Forward Link

A DVB-S2 forward link transmission can be spread in bandwidth using the provisions adopted in [i.1]. The following diagram is representative of a possible implementation of the spreading functionality that can be added to a DVB-S2 compliant modulator and demodulator. This is considered representative of practical implementations (alternative schemes are possible provided the signal transmitted is equivalent).

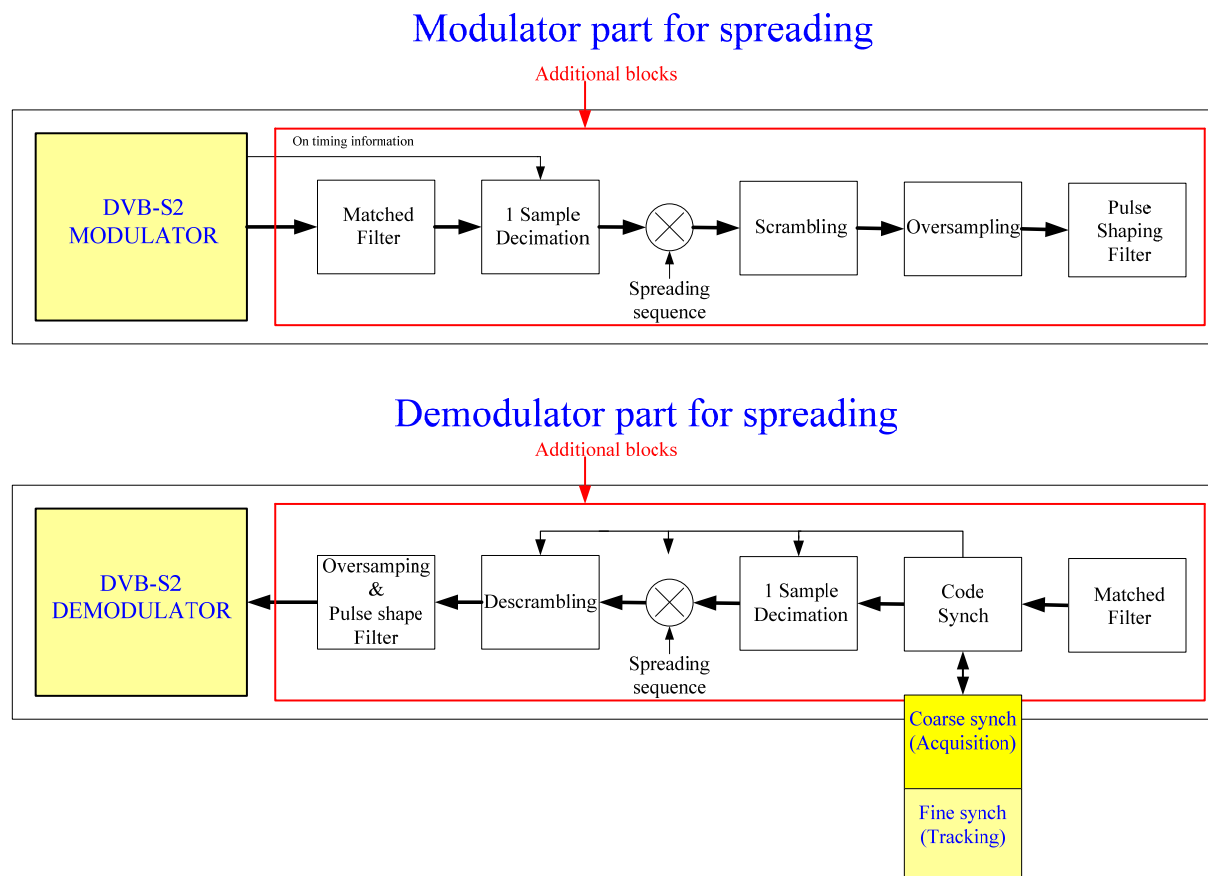


Figure 4: Block diagram of the demodulator with the spreading functionality

The block diagram in the Figure 4 represents the typical modulator according to the DVB-S2 specification [i.2] with the addition of the spreading functionality. The signal outgoing from the first pulse shaping is the classical DVB-S2 signal and it is the input of the spreading chain. The first two blocks in the transmitter spreading chain, which are a Matched Filter and a Sample Decimator, aim at producing the same symbol sequence generated by the DVB-S2 constellation mapper. Symbols are then multiplied by the spreading sequence and scrambled, by using the same scrambling sequence defined for physical layer scrambling in DVB-S2 [i.2]. The Scrambler Block is needed because it allows obtaining an averaging effect on the spectral signal properties. Indeed, although the Spreading Factor is at most 4, the use of scrambling makes the whole spreading effect similar to that obtained using a long spreading sequence. After scrambling, chips are oversampled and filtered to produce a signal according to the DVB-S2 specification [i.2]. At the receiver side, dual operations must be performed, and the despreading chain must be implemented. In this way, the signal coming out from the receiver spreading chain can be processed by a classical DVB-S2 receiver. The adoption of the Direct Sequence (DS) spreading technique imposes a code acquisition in order to perform efficient despreading. In the following, the design and performance analysis of the acquisition subsystems are reported for both forward and return links.

5.1.1 Spectrum Spreading in the Forward Link: code acquisition sub-system

The adoption of the Direct Sequence spreading technique imposes a code acquisition in order to perform efficient despreading. Because the receiver typically operates under different conditions, the high-level algorithm that drives the

code acquisition sub-system can be represented by the finite state machine shown in Figure 5. In particular, the following operation modes can be identified:

1. **S¹ - Cold start acquisition**, which is verified at terminal switch-on;
2. **S² - Verification mode**, which verifies the correctness of the frame acquisition decision;
3. **S³ - Frame tracking**, which is in charge of continuous verification and detection of deep fade events;
4. **S⁴ - Re-acquisition after short interruptions**, which is the procedure put in place to recover the code alignment after a deep fade;
5. **S⁵ - Warm start acquisition**, which takes place after long fading events.

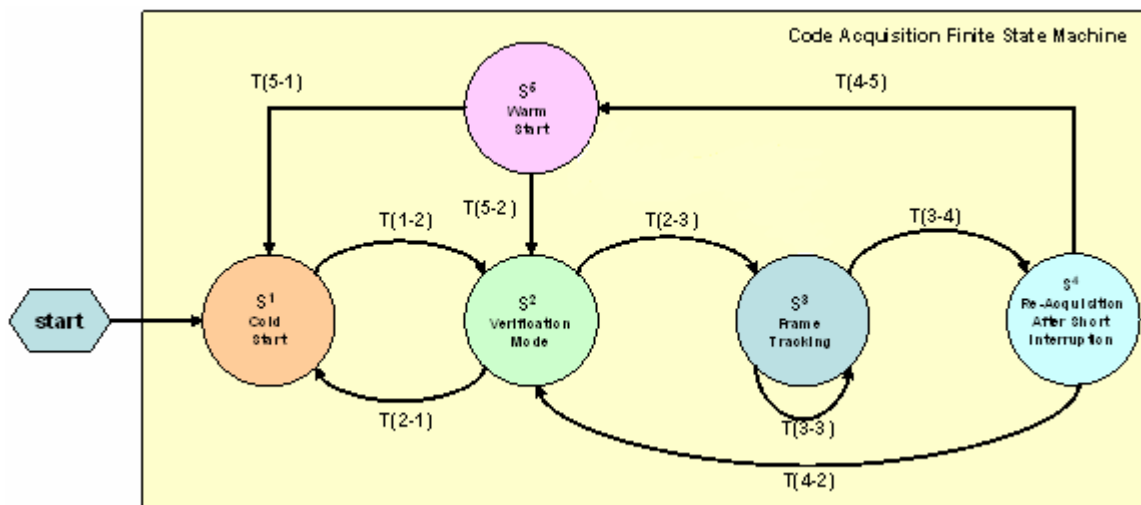


Figure 5: Code acquisition finite state machine

The transitions between the different states are the following:

- **T(1-2)** occurs when the cold start frame synchronization procedure is terminated with the indication of a certain frame alignment, and the control is transferred to the verification mode that has to verify the correctness of that outcome;
- **T(2-1)** occurs when S² reveals the incorrectness of the cold start frame acquisition decision, classifying the outcome of S¹ as a false alarm event;
- **T(2-3)** occurs when S² has completed the verification phase, having verified the correctness of the outcome of S¹;
- **T(3-3)** is the loop transition of S³, which characterizes the normal operating state of the frame synchronization subsystem;
- **T(3-4)** occurs when the lock to the frame alignment is lost, e.g. as a result of a fade event;
- **T(4-2)** occurs when S⁴ has managed to recover the frame alignment, and thus verification can restart;
- **T(4-5)** occurs when S⁴ cannot recover the frame alignment with a limited delay, and thus the last synchronization lock is no more reliable. In this case, warm start synchronization has to be initiated;
- **T(5-2)** occurs when the warm start frame synchronization procedure is terminated with a certain frame alignment indication to be verified by the verification state;
- **T(5-1)** occurs when the warm start acquisition fails to recover the frame alignment within a tolerable time delay, and thus cold start synchronization must be re-initiated.

For any state, the code acquisition subsystem is composed by a code sequence detector, a decision criterion, and a controller that implements the control logic necessary to perform the acquisition procedure. Because of the large uncertainty region, a Threshold Crossing (TC) criterion is selected, which compares the decision variable with a threshold and declares the testing hypothesis to be synchronous when the threshold is crossed.

According to a conventional pilot-aided approach, code alignment is achieved by detecting in the transmission flow the presence of the known preamble formed by the Start of Frame (SOF) spread with the spreading code and scrambled by the scrambling sequence. The total preamble length is thus $L_{preamble} = L_{SOF} SF$, where $L_{SOF} = 26$ according to the DVB-S2 standard [i.2], and SF is the spreading factor.

5.1.1.1 Acquisition performances in the railway scenario

This section presents solutions to enhance the code acquisition sub-system and shows the performance of these techniques in an environment representative of the railway scenario. For the performance assessment, the following assumptions are considered:

- Modulations: QPSK 1/4 and QPSK 1/2
- FECFRAME size 64K
- Chip rate: 28Mcps
- Symbol rate: 7Msps ($SF=4$) and 14 Msps ($SF=2$)
- Channel: AWGN
- Linearized TWTA as in [i.29]
- OMUX as in [i.29]
- Phase noise mask representative of a professional equipment as in Table 2.

Table 2: Phase noise mask for professional equipment

100 Hz	-45dBc/Hz
1000 Hz	-65dBc/Hz
10000 Hz	-80dBc/Hz
100000 Hz	-95dBc/Hz
1e+006 Hz	-105dBc/Hz
>1e+007 Hz	-115dBc/Hz

For the cold start acquisition:

- Carrier frequency error: 3 MHz,
- Timing frequency error: 100ppm
- Doppler rate = 1300 Hz/s (representative of a Ka-band aeronautical scenario, i.e. worst case Doppler rate),

For the warm acquisition:

- Carrier freq. error: 100 KHz
- Timing freq. error: 100 ppm,
- Doppler rate = 1300 Hz/s (Ka-band aeronautical)

For the acquisition after a short interruption (up to 1 sec) in the railway scenario:

- Doppler rate = 200 Hz/s,
- Carrier freq. error normalized to the symbol rate = 2×10^{-4} , (the first stage coarse carrier recovery loop which is assumed almost frozen during steady state)
- Timing freq. error: 1 ppm, (three times the standard deviation of a 2nd order Gardner loop with $BW = 10^{-4} \times R_s$ and critical damping)
- Random timing phase (lost timing lock)
- It is considered that, after a short interruption, the RCST needs to recover: timing, carrier frequency and phase.

5.1.1.1.1 Cold start code acquisition design

Code acquisition in the cold start state is the most critical procedure due to the maximum observed frequency offset. Also, cold start acquisition is characterized by the maximum uncertainty region for the code epoch domain, which is equal to $SFL_F h$, where L_F is the frame length, and h is the number of hypotheses per chip. Due to the low SNR before despreading, chip timing recovery is not feasible with satisfactory performance before code acquisition, thus at least $h = 2$ hypotheses per chip are tested in the synchronization subsystem.

In order to enhance the subsystem robustness against frequency offsets, two detectors based on Post Detection Integration (PDI) can be adopted; namely, Generalized PDI (GPDI) and Differential GPDI (D-GPDI) [i.10]. In general, PDI techniques foresee coherent integration over a code subsequence of length M , to increase SNR before applying non-linear processing. In fact, the presence of a normalized frequency offset $\nu = \Delta f T_c$ (here equal to $3\text{MHz}/27.5\text{Mcps} = 0.11$) determines an energy degradation equal to $\text{sinc}^2(M \nu)$ after coherent accumulation over M chips, which can be contained by appropriately selecting M , given the frequency offset. Thus, for all PDI-based detectors, correlation over the code sequence is split in two parts: coherent accumulation over M chips followed by PDI over the residual length $L = L_{\text{preamble}}/M$. In particular, GPDI derives from the application of the generalized likelihood ratio test and, as illustrated in Figure 6, it results in the combination of Non Coherent PDI (NCPDI) and several differential terms, identified as n -Span DPDI, which take the phase reference over the n th predecessor. The block diagrams of NCPDI, n -Span DPDI, GPDI, and D-GPDI are reported in Figure 6. D-GPDI is a structured extension of GPDI able to enhance the robustness against particularly penalizing frequency errors that force the selection of $M = 1$.

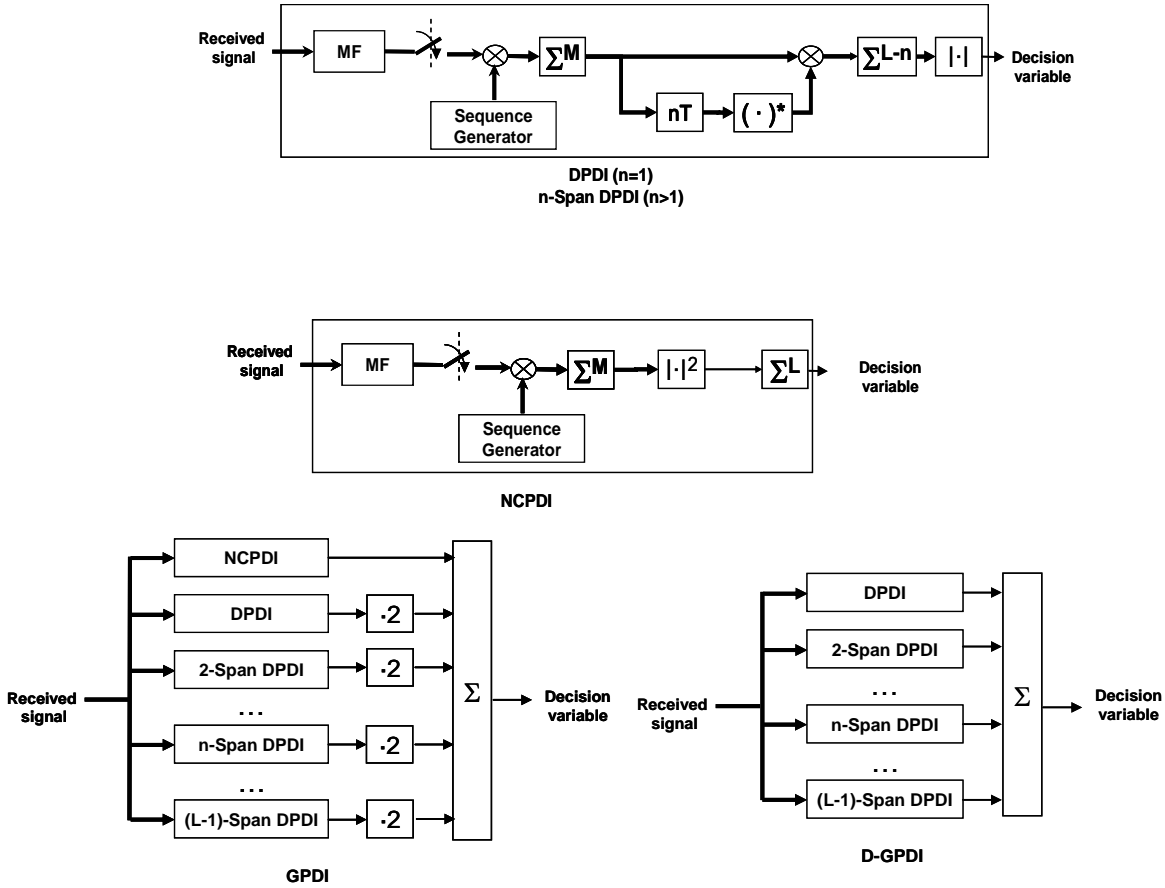


Figure 6: Cold start code acquisition detectors block diagrams

A single dwell code acquisition procedure with serial search is considered to cope with the variable frame length foreseen by the DVB-S2 standard [i.2], which can be modeled as a Markov chain as depicted in Figure 7.

The overall false alarm state (FA) is modelled as non-absorbing, the recovery from which requires a penalty time T_p that has been assumed to be constant. Uniform a priori probabilities $\pi_1 = 1/(L_f h)$ are considered. In order to limit the delay of the acquisition procedure, passive implementation of the detectors is selected, at the cost of increased complexity. This choice allows to reduce the testing time to half of the chip period if $h = 2$ cells per symbol are exploited by the receiver. The large frequency error typical of cold start operation (3MHz) suggests using D-GPDI with $M = 1$ or GPDI with $M = 2$.

Finally, Figure 7 also shows the flow chart corresponding to the control logic implementing the single-dwell TC procedure described in the flow graph (Figure 5). The Cells Selector serially selects the cells to be tested in order to explore the entire uncertainty region. The procedure stops as soon as the spread SOF is detected.

Solving the flow-graph, the analytical expression for the mean and variance of the acquisition time can be obtained as a function of the single cell missed detection and false alarm probabilities, according to the following expressions:

$$\bar{T}_A = \frac{1}{P_D} \left[T_d \left(1 + (N_c - 2) \left(1 - \frac{P_D}{2} \right) \right) + T_p P_{fa} (N_c - 2) \left(1 - \frac{P_D}{2} \right) \right] \quad (1)$$

$$\begin{aligned} \text{Var}\{T_A\} = & \frac{1}{12P_D^2} \left\{ \left[-12(P_D - 1) + N_c(N_c - 2)(12 + P_D(P_D - 12)) \right] T_d^2 + \right. \\ & + 2 \left[-12 + 2(N_c - 2)(P_D - 3) + (N_c - 2)^2(12 + P_D(P_D - 12)) \right] P_{fa} T_d T_p + \\ & + P_{fa} T_p \left[12(-1 + (N_c - 2)(N_c - 4)) P_{fa} T_p + (N_c - 2) P_D^2 (-6 + (N_c - 6) P_{fa}) T_p + \right. \\ & \left. \left. + 12P_D(1 + (N_c - 2)(1 - (N_c - 2)P_{fa})) T_p \right] \right\} \quad (2) \end{aligned}$$

where $P_D = P_d(2 - P_d)$ is the correct detection probability of the collective synchronous state, P_d is the single test correct detection probability, P_{fa} is the single test false alarm probability, $N_c = h L_F SF$ is the number of cells in the uncertainty region, $T_d = T_c/h = T_c/2$ is the dwell time, i.e. the time needed to perform a single test, $h = 2$ is the number of hypotheses per symbol used by the receiver.

In order to minimize the acquisition time, the optimization of the decision threshold plays a fundamental role. In particular, because the detection problem is strongly unbalanced (2 H_1 hypotheses against $N_c - 2 H_0$ cells) and the detector has to test a very large number of H_0 cells, P_{fa} has to be reduced in order to avoid an excessive number of erroneous synchronizations. This optimization is performed by searching for the decision threshold that minimizes the mean acquisition time. However, it is not possible to analytically express the mean acquisition time dependence on the decision threshold, because the analytical characterization of GPDI and D-GPDI single cell performance is not available. Thus, a semi-analytical approach is adopted that uses the analytical expression of the mean acquisition time along with the simulated single cell performance P_{fa} and $P_{md} = 1 - P_d$.

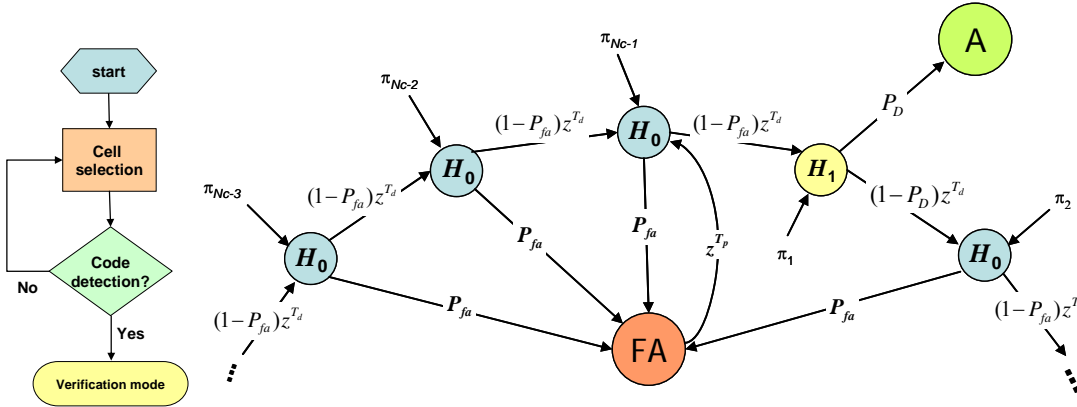


Figure 7: Cold start acquisition procedure

Figure 8 shows the performance in terms of mean acquisition time for GPDI with $M = 2$ and D-GPDI with $M = 1$, considering $SF = 2$ and $SF = 4$, with $E_c/N_0 = -8.35$ dB and -5.35 dB respectively (corresponding to the DVB-S2 worst case of $E_s/N_0 = -2.35$ dB). Non ideal chip timing recovery is assumed by accounting for a fixed worst case fractional chip misalignment $\delta T_c = 0.25T_c$ with two hypotheses per symbol processed by the receiver ($h = 2$). The penalty time is conservatively assumed equal to $2T_F$.

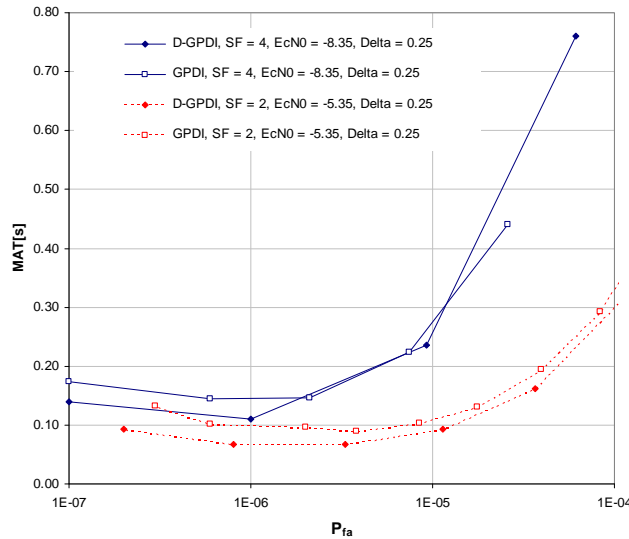


Figure 8: Mean acquisition time performance

Notably, with the large uncertainty region of the scenario analysed here ($N_c = 133128$ for $SF = 2$ and $N_c = 266256$ for $SF = 4$) the optimal mean acquisition time occurs with a very low false alarm probability P_{fa} , in the order of $10^{-(6)}$, and

for D-GPDI it is equal to 0.067 s. with $SF = 2$ and 0.111 s. with $SF = 4$, while with GPDI it equals 0.090 s. with $SF = 2$ and 0.144 s. with $SF = 4$. The thresholds corresponding to these values are selected as operational points.

To provide more conservative performance figures, the 99% acquisition time percentile is also investigated. To this aim, the probability density function (pdf) of the acquisition time is modeled using the one-sided central limit theorem which yields a Gamma pdf [i.11]

$$p_{T_A}(T_A) = \frac{T_A^{a-1} e^{-T_A/b}}{b^a \Gamma(a)} \quad T_A \geq 0 \quad (3)$$

where

$$a = \frac{\bar{T}_A^2}{\text{Var}\{T_A\}} \quad b = \frac{\text{Var}\{T_A\}}{\bar{T}_A} \quad (4)$$

and $\Gamma(x)$ is the Gamma function.

The computation of the Gamma pdf in correspondence of the optimal operational point discussed above yields the cumulative distribution functions (cdf) shown in Figure 9, where also the 99% specification is highlighted.

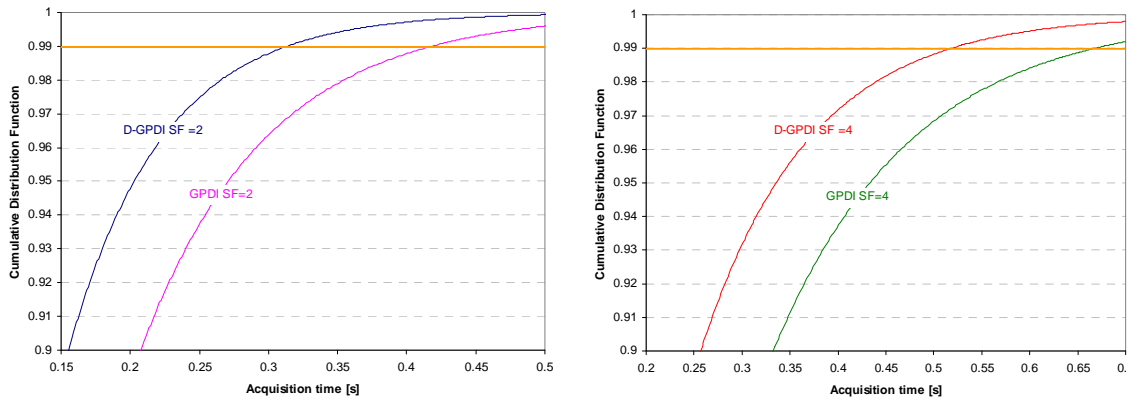


Figure 9: Acquisition time cdf for D-GPDI and GPDI with $SF = 2$ and $SF = 4$

The abscissa points satisfying the 99% constraint are finally summarized in Table 3. GPDI allows to achieve a 99% acquisition time equals to 0.419 s. with $SF = 2$ and 0.669 s. with $SF = 4$. A further improvement is reached by D-GPDI, which allows for 0.314 s. with $SF = 2$, and 0.518 s. with $SF = 4$.

Table 3 – Acquisition time performance

SF	Detector	Pfa	Pmd	Mean Acq Time [s]	Var Acq Time	99% Acq Time [s]
2	D-GPDI	3.30E-06	0.96658	0.067	0.005	0.314
2	GPDI	3.80E-06	0.97317	0.090	0.008	0.419
4	D-GPDI	1.00E-06	0.96707	0.111	0.013	0.518
4	GPDI	6.00E-07	0.97803	0.144	0.021	0.669

5.1.1.1.2 Acquisition after a short interruption

Acquisition after a short interruption is represented by the S^4 state in Figure 6. The maximum fade event duration is assumed equal to 1s. (railway scenario), i.e. 500 frames with $SF = 2$ or 250 with $SF = 4$. After this period, considering that all tracking circuits have been frozen in their last steady state operation, the frequency error is equal to 5700Hz (clock instability, 5500Hz, plus Doppler rate, 200Hz), i.e. 0.0002 normalized to the chip rate. In addition, the clock drift introduces a time uncertainty equal to ± 28 chips. Accordingly, the code epoch domain is equal to:

- 56 chips for constant QPSK modulation

- The entire longest frame length for variable modulation, i.e. $33282 \times SF$ chips

No chip timing recovery is feasible before despreading, thus a fractional displacement $\delta = 0.25$ is assumed along with $h = 2$.

A typical performance requirement for the acquisition time is 10 % of the fade duration at 99 % percentile, i.e. 0.1 s.

In line with cold start acquisition, GPDI and D-GPDI are considered with the same (M, L) , although this approach leads to sub-optimum performance. Optimized performance is provided by $M = L_{SOF} \times SF$ and $L = 1$, i.e. full coherent correlation followed by energy detection (NCPDI).

With variable modulation, the single dwell TC procedure of [i.12] is adopted. The penalty time is assumed equal to one frame, assuming that verification is done by exploiting the pilot fields, i.e.:

- $T_p = 8$ PSK frame duration ($22194 \cdot SF$) with variable modulation;
- $T_p =$ QPSK frame duration ($33282 \cdot SF$) with constant QPSK modulation.

Considering all possible impairments (non linear distortion, IMUX-OMUX linear distortion, phase noise, frequency error, clock time drift) and pre-distortion, the mean acquisition time and variance are reported in Table 4 along with the 99% percentile acquisition time. Notably, all solutions provide in-spec performance, except for D-GPDI with $SF = 4$, because of the reduced frequency error.

Table 4 – Acquisition performance after deep fade events, variable modulation

SF	Detector	Pfa	Pmd	Mean Acq Time [s]	Var Acq Time	99% Acq Time [s]
2	D-GPDI	2.60E-06	0.93	0.01993	0.00040	0.0930
2	GPDI	1.50E-06	0.91	0.01540	0.00024	0.0705
2	NCPDI	1.04E-05	0.49	0.00382	0.00001	0.0180
4	D-GPDI	1.20E-06	0.91	0.03147	0.00228	0.2250
4	GPDI	3.40E-06	0.85	0.02466	0.00063	0.1155
4	NCPDI	1.00E-06	0.57	0.00555	0.00003	0.0240

With constant QPSK modulation, performance can be further improved by exploiting a single dwell TC procedure with a limited uncertainty region centered around the last frame lock, making the scenario less critical. The achievable mean acquisition time is shown in Table 5.

Table 5 - Acquisition performance after deep fade events, constant QPSK modulation

SF	Detector	Pfa	Pmd	Mean Acq Time [s]
2	D-GPDI	2.60E-06	0.92785	0.00190
2	GPDI	0.001674	0.49016	0.00113
2	NCPDI	0.000352	0.17402	0.00013
4	D-GPDI	0.000768	0.67625	0.00463
4	GPDI	0.001693	0.48872	0.00225
4	NCPDI	0.000332	0.13015	0.00018

5.1.1.1.3 Warm Start Acquisition

Warm start acquisition takes place after long fading events or time slicing. The frequency offset is in this case increased with respect to S^4 , albeit still strongly reduced with respect to cold start operation. In particular, a 100kHz carrier frequency error is considered, i.e. 0.0036 normalized to the chip rate. In addition, the typical 100ppm clock drift introduces a time uncertainty equal to ± 2750 chips. Accordingly, the code epoch domain is equal to:

- 5500 chips for constant QPSK modulation
- the entire longest frame length for variable modulation, i.e. $33282 \cdot SF$ chips

Again, no chip timing recovery is feasible before despreading, thus a fractional displacement $\delta = 0.25$ is assumed along with $h = 2$.

In line with cold start acquisition, GPDI and D-GPDI are considered with the same (M, L) , although this approach leads to sub-optimum performance. Similarly to re-acquisition after short interruptions, optimized performance is provided by $M = L_{SOF} \cdot SF$ and $L = 1$, i.e. full coherent correlation followed by energy detection (NCPDI).

The single dwell TC procedure of [i.12] is considered with variable modulation, with a penalty time equal to one frame, assuming that verification is done by exploiting the pilot fields, i.e.

- $T_p = 8$ PSK frame duration ($22194 \cdot SF$) with variable modulation;
- $T_p =$ QPSK frame duration ($33282 \cdot SF$) with constant QPSK modulation.

Considering all possible impairments (non linear distortion, IMUX-OMUX linear distortion, phase noise, frequency error, clock time drift) and pre-distortion, the mean acquisition time and variance are reported in Table 6 along with the 99% percentile acquisition time.

Table 6 – Acquisition performance for warm start operation, variable modulation

SF	Detector	Pfa	Pmd	Mean Acq Time [s]	Var Acq Time	99% Acq Time [s]
2	D-GPDI	8.60E-06	0.90	0.02004	0.00041	0.0945
2	GPDI	3.10E-06	0.89	0.01369	0.00019	0.0630
2	NCPDI	6.60E-06	0.66	0.00486	0.00002	0.0225
4	D-GPDI	2.10E-06	0.91	0.03522	0.00126	0.1635
4	GPDI	3.20E-06	0.85	0.02379	0.00058	0.1110
4	NCPDI	2.40E-06	0.86	0.02365	0.00057	0.1095

Similarly to S^4 , with constant QPSK modulation, performance can be further improved by exploiting a single dwell TC procedure with a limited uncertainty region centered around the last frame lock. The achievable mean acquisition time is shown in Table 7.

Table 7 - Acquisition performance for warm start operation, constant QPSK modulation

SF	Detector	Pfa	Pmd	Mean Acq Time [s]
2	D-GPDI	2.36E-05	0.86	0.009
2	GPDI	3.40E-05	0.79	0.006
2	NCPDI	3.09E-05	0.52	0.002
4	D-GPDI	1.32E-05	0.86	0.016
4	GPDI	4.38E-05	0.74	0.010
4	NCPDI	4.54E-05	0.70	0.008

5.2 Fast re-synchronization in mobile scenarios

Some mobile channels are characterised by the presence of obstacles that can cause link outage (see section 4.2). In these circumstances, it is important to analyse the re-synchronisation performances of the RCST. In the following, the focus is on the railway scenario.

5.2.1 Forward link re-synchronization issues in the Railway scenario

As seen in sec 4.2.1, the railway channel is characterised by periodic fading. This section analyses the synchronisation sub-system behaviour in the presence of this frequent fading.

5.2.1.1 System assumptions for the analysis

The following analysis considers only one antenna in order to minimize terminal cost and installation. The DVB-S2 forward link symbol rate ranges from 5 Msps to 25 Msps. It is assumed that pilot symbols are used. This analysis considers that no spreading is required neither on the forward nor in the return links.

It is considered that the terminal receiver is equipped with a signal loss detector which is able to estimate the loss of the signal within a short period of time (typically one frame if pilots are used). Additionally, a second order timing recovery loop is considered in the receiver terminal

5.2.1.2 Forward link carrier stability analysis

As long as the normalized frequency error (with respect to the symbol rate) accumulated during a signal loss is below $\sim 3.5 \times 10^{-4}$ (1750 Hz at $R_s=5$ Msps–8750 Hz at $R_s=25$ Msps) no carrier frequency re-training is needed. With a Doppler rate of 100 Hz/s, an accumulated frequency error of 1750 Hz is reached after 7.5 seconds, while 8750 Hz is reached after 87.5 seconds. Phase tracking is always performed on a frame-by-frame basis which implies a fast acquisition. Under these conditions, it is considered that carrier re-training is not needed after interruptions due to trellises and bridges. Carrier stability could be considered as a problem only in the case of 5 Msps carriers at low speeds (i.e., below 30 km/h).

5.2.1.3 Forward link timing instability analysis

5.2.1.3.1 Timing instabilities due to terminal acceleration

Assuming a worst- case acceleration of 2.5 m/s^2 , the timing error accumulation during a signal outage would amount to $0.8 \times \pi \times 10^{-8} \times R_s \times T_{\text{out}}^2$ radians where T_{out} is the outage duration.

Timing errors within $0.05/R_s$ are acceptable to limit the performance degradation. Taking into account these conditions, the maximum outage time T_{out} can be calculated as follows:

$$T_{\text{out}} < \sqrt{1.25 \times 10^7 / R_s},$$

That is 1.58 seconds at $R_s=5$ Msps and 0.7 seconds at $R_s=25$ Msps.

5.2.1.3.2 Timing instabilities due to residual frequency jitter

Even using a second-order timing recovery loop, the estimated clock frequency before signal interruption may be offset with respect to the true frequency by several Hz. This frequency error would make the timing error to accumulate very fast during the signal interruption even if no Doppler rate is present.

With a normalized bandwidth of $10^{-4} \times R_s$ and a critical damping $\xi=0.707$, the jitter of the frequency state variable of the loop is in the order of $3 \times 10^{-7} \times R_s$ Hz at an input SNR of the clock wave of 0 dB. Considering a max error of 3 times the standard deviation, the error of $0.05/R_s$ is reached after 11 ms at $R_s=5$ Msps and 2.2 ms at $R_s=25$ Msps.

A better choice of the damping factor ($\xi=10$) would allow for a reduction of the steady state timing frequency error by about a factor of 100 and still be able to track the timing frequency ramps typical of this scenario. Signal losses of duration up to 1 second and 200 ms (for 5 Msps and 25 Msps, respectively) could be handled without re-synchronization.

5.2.1.3.3 Conclusions on timing instabilities

Considering the two effects (terminal acceleration and residual frequency jitter) it turns out that timing re-training is not needed for interruptions of duration less than 1 second for $R_s=5$ Msps and 200 ms for $R_s=25$ Msps. Therefore retraining of the receiver clock is not needed to cope with trellises interruptions.

Clock retraining is needed after interruptions due to bridges at all speeds for $R_s=25$ Msps and for speeds lower than about 200 Km/h at $R_s=5$ Msps. Re-training of the clock takes up to 5×10^4 symbols, i.e. 100 ms at 5 Msps and 2 ms at $R_s=25$ Msps

5.2.1.4 Forward link frame acquisition analysis

In CCM mode, if correct timing is not lost, there is no need to recover the right frame alignment. The receiver only needs to correctly count the symbol periods during a signal outage. This guarantees that after the signal comes back up the right frame alignment is maintained.

In ACM mode, regardless of whether the receiver clock has to be retrained or not, the right frame alignment has to be re-acquired, because in ACM mode the frame alignment relies on the successful decoding of the PL header. In case only one of these is lost, correct re-alignment is needed. Two situations can be considered in this case:

- If timing is maintained, frame resynchronization can be carried out within one frame. This may be done by modifying the frame acquisition process in case a short interruption is detected. In such a case, taking into account the smaller frequency error with respect to cold acquisition, coherent correlation could be used on the SOF plus a number of pilot fields.
- If timing is not maintained, clock retraining has to be performed first, followed by the procedure listed in the preceding bullet point.

5.2.1.5 Conclusions on Forward link re-synchronisation

With a proper demodulator design, interruptions due to trellises do not imply any need for the demodulator re-acquisition. Interruptions due to bridges may require timing and/or frame re-acquisition for a maximum duration of about 100-120 ms at 5 Msps, 5-8 ms at 25 Msps.

The use of the optional pilot symbols specified in [i.2] will increase system performance and is highly recommended in the mobility scenario to help the channel state estimation, carrier recovery and frame acquisition in ACM mode.

The terminal demodulator should be designed to achieve a fast signal level detector to establish when to freeze and re-activate the demodulator.

5.3 Forward link countermeasures for Non-LOS mobile scenarios

5.3.1 Proactive retransmission on the forward path for TCP traffic in Railway scenarios

In the absence of link-layer FEC-based mitigations (analysed in section 5.3.2), deep fades caused by regularly placed power arches along railways will significantly reduce the maximum achievable forward throughput by TCP connections. This is due to the TCP congestion control algorithm's inability to distinguish corruption-caused segment losses from congestion-caused segment losses. The mitigation explained in this section dictates transmitting each TCP segment twice at a retransmission delay, denoted as `rtx_delay`. The mitigation does not require any modification on the underlying satellite system. Performance of this mitigation is evaluated in [i.13] for TCP variants, Reno, Newreno, and SACK. The analyses in [i.13] consider various train velocities with corresponding fade durations and inter-fade durations (see section 4.2).

The simulation analyses presented in [i.13] used ns-2.30 TCP variants with DVB-RCS extensions in MAC layer. The analyses considered zero FEC protection for data as well as control messages. Best performance was observed for TCP SACK variant with delayed acknowledgment option. Retransmission delay, `rtx_delay`, values of 10ms and 40ms were considered with 3 and 5 duplicate acknowledgment options. According to the results shown in [i.13], this technique brings improvement when the fade duration is comparable with the `rtx_delay`, however this mitigation should only be used with `rtx_delay` longer than the longest expected fade duration. Further throughput increase was attained by using 5 duplicate acknowledgement option instead of 3.

The throughput improvement achieved by this mitigation is obtained at 100% redundancy, and hence, its use can only be recommended in the absence of more sophisticated forward error correction mechanisms that operate at a lower redundancy.

The implementation of this technique does not require modifications at layers below TCP. Neither does it need modifications at the RCST-side TCP SACK protocol stack, unless 5 duplicate acknowledgement option is to be used. Most TCP PEPs already use TCP variants that are based on TCP SACK. Forward proactive retransmission of TCP segments should best be realised at TCP PEPs in gateways.

5.3.2 Link layer FEC protection

In clause 6.4.5 of [i.1] countermeasures for non-line-of-sight channels have been specified. RCST's that declare support for non-line-of-sight (NLOS) countermeasures in the CSC burst shall be able to receive and process Link Layer Forward Error Correction (LL-FEC). This technique can also be applied to the optional continuous return link carrier transmissions defined in [i.1].

LL-FEC has been introduced to support reception in situations of high Packet Loss Ratio (PLR) at the MPE section or GSE packet level. Such high PLR may occur for example on mobile channels when the speed is too high and/or the signal-to-noise ratio is too low and/or multi-path propagation results in fading. It may also occur due to obstruction, blockage, or other situations in which the line of sight is interrupted. With the LL-FEC, a variable amount of capacity is allocated to parity overhead.

Transmissions employing LL-FEC use the same basic data structures as other MPE/GSE transmissions. The use of LL-FEC is optional and is defined separately for each elementary stream in the transport stream or GSE-FEC stream in the generic stream. Each elementary/GSE-FEC stream may configure different code parameters, resulting in different delays, level of protection and FEC overhead.

Hereafter, guidelines are given for the use of LL-FEC with a particular focus on the railway scenario.

5.3.2.1 General LL-FEC architecture

The use of LL-FEC is optional and is defined separately for each elementary stream in the transport stream, i.e. per PID, in this way it is possible to choose whether or not LL-FEC is applied on each elementary stream. Each elementary stream may be configured with different LL-FEC code parameters (LL-FEC code, LL-FEC frame size) resulting in different levels of protection, as well as different delays and FEC overheads. The application and parity data from a LL-FEC Frame are carried over the same elementary stream. For each elementary stream implementing LL-FEC, a cyclic LL-FEC frame index, called `fec_frame_number`, is managed. It is incremented for each new LL-FEC frame transmitted on the stream.

Similarly, for GSE generic streams, the use of LL-FEC is defined separately for each GSE-FEC stream in the generic stream, and each GSE-FEC stream may be configured with different code parameters. A GSE-FEC stream represents a sequence of GSE packets transmitted on the generic stream and carrying the application and parity data from the same LL-FEC Frames. A GSE-FEC stream is identified by a 14-bit identifier called `gse_fec_id` (equivalent to the PID for elementary streams). This identifier is present in the GSE packets carrying application and parity data. A cyclic LL-FEC frame index, called `fec_frame_number`, is managed within each GSE-FEC stream. It is incremented for each new LL-FEC frame transmitted on the stream.

Transmissions not requiring LL-FEC should use elementary streams in the transport stream different from the elementary streams using LL-FEC. Indeed transmissions employing LL-FEC use the same basic data structures as other MPE transmissions but with certain fields in the header interpreted in different ways. The possibility could exist that a LL-FEC header may be interpreted as a regular DSM-CC header, addressed to a terminal that does not support LL-FEC

For GSE generic stream, transmissions not requiring LL-FEC will be transmitted in a regular way in the generic stream.

The level of protection provided by LL-FEC on an elementary/GSE-FEC stream is dependant on the code parameters configured on this stream (the code parameters are signalled in the LL-FEC descriptor). This level should be selected in accordance with the level of protection required by the traffic transmitted on this stream. It is therefore recommended to transmit on a same elementary/GSE-FEC stream services requiring a similar QoS class.

LL-FEC aims at protecting interactive unicast and multicast traffic data transmitted over the DVB-S2 forward link against channel impairments such as short interruptions and shadowing. Unicast and multicast traffic requiring FEC protection will be distributed over the different elementary streams implementing LL-FEC or GSE-FEC streams.

Assigning an Elementary Stream/ GSE-FEC stream on a RCST basis (e.g. one stream per RCST and per protection level) is beneficial for RCST since the processing in the RCST is limited to its own traffic. However, this option is clearly limited by the number of available PIDs/gse-fec-ids, which limits the number of terminals. Moreover this low traffic aggregation may have several drawbacks such as an increase of the total buffering in the gateway, more overhead generated by LL-FEC, longer delay and jitter.

It is therefore highly recommended to aggregate several traffic flows, i.e. flows destined to different RCSTs or from different multicast groups, per elementary/GSE-FEC stream. This traffic aggregation should be performed per required protection level as mentioned above. Concerning the level of traffic aggregation, the standard does not enforce any rules. This depends on the targeted system, and many solutions may be considered; e.g. traffic aggregation per MODCOD and QoS. For the terminals, this traffic aggregation implies that they must decode more than their own traffic. A terminal shall thus first filter traffic on PID/gse_fec_id identifier and perform LL-FEC decoding (it shall be configured with the PID(s) and gse_fec_id identifier(s) of the corresponding elementary and GSE-FEC streams on which its traffic is transmitted). It should then filter on the `NLOS_RCST_ID` for MPE sections and on the Label for GSE packets. The `NLOS_RCST_id` has been defined to allow receivers to filter MPE section. This 22-bit field is a unique identifier which is locally or statically configured in each NLOS-capable RCST in the network. It can be used

for filtering of the received data instead of the conventional MAC address, which is not available due to the carriage of the LL_FEC real-time parameters.

5.3.2.2 Guidelines on Link Layer FEC parameters selection

The FEC can be configured by making use of the ll_fec_identifier descriptor as specified in 8.5.5.10.24 in [i.1]. This descriptor (see Table 8) defines the characteristics of one or more link layer FEC frames.

The main configuration parameters for the LL_FEC identifier descriptor

- ll_fec: indicates whether the referenced elementary stream uses LL-FEC, and which algorithm is used, namely Reed-Solomon or Raptor code.
- frame_size: This field indicates the exact number of rows in each LL-FEC Frame. In addition, by using this field the address granularity and the maximum LL-FEC ADT Size are signalled. For reference, the coding of the field is repeated in Table 8. In case of any ambiguity, the Table 82 in [i.1] takes precedence.

Table 8: LL_FEC frame size coding

Value	LL-FEC Frame rows (RS)	LL-FEC Frame rows (Raptor)	Address Granularity (Raptor)	Max LL-FEC ADT Size (Raptor, Informative)
0x00	256	256	2	16777216 bits = 16 Mbits
0x01	512	512	4	33554432 bits = 32 Mbits
0x02	768	768	6	50331648 bits = 48 Mbits
0x03	1024	1024	8	67108864 bits = 64 Mbits
0x04	reserved for future use	64	1	4194304 bits = 4 Mbits
0x05	reserved for future use	2048	16	134217728 bits = 128 Mbits
0x06	reserved for future use	4080	32	267386880 bits = 255 Mbits
0x07	reserved for future use	reserved for future use	reserved for future use	reserved for future use

Note: The address granularity is 1 for all Reed-Solomon code options.

- **buffer_timeout:** This field indicates in milliseconds the maximum time interval between the transmission of the first section with a given fec_frame_number (in general a data section) and the transmission of last section with the same fec_frame_number (in general a parity section).

In addition to this, for each LL-FEC frame generated in the operation the transmitter can flexibly decide on the number of LL-FEC ADT columns, no_adt_columns, to use for this specific LL-FEC frame and the number of LL-FEC FDT columns, no_fdt_columns, to use for this specific LL-FEC frame.

Table 9 summarizes the main parameters for the LL-FEC for different codes.

Table 9: Summary of parameters for LL-FEC configuration

Value	RS Code		Raptor Code	
	Minimum	Maximum	Minimum	Maximum
LL-FEC Frame rows no rows	256	1024	64	4080
no_adt_columns	1	191	4	8192
no_fdt_columns	0	64	0	65536 - no_adt_columns
LL-FEC ADT Size	-	1528 kBit	-	255 Mbit
LL-FEC Frame Size	-	2040 kBit	-	2040 MBit

For the appropriate selection of the FEC code and the parameter, the following parameters should be taken into account:

- The maximum service bit rate referred to as $R_{service}$
- The maximum permitted latency for the service referred to as $\Delta_{service}$. The maximum permitted latency depends for example on the QoS requirements of the service. For guidelines on the selection of this parameter refer to clause 10.
- The expected channel conditions and some resulting target code rate for the link layer referred to as r_{ll} . More details on the selection of appropriate code rates are discussed below.

The LL-FEC obviously performs best if the code can make use of maximum time diversity. Therefore, the LL-FEC should expand the size of the LL-FEC ADT Size to be as large as possible to get highest diversity. The maximum tolerable LL-FEC ADT Size is obtained as the product of the service bit rate and the maximum permitted latency as $R_{service} \cdot \Delta_{service}$.

If this product of the service bit rate and the maximum latency is less than or equal to 1528 kBit, then either code, Reed-Solomon or Raptor, may be chosen. If this product of the service bit rate and the maximum latency is greater than 1528 kBit, then the Raptor code should be chosen. Furthermore, due to the restrictions of the LL-FEC Frame Size and the `no_fdt_columns` only a certain set of minimum link layer code rates $r_{ll, min}$ can be supported for specific codes. The minimum link layer code rate results in

$$r_{ll, min} = \min(\text{no_adt_columns}_{max}, \text{ceil}(R_{service} \cdot \Delta_{service} / (\text{no_rows}_{max} * 8)) / (\text{no_adt_columns}_{max} + \text{no_fdt_columns}_{max})),$$

and is shown in Table 10 for different codes and different combinations of $R_{service}$ and $\Delta_{service}$. Table 10 highlights in green the combinations that can be supported with a minimum code rate below 2/9 and in yellow the ones between 2/9 and 1. For white areas indicated with “na”, the code cannot support the corresponding combination. The Reed-Solomon codes are not applicable when long latency (long protection time) and high data rates are considered. The Raptor code allows the extension to use cases of higher bit-rates and larger latencies and covers the entire range of service bit-rates up to 32 MBit/s and latencies of 10 seconds.

Table 10: Supported link layer code rates r_{ll} (in green if below 2/9, in yellow if between 2/9 and 1) for different maximum service bit rates $R_{service}$ and latency $\Delta_{service}$ in ms for RS codes and Raptor codes.

RS Code		Latency in ms													
		10	20	40	80	160	320	640	1280	2560	5120	10240	20480	40960	81920
32		0.02	0.02	0.02	0.02	0.02	0.03	0.04	0.07	0.14	0.24	0.38	na	na	na
64		0.02	0.02	0.02	0.02	0.03	0.04	0.07	0.14	0.24	0.38	na	na	na	na
128		0.02	0.02	0.02	0.03	0.04	0.07	0.14	0.24	0.38	na	na	na	na	na
256		0.02	0.02	0.03	0.04	0.07	0.14	0.24	0.38	na	na	na	na	na	na
512		0.02	0.03	0.04	0.07	0.14	0.24	0.38	na	na	na	na	na	na	na
1,024		0.03	0.04	0.07	0.14	0.24	0.38	na	na	na	na	na	na	na	na
2,048		0.04	0.07	0.14	0.24	0.38	na	na	na	na	na	na	na	na	na
4,096		0.07	0.14	0.24	0.38	na	na	na	na	na	na	na	na	na	na
8,192		0.14	0.24	0.38	na	na	na	na	na	na	na	na	na	na	na
16,384		0.24	0.38	na	na	na	na	na	na	na	na	na	na	na	na
32,768		0.38	na	na	na	na	na	na	na	na	na	na	na	na	na
65,536		na	na	na	na	na	na	na	na	na	na	na	na	na	na
131,072		na	na	na	na	na	na	na	na	na	na	na	na	na	na
262,144		na	na	na	na	na	na	na	na	na	na	na	na	na	na
524,288		na	na	na	na	na	na	na	na	na	na	na	na	na	na
1,048,576		na	na	na	na	na	na	na	na	na	na	na	na	na	na

Raptor Code		Latency in ms													
		10	20	40	80	160	320	640	1280	2560	5120	10240	20480	40960	81920
Maximum Service Bitrate in kbit/s	32	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	64	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	128	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	256	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01
	512	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02
	1,024	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04
	2,048	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08
	4,096	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na
	8,192	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na	na
	16,384	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na	na
	32,768	0.00	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na	na	na	na
	65,536	0.00	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na	na	na	na	na
	131,072	0.00	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na	na	na	na	na	na
	262,144	0.00	0.00	0.00	0.01	0.02	0.04	0.08	na	na	na	na	na	na	na
	524,288	0.00	0.00	0.01	0.02	0.04	0.08	na	na	na	na	na	na	na	na
1,048,576	0.00	0.01	0.02	0.04	0.08	na	na	na	na	na	na	na	na	na	

In addition to the code selection, also the LL_FEC frame size coding needs to be selected appropriately taking into account the parameters maximum service bit rate R_{service} , maximum permitted latency for the service Δ_{service} , the target link layer code rate r_{Π} .

For Reed-Solomon codes, the LL_FEC frame size coding should be chosen such that the three parameters can be supported with the smallest available number of rows, no_rows , i.e. the smallest no_rows such that

$$\min(191, \text{ceil}(R_{\text{service}} \cdot \Delta_{\text{service}} / (\text{no_rows} * 8))) / 255 \leq r_{\Pi}.$$

For Raptor codes, it is recommended that the maximum number of LL-FEC ADT columns is in the range between 1000 and 2000. The LL_FEC frame size coding should be chosen such the three parameters can be supported with the smallest available number of rows, no_rows , i.e. the smallest no_rows such that

$$R_{\text{service}} \cdot \Delta_{\text{service}} / (\text{no_rows} * 8) \geq 1000.$$

Note that if very small or very large values of the combinations of maximum service bit rate R_{service} and maximum permitted latency for the service Δ_{service} are to be supported, the maximum number of LL-FEC ADT columns may also be selected outside the range of 1000 and 2000 between 4 and 8192. For very small values the frame size coding should be 0x04, i.e. the number of rows is 64, and for very large values the frame size coding should be 0x06, i.e. the number of rows is 4080.

For both codes, the exact no_adt_columns may be different for each LL-FEC frame and usually depends on the IP datagram inserted. The no_fdt_columns should be selected such that the target code rate for each LL-FEC frame is met. Both values, no_adt_columns and no_fdt_columns can be signaled individually for each LL-FEC frame.

For Reed-Solomon codes, each column of the LL-FEC FDT shall be carried in a separate section/packet. For Raptor codes, the number of columns carried in one section/packet may be one or several. The number of columns carried in one section/packet can be selected individually for each section/packet and may be determined based on some mappings onto lower layer units, or based on some overhead versus coding performance metrics. In general the header overhead should not exceed 2-3% for each section and packet.

5.3.2.3.1 Guidelines for the selection of the LLFEC profile

The following methodology is proposed to select the most adequate LL-FEC profiles and it should be applied to size each of the LL-FEC instances active in the satellite gateway. The number of LL-FEC instances depends on several architectural choices and on the way QoS support is implemented and it represents a major issue for the system complexity and scalability. In the following, it is assumed that there are different LL-FEC instances per QoS class (or groups of QoS classes) and per ModCod. The first requirement permits to ensure that the maximum extra latency is kept under control; the second requirement avoids wasting the LL-FEC correction capabilities.

The following parameters are defined. Index i indicates the ModCod, whereas index j the QoS class (or group of classes).

- $P_{\text{out}}^{(i,j)}$ [s] desired protection for ModCod i and QoS class j
- $\Delta^{(i,j)}$ [s] extra maximum latency due to the LL-FEC
- R_s [baud] symbol rate
- η_i [bit/s/Hz] physical layer efficiency of ModCod i according to Table 13 of [AD-1]
- $\bar{\eta}_2$ [%] average layer 2 encapsulation efficiency which will depend on the adopted layer 2 encapsulation scheme (MPE/MPEG or GSE) and on the typical size of IP packets
- S_i [%] bandwidth share of ModCod i
- $N_R^{(i,j)}$ [bytes] number of rows in the ADT for ModCod i and QoS class j
- $N_C^{(i,j)}$ number of columns in the ADT for ModCod i and QoS class j
- $N_{CR}^{(i,j)}$ number of redundancy columns in the ADT for ModCod i and QoS class j

- η_c [%] efficiency of the LL-FEC coding scheme. This is 100% for MDS codes like RS, typically around 98% for Raptor or LDPC codes.

With the above parameters, the following definitions apply:

- $\bar{R}_i = S_i R_s \eta_i \bar{\eta}_2$ [bps] average equivalent data rate per ModCod after LL-FEC encoding and prior to layer 2 encapsulation (assuming that BBFrames of different ModCods are multiplexed uniformly – if a different scheduling policy is implemented, the previous equation might not be valid).
- $r^{(i,j)} = \frac{N_C^{(i,j)}}{N_C^{(i,j)} + N_{CR}^{(i,j)}}$ LL-FEC code rate for ModCod i and QoS class j
- $ADT^{(i,j)} = 8N_R^{(i,j)} N_C^{(i,j)}$ [bits] size of ADT for ModCod i and QoS class j
- $MTBL^{(i,j)} = \eta_c \frac{ADT^{(i,j)}}{r^{(i,j)}} (1 - r^{(i,j)})$ [bits] maximum tolerable burst length for ModCod i and QoS class j.

Using the previous 2 definitions, the latter can be also written as:

$$MTBL^{(i,j)} = 8\eta_c N_R^{(i,j)} N_{CR}^{(i,j)}$$

- $FECFR^{(i,j)} = 8N_R^{(i,j)} (N_C^{(i,j)} + N_{CR}^{(i,j)})$ [bits] size of LL-FEC Frame for ModCod i and QoS class j
- $\bar{R}^{(i,j)} = \bar{R}_i w^{(i,j)}$ [bps] average equivalent data rate per ModCod and QoS class after LL-FEC encoding and prior to layer 2 encapsulation. The factor $w^{(i,j)}$ account for different possible scheduling strategies:
 - $w^{(i,j)} = 1$ means that the scheduler selects entire FEC Frames for transmission
 - $w^{(i,j)} = 1/N_Q$ (being N_Q the number of QoS classes) corresponds to a round robin approach, resulting in a uniform multiplexing of packets belonging to FEC Frames of different QoS classes, under the assumption that the granularity of the scheduler is much lower than the size of the smallest ADT.

With the definitions given above, and omitting indices i and j in the following for the sake of clarity whenever this is not going to cause ambiguity, the problem of properly sizing the LL-FEC process (i.e. of selecting N_R , N_C and r) can be analytically formulated by means of the two following equations:

$$MTBL \geq P_{out}^{(i,j)} \bar{R}^{(i,j)} \Rightarrow N_R N_{CR} \geq P_{out}^{(i,j)} \bar{R}^{(i,j)} / 8\eta_c$$

$$FECFR \leq \Delta^{(i,j)} \bar{R}^{(i,j)} \Rightarrow N_R (N_C + N_{CR}) \leq \Delta^{(i,j)} \bar{R}^{(i,j)} / 8$$

The following considerations are in order:

- The desired protection level and the extra maximum latency are guaranteed on an **average** basis by the above equations, due to the fact that average data rates values are used because, in order to avoid a loss of generality, an average value for the layer 2 encapsulation efficiency $\bar{\eta}_2$ (which in reality depends on the size of the IP datagrams) has been used. This does not make the present methodology especially suited for jitter-sensitive applications, unless constant size IP datagrams can be assumed for the corresponding QoS classes.
- N_R shall be kept as low as possible to reduce the number of encoding/decoding processes running in parallel at the gateway/terminal side, thus reducing the system complexity
- The code rate shall be maximised in order to increase the system efficiency. This means that N_C/N_{CR} shall be as large as possible.
- The fulfilment of the extra latency requirement expressed by the second equation is subject to the assumption that the system is fully loaded (i.e. the service rate is roughly equal to $\bar{R}^{(i,j)}$). If this is not the case, N_R shall be reduced so that the extra delay requirement is still met (at the expenses of a lower efficiency). According to [i.1], N_R can be reduced on a frame-by-frame basis.

- The methodology assumes a constant share of capacity between ModCods. When a significant change in the share of capacity between ModCods occurs, the FEC Frame dimensioning needs to be adjusted. This implies that the NCC shall distribute new TIMs whenever N_C or N_{CR} shall be changed.

The type of code shall be selected taking into account the parameters range reported in the Table 9 in accordance with [i.1].

1. The type of code shall be selected considering that RS codes can be used only if:

$$P_{out} \bar{R}^{(i,j)} \leq 64 \cdot 1024 \cdot 8 = 512 \text{ kbps}$$

otherwise Raptor shall be used.

2. N_R shall be hence initially determined as the minimum value according to those listed in the tables above which is still greater or equal than

$$P_{out} \bar{R}^{(i,j)} / 8\eta_c \max(N_{CR})$$

where $\max(N_{CR})$ is 64 for RS and 65535 for Raptor.

3. N_{CR} shall be determined as the minimum value according to the ranges listed in the tables above which is still greater or equal than

$$P_{out} \bar{R}^{(i,j)} / 8\eta_c N_R$$

4. N_C shall be determined as the maximum value according to the ranges listed in the tables above which is still lower or equal than

$$\Delta \bar{R}^{(i,j)} / 8N_R - N_{CR}$$

In this way, the highest possible code rate is selected, thus maximising the overall efficiency.

In the case of Raptor codes, N_C should lie in the range between 1000 and 2000 to achieve optimal performance. It might be then necessary to re-iterate steps 1 to 4 by increasing N_R in order to achieve this optimal configuration.

5.3.2.4 Selection of an elementary stream carrying LL-FEC

LL-FEC is applied individually in the context of a TS or GS elementary stream. This allows some elementary streams to apply LL-FEC and some to apply MPE/GSE.

[i.14] indicates directly and indirectly two methods that an RCST can use for selecting an elementary stream:

- a) By recognition of the elementary stream type in the PMT section in the Forward Link Signalling (FLS) service;
- b) By explicit assignment of an elementary stream through the Forward Interaction Path (FIP) descriptor.

In addition, a Multicast Mapping Table (MMT) may be used to select elementary streams for multicast.

LL-FEC uses the MPE parameter fields in a way that is incompatible with standard MPE and may cause legacy RCSTs to fail. It is thus necessary to avoid that a legacy RCST selects and attempts to decode an elementary stream that applies LL-FEC, whereas an LL-FEC- capable RCST must be able to automatically acquire a stream that applies LL-FEC.

A legacy RCST cannot be expected to receive transmissions that apply LL-FEC due to possible 48-bit MAC discrimination. Additionally, irrelevant IP traffic may pass the MAC address filter due to MAC address aliasing.

NOTE: The same type of MAC address incompatibility will not occur with GSE.

5.3.2.4.1 Assignment of LL-FEC TS elementary stream to RCST.

Primarily, the explicit method for assigning an elementary stream seems most feasible when applying LL-FEC for the elementary stream. The RCST, even if LL-FEC compatible, should not be expected to autonomously select and receive the LL-FEC elementary streams indicated in the PMT service description, and may initially choose not to receive any LL-FEC elementary stream until explicitly instructed. There may be LL-FEC elementary streams listed in the PMT service description that are not relevant for a specific RCST at a given time as they may be listed purely for the declaration of the stream type that indicates that LL-FEC is applied.

An elementary stream applying LL-FEC can be recognized in the PMT section of the applicable FLS services by the association of the specific stream type with the specific elementary stream. It is recommended that the value 0x8D (user defined) is interpreted as indicating a LL-FEC stream type; and any value within the range administrated by the normative documents assigned to indicate the LL-FEC stream type should be understood as such an identifier as well, if the meaning of the value is known.

The NCC may recognize the LL-FEC capability of the RCST at the logon request. It may then explicitly assign an elementary stream carrying LL-FEC by use of the FIP in the logon response TIM. An associated declaration of the stream type of each elementary stream in the PMT section with the associated FLS service description indicates explicitly that an elementary stream applies LL-FEC, and hence, the RCST can configure the MAC discrimination accordingly. Alternatively, the MMT may be used to indicate elementary streams used to carry LL-FEC. An associated declaration of the stream type of each elementary stream in the MMT section with the associated FLS service description indicates explicitly that an elementary stream applies LL-FEC, so that the RCST can configure the MAC discrimination accordingly. By default, an RCST will assume that the MMT indicates elementary streams with stream type 0x0D, i.e. carrying MPE with full 48 bit MAC address.

The reduced MAC address size indicated by the combination of an applicable broadcast descriptor and the specific conventions for non-LOS use applies only for the elementary streams that are of a stream type that applies LL-FEC. This concerns multicast traffic as well as unicast traffic.

5.3.2.4.2 Assignment of LL-FEC stream gse_fec_id to RCST.

It is assumed that a GSE receiver will parse all GSE traffic and discriminate non-compatible protocols and irrelevant MAC addresses. A specific assignment mechanism is not required.

5.3.2.5 Use of LL-FEC on GSE generic streams

5.3.2.5.1 Application data transfer over GSE-FEC streams.

For the carriage of application data over GSE-FEC streams, the standard defines a new optional GSE extension header referred to as LL_RCS_FEC_ADT. This extension header carries the real-time parameters as well as the gse_fec_id identifier of the GSE-FEC stream.

This extension header is inserted in GSE packets carrying a full datagram or, in case of fragmentation of the datagram, in the GSE packet carrying the first fragment of the datagram (c.f. Figure 10), as specified by the GSE specification [i.14]. The real-time parameters always make reference to the full datagram: the dt_position field indicates the position in the LL-FEC frame of the first byte of the PDU, the table_boundary flag indicates whether or not the PDU is the last PDU in the Application data table.

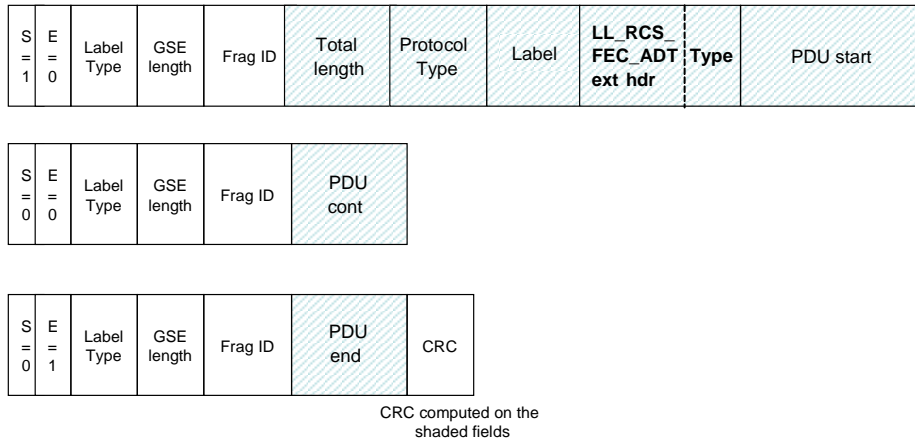


Figure 10 : PDU fragmented over 3 GSE packets

A 2-Byte Type field will always be appended to the LL_RCS_FEC_ADT header (see Figure 11), as for any GSE optional extension header [i.15]. This Type field is identical to the Protocol Type/Extension field of the GSE Header (with the same rules and semantic for values assignment). It indicates the type of the next header or the protocol type of the encapsulated PDU for the last extension header in the packet.

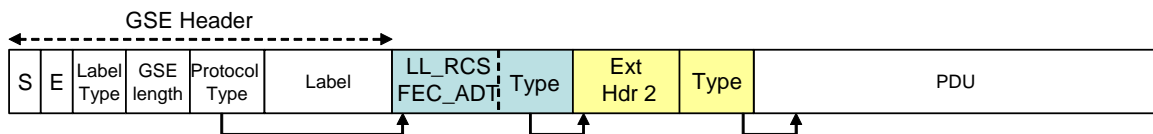


Figure 11: Use of the Type field in the LL_FEC_ADT_extension

5.3.2.5.2 Carriage of parity data over GSE-FEC streams.

For the carriage of parity over GSE-FEC streams, the standard defines a new mandatory GSE extension header referred to as LL_FEC_FDT_extension. This extension header carries information related to the FEC frame and the location of parity data in the FEC frame as well as the gse_fec_id identifier of the GSE-FEC stream.

The format of GSE packets carrying parity data is described in the Figure 12.

As described, the label field is not present in the header of GSE packets carrying parity data (the label_type indicator is set to "10"). Indeed Receivers only need to filter on the gse_fec_id identifier present in the extension header. They will then perform LL-FEC decoding.

1. Parity Data carried over one GSE packet

S = 1	E = 1	Label Type	GSE length	Protocol Type = LL_RCS_FEC_FDT	LL_RCS_FEC_FDT info (gse_fec_id, RS or Raptor info)	Parity Data	Parity_data_CRC_32
-------	-------	------------	------------	--------------------------------	---	-------------	--------------------

CRC computed on the shaded fields

2. Parity Data fragmented over several GSE packets

S = 1	E = 0	Label Type	GSE length	Frag ID	Total length	Protocol Type = LL_RCS_FEC_FDT	LL_RCS_FEC_FDT info (gse_fec_id, RS or Raptor info)	Parity Data Start
-------	-------	------------	------------	---------	--------------	--------------------------------	---	-------------------

S = 0	E = 0	Label Type	GSE length	Frag ID	Parity Data cont
-------	-------	------------	------------	---------	------------------

S = 0	E = 1	Label Type	GSE length	Frag ID	Parity Data end	CRC
-------	-------	------------	------------	---------	-----------------	-----

CRC computed on the shaded fields

Figure 12: Carriage of parity data over GSE packets

The carriage of parity data will be realised as follows:

- For Reed-Salomon parity data, LL_FEC will provide the GSE encapsulator with FDT columns (seen by the encapsulator as separate PDUs). The GSE encapsulator will fragment if necessary, and encapsulate independently each FDT column in GSE packets.
- For Raptor code parity data, LL_FEC will provide the GSE encapsulator with repair symbols or groups of repair symbols, i.e. one FDT column or a group of several consecutive FDT columns. The GSE encapsulator will fragment if necessary, and encapsulate independently each repair symbol or group of repair symbol.

In case of fragmentation, the information related to the location of the Parity Data in the FDT table (which are inserted in the GSE packet carrying the first fragment) always make reference to the “full parity data” (i.e. FDT column, repair symbol or group of repair symbols).

5.3.2.5.3 Use of the CRC_32 extension header

GSE assumes that the physical/link layer below can ensure a sufficient error detection probability like in DVB-RCS/S2 fixed systems (DVB-S2 FEC code, based on BCH+LDPC, allows Quasi-Error-Free operation). However, in NLOS conditions this assumption is not valid, and GSE packets may contain bit errors.

When received parity and application data are inserted by Receivers in the LL_FEC frame, they are marked by the LL-FEC process as reliable. However as GSE packets may contain bit errors and the PDU reliability is not checked in all cases, bit errors may be inserted (and marked as reliable) in the table. This could impair the LL-FEC performances.

The only case where the reliability of GSE packets is not checked is for packets carrying non-fragmented application data. For fragmented application and parity data, the GSE native CRC-32, which is appended at the end of the GSE packet containing the last fragment, allows checking the reliability of these data. For non-fragmented parity data, a CRC-32 has been defined in the GSE packet format, part of the LL_RCS_FEC_FDT extension header, and allows checking the reliability of these data.

In order to counteract this issue, the standard defines a new optional GSE extension header referred to as LL_CRC32_extension. This extension header may be used in GSE packets carrying Application data, but only if they carry a non-fragmented PDU.

This CRC_32 value is computed using a method equivalent to that used for the CRC_32 defined by GSE and with a similar scope (it is computed over all bytes be starting from the GSE length field (included) to the end of the PDU, but not including the CRC extension header field).

Similar to the case the LL_RCS_FEC_ADT extension header, a 2-Byte Type field will be always appended to it. It will indicate the type of the next header or PDU. This extension header may be equally inserted before or after the LL_RCS_FEC_ADT extension header.

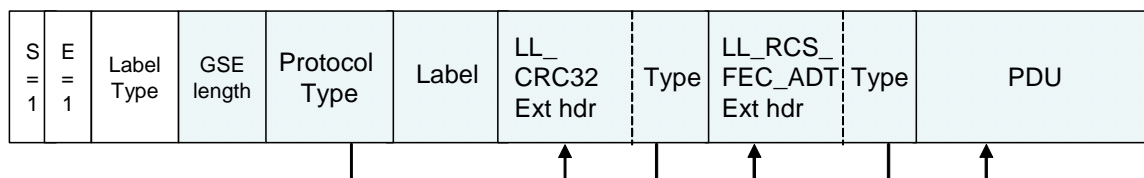


Figure 13. Use of the LL_CRC32 extension header in GSE packet carrying a non-fragmented datagram (from application data)

Note: The CRC 32 defined by GSE as well as those from the LL_CRC_32 and LL_RCS_FEC_FDT extension headers (which have been defined with a scope of protection similar to the GSE CRC's scope) are not computed over all the bytes of GSE packets (as shown on the previous figures). This may induce residual errors (i.e. not detected), however this should barely affect the LL-FEC performances.

5.4.3 Protection of the signalling

No channel countermeasures have been adopted for the protection of signalling. As a general rule, the adoption of a link margin of the same magnitude as the Rice channel amplitude variations is recommended in order to protect the signalling from the losses due to these Rice channel variations both on forward and return paths.

QoS requirements for signalling are presented in section 10.2 .

5.4 Guidelines for the use of VCM/ACM in mobile scenarios

Fade Mitigation Techniques (FMTs) are a set of techniques that help a system adapt its physical layer to the propagation channel variations due to tropospheric attenuation (e.g., rain, clouds etc.). These techniques (such as ACM) allow maximising the link capacity in clear sky while maintaining the required availability during unfavourable propagation conditions. FMTs require a control mechanism, called "FMT control loop", to monitor the state of the channel and to command the activation of the technique itself.

The mobility of the RCST introduces an additional time-dependency in the propagation channel. This needs to be taken into account when designing the FMT control loop. In principle, FMTs cannot be used to counteract mobility effects such as multipath, blockage and shadowing encountered due to the local environment in the vicinity of the mobile RCST. Therefore, in mobile DVB-RCS reference scenarios, the FMTs can be applied to counteract only the tropospheric attenuation, such as rain attenuation. Particularly, in the aeronautical scenario, where rain attenuation is not relevant at cruising altitude, FMTs can be applied to track the satellite antenna EIRP variation over the coverage.

The design of a reliable FMT control loop is based on the knowledge of the actual dynamics (that is, second-order statistics) of fading, such as the fade slope and fade duration. The ITU-R Recommendation P.1623-1 [P.1623-1] proposes a model for the prediction of the rain fade slope in classical Ku-band and Ka-band FSS. Based on this analysis and its extrapolation to mobile conditions [i.17], it is possible to transform the second-order rain attenuation statistics from a fixed to a mobile satellite system. This transformation is possible through the following extrapolation factor, here termed as Slope Ratio (SR):

$$SR = \frac{\text{fade slope in mobile}}{\text{fade slope in fixed}} = \frac{|v_M - v_R \cdot \cos \phi|}{v_R}$$

where $\vec{v}_M = v_M \exp(i\theta_M)$ is the velocity vector of the mobile RCST, $\vec{v}_R = v_R \exp(i\theta_R)$ is the velocity vector of the raincells movement (v_R also called “advection” or “front speed”) and ϕ is the angle between these two vectors.

Therefore, for the prediction of the FMT control loop margins in a mobile DVB-RCS system, it is important to know the following items:

- i) the second-order rain attenuation statistics of a fixed satellite system operating under the same conditions (i.e., geographic location and system characteristics);
- ii) the front speed, the mobile RCST speed, the relative direction of their motions and the kind of their motions (i.e., rectilinear, zig-zag, circular, etc).

Based on the results shown in [i.18] (specifically for the railway scenario), the FMT control loop margin needs to be increased in a mobile DVB-RCS system under LOS conditions, due to the relative increase of the rain fade slope by a factor of $SR=6$. Moreover, the FMT control loop margin required due to mobility can be adaptive based on the actual train speed and the rain attenuation level. In non-LOS conditions referring, particularly, to fast fades due to power arches, the absence of channel estimation for few tens of milliseconds will not affect the FMT control loop. However, in the case of long blockages due to tunnels, the use of a short-term rain fade prediction method allows estimation of rain fading statistics after few seconds of outage. Assuming the worst case when the rain fading always keeps increasing while the train is in blockage, the FMT control loop margin at the moment of the train’s exit from the tunnel has been calculated and shown to increase as the train speed, or the initial actual rain attenuation levels (at the train’s entrance to the tunnel), or the blockage duration increases. For example, for a train speed $v_M=250$ km/h, blockage of length 350 m (that is, duration of 5 s) and 3 dB of actual rain attenuation level at the moment of the train’s entrance to the tunnel, the required FMT control loop margin at the moment of the train’s exit from the tunnel is 1.5 dB.

In maritime scenario, rain fade can be counteracted in the same way as it is done for the FSS. The low speed of the mobile RCST in this scenario leads to a slight increase in fade slope by a factor $SR=1.5$. Therefore, only 0.5 dB of margin due to this fade slope increase is required in the FMT control loop.

In aeronautical scenario, the link can be adapted to the different antenna gain over the coverage. The C/N variation caused by the antenna roll-off is slow enough to be safely counteracted with a control loop based on the C/N estimation without the need of additional margins. If the RCST position is known, this information can be used by the GW or by the terminal to activate the appropriate MODCOD without the need of channel estimation or an FMT control loop.

6 Return link

6.1 Doppler drift and time correction

Annex L to the general DVB-RCS guidelines [i.3] presents a feasibility analysis of mobile use of the standard. As suggested there, few if any special considerations for mobility are necessary in the forward link. There are however some considerations for the return link; these are addressed in general terms in [i.3] and in more detail in the following.

6.1.1 Carrier Frequency Doppler Shift

6.1.1.1 Log-On

Annex L [i.3] derives lower limits for the symbol rates that can be supported without Doppler compensation. These are determined by the carrier frequency tolerance of the hub, and are in some cases quite high. On the other hand, upperbounds the symbol rates are set due to the EIRP limitations in combination with the anticipated space segment characteristics. In some cases, there is only a narrow range of possible rates left. This is clearly unattractive in terms of system design flexibility. It is difficult to increase the receiver’s frequency tolerance substantially beyond the 3% assumed in [i.3]; hence, we recommend the use of Doppler pre-compensation in the return link where appropriate. This can be implemented with relative ease, provided that navigation information is available. With such compensation, bound on the symbol rate are set e.g. by phase noise considerations, just as they are in fixed implementations.

The Doppler shift in the return link can be determined from the link geometry and the velocities of the terminal and satellite, using conventional solid geometry. The Doppler shift is proportional to the relative speed of the satellite and terminal. It should be noted that the total frequency error at the hub receiver is determined not only by the Doppler shift in the return link, but also has a contribution from the forward link, resulting from the Doppler shift on the NCR clock.

As shown in [i.3], the worst-case Doppler is experienced by aeronautical terminals. For these, the terminal motion dominates the Doppler shift. In the worst case, the Doppler shift is therefore given by $f v/c$, where f is the frequency, v is the terminal speed and c is the speed of light. With an initial frequency tolerance of 3% as suggested in [i.3], a 100 kBaud carrier can tolerate an offset of 3 kHz. In turn, this corresponds to an error in the terminal's speed estimate of 30 m/s or 108 km/hr. This is well in excess of the error that can be expected, for example, from satellite navigation equipment. Estimating the initial frequency offset due to Doppler with sufficient accuracy should therefore not be a problem in practice.

6.1.1.2 Carrier Frequency Synchronisation Maintenance

Following logon, the terminal's transmit frequency can be controlled by closed-loop mechanisms already present in DVB-RCS. The worst-case Doppler rate identified in [i.3] is 1700 Hz/s. A closed-loop frequency control can therefore maintain the carrier frequency within 3 kHz, provided the latency (update interval plus two-way propagation delay) of the loop is less than approximately 1.75 seconds.

For typical implementations, these tolerances are relaxed in proportion to the return link symbol rate.

6.1.2 Symbol Rate Doppler Shift

According to [i.3], the worst-case relative Doppler shift, which is experienced by the aeronautical terminal, is 1 ppm. This is considerably smaller than the 20 ppm symbol rate tolerance allowed by the normative document (clause 6.1.4), and is therefore not an issue for system synchronisation.

6.1.3 Burst Timing

6.1.3.1 Log-On

The ordinary log-on process for DVB-RCS RCST's relies on accurate knowledge of the position of both the RCST and the satellite, in order to allow transmission of the CSC burst within a window that is at most one or two milliseconds long. In certain mobile applications, it is highly desirable to allow a much wider acquisition window. Reasons for this include a desire to be independent of for example accurate satellite ephemeris data and/or positioning information. It can be noted that the timing uncertainty applies to all types of mobile terminals, including for example maritime terminals that may not need the navigation information for correction of carrier frequency Doppler shift.

Large initial timing uncertainty may also arise where accurate ephemeris data is readily available. This includes, but is not always limited to inclined-orbit satellites. It can be noted that use of inclined-orbit satellites is particularly attractive for mobile applications, where the RCST most often needs a tracking antenna anyway.

The standard has some limitations that impede a straightforward extension of the initial timing tolerance. The leading-edge guard interval (`burst_start_offset`) of the CSC slot is limited to ~ 1.42 ms, because the corresponding field in the Timeslot Composition Table is only 16 bits wide. While the time slot can be up to 364 ms long, the leading-edge guard interval limitation effectively limits the tolerance. In addition, the maximum time correction that is possible using the Correction Message Descriptor is 127×2^7 PCR ticks, or ~ 439 μ s.

Assuming that any NCC/satellite path delay variation is handled either by making use of the optional NCR packet payload or by appropriate adjustments of the NCC timing, the uncertainty in the absence of positioning or ephemeris information is determined by the path delay variation between possible terminal locations and that caused by satellite motion. The one-way delay variation between a satellite directly overhead and one on the horizon is ~ 20 ms. The daily delay variation of a satellite in a 10° inclined orbit is around 7.5 ms (for a user location that has the satellite on the horizon). The worst-case uncertainty is therefore around 28 ms — i.e., much bigger than what can be handled by the current method.

In the past, TDMA systems have employed techniques such as dedicated log-on carriers with completely open acquisition windows in order to overcome the initial uncertainty. Adding such a feature is unattractive for DVB-RCS, both in terms of modification of the standard and equipment, and because it requires dedicated bandwidth, which does not generate revenue. Instead, the method described in the following has been adopted. Details are defined in clause 7.3a of the normative document.

The method is illustrated in Figure 14. The system is configured with a number of contiguous, normal-length CSC slots that together cover an interval at least as long as the anticipated uncertainty. When operating in this mode, the RCST always aims for a CSC slot as close to the middle as possible. Due to the uncertainty, the burst may however be

received in another slot. To resolve this ambiguity, the Correction Message Extension descriptor (clause 8.5.5.10.20 of the normative document) is sent in the logon TIM. This descriptor identifies the slot in which the CSC burst was received. It is sent to the RCST in addition to the usual Correction Message Descriptor. This information enables the RCST to compute the overall timing correction required before proceeding to coarse or fine acquisition.

In order to prevent RCST's that require this large initial uncertainty from attempting to log on to systems that do not support it, NCC support for this feature is signalled in the Superframe Composition Table. The feature can thus be made available on a per-superframe basis.

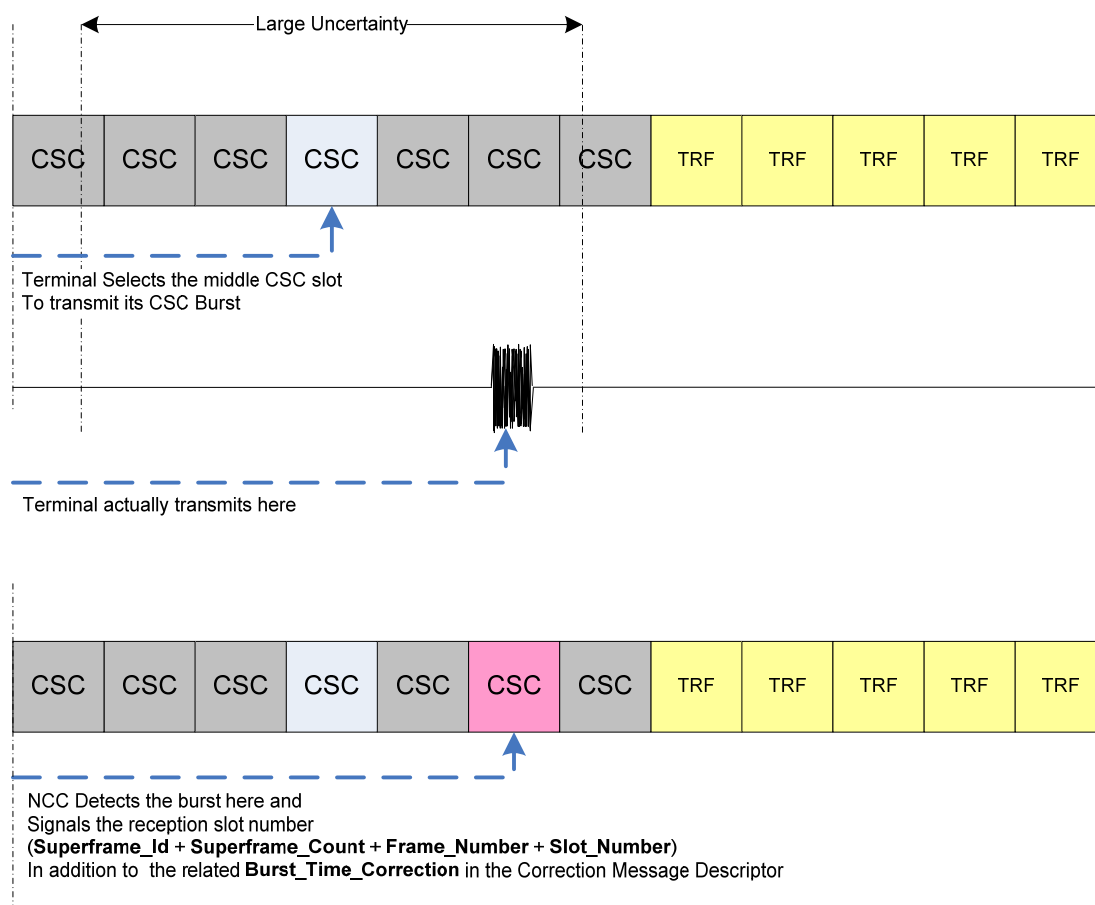


Figure 14: Log-on method for terminals with large initial uncertainty.

The method is completely backwards compatible. RCST's that do not need the extra tolerance can log on in exactly the same way as they currently do, and will simply ignore the Correction Message Extension descriptor that tells them which slot they hit.

It may happen that the CSC burst is transmitted across a slot boundary. Depending on the NCC receiver implementation, this may result in the loss of the burst. This situation is handled by the regular CSC retransmission scheme; however, the RCST should ideally use a back-off that is not an integer multiple of the slot duration, in order to avoid continually selecting locations that cross slot boundaries.

6.1.3.2 Timing Synchronisation Maintenance

Once initial acquisition has been performed, the closed-loop timing of DVB-RCS can maintain synchronisation. The worst-case timing drift identified in [i.3] is 1100 ns/s. For a 4 Msps carrier and a loop latency (update interval + two-way propagation delay) of 1 second, the maximum drift is approximately 4 symbol periods. Uncertainties of this magnitude can be accommodated by minor adjustments of the guard intervals, without dramatic effect on the spectral

efficiency. Expressed in symbol periods, the uncertainty is proportional to the data rate, so it is correspondingly smaller for carriers with symbol rates lower than 4 Msps.

6.2 Spectrum spreading in the return link

In Mobile Line-of-sight scenarios, return link carriers may require the use of spectrum spreading in order to reduce the spectral density and in particular off-axis.

Indeed, the use of very small antennas necessary to the mobile applications in Ku Band is incompatible with the off-axis emissions defined in the ETSI and FCC rules with the existing DVB-RCS waveform.

The spreading technique to be used is described in clause 6.5.5 of [i.1].

6.2.1 Spreading description

The spectral spreading is achieved by two means. The first consists of the use of $\pi/2$ -BPSK modulation. This is equivalent to spreading a QPSK modulated signal by a factor 2.

The second solution consists in increasing the symbol rate of the signal by a factor N without increasing the power. This modification reduces the E_s/N_0 at the receiver side. In order to recover the required E_s/N_0 , the signal is repeated N times at the transmitter side.

In the DVB-RCS application, the signal is repeated in a burst by burst basis.

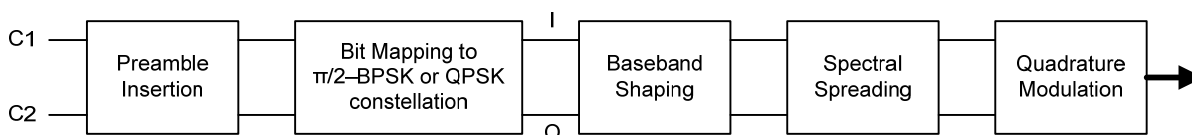


Figure 15: Post-encoder processing with spectrum spreading

The signal after spreading looks like in the following figure :

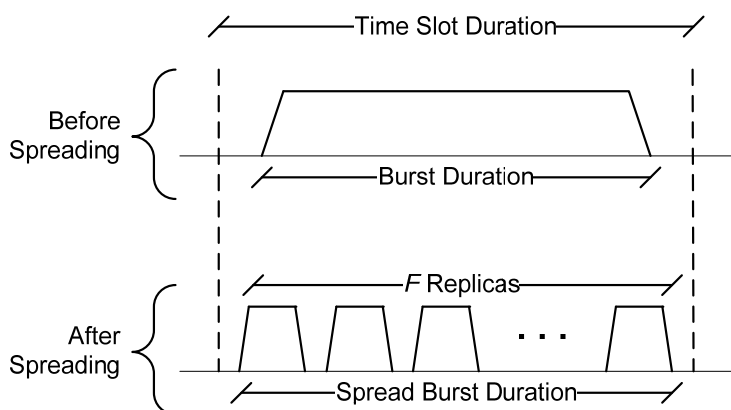


Figure 16: Spectrum spreading by burst repetition

In order to avoid discrete lines in the output spectrum, a random phase shift shall be applied to each replica before transmission. This has no impact on the despreading since the relative phases between replicas are estimated in the receiver.

6.2.2 Despreading description

Despreading is achieved by summing together homologous signal samples of successive replicas.

Prerequisite of the approach is that the timing error and Doppler of the transmitter with respect to the gateway is lower than what would cause an accumulated timing phase error during the duration of the burst replicas. This is actually the

case in typical operational scenarios. In fact, even considering the worst case Doppler experienced in the aeronautical domain (1100 ns/s), more than of 45000 symbols would be required for timing errors to accumulate up to a value larger than 10% of the symbol period.

Obviously, the combining shall be such that homologous samples are added in phase. In this regard, a phase alignment is required before combining.

The principle for the recombination is shown in Figure 17 :

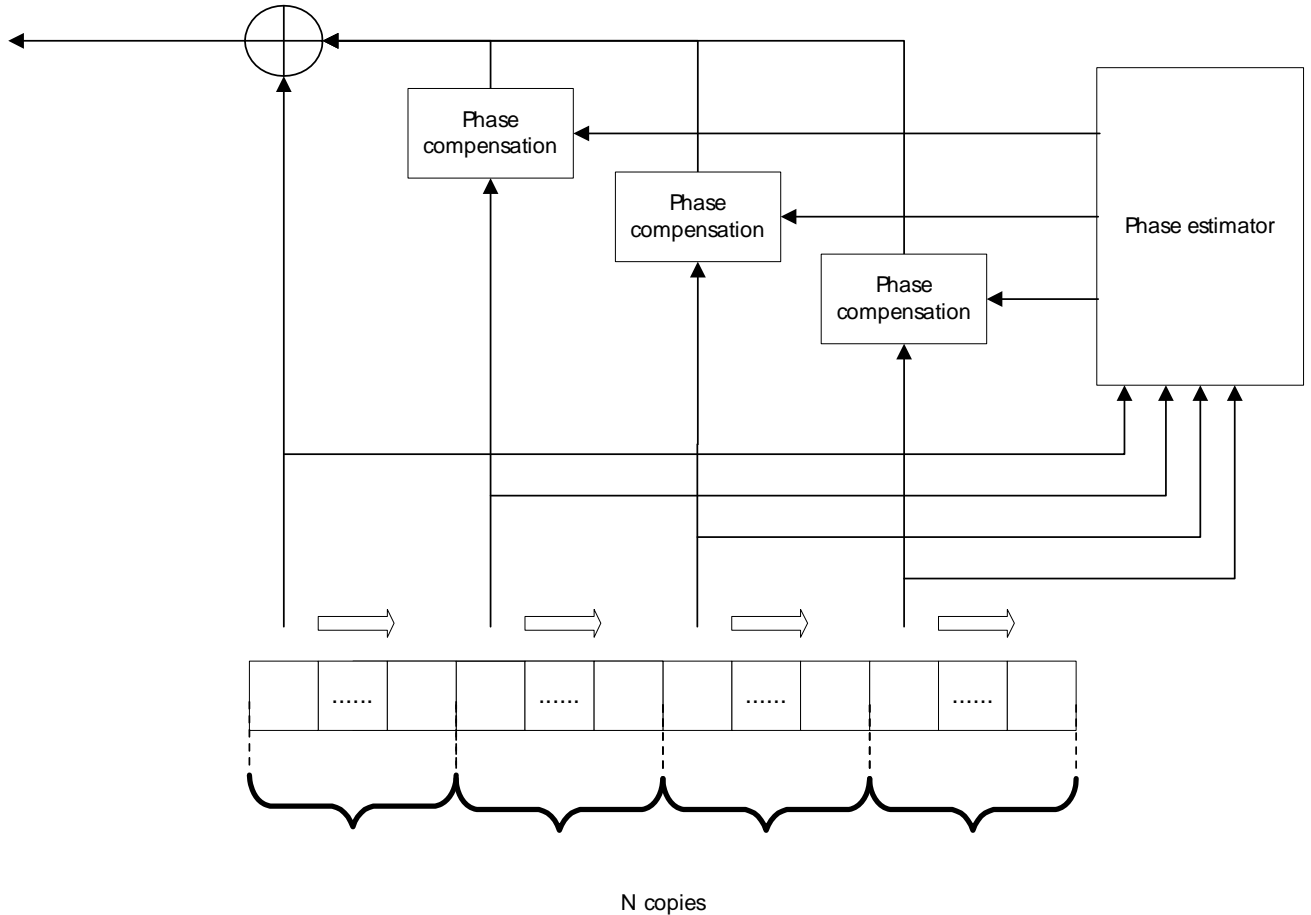


Figure 17: Coherent recombination

The despreading solution by recombination needs to memorize at least 2 replicas in order to perform the relative phase estimation. This estimation can be done via a block correlator because what we need is $\exp(j\Delta\phi)$ and not $\Delta\phi$ itself. An iterative recombining scheme will minimize the required memory.

In order to have a good phase alignment, the differential frequency shift between replicas shall be as low as possible. The corresponding frequency shift tolerance will depend on the burst size.

The frequency tolerance and the phase-shift tolerance are related by the following formula :

$$\Delta f \cdot T_s = \frac{\Delta\phi}{\pi \cdot N_s} \quad (5)$$

where N_s is the number of symbols inside a replica, T_s is the symbol duration after spreading, $\Delta\phi$ is the maximal phase tolerance and Δf is the frequency tolerance between replicas.

The longer the burst, the lower the frequency tolerance.

Typically, in order to have negligible degradations, a 15° error phase between replicas have been considered. Considering a 512 kBaud carrier and a 15° tolerance, the maximal frequency tolerance between 2 consecutive replicas is 45 Hz for a burst of 1000 symbols length. This value is compatible with a 1 kHz/s frequency drift in Ku-Band.

After the despreading, the estimations of the absolute delay, frequency and phase are done by conventional timing and carrier recovery algorithms, so a classical DVB-RCS demodulator can be used as shown in Figure 18 :

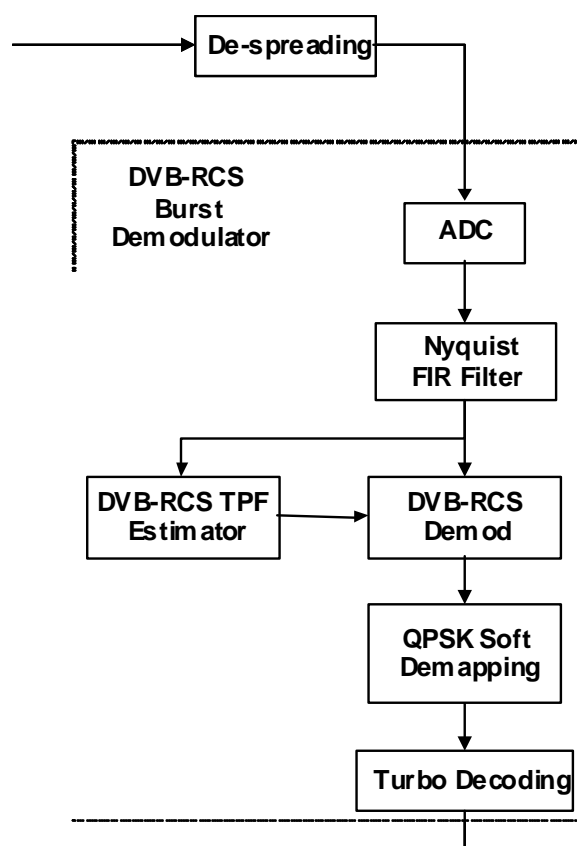


Figure 18: Receiver architecture including despreading

6.3 Return link countermeasures for Non-LOS mobile scenarios

The following techniques can be used to counteract signal blockage in Non-LOS scenarios. The first technique, proactive retransmission, is applicable for the MF-TDMA mode of DVB-RCS. The second technique, LLFEC, is applicable for the continuous carrier mode operation.

6.3.1 Proactive retransmission technique

The Proactive Retransmission technique is an outage countermeasure suitable for the return link of a DVB-RCS system operating in mobile scenarios.

In its basic form, it consists in the disabling of the RCS transmitter when an outage event is detected. During the fading event, data are buffered in the transmitter and only transmitted when the fading event is over. In this form of the technique only the physical layer of the terminal is involved. Variants to this mechanism, involving multiple protocol layers, can be foreseen to further improve the performances and will be described briefly.

Proactive retransmission is made possible by the correlation of fading due to signal blockage or shadowing between FL and RL. As the FL fading level can be readily estimated by suitable processing of the received DVB-S2 carrier, a threshold on the measured FL fading can be set to trigger the Proactive Retransmission mechanism. In this regard, the

fading level measurement can be done by monitoring the FL carrier SNIR. A SNIR measurement averaged over a single DVB-S2 frame (assuming the availability of the optional pilot symbols) is typically sufficient for the purpose as evident in Figure 19, which shows the mean and standard deviation of the DA-SNORE [i.16] SNIR estimator operating on the pilot symbols of a single PLFRAME. In fact, an error standard deviation of about 0.27 dB would be achieved at 0 dB input SNR. The standard deviation would increase to about 0.4 dB at an input SNR of -5 dB and 0.6 dB at -10 dB input SNR.

Since the multipath component of the fading is uncorrelated between FL and RL, the error in the RL fading estimation is obviously larger than the FL SNIR estimation error. However, for a C/M ratio of 17 dB the probability that the multipath fading exceeds 2 dB is well below 10^{-3} . Regardless, a suitable margin to overcome the multipath fading on the RL shall be available to minimize the packet loss due to the multipath fading, which cannot be countered by the Proactive scheme.

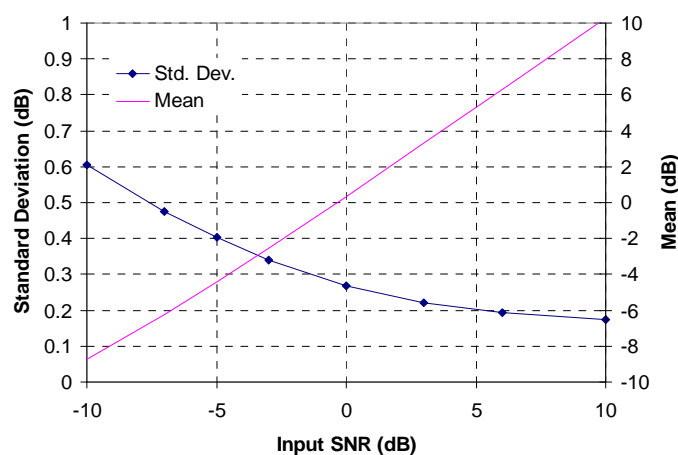


Figure 19: DA-SNORE estimation error standard deviation (for QPSK modulated normal FECFRAME).

The other condition necessary to enable this technique is that the FL DVB-S2 signal should be rapidly re-acquired after a short deep fading event. If timing is maintained, frame resynchronization can be carried out within one frame with suitable re-acquisition techniques implemented at the demodulator level. As the DVB-S2 frame time is in the order of 1 to 2 ms for typical system parameters we can neglect the time for DVB-S2 signal re-acquisition after short link interruptions. In fact, re-acquisition time is less than or comparable with the typical length of a DVB-RCS burst.

The main advantage of the Proactive retransmission technique is that it may allow faster retransmission with respect to what is possible if either standard TCP or I-PEP recovery mechanisms are relied upon instead. In particular, it is well known that TCP recovery mechanisms may negatively affect the connection throughput and experienced latency particularly if congestion control mechanisms like slow-start or congestion avoidance are triggered due to the packet losses caused by the link fading instead than true congestion. Proactive retransmission has the capability to minimize such events particularly when frequent short fading events are foreseen as it typically happens in the railway scenario due to the effects of the power arches.

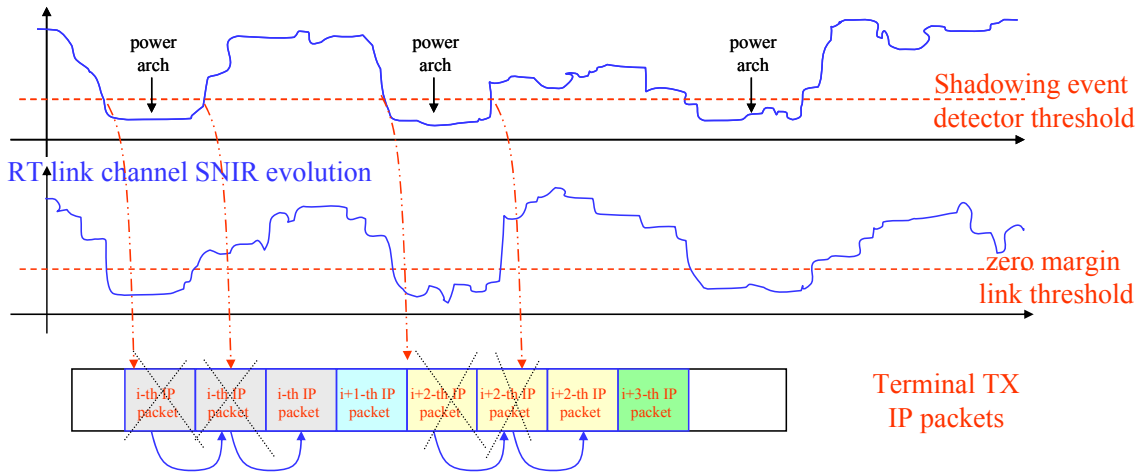


Figure 20: Pictorial description of the proactive retransmission scheme in the railway scenario

6.3.1.1 Proactive retransmission performance in the railway scenario

In order to test the effectiveness of the proactive retransmission scheme a simple railway scenario has been simulated. Figure 20 shows a pictorial description of the technique.. A single GW station managing multiple trains was considered in the following simulations. Also, in order to minimize the simulation time, a single DVB-RCS carrier is assumed available in the system. The Reno TCP with SACK option has been used in addition to the ECN and large window options.

An example result obtained in a scenario consisting of six terminals (trains) and a gateway is presented in Figure 21 and Figure 22. A single MF-TDMA carrier was (512 Kbit/s bit rate) considered. Moreover, each train terminal was assumed running an FTP application.

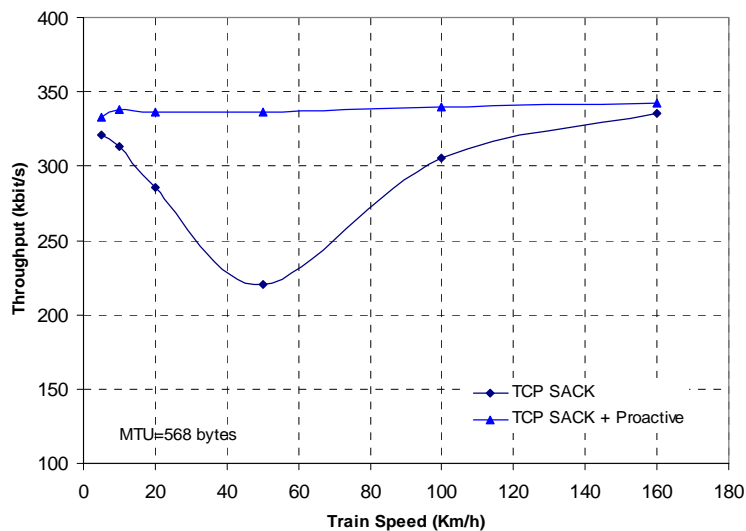


Figure 21: Throughput Comparison of proactive versus conventional scheme. TCP-SACK (MTU=568 bytes). Link Margin: 3 dB. RCS burst size: ATM-2

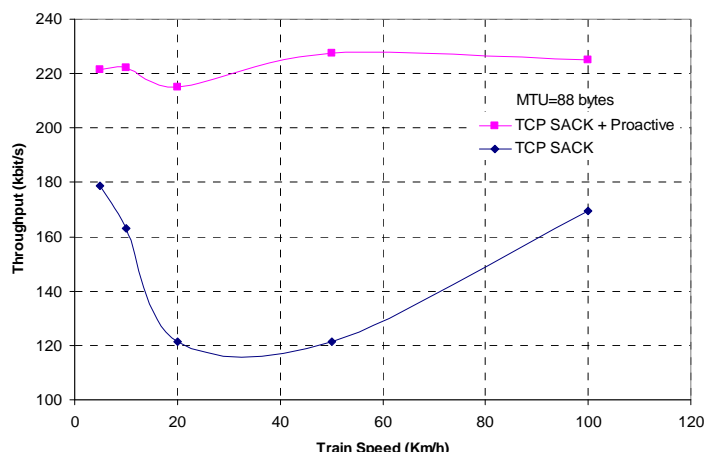


Figure 22: Throughput Comparison of proactive versus conventional scheme. TCP-SACK (MTU=88 bytes). Link Margin: 3 dB. RCS burst size: ATM-2

It shall be observed that, given the possible segmentation of an IP packet on multiple ATM cells / bursts and the lack of mechanism in ATM/AAL5 IP encapsulation for supporting reassembly of IP packet in presence of out of order ATM cells, proactive retransmission shall be performed in such a way that the whole IP packet is retransmitted even if a part of that IP packet has been already successfully delivered to the receiving end.

The worst performance achieved with the shorter MTU (88 bytes) are due to the losses produced by multipath and are thus related to the link margin on the up-link. Although the number of lost packets due to multipath is not higher than those lost with MTU=568, the effects of the loss are higher as the TCP slow start mechanism require more time for shorter MTU to achieve the maximum link speed.

According to these results, proactive retransmission provides a clear advantage compared to the classical TCP. However, to further improve the performance of the scheme when long outages are experienced some cross-layer optimization is required. In fact, the efficiency of the Proactive retransmission can vanish in presence of long fading events due to the fact that TCP retransmission timers will expire and trigger retransmission at the TCP layer. This will trigger the congestion avoidance mechanisms which will reduce the throughput and hence the recovery when the outage is over). An improved Proactive scheme can thus also envisage the freezing of the TCP retransmission timer to avoid the kick-off of the congestion avoidance mechanisms.

6.3.2 Link Layer FEC protection in the return link

In continuous carrier mode, Link layer FEC can be used to counteract the effects of signal fading according to the provisions described in [i.1]. Appropriate tailoring of the LL-FEC parameters is required to match the return link symbol rate and channel dynamics.

6.3.3 Protection of signalling in the return link

No specific countermeasures have been adopted for the protection of signaling. As a general rule, the adoption of a link margin of the same magnitude as the Rice channel amplitude variations is recommended in order to protect the signalling (CSC, ACQ, SYNC) from the losses due to these Rice channel variations both on the forward and on the return path.

QoS requirements for signalling are presented in section 10.2 .

7 Synchronization procedures

7.1 Logon in the presence of a large timing uncertainty

A procedure that allows log-on in the presence of a large initial timing uncertainty, in a completely backwards compatible manner, is explained in section 6.1. It should be noted that the method may have applicability beyond mobile systems; e.g., to simplify terminal installation.

7.2 Logoff procedure

Loss of a number of consecutive CMT's is considered a failure of the synchronisation maintenance procedure and causes the RCST to log off. A typical setting for fixed systems is to log the RCST off after three consecutive losses. The number is primarily determined by the tolerable time and frequency drifts, which are corrected by the messages contained in the TIM. For mobile systems, a suitable number of allowed losses before log-off is determined by the equipment stability and factors such as the maximum terminal speed, which in turn determines Doppler shift and timing drift rate.

8 Control and management

8.1 Control and monitoring functions for mobile terminals

The primary purpose of which Control and Monitoring functions (CMF) is to ensure that there is no harmful interference to other MSS terminals or other services. Such interference could arise from either equipment malfunction or from MSS terminal operation in proximity to other services with which it shares the spectrum allocation on secondary basis in the K_u band, and both secondary and primary basis in the K_a band, depending on the radio region.

The mobile architectures and proposed methods to perform the required control and monitoring functions is described in this guidelines section. As background, this section begins with a review of the interference scenarios from which the requirement for CMF arises.

8.1.1 Interference Scenarios

The interference scenarios for the broadband MSS stems from the regulatory environment established by the ITU-R, in particular for the Aeronautical Mobile Satellite Service (AMSS) [i.23]. Based on these regulatory requirements, the resulting interference scenarios can be grouped into two general categories:

- interference to FSS services, including VSAT's
- interference to other services, including terrestrial fixed service (FS) and specialised scientific services.

The key proviso of these satellite mobile broadband services, especially for K_u band, is that they operate on a strictly non-interference basis, secondary to FSS as well as all other primary or secondary services in the particular frequency segment. For the aeronautical mobile satellite service in particular, ITU-R recommendation M.1643 [i.23] specifically requires the use of interference mitigation measures including continuous monitoring and control by a Network Control and monitoring facility, as well as mobile terminal self-monitoring, to prevent any harmful interference.

8.1.1.1 FSS Interference Environment

Sharing of the FSS spectrum on a secondary basis effectively places the equivalent requirement of fixed VSAT terminals onto the mobile terminals; namely to operate within the off-axis EIRP density limits which in turn impacts on antenna pointing and sidelobe performance. In general, the recommendations are based on 3° adjacent satellite spacing for Europe and Asia and 2° for North America, although the interference levels are ultimately determined on a case by case basis as part of the coordination process. For K_a band, the reference spacing is universally set at 2°. The applicable FSS off-axis emission limits are specified by the ETSI mobile standards for 3° spacing [i.19][i.20][i.21][i.22] the ITU-R FSS recommendations for 2° and 3° orbit spacing [i.24], and the FCC CFR Title 47 part 25 [i.25].

In general, spectrum sharing for the service-link involves the return path uplink and forward path downlink. The geometry of the FSS interference scenarios is illustrated in Figure 23, using an aeronautical terminal (Aeronautical Earth Station, AES) for the mobile. However, this geometry applies equally to other types of mobiles.

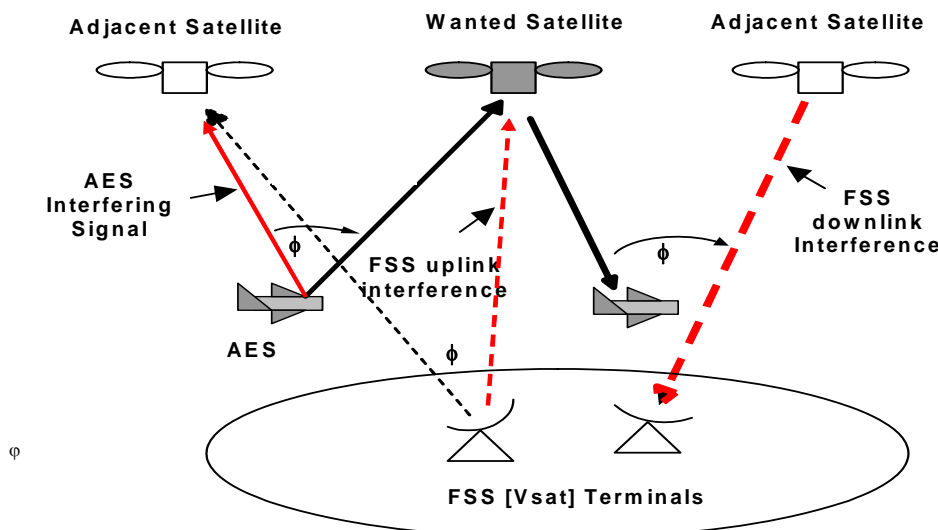


Figure 23: Mobile FSS interference Geometry

The uplink interference paths include the mutual off-axis emissions from the mobile and VSAT terminals into each others' satellites. Since licensed fixed VSAT type terminals comply with the FSS off-axis emission limits, there will not be any harmful interference into the mobile satellite. However, the use of mobile terminals with compact asymmetrical apertures can result in harmful off-axis interference into the adjacent satellites, depending on the effective size of the mobile transmitting aperture parallel to the geostationary arc. This is due to reduced sidelobe discrimination and to the accuracy of antenna tracking and pointing.

On the downlink, the mobile terminal is susceptible to the interference from the adjacent satellite into the mobile terminal, mainly due to the reduced sidelobe discrimination of the receiving mobile antenna aperture.

In order to mitigate the occurrence of harmful interference, the use of interference mitigation techniques are required to reduce both the EIRP density and the interference susceptibility. One such technique is spectrum spreading. This is the primary reason for the inclusion of this technique in the standard.

In order to prevent the occurrence of harmful interference due to the mobile return path uplink, even with interference mitigation measures, a Network Control Center (NCC) continuously monitors and controls the operation of the mobile terminals as described in section 8.1.2.

8.1.1.2 Terrestrial and Scientific Interference Environment

The other category of interference scenarios involves the sharing by MSS with the other services, particularly in K_u band, on a secondary or co-primary basis. The rules are particularly strict for the aeronautical mobile satellite service (AMSS). All other services sharing the FSS allocation are protected, including Fixed Service (FS), Radio Astronomy Service (RAS) and Space Research Service (SRS), as identified in [i.23]. These services – in particular the scientific services – have strict Power Flux Density (PFD) limits. These services are not present in the K_a -band FSS segment.

The results of previous studies of these scenarios reveal that significant additional isolation is required to avoid harmful interference into these other services from an MSS terminal operating in proximity to their stations. This is particularly the case for aeronautical mobiles — even for aircrafts at cruising altitude. However, a key mitigating factor in these interference scenarios is the fact that these other services are localised geographically. Moreover, in cases of RAS and SRS, they are even limited to specific frequencies at specific time intervals.

Based on the previous studies, it is concluded that the most effective method of avoiding harmful interference is by defining an exclusion zone around the stations of these services. Within the exclusion zones, all transmissions will cease or – if possible – switch to other frequencies sufficiently separated from the frequencies used by these other services. Note that the RAS interference scenario is not co-channel; and hence, using another frequency may not be sufficient. The change in frequency can be augmented by a reduction in power. However, this will result in a reduction in supported data rates. The signalling adopted in the standard however allows for all these possibilities.

Obviously, another approach is to avoid the exclusion zones altogether if it is possible for the mobile terminal to alter its path accordingly. Since the routes for many mobile systems are predetermined (such as railways and air traffic flight paths), this may not in general be an effective measure.

8.1.2 CMF and Interference Mitigation Techniques

The reference architecture of the mobile satellite system consists of individual satellite sub-networks forming a global network as illustrated in Figure 24 for a two-satellite network.

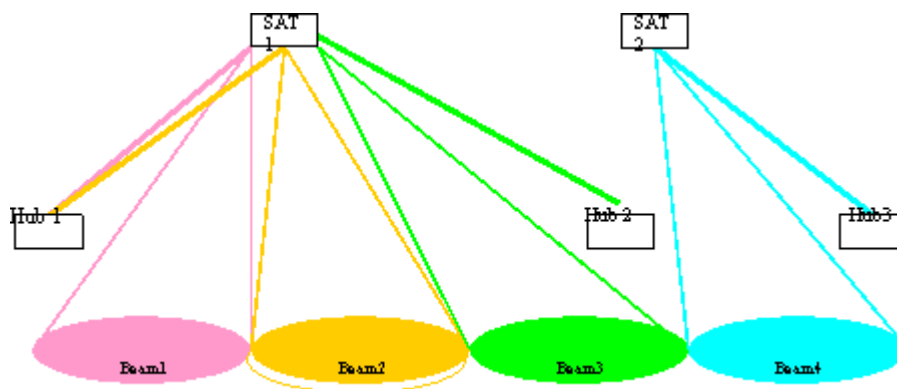


Figure 24: Reference Global Mobile Architecture (2 satellites)

This figure illustrates the different types of satellite sub-networks, including Sat1 with multi-beam coverage and two hub stations, while Sat 2 has a single beam and one hub station. Each satellite sub-network includes its own hub stations that perform the traffic gateway and network control functions. As for fixed DVB-RCS networks, the network control functions are performed by an NCC (Network Control Center) or NCF (Network Control Facility) and include network synchronisation, dynamic bandwidth allocation and the minimum control and monitoring functions as required for fixed terminals. In particular, this includes the monitoring and control of the transmit power of the satellite terminal.

For mobile satellite networks, the CMF will need to be enhanced to be more dynamic and monitor the terminal movement with respect to:

- the satellite (antenna pointing)
- the satellite beam coverage
- the location of the stations of the other services that need to be protected.

As described below, the baseline is to perform the control and monitoring functions in a distributed architecture which includes the Network Control Facility and a CMF agent resident in the mobile terminal.

8.1.2.1 Antenna Pointing and EIRP

The antenna system of mobile terminals need to accommodate the movement of the mobile platform and maintain pointing to the satellite within a given pointing accuracy. This tracking of the satellite is required to ensure that the depointing losses are limited for the wanted link and to prevent excessive off-axis EIRP toward an adjacent FSS satellite. The antenna positioning mechanism can either report its orientation continuously to the NCF, or it can keep track of it locally and report any excessive pointing deviations and corrective actions taken locally.

In order to mitigate potentially harmful interference due to off-axis emissions into adjacent satellites, an interference mitigation technique may need to be used in mobile DVB-RCS. The amount of mitigation will in general depend on a number of factors, including: the terminal effective aperture size and on-axis EIRP, antenna pointing accuracy, the orbital spacing and occupancy of other satellites adjacent to the broadband mobile satellite.

The NCC may need to control this amount of mitigation, so that the terminal transmission does not exceed the prescribed mask for the off-axis EIRP density level toward the adjacent satellites. For a given antenna pointing error and terminal aperture antenna pattern, the off-axis EIRP density of the return link transmission for the entire range of

off-axis angles can be characterised by the on-axis EIRP density, and the antenna orientation. The NCF can calculate the EIRP density by using the received power measurement in the SYNC burst for a given terminal. The NCC can control the EIRP density by adjusting either the transmit power level or mitigation level, such as the spreading factor, or both.

The baseline approach is for the local control of antenna pointing and status monitoring by the NCF agent in the terminal, coupled with the NCC monitoring of terminal transmit EIRP of received (SYNC) bursts. A polling/reporting mechanism for transmission power and antenna pointing is included in the standard to support this function. The reporting mechanism uses the Mobility Control Descriptor in the TIM for queries and the Mobility Control Message in the SAC field for reporting.

8.1.2.2 Exclusion Zones

The stations of the other services that share the K_u FSS band include the fixed service in Regions 1 and 3, and earth stations for the Radio Astronomy Service (RAS) and Space Research Service (SRS). The locations of these stations are known. In order to prevent interference to these stations, a contour is determined that enclose their locations and establishes the minimum range required to ensure that the received PFD of the interference signal from the mobile terminals does not exceed the prescribed limits. One possible approach is to define a reference contour for a reference mobile terminal, and then, adjust the size of this contour according to the actual EIRP density mask of the mobile terminal. However derived, this contour defines an "exclusion zone" around the station within which some form of mitigation is necessary.

The control of transmissions involving exclusion zones is part of an overall control and monitoring facility (CMF), including a centralised portion of the NCC but also supported by a semi-autonomous NCF agent in the terminals. A key element of the CMF is the monitoring of the terminal location with respect to the exclusion zones, by the NCC in conjunction with the NCF agent.

Since DVB-RCS terminals maintain a periodic synchronisation loop also used for "stay alive" signalling and terminal status of health even for a fixed installation, the cessation of transmission in an exclusion zone may require the complete shutdown of the terminal transmissions, thereby preventing the terminal from reporting its position to the NCC while in the exclusion zone.

The baseline CMF technique uses position based detection involving the NCC and NCF agent in the mobile terminal. It should be noted that the standard does not exclude other methods, as long as these can be implemented using the signalling that is mandated by the standard.

Position based detection relies on the measurement of the position of the mobile terminal (based on GPS/Galileo and/or the navigational system of the mobile platform) and on the real-time knowledge of the service area, including the beam coverage and the exclusion zones.

Position measurements take place in the terminal. A number of implementation alternatives are possible regarding the detection process. A completely centralised implementation would require that the terminals send regular position reports; with all the necessary processing located at the NCC.

A distributed approach is the baseline, since this provides the best compromise between the processing requirements and the signalling overhead. The processing is distributed across the mobile terminals, while the signalling is reduced to the signalling of the exclusion zone event. This arrangement is additionally justified by the fact that a terminal that has been forced to terminate transmission when entering an exclusion zone will need some form of autonomous means of determining when it is leaving the exclusion zone and can resume transmissions. We anticipate that this will be implemented by carrying a database of exclusion zones in the terminal and perform entry/exit detection based on position information that is available at the terminal.

The resources required for storage and maintenance of exclusion zone definitions in the terminal have been investigated and are considered consistent with current technology and existing standard protocols for software updates (for example using SatLabs Harmonised Management and Control recommendations).

8.1.2.3 Fault Detection

The requirements for fault detection and remedial action in mobile terminals are essentially the same as for fixed VSAT systems. DVB-RCS already implements mechanisms and a terminal state diagram that comply with the relevant

regulations. Aside from the special interference mitigation mechanisms discussed above, we do therefore not anticipate any need for special mechanisms to support these functions.

8.2 Handover in Mobile Systems

8.2.1 Reference Network

Figure 25 below illustrates all mobility handover scenarios, including beam, gateway and satellite handover. Satellite handover always entails beam and gateway handover. Gateway handover always entails beam handover, but can take place within the same satellite delivery network.

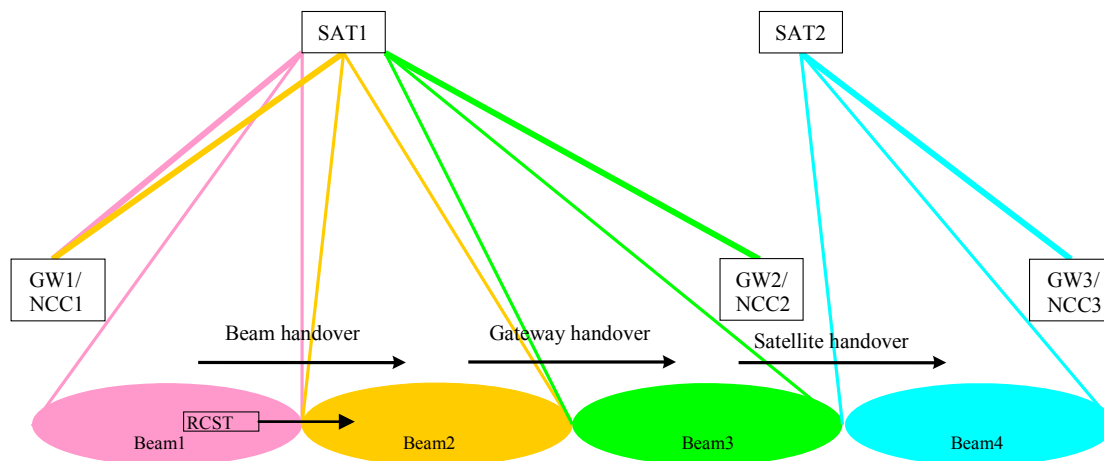


Figure 25: Handover Scenarios

8.2.2 Beam Handover in Mobile Systems

Beam handover (HO) is the most common handover scenario, as part of the RCST mobility handover. Other scenarios may include gateway handover, satellite handover and handover to a terrestrial network (gapfiller). Gateway and satellite handovers are always entailing beam handover. They also entail network mobility handling aspects, e.g. routing of traffic between gateways; these aspects are not relevant to beam handover and therefore not considered in this section.

Beam handover is the preferred technique for mobility management in the case of Line of Sight (LOS) environment specific to aeronautical applications, and also applicable in some cases to maritime applications and even to land applications. Nevertheless it can also be used in the case of non-LOS environment. However, in non-LOS environment mobility management would typically also include countermeasure techniques intended to combat the mobile channel impairments, while the RCST remains within the same beam.

This section concentrates on beam handover in LOS environment, but some of the proposed technical solutions will also be applicable to the non-LOS environment. The section provides background information and an example of a possible implementation of the beam handover. It includes details on various aspects of the beam handover management, such as handover detection/decision, handover execution; event synchronisation and the associated signalling.

For the purpose of the discussion in this section, the term "area" is used to refer to a segment of return link capacity that is managed by the NCC as a single entity, such that time slots within one area are equivalent. An area is thus in general a subset of the return link bandwidth in one beam or of one transponder in one beam.

Also for the purpose of this section, it is assumed that in general the NCC (integrated with the Gateway and also referred to as Hub) includes a Forward Link SubSystem (FLSS), a Return Link SubSystem (FLSS), an IP SubSystem (IPSS) and a Network Management System (NMS); their implementation is system specific and references to these sub-components of the NCC are for illustration purposes only.

This section concentrates on beam handover within the same interactive network. The interactive network is a conventional regional multi-beam access network with a single Gateway (GW1), typically co-located / integrated with the NCC (NCC1). Only two beams of this interactive network (carried by SAT1) are illustrated – Beam 1 and Beam 2.

Beam handover involves both forward path and return paths handover. Each path is associated with a transponder, therefore beam handover is equivalent to a transponder handover and can ultimately be reduced to carrier switching (forward path) / frequency band switching (return path). Note that Transponder handover can also take place within a single beam, if the beam is equipped with multiple transponders (e.g. for different levels of protection against channel impairments); this is not triggered by RCST motion across beams and is out of scope of the current discussion, though the same mechanisms can be used to carry out an intra-beam transponder handover.

Recall that the operation in mobile environment sometimes requires bandwidth spreading of the forward link TDM carriers and return link MF-TDMA carriers; this will have implications on RCST synchronisation. Each forward link carrier is associated with either a MPEG2 transport stream (TS) or - in the case of DVB-S2 – one or several MPEG2 transport streams or generic streams (GSs) within the DVB-S2 multiplex.

A key feature of the beam handover is that the RCST remains attached to the same NCC as it moves from beam to beam through the coverage area.

8.2.2.1 Handover Strategy

Beam handover involves the control and management of RCST handover from one beam to another, while trying to preserve service continuity. The handover process includes:

- Handover detection/recommendation
- Handover decision
- Handover execution
- Resource release / allocation associated with RCST handover
- Synchronisation of events taking place in RCST and NCC, and associated signalling

The *detection* determines the need for the mobile RCST to be handed over and typically also determines a list of handover beam candidates. It can take place in RCST or NCC and can be based on position measurements or link quality measurements.

Handover *decision* is based primarily on the handover recommendation but also on considerations related to the higher layers, in particular with regard to resource management, load balancing and QoS support. It consists of selecting the target beam from the list of beam candidates and issuing the handover command. The handover decision is typically implemented at the NCC, where all the relevant information can be easily made available.

Handover *execution* starts with the issuing of the handover command and consists of moving the RCST from a set of resources in the current beam (which are released), to another set of resources allocated in the target beam. The allocation of resources (physical, logical and bandwidth resources) relies on the forward link signalling (DVB-RCS FLS service).

Handover execution implies the handover of forward link, return link or both. It is assumed that the Gateway is unique in the interactive network and supports multiple forward links, at least one in each user beam. It is also assumed that the return link resources in all user beams are controlled by the same NCC. The immediate consequence of these assumptions is that the terminal remains attached during handover to the same Gateway/NCC (as point of access to the satellite network).

Forward link handover entails switching the forward traffic and signalling from one transport stream / forward link carrier to another (within the same Gateway). The streams can be of either DVB-S type or DVB-S2 type (CCM, VCM or ACM — the DVB-S2 mode is not relevant as far as beam handover is concerned, since the handover always implies carrier switching). With regard to DVB-S2, the forward link traffic can be carried in either MPEG2 TS format or Generic Stream (GS) format.

Return link handover entails moving the RCST from one set of resources (scheduling area) to another, both under the control of the same NCC.

The events taking place in NCC and RCST need to be tightly synchronised in order to attain a handover as seamless as possible. Event synchronisation relies on return link and forward link DVB-RCS signalling.

8.2.2.2 Position Based Detection/Recommendation

Position based detection relies on the measurement of the position of the mobile RCST (based on GPS/Galileo and/or the navigational system of the mobile platform) and on the real-time knowledge of the service area (beam coverage). Though rather complex, it is a preferred detection approach and considered as baseline, since it eliminates the influence of the propagation environment which affects the link quality based detection. However, the normative handover process is independent of the criteria used to trigger the request.

Position measurements take place in the RCST. A number of implementation options are available regarding the detection process. At one extreme, a centralised implementation requires that the RCST's send regular position reports; all processing being done at the NCC. At the other extreme, in a distributed implementation, the RCST's themselves perform the analysis of the position information and signal only their handover recommendation, together with a list of beam candidates, to be used in the handover decision process (section 8.2.2.4). A distributed approach is proposed as baseline, since it provides the best compromise between the processing requirements and the signalling overhead. The processing is being distributed across the mobile RCSTs, while the signalling is reduced to the handover recommendation and the beam candidates. The fact that the coverage area (beam patterns) needs to be stored and maintained (updated) in the RCST database is consistent with the current provisioning of storage resources in RCST's and the existing standard methods for software updates such as those provided by the SatLabs group. The SatLabs Group is an international, not-for-profit association whose members are committed to bringing the deployment of the DVB-RCS standard to large-scale adoption. SatLabs membership is comprised of service providers, satellite operators, system integrators, terminal manufacturers and technology providers with an interest in DVB-RCS.

The key aspects / parameters associated with the position based detection include:

- Position measurement (basic accuracy, rate of change, link margin allowance for handover, heading information)
- Geometrical considerations
- Detection algorithm
- Signalling of the handover information

They are briefly reviewed in the following sub-sections.

8.2.2.2.1 Position Measurement

The primary parameter related to the position based detection is the accuracy with which the positions need to be known. For a moving RCST, this is determined by the basic accuracy of the position fixes, as well as by the age of the position reports: the older the report, the more the signal quality might have deteriorated.

Basic Accuracy

For a mobile RCST, assumed of professional or at least semi-professional quality, the assumption is made that positioning information is available with an accuracy equivalent to that achievable with a civilian GPS receiver or, in near-future, with a "Basic Service" Galileo receiver. For aircraft or ships, this information is likely to be available from the vehicle's navigation system if not provided by sub-systems within the RCST itself.

Current typical GPS accuracies are of the order of 10m. However, an allowance of 200m has been suggested in previous studies for this component, to cover situation where the artificial degradation of the civilian GPS signal ("Selective Availability" — "SA") is re-introduced; this proved not to be a highly significant factor in the accuracy budget.

A typical civilian GPS receiver is capable of producing a new position fix once per second.

Rate of Change

The worst-case antenna gain variation with position at the edge of a satellite beam is considered to be in the order of 0.068 dB/km. This is easily estimated by considering feasible beam sizes and main lobe roll-off patterns. The derivation of this figure can be found in Annex A.

Link Margin Allowance for Handovers

The link budget allowance for the handover process can be determined from the error budget constructed from RCST characteristics and the handover time; a value of 10 seconds has conservatively been assumed for the latter parameter for illustration purposes. The link budget allowance can then be used to derive the required position update interval; the relationship is parametrically represented in Figure 26 for four different classes of mobile RCSTs.

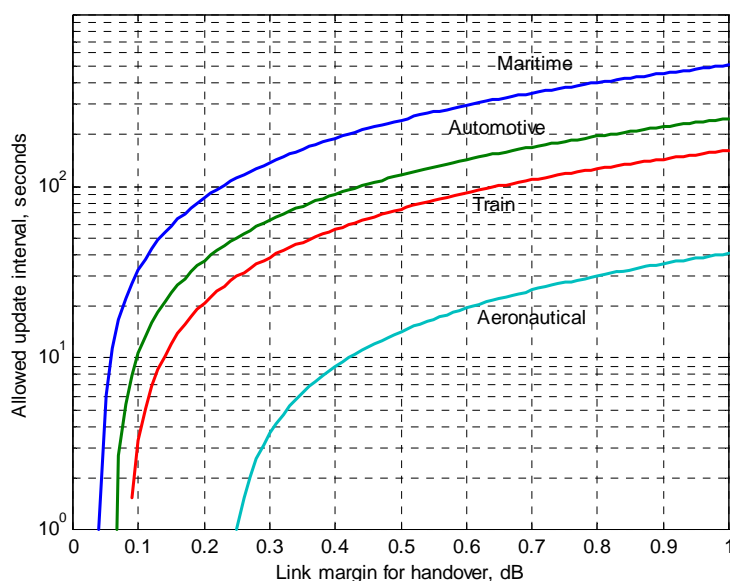


Figure 26: Allowable position update interval as a function of link margin.

For the illustration of the derivation process we consider the case of an aeronautical RCST. Given the worst-case gain change for this RCST as it moves through the antenna coverage (0.02 dB/s, given roll-off rate and velocity), we need an allowance of 0.2 dB when allowing 10 seconds for completion of the handover process. Considerations of the required time are presented in section 8.2.2.5.5. If new positions are available every 3 seconds, the maximum age of a position report is $3 + 0.848 + 0.5 = 4.348$ s, where 0.848s is the assumed SYNC update interval and 0.5s is the one-way propagation/processing time. This gives a maximum change in gain of 0.082 dB.

Since the position accuracy of 200m corresponds to an uncertainty of only 0.014 dB, the total required link margin allowance for the handover process is dominated by the handover processing itself, and is equal to $(0.2 + 0.082 + 0.014) \sim 0.3$ dB for the assumed example.

As shown in Figure 26, a link margin of 0.3 dB requires position updates every 3 seconds for aeronautical RCST (as calculated above), every 38 seconds for train RCSTs, every 63 seconds for cars and every 138 seconds for ships. These figures include allowances for accuracy; therefore, there is no need for averaging or prediction in-between position updates.

Other Supporting Information

As will be discussed in section 8.2.2.2.2, the mobile heading information is desirable in order to facilitate certain aspects of the handover detection/decision. In a distributed approach, the heading information is typically available directly from the navigation equipment.

It is possible to add increased sophistication to the handover recommendation by including consideration of any other available relevant information, such as for example programmed trajectories (flight plans) or known constraints (railway line trajectories). Such sophistication may improve performance but should not create interoperability issues. They are therefore left to the implementers of RCST's and/or systems.

8.2.2.2.2 Geometrical Considerations

The previous sub-sections have demonstrated that a position-based handover algorithm is feasible from the point of view of position measurement. This section provides considerations of the geometry involved, leading to the detailed definition of a possible detection algorithm in the next section.

A quasi-seamless handover between beams inherently assumes some overlap between their coverages, so that transmission in the “destination” beam can begin before it becomes unreliable in the “source” beam. An example of beam arrangement in a “triangular grid” is illustrated in Figure 27. This arrangement is in some respects a worst case and is selected for convenience only; the example algorithm presented below does not make any assumptions in this respect.

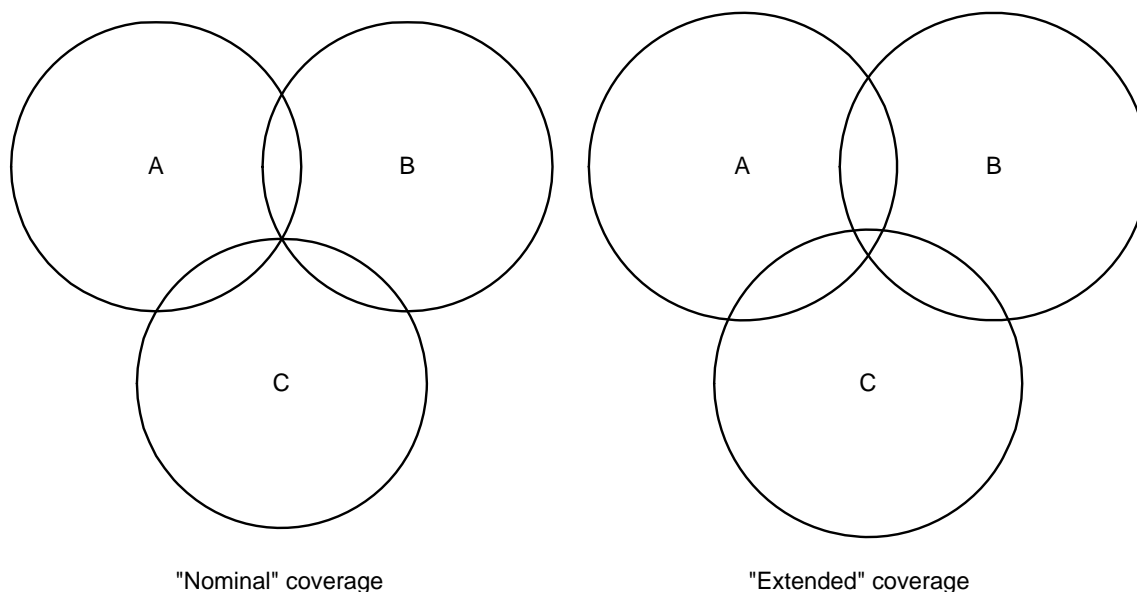


Figure 27: Example spot beam arrangements.

For fixed service, the beams are usually arranged as shown on the left of the figure; i.e. with the minimal overlap possible, given the need for contiguous coverage with minimum number of beams and minimum inter-beam interference. In general, handovers can be carried out while the RCST is in one of the lens-shaped overlap regions. There are however “singularity” points where the edges of three beams coincide, and where there is no overlap.

For mobile service a “handover margin”, as determined in the previous section, is made available when the RCST is located at the nominal edge of a beam, so that sufficient time is available to carry out the handover. This is equivalent to defining an “extended coverage”, as illustrated on the right of Figure 27: All routes between the interior of the beams pass through “handover regions” and there is even a small, near-triangular, region where handover to more than one beam is possible.

The following considerations can be made in order to determine some basic rules for handovers:

- No handover is necessary as long as the RCST is inside the nominal coverage of a beam.
- A handover is necessary as soon as the RCST leaves this coverage area.
- A handover is possible whenever the RCST is inside one of the overlap regions inside the “extended coverage”.

More specifically, the region in which a handover between two beams is possible is bounded by the extended coverage of the current beam and by the nominal coverage of the candidate destination beam. An example is shown in Figure 28; handovers from the left (current) beam to the right (candidate) beam are possible when the RCST is in the hatched area. Note that the area for the inverse handover (right to left, not shown) will be different.

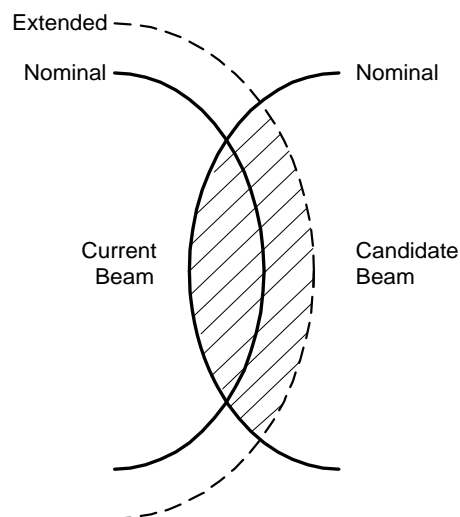


Figure 28: Handover region

There are some special situations:

- If the RCST leaves the nominal coverage of the beam without being in a handover region, it is possible that it is leaving the system coverage and that service will be lost. This is unavoidable; however, the algorithm should issue a warning so that the RCST can be logged off or other appropriate action taken.
- There can be several candidate destination beams; i.e., the RCST can be inside more than one handover region at a given time, the maximum number is likely 6 – as encountered in regular-grid beam patterns. In that case the algorithm should determine the most appropriate candidate from a geometrical point of view; this should be the beam in which the RCST can expect to remain for the longest period. If the trajectory of the RCST is known (e.g. in the case of fast trains), the candidate beam can be selected based on this knowledge. Since in general the trajectory is not known, the beam selection can be based on an extrapolation of the past trajectory (i.e. prediction), so that the number of beam candidates can be reduced (e.g. to three). As a simple initial rule, handover destination candidates could be ranked and ordered according to how closely the recent heading of the RCST points towards a defined “centre” of each candidate beam.
- Immediately following a successful handover, the RCST will typically be in the handover region of the new beam, which would allow the RCST hand back to the beam it just left. This could give rise to repeated back-and-forth handovers (especially for slow-moving RCSTs) known as “ping-pong effect”, which is highly undesirable. The effect can be avoided by not recommending handovers unless the RCST is outside the nominal coverage area and by proper design of the handover detection algorithm.
- If the RCST happens to be stationary, the algorithm should not recommend handovers. This situation is also covered by the above rule (i.e., do not recommend handovers when inside the nominal coverage).
- The uplink and downlink beams can be different. Separate coverage geometries should be considered in this case and the detection algorithm should issue separate handover request/recommendation for the forward link and return link. This is a special handover situation; while possible, it is not considered very likely.

In summary, an appropriate algorithm can take as input the current position and headings of the RCST, identification of the current beam and the database that defines the system coverage configuration. From this, it can produce an ordered list of candidate beams, with the following properties:

- As long as the RCST is inside the nominal coverage of the current beam, the recommendation shall be to remain there.
- Other handover candidates shall be ordered according to the rules as described above.
- If the RCST is outside the nominal coverage with no handover destination candidates available, a loss-of-service warning shall be issued. With the chosen signalling format, this will take the form of a handover request with no valid candidate destination beams.

8.2.2.2.3 Data Structures

In our purely position-based example, we assume that the RCST and NCC each have copies of a database which describes the system coverage.

The fundamental unit of information in the database needed for the handover algorithm describes a beam. The following information is needed:

- An identifier (so that the algorithm can distinguish between the current and handover candidate beams)
- A definition of the outline of the nominal coverage area, for example in the form of a sequence of polygon vertices
- Definitions of handover regions. Each definition describes the handover region to one beam, and contains
 - The identity of the candidate beam, including a “short-form” identifier which is used to minimise the amount of signalling. This short-form identifier is unique only within the current beam.
 - The outline of the handover region, for example in the form of a sequence of polygon vertices. As described in Section 8.2.2.2.2, this region should be bounded by the extended coverage of the current beam and the nominal coverage of the candidate beam. Note that two or more handover regions may overlap.

The database may, of course, contain additional information which can aid the detection and prioritisation of candidate beams. This is system-specific.

Maintenance of the database is outside the scope of the standard. It can for example be carried out via the software update mechanisms defined for Harmonised Management and Control by the SatLabs group.

8.2.2.2.4 Signalling of Handover Recommendation

The baseline for the beam handover scenario is to use the MAC layer signalling defined in the standard. For gateway/satellite handover scenarios, which may involve additional network components outside the NCC/NCC (e.g., a global NOC), higher-layer protocols such as SNMP might be more appropriate for the communication between NCCs and NOC.

The beam handover request can be communicated in a very compact form, as defined in the Mobility_Control_Message SAC field. It is recalled that the recommendation takes the form of a flag that defines the type of handover required (forward link, return link or both), plus an ordered list of candidate beams (limited in number). The candidate beams can be defined in terms of the handover region within the current beam. 4 bits are allowed for the encoding of each candidate beam.

With the candidates encoded in this manner, the 4-bit field can convey up to three candidate beams. This is considered adequate, assuming some form of trajectory prediction. The value ‘1111’ represents an invalid value; it can be used as filler when there is fewer than 3 candidates altogether.

If there are no candidate beams at all, a loss of service is likely imminent. In such situations, transmission of the handover request should be accompanied by a logoff request.

For robustness, it is suggested to repeat the handover request / recommendation until a TIM message with the handover command has been received. The handover (execution) algorithm must accept multiple handover requests from a RCST without inducing system instability.

It should be noted that the Mobility_Control_message field will normally be located in the SAC field carried in SYNC bursts. These typically have sufficient room for a message of this nature, so there is in practice no additional overhead associated with this type of signalling for mobility management purposes.

8.2.2.3 Centralised Handover Detection

The baseline handover detection method described in section 8.2.2.2 uses a distributed approach and only generates handover requests when the RCST determines that this is necessary. The standard however also allows a centralised approach, in which the NCC can poll the position of the RCST using the Mobility Control Descriptor for the query and

the Mobility Control Message SAC field as the reply; and subsequently use this information to make handover decisions. Support for both methods is mandated for all RCST's with mobility support, so the use of either method, alone or in combination with the other, is a system implementation decision.

8.2.2.4 Example Handover Decision

The handover decision is typically based primarily on the handover recommendation arising from physical layer considerations, whether this is made in the RCST (Section 8.2.2.2) or in the NCC itself (section 8.2.2.3). However, the decision can also include considerations related to the overall management of network resources and of the RCST during handover. From this latter point of view, the assumption is made that in general the beam handover entails both forward link handover and return link handover; handing over only one of the links is a special case applicable to some particular situations (e.g. broad area downlink beams).

The process described here is an illustrative example only; intended to demonstrate the feasibility of the proposed approach and associated signalling. Details are system-dependent. In any case, the example includes descriptions of elements that are outside the scope of the DVB-RCS standard, but assumed present in all practical implementations.

Once the handover recommendation has been made and before the decision is taken and the handover command sent to the RCST, the decision algorithm typically performs a number of tasks associated with the higher layers, in particular with regard to resource management and handover start timing. A preliminary list may include the following steps:

- 1) Check if the handover recommendation applies to the forward link, or return link, or both
- 2) Check the availability of resources in the list of ranked beam candidates, in order to preserve the level of service for the RCST (i.e., its users). This may require the invocation of a connection admission control (CAC) function for the existing connection, which could use either the current service commitments to the corresponding application(s) or the requirements configured in the current Service Level Agreement (SLA).
- 3) If the beam candidates are served by multiple forward and/or return link transponders, select the transponder (forward link and/or return link) based on the following considerations:
 - 4) The characteristics of the carriers configured for the transponder match RCST's characteristics
 - 5) Load balancing
 - 6) Select the target beam from the list of candidate beams matching RCST's characteristics, based on the following criteria (in their priority order):
 - Rank of the beam in the list
 - Availability of resources
 - Load balancing when multiple options are available
 - The selected target beam should be the highest in rank with matching characteristics, best satisfying the service requirements and possibly complying with the load balancing needs.
- 7) If resources are not fully available in the selected target beam in order to maintain the current service level, advise the users/passengers about the service degradation. Update the affected service parameters in the relevant network components.

As a result of the above steps, the decision algorithm will typically return a data structure including the identity of the target beam and the associated resources in this beam (e.g. Beam ID, forward link transponder ID / forward link frequency, return link transponder ID / Superframe ID / frequency offset), and possibly a service degradation warning flag that indicates that the current level of service cannot be maintained.

- 8) Once the handover decision has been taken, delay the handover command until a certain event or combination of events occurs; this is intended to reduce the handover time. Potential events may include:
 - The occurrence of the composition tables distribution in the current beam
 - The occurrence of a SYNC assignment to the RCST being handed over
- 9) Send the handover command to the RCST, together with the appropriate information for operation in the target beam, by relaying on a unicast TIM.

The handover command is sent by using the Mobility Control Descriptor in the TIM, while the information relative to the target beam can be included in various descriptors, also in the TIM.

The above steps are captured as part of the overall handover protocol. They should be carried out for each logged-on RCST for which the handover has been recommended.

The above task description assumes an NCC implementation of the decision algorithm, consistent with a distributed approach to handover detection/decision. The choice of a distributed approach appears as the preferred approach when the signalling (volume and rates, protocol), storage and processing needs are analysed. Furthermore, it is suggested to implement the decision algorithm in NMS, which is the natural component for the storage of beam coverage databases and for the execution of the Connection Admission Control (CAC) for the target beam, triggered by the detection algorithm.

8.2.2.5 Handover Execution

Handover execution starts with the issuing of the handover command and consists of moving the RCST from a set of resources in the current beam to another (new) set of resources corresponding to the target beam. For the purpose of analysis in this section it is assumed that the beam handover entails both forward link handover and return link handover at the same time, therefore the resources in the target beam refers, respectively, to a forward link transponder and a return link transponder in that beam.

8.2.2.5.1 Key Architectural Features / Assumptions

The single most important aspect regarding the beam handover is that there is a unique IP point of attachment of the mobile RCST to the satellite interactive network, which is the GW/NCC (Hub). There are two major consequences of this:

- The RCST remains under the control of the same GW/NCC
- All network mobility issues are resolved by configuring RCST specific routes within the GW/NCC

Since the population of mobile RCSTs is expected to be rather small, one can assume that all RCSTs are managed (for the purpose of return link resource allocation) by a unique RLSS. The RLSS will control the return link resources in all beams / transponders and the resources corresponding to each transponder can be configured as one or more scheduling areas.

With regard to the forward link, the gateway should include at least one forward link processing chain (at IP layer and MAC layer) for each user beam or for each transponder in a user beam.

Since the RCST can log-on from any beam of the interactive network, it should be configured with a set of start-up forward link transponders (carrying the NIT), one for each beam. The RCST will acquire first the forward link in the beam in which it is located (via the conventional table linkage process), and then it will acquire the return link burst synchronisation. The NMS in the NCC can derive the location of the RCST (i.e., the return link transponder/beam) from the Superframe_ID – Beam_ID table in its database, using as entry the Superframe_ID of the acquired return link. Beyond these particularities, the logon and logoff processes are as in standard operation; it is noted, however, that the logoff could be triggered by the RCST mobility, e.g. in case where no valid target beam has been identified.

8.2.2.5.2 Overall Handover Procedure

The beam handover involves a number of processes which take place in NCC and RCST. They are captured in flow chart form in Figure 29, which also includes the handover detection/decision processes. The flow chart reflects the distributed approach suggested for detection/decision, with the (position based) handover detection implemented in RCST and the handover decision implemented in NCC. This implies that the handover recommendation is signalled by RCST to the NCC. After the handover decision has been made, the NCC initiates the handover of traffic and signalling in Gateway and informs the RCST about the physical and logical resources to be used in the target beam. The RCST responds by re-tuning its transmitter and receiver to the carriers in the new beam. During RCST re-tuning, the traffic and signalling to/from RCST are discontinued. Signalling discontinuity will affect RCST frequency and burst timing synchronisation and may trigger an RCST logoff, unless the synchronisation scheme is upgraded to accommodate the discontinuity in PCR and the Doppler effects induced by RCST motion. Once the synchronisation in the new beam has been achieved, the transmission/reception of the forward and return traffic and signalling can resume.

Overall Beam Handover (HO) Procedure

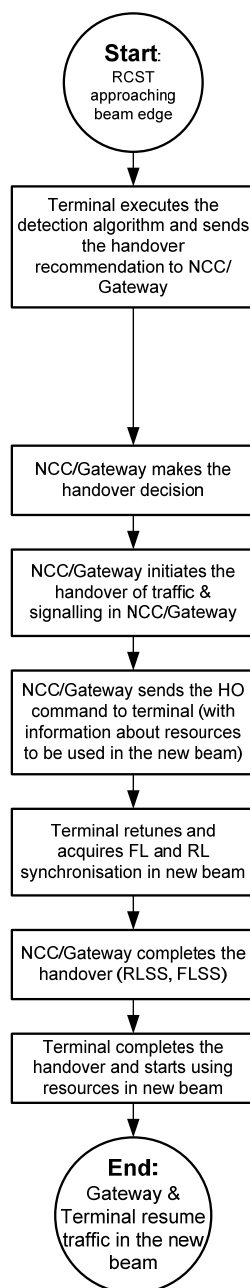


Figure 29: Overall Beam Handover Protocol

The beam/transponder handover entails the handover of both forward link and return link, which translates into forward link carrier and return link MF-TDMA carrier group switching. Since forward link and return link handover are done in parallel, there will be interactions/dependencies between the two processes, primarily at the physical layer but also at MAC and upper layers. These interactions will be considered after reviewing the key aspects associated with the forward link handover and return link handover.

8.2.2.5.3 Forward Link Handover

For illustration purposes the MPEG2 TS format is assumed in the discussion below.

Forward link handover consists primarily of switching the forward link traffic and signalling from one TS/carrier to another. This will have a number of implications on the system operation:

- The acquisition by the RCST of the new carrier/TS on which FLS service is distributed.

- The acquisition consists of the re-tuning of the RCST receiver to the new forward link carrier and the extraction of the FLS service (i.e. the DVB-RCS tables).
- The carrier frequency (associated with a forward link transponder, uniquely identified in the interactive network) is conveyed via a unicast TIM in the Satellite Forward Link Descriptor, which also includes the identity of the new beam.
- The extraction of the DVB-RCS tables relies on PID filtering of TS packets. A number of options have been identified for conveying to RCST the signalling PID values for the new beam, namely the use of RMT/PAT/PMT standard mechanism, the use of the Network Layer Information descriptor in TIM and the configuration at logon time of the signalling PID values for all beams. Options 2 and 3 are faster than option 1, therefore they are both good candidates; selecting one over the other will finally depend on implementer's preferences. Regardless of the option used, the new PID values can be used as soon as the RCST has synchronised to the new forward link.
- Changes to the traffic PID values (if applicable, depending on traffic and signalling handling in Gateway at handover time)
- Traffic PID values can be changed, if needed, via the Forward Interaction Path descriptor in the TIM.
- Whenever possible, it is desirable to maintain the same traffic PID values on all TSs in all beams. However, if the PID values are used for local routing in Gateway (depending on the routing option), this might not be possible unless the PIDs are re-stamped after routing.
- All PID values used in the system for either traffic or signalling should be consistent with the system PID plan.
- Buffering the traffic for the duration of handover (optional, depending on application/service)
- Re-routing (within Gateway) of forward traffic and signalling from the current TS/beam to the target TS/beam
- Traffic re-routing can take place at either network layer or MAC layer. Both can be achieved with minimum impact on traffic continuity. The choice of one option over the other is largely a system implementer's decision.

8.2.2.5.4 Return Link Handover

Return link handover consists primarily of changing the MF-TDMA carrier group the RCST is authorized to use in the target beam. This will have the following implications on the system operation:

- Re-tuning of the RCST transmitter to the new MF-TDMA carriers
- The new MF-TDMA carrier group is associated with the Superframe_ID corresponding to the return link transponder in the new beam. The RCST is notified about the new Superframe_ID and the corresponding frequencies prior to the handover, via the Satellite Return Link Descriptor conveyed in a unicast TIM.
- After re-tuning, the RCST will also need to acquire the return link synchronisation, which requires prior establishment of the forward link signalling in the new beam (see section 8.2.2.5.3).
- Changes to the frame organisation to match the new Superframe_ID, i.e. the SCT, FCT and TCT, collectively referred to as (static) Burst Time Plan (BTP).
- A number of options are available, including static configuration in the RCSTs database of all composition tables for all beams, downloading at the handover time (after re-tuning to the new TS/carrier) or distribution of all BTPs in the system on all TSs:
 - Option 1, at one extreme, is storage intensive but fast (in terms of impact on the handover time).
 - Option 2, at the other extreme, will have minimum storage requirements but will increase the handover time, since the BTP can only be downloaded after the RCST has been advised about the target beam (via the TIM carrying the HO command) and has acquired the TS/carrier in that beam; the handover time may be further degraded since the BTP is typically distributed every 10 seconds.
 - In the Option 3, the RCST can download the BTPs for the beam candidates (e.g. three beams) prior to the start of the handover process, i.e. while still operating in the current beam.

area/beam), while the right-hand side is the new configuration after handover (target area/beam). The handover itself requires the new dashed connections to be established, while the corresponding mirror-image solid lines are released. Various linkages in the RLSS / DAMA Scheduler need to be changed in a synchronous fashion to match the delayed reaction of the RCST. For a short period of time the RCST will continue to receive assignments in the old area but will make requests (IBR and OBR) for capacity that will be considered for scheduling in the new area. The requests are temporarily stored in the request table in order to be transferred between areas; they are used to update the RCST attributes table (i.e. the VBDC Queue, last RBDC request and possibly the CRA parameter).

Since the RCST remains under the control of the same RLSS / DAMA Scheduler, the handover does not typically entail changes to the return traffic and signalling outside the RLSS (i.e. between physical modules) and can therefore be completed almost seamlessly. However, discontinuity regarding return traffic handling will happen due to the disruptions in the distribution of assignments (TBTP), resulting from the forward link handover. During the disruption neither capacity requests nor return link traffic can be sent. Nevertheless, the traffic could be buffered in RCSTs and released after the links are restored in the new beam.

8.2.2.5.5 Beam Handover: Event Synchronisation / Handover Signalling

The management of the beam handover process relies on DVB-RCS specific signalling, i.e. DVB-RCS tables/descriptors on the forward link and ACQ/SYNC bursts on the return link. Of particular relevance in the unicast TIM, used to carry the handover command to RCST. The TIM also includes (in various descriptors) all the information required for the RCST operation in the target beam.

The majority of tables/descriptors are used as in standard operation, with regard to their distribution and timing. However, some changes/optimisations of the signalling are required, in order to improve the handover timing performance. They will be identified as part of the event synchronisation, aimed at minimising the handover time.

An overview of the handover events is provided in flowchart form in Figure 31. The detailed analysis of signalling / event synchronisation is discussed in the following. The assumptions used in the analysis can be summarised as follows:

- The beam handover implies both forward link handover and return link handover.
- A beam can be provisioned with several transponders
- The gateway supports multiple forward link carriers and multiple MF-TDMA carrier groups, associated with different forward link and return link transponders, respectively.
- Traffic routing in gateway is performed without changing the RCST IP addresses, so that the network and upper layer are not affected. The result of this is that any IPsec and/or RADIUS sessions remain open.
- Each forward link carrier transports packets of one MPEG-2 Transport Stream (DVB-S) or of a DVB-S2 multiplex stream in either MPEG2 TS format or Generic Stream format.
- The handover detection/recommendation takes place in the RCST, while the handover decision is taken by the NCC, which also controls the overall handover process.
- For event sequencing/synchronisation and for the assessment of the handover time, the starting point is considered the moment when the handover decision is taken. Prior to this moment the NCC executes the decision algorithm, aimed at determining the optimum start of the handover.

Flowchart of Handover Events

The flow chart in Figure 31 summarises in graphical form the main activities to be performed in NCC and RCST during handover. The activities (events) in the NCC are staggered with respect to those occurring in the RCST, to suggest the approximate sequencing.

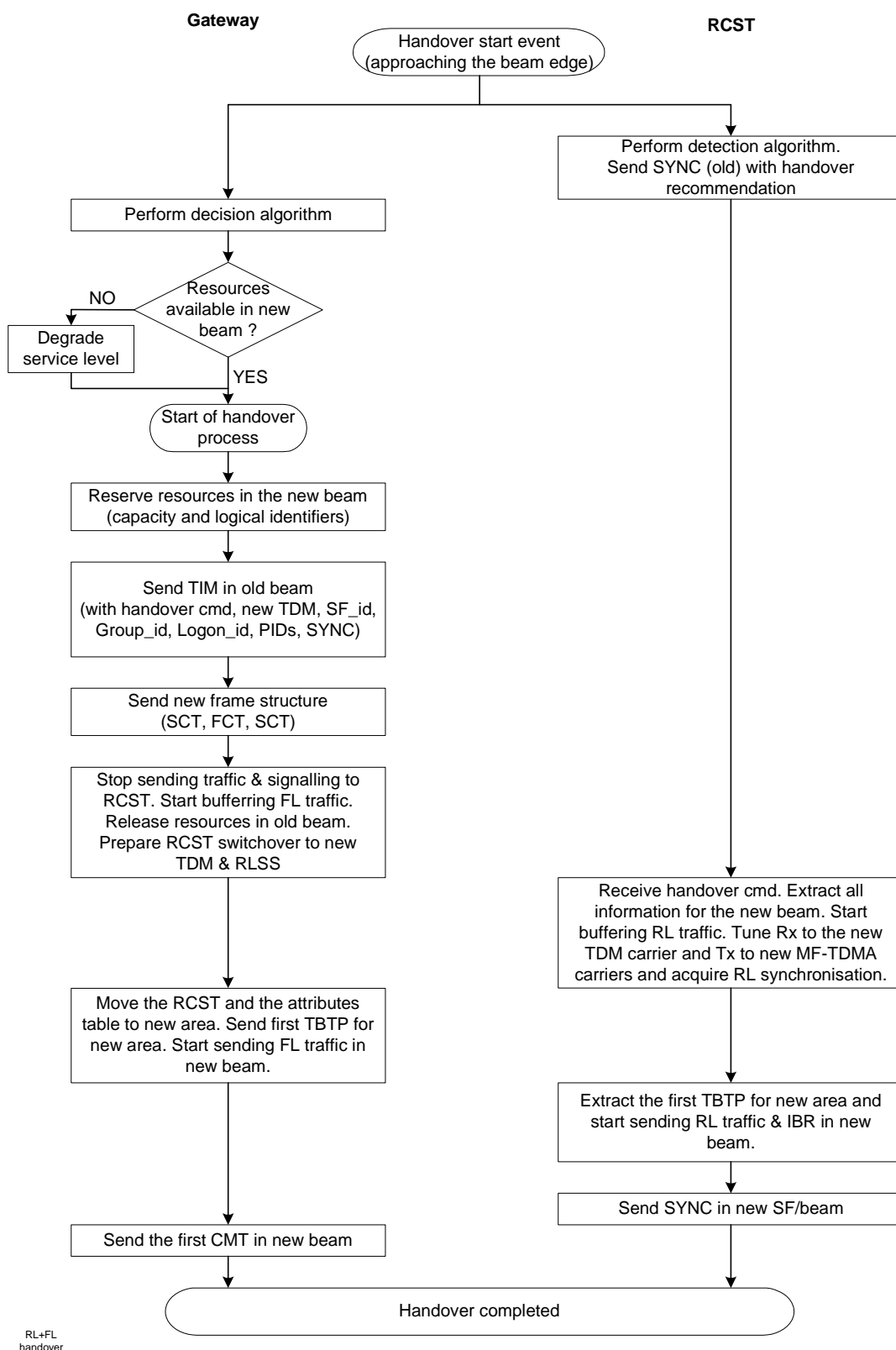


Figure 31: Flow chart for beam handover.

Example Event Synchronisation, Signalling and Timing Analysis

A possible sequencing of events for the purpose of signalling and timing analysis is illustrated in Figure 32. It depicts the activities in RCST and NCC and the associated times (not necessarily drawn to scale).

The illustration in Figure 32 applies to a particular implementation of a DVB-RCS system, characterised by short superframe/frame duration (e.g. 26.5 ms) and a single frame per superframe. In such implementations, the superframe is

the elementary period for all timing processes and for the control of return link resources. The occurrence of various events is expressed in terms of superframe counter values (in PCR counts). The assumption is made that all superframes in the system have the same duration (regardless of the Superframe ID), so that the same superframe counter applies to all. The superframe counter (modulo 65536) is included in the SCT, TBTP and SYNC Assignment descriptor, in order to remove any ambiguity concerning the superframe to which the requests, assignments and SYNC bursts apply.

The sequencing of events / timing may be different for other implementations and may lead to different handover timing performance. The timing analysis in this contribution will therefore be performed in terms of relations/equations rather than in terms of specific values for various parameters, with the emphasis on the events/processes and the corresponding signalling; consequently no timing performance figure will be assessed.

The event sequencing/synchronisation and the required signalling are generically defined in terms of "old" (for current) and "new" (for target) beams/transponders and associated resources. The signalling is indicated in the figure as annotations to the propagation paths.

The approach described in the sequel is conservative in the sense that it ensures that the RCST is completely synchronised in the target beam before even resuming forward link transmissions. Optimisations may be possible in some situations; for example, it may be advantageous to resume forward link transmissions before return link synchronisation is completed. This can be particularly useful for real-time, non-acknowledged traffic such as VoIP, which would in any case be discarded during the handover.

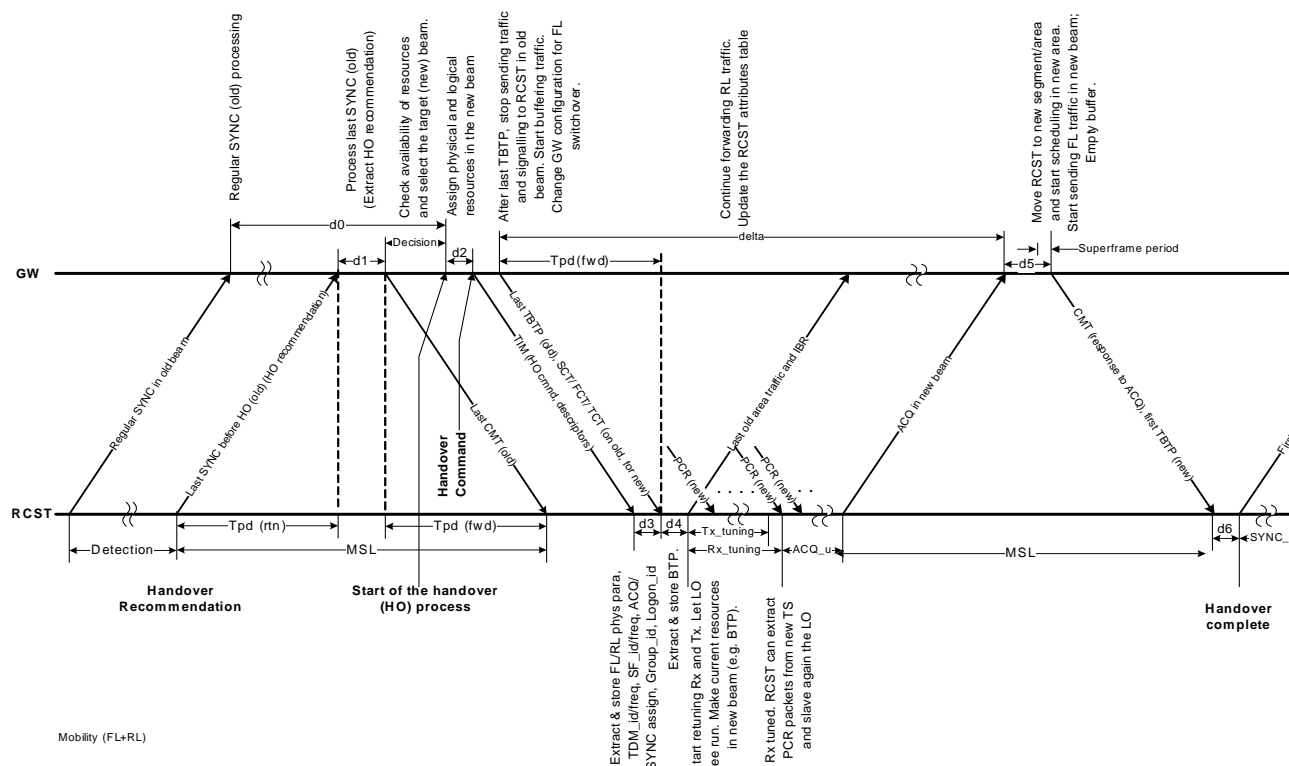


Figure 32: Event sequencing and synchronisation.

$Tpd(fwd)$ and $Tpd(rtn)$ in Figure 32 refer to the forward link propagation delay and return link propagation delays, respectively. Various processing times in the NCC and RCST are designated by d_i and expressed in number of superframes.

MSL in Figure 32 stands for the Minimum Scheduling Latency – the minimum time elapsed from issuing a capacity request in RCST until the RCST can dispatch traffic in the slots assigned in response to that request. The MSL thus includes the propagation times on the forward link and return link and the processing times in RCST and RLSS. Considering typical processing times:

$$MSL = Tpd(fwd) + Tpd(rtn) + \text{processing time}$$

The handover is initiated by the handover recommendation sent by the RCST (in a SYNC burst). The NCC then executes the decision algorithm (section 8.2.2.4), in order to select the target beam from the list of candidate beams.

The handover detection / decision time is labelled as d_0 in Figure 32 and can take values from a few superframes to a few seconds, depending on the method used, velocity of the mobile RCST etc. The last SYNC, before the start of the handover process, carrying the handover recommendation, is shown in Figure 32 together with the corresponding CMT carrying the results of power measurements and time / frequency errors. The time d_1 represents the time to demodulate a SYNC burst, to extract the handover recommendation, to perform return link channel measurements (part of the routine operation) and to encapsulate the results in the format appropriate for inclusion in the CMT. SYNC processing is done in the RLSS.

As a precaution, the handover recommendation could be repeated in subsequent SYNC bursts, until the TIM carrying the handover command is received (see below).

Once the decision algorithm has selected a target beam, the handover process can be triggered. The start of the handover process is in general not synchronized with the transmission of the last SYNC/CMT. The delay between the start of the d_1 interval and the start of the handover process can be as high as the SYNC repetition interval, if the handover process were to start just prior to the arrival of the next SYNC in the old beam. This delay is not relevant; it is part of the detection/decision time, which typically is not included in the handover time since the system is operational during handover detection/decision. It may instead be beneficial to synchronise the start of the handover process (and the subsequent issuing of the handover command) with the broadcast time of the composition tables in the old beam; the benefit will be the availability when needed of the composition tables for the target beam. Alternatively, as mentioned elsewhere, all composition tables could be distributed at the same time (same superframe count) in all beams

The sequencing of events (activities) associated with the execution of the handover process is as follows:

- 1) The Gateway assigns physical resources and logical resources (identifiers) to the RCST in the new (target) beam (see step 2 below). This activity is associated with the delay d_2 in Figure 32 and is estimated at 1-2 superframes.
- 2) The Gateway sends to the RCST the last unicast TIM in the old beam with the handover (HO) command and a number of descriptors defining the changes to the forward link and return link associated with the handover. The changes will apply to a future time (see below). The assumption is made that the RLSS can generate and transmit a TIM in the current beam with information pertinent to the resources in the target beam. The following descriptors are included in the TIM:
 - Mobility Control descriptor, carrying the handover command
 - Network Layer Information (NLI) Descriptor, for changes to the signalling PID values (if needed, depending on the PID addressing scheme). The management of signalling (and also traffic) PIDs should be under the responsibility of the NMS. They are assigned during the d_1 interval (i.e. before the TIM is sent), as part of resource reservation following the resource availability check.
 - Satellite Forward Link Descriptor, containing the new beam ID and the physical characteristics (frequency, polarisation, modulation/coding) of the new forward link carrier. The knowledge of beam ID / forward link carrier frequency could be used to derive the new TS ID. During RCST re-tuning the RCST cannot receive traffic and signalling on either the old or the new forward link carriers.
 - Satellite Return Link Descriptor, containing the new beam ID, new Superframe ID with the associated TX Frequency Offset; the remaining descriptor parameters must not change.
 - The new Superframe ID is associated with a new frame structure, defined by the composition tables (SCT, FCT, TCT) corresponding to the new beam/transponder.
 - SYNC Assign descriptor, which gives the location (i.e. slot number), frame and start superframe of a new static SYNC assignment (applicable to the new frame structure), together with the repeat period. This automatically cancels any existing static assignment, from the SYNC_start_superframe on. The SYNC_start_superframe should point to a time posterior to the transmission time of the handover command. This descriptor is not used when SYNCs are assigned dynamically (i.e. via TBTP).
 - ACQ Assign descriptor, which gives the location (i.e. slot number), frame and start superframe of a temporarily assigned ACQ slot (applicable to the new frame structure), together with the number of repeats and the repeat period. The ACQ_start_superframe should point to a time posterior to the transmission time of the handover command. For the need and usage of ACQ assignment during beam handover please refer to step 9) below.
 - Forward Interaction Path Descriptor, containing the traffic PID values.

- The Forward Interaction Path Descriptor is only used if not all PID values for all TSs are stored in the RCST database at logon time. The RCST should start using the new PID(s) as soon as its receiver is re-tuned to the new carrier.
 - Logon Initialise descriptor, containing the new Group ID, Logon ID. There is no time associated with these logical identifiers in the descriptor. In standard operation they are assigned at logon time and used throughout the duration of the connection. When they are changed as a result of RCST handover, they should be used in association with the new frame structure (new Superframe ID), starting from a given event (see step 4) below). The RCST will be required to handle multiple sets of logical identifiers (Superframe ID, Group ID, Logon ID,) and switch transparently from the current one to the next one. The Group ID will be used by the RLSS for grouping the RCST capacity assignments and correction messages in the appropriate sections of the TBTP and CMT, respectively; RCST-specific information from the relevant CMT section will then be extracted based on the Logon ID. The management of the Group ID, Logon ID is under the responsibility of the RLSS. They are assigned during the d1 interval, just before the TIM is sent. The RCST needs to store all parameters received via the TIM in its database, for use at various moments during the handover process (see below).
- 3) A few superframes after sending the last TIM in the old beam, the Gateway sends the last TBTP for the old beam/area and the composition tables (SCT, FCT, TCT) with sections for all superframe IDs, including the Superframe ID for the target beam. The RCST should be allowed the time (interval d3) to extract, from the Satellite Return Link Descriptor in TIM, the Superframe ID for the target beam (as per step 5)), so it can afterwards extract from the SCT/FCT/TCT (during d4 interval) the section relative to this Superframe ID and store it temporarily in the RCST database (as per step 6)). This presumes that the transmission of the TIM carrying the handover command and all other descriptors is synchronised with the broadcast time of the composition tables in the old beam (i.e. it is sent d3 superframes in advance). The implicit assumption is that the handover command can be delayed for up to 10 seconds (SCT typical periodicity) after the handover recommendation, without compromising the quality of the links in the old beam.
- 4) Immediately after the last broadcast of the composition tables and TBTP in the old beam, the Gateway stops sending forward link signalling and traffic to the RCST in the old beam, prepares for the forward link and return link switch-over and starts buffering the traffic to the RCST (optional). These activities are associated with the interval delta, from the transmission of the last TBTP (old) until a few superframes before the transmission of the first TBTP (new). The delta interval is rather large and includes the uncertainties associated with RCST re-tuning and ACQ/SYNC assignments (see steps 8) and 9) below). During this interval the Gateway broadcasts in the usual manner all general DVB-RCS tables in the new beam, including the PCR Insertion packets. The preparation for forward link traffic and signalling switchover to the new TS consists in the reconfiguration (by the NMS) of the relevant components for routing the forward link traffic and signalling to the new forward path, as follows:
- Notifies the Buffering System to start buffering traffic with the RCST IP Address
 - Changes the routing table in the Edge Router, to re-route RCST traffic to the Buffering System (for the duration of the handover)
 - Removes the triplet IP/MAC/PID from the IP Encapsulator in the old forward path (FLSS) and configures the triplet in the IP Encapsulator in the new forward path.
 - The PID values should be changed according to the PID plan. The changes may or may not need to be signalled to the RCST, depending on the implementation. Nevertheless, the signalling PIDs need to be known in the NCC, which should generate the DVB-RCS tables for all TSs with the appropriate PID values.

All forward link changes within the Gateway associated with step 4 will typically take 2-3 superframes, well within the delta interval. Traffic buffering should continue for the entire duration of delta. With regard to the return link, after sending the last TBTP (old) the RLSS:

- Continues to receive traffic and capacity requests to the old area/beam for another MSL interval (i.e. until the reception of the last traffic burst for the old area), but stops scheduling in the old area. The traffic is passed to the IP components as usual, while the requests are used to update the RCST attributes table (i.e. the VBDC queue is updated, while the last RBDC request is updated; these requests are tagged with the Group ID and Logon ID corresponding to the old area but will be used for assignments in the new area, therefore they need to be re-tagged. At the end of this interval, return link traffic and signalling are discontinued for a period commensurate with the return link acquisition time (including RCST TX re-tuning and coarse synchronisation - see below).

- Prepares to moves the RCST and its attributes table to the new area but does not start scheduling in the new area until a well defined event/time (the end of delta interval, as signalled by the reception of the first ACQ or SYNC burst in the new beam, see step 9)). The new area is laid over the new superframe structure (static BTTP). The preparations include the de-registration of the RCST from the old area and the registration with the new area, with the new Group ID and Logon ID. They also include the transfer of the RCST attributes table to the new area. All these activities can be completed in general in a few superframes (interval d5), prior to the start of scheduling in the new area and the transmission of the first assignments in the new TBTP/beam. With regard to the RCST attributes table, it is noted that while the VBDC queue is updated as long as the VBDC requests continue to arrive (up to a certain limit, given by the buffer size in the Scheduler), all RBDC requests received in the same interval but the last one will be discarded. Moreover, there will be no requests during the return link re-tuning / synchronisation, therefore the impact on return link traffic will be rather serious; it can be reduced to a certain extent by traffic buffering in RCST.
- 5) After receiving the TIM, the RCST extracts and stores temporarily in its database the information needed for operation in the new beam on both forward link and return link; this includes the identity and physical characteristics of the new forward link carrier (frequency, polarisation, modulation/coding) and of the new superframe (Superframe ID, TX_frequency_offset), the new ACQ and SYNC assignments and the new RCST logical identifiers (Group ID, Logon ID, PIDs). The extraction of this information takes 2-3 superframes (d3). At this time the RCST is ready to start re-tuning its receiver, but this is delayed until the reception and extraction of the last TBTP and the new frame structure (step 6)).
 - 6) After receiving the last TBTP (old) and the new SCT (and FCT, TCT), the RCST extracts from SCT the section relative to the Superframe ID derived in the previous step and stores temporarily the composition tables for the new beam in the RCST database. The extraction of this information takes 2-3 frames (d4 interval). After this step the RCST shall stop sending SYNC bursts to the old area/beam.
 - 7) The RCST sends the last traffic and capacity requests (IBRs) for the old area/beam and starts buffering the traffic in the RCST queues.
 - 8) After d4 superframes the RCST starts tuning its receiver and transmitter and prepares for the handover. More specifically, the RCST:
 - Tunes its receiver to the new forward link carrier frequency extracted in step 5) from the Satellite Forward Link Descriptor.
 - Tunes its transmitter to the new TX frequency, determined by the centre frequency of the new MF-TDMA carrier group (extracted in step 6) from the SCT), adjusted by the TX_frequency_offset obtained in step 5) above from the Satellite Return Link Descriptor, in order to send traffic (and requests) for the new area.
 - Since the knowledge of the centre frequency of the new MF-TDMA carrier group is a must for TX re-tuning, transmitting the composition tables (for the new beam) in the old beam appears even more beneficial: the TX re-tuning can start in parallel with RX re-tuning, so that the overall re-tuning time can be reduced.

Both TX and RX tuning times are subject to uncertainties. Typical RX re-tuning time (including the spectrum spreading code acquisition) is about the same as the typical TX re-tuning time, probably about 500 ms. However, RX re-tuning may take longer (up to 1-2 seconds, depending on the interference environment) than TX re-tuning, which should never exceed 1 sec (according to the requirements in the DVB-RCS standard). This is the situation illustrated in Figure 32, but the other situation can also happen.

The next step of the event synchronisation process can only be taken after completing both TX and RX re-tuning (see step 9)), so the largest of the re-tuning times should be considered for the handover timing performance.

TX re-tuning time refers to the settling time of the transmit oscillator, i.e. the time needed to adjust the TX frequency to the programmed value with a predefined tolerance. For burst transmission the RCST must also achieve burst (timing) synchronisation (step 9)).

During TX re-tuning the RCST activates the composition tables and all other parameters associated with the new beam.

- 9) Once the receiver re-tuning and transmitter re-tuning have been completed, the RCST acquires the burst (timing) synchronisation.

Burst synchronisation relies on a closed loop involving PCR packets and CSC/ACQ/SYNC bursts and CMT replies; this implies the existence of an operational forward link (in order to receive PCR Insertion packets and CMTs) and a return link capable of transmitting return link signalling bursts. The PCR are used for fine synchronisation of the local PCR clock, while the return link signalling bursts are used, together with CMT, to track the variations in RCST-satellite range.

After achieving RX synchronisation, the RCST can start decoding the PCR Insertion packets and use the global PCR to resynchronise (slave) its local PCR clock. A flag in the RCST data base is set after achieving the required accuracy. The corresponding time is considered part of the TX_tuning time. The RCST is now ready to transmit return link signalling bursts, in order to complete the burst synchronisation.

Due to RCST movement it is possible that SYNC bursts, for which the guard times are rather small, would fall outside the receive window in the RLSS; ACQ bursts, for which the guard time is larger, could be used instead for coarse synchronisation. The need for ACQ bursts (temporarily assigned via the TIM) will depend on the accuracy with which the effects of RCST movement are compensated. For a generally applicable solution, it is considered safer and therefore suggested to use ACQ burst for time synchronisation (as illustrated in Figure 32); it is expected that one ACQ burst will be enough to bring the system within the fine synchronisation range.

The accurate estimation of the return link synchronisation time is further complicated by the uncertainty related to ACQ assignment (ACQ_u in the figure) in the new frame organisation with regard to start_superframe, frame_number and slot_number within the repeat_period. Since the ACQ_start_superframe is difficult to anticipate (due to uncertainties in TX/RX tuning times), any time after the start of re-tuning is possible. The uncertainty could therefore be as high as the repeat_period, which should be equal or greater than MSL. Similar uncertainty is defined with respect to the SYNC assignment (SYNC_u) (to be used in a future step). The SYNC_repeat_period is typically not smaller than approximately one second; one can adopt the same value for the ACQ_repeat_period. The number of ACQ_repeats in the ACQ Assignment Descriptor should be set large enough (e.g. 5-6) to account for the largest re-tuning time (e.g. 1-2 seconds) and to allow for multiple coarse synchronisation attempts, if needed. Note that any ACQ slot assigned during the re-tuning interval will be wasted.

The RCST will get the CMT reply in response to the ACQ burst after MSL (which includes the processing time d5), so that the total re-tuning / return link synchronisation time is given by

$$T_s = \max \{TX_tuning, RX_tuning\} + ACQ_uncertainty + MSL$$

One ACQ_repeat_period should be added for each supplementary ACQ burst, if more than one is needed.

- 10) The Gateway transmits the first assignments to the RCST in the TBTP corresponding to the new area/beam, at the end of delta interval (i.e. d5 superframes after the receipt of the ACQ burst), and also resumes the transmission of forward link traffic (in the new beam). As already mentioned, during the d5 interval the Gateway transfers the RCST to an area in the new beam, with new Group ID and Logon ID and new assignment of SYNC space. In the same time the RCST attributes table is transferred to the new area. The implicit assumption is that fine synchronisation has been achieved. The Gateway can derive the synchronisation status from the results of measurements performed in the RLSS on the ACQ (Burst_time_correction, to be included in the CMT) and the knowledge of the fine synchronisation threshold (as defined/used in the coarse synchronisation procedure). The fine synchronisation threshold can be obtained from the RCST database in Gateway (NMS). The Gateway may decide to include the first assignments to the RCST in the new area/TBTP even if the Burst_time_correction is not within the fine synchronisation threshold, but close enough so that it can be expected that the RCST will bring the burst time within the fine synchronisation threshold before the transmission of TRF and/or SYNC bursts, by applying the Burst_time_correction. If the Burst_time_correction is judged to be too high, the Gateway may decide to postpone the assignments in the new TBTP/beam, and the correction message will be used to trigger additional ACQ bursts; this will automatically add one ACQ_repeat_period to the handover time for each transmitted ACQ burst.

The above description refers to a somewhat unconventional NCC behaviour. If the synchronisation status is not checked by the Gateway, the Gateway will typically wait for the first SYNC burst with the fine_synchronisation_achieved flag set in the SAC field before including the first assignments in the TBTP for the new area. This would add a substantial delay of (MSL + SYNC_uncertainty).

It is noted that the new area (and the corresponding TBTP) are new only with regard to the RCST being handed-over to that area/beam. The TBTP was being transmitted since the activation of that area/beam; new is the fact the handed-over RCST starts getting the assignments in this TBTP.

The resumption of the forward link traffic will start with the depletion of the Buffer System and the inclusion of new traffic (especially of real time nature).

- 11) Upon the receipt of the first TBTP and after the processing delay d_6 (2-3 superframes), the RCST sends the first traffic bursts (and IBRs) for the new area.

With the transmission of the first packets in the assignments provided by the TBTP for the new area/beam the handover process itself is completed.

- 12) After the reception and processing of the CMT (generated in response to the ACQ burst) the RCST will start sending SYNC bursts for fine synchronisation maintenance. The first SYNC is sent after SYNC_uncertainty, which can be as high as SYNC_repeat_period.

This step is needed in order to check that the RCST has not been deprived of SYNC burst / CMT replies for longer than tolerated by the RCST synchronisation scheme.

Based on the above analysis, the handover time (i.e. the interval during which the forward link and/or return link traffic are discontinued) is given by:

$$T_{HO} = d_4 + \text{Max} \{TX_tuning, RX_tuning\} + ACQ_uncertainty + MSL + d_6$$

Typically, best and worst-case values can be derived based on the typical values/ranges of the parameters in this equation.

If the Gateway only sends the first TBTP after receiving a SYNC with the fine_sync_achieved flag set (which is recommended for system operational robustness), an additional term (SYNC_repeat_period + MSL) should be included in T_{HO} . The timing performance will be further degraded if the return link synchronisation cannot be achieved with only one ACQ burst / CMT reply.

In summary, the beam handover cannot be seamless in the described scenario, where the RCST is assumed to have only one transmitter and one receiver. The handover time is primarily determined by the RCST's re-tuning time and return link fine synchronisation time and takes a few seconds. During this time some packets can be lost (even with buffering) or delayed. This will have impact in particular on real-time applications, but may be less noticeable for non-real time applications relying on best-effort service. Buffering does not make much sense for delay/jitter-sensitive applications; since the off-time is rather high, it is difficult if not impossible to compensate for the delay introduced by buffering, therefore it is felt that the real time traffic should not be buffered at all (but discarded) and the Buffer System is considered optional. As mentioned above, it maybe possible to reduce the handover duration for real-time traffic that would be discarded in any case, at least in the forward link.

One way to circumvent the inconveniences resulting from a handover is to advice the users (passengers) of the imminence of the handover, so that they can refrain momentarily from being engaged in real time sessions; this can be achieved via some broadcast messages from the mobile platform controller.

8.3 Gateway Handover in Mobile satellite systems

Figure 33 shows the scenario for gateway handover. It is considered that the source gateway and the target gateway are associated with the same NCC and with the same satellite. A gateway handover always entails a beam handover, so the beam handover procedure is tightly integrated and synchronized with the gateway handover.

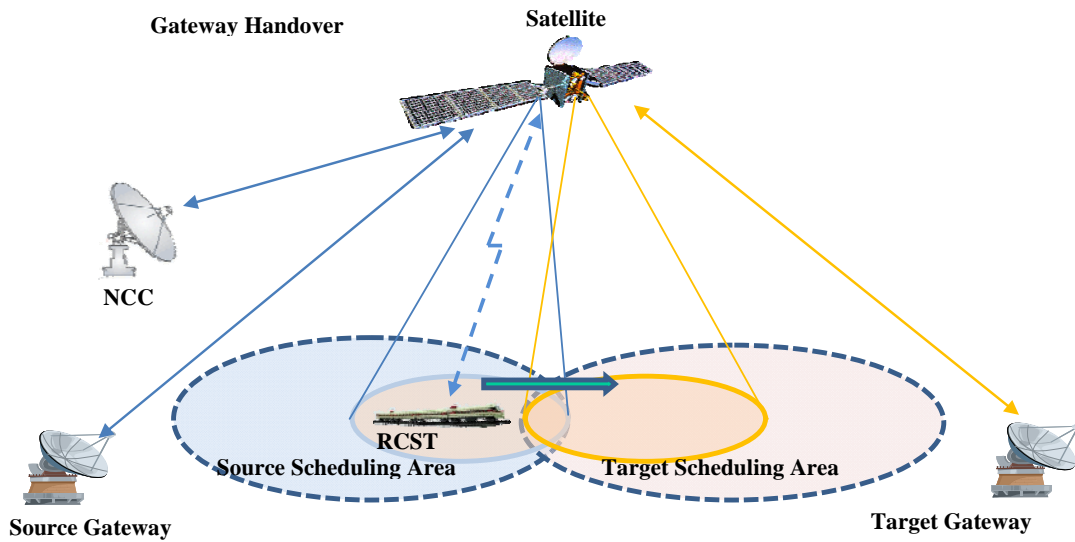


Figure 33 : Gateway Handover scenario

The flow chart of a gateway handover procedure is shown in Figure 34. The following subsections explain the whole handover procedure.

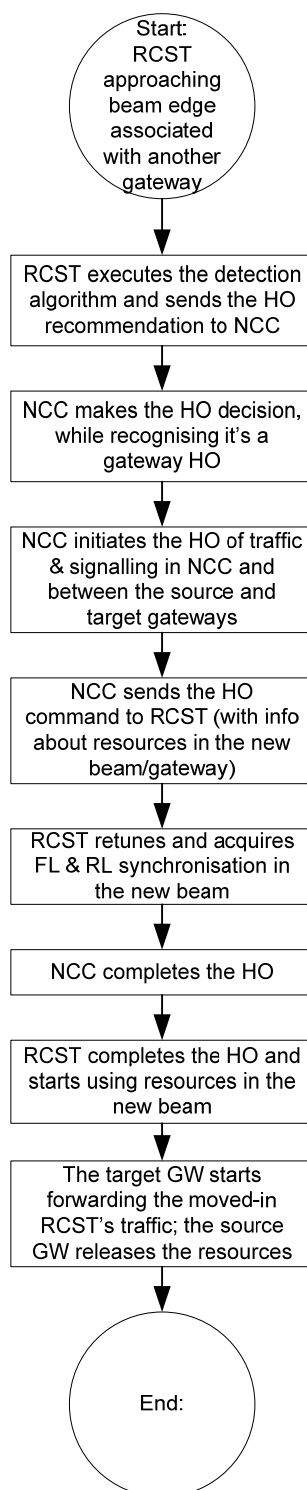


Figure 34: Gateway Handover procedure

8.3.1 Handover Detection

The handover detection is the same as that of beam handover. Various approaches can be used for the detection of a beam handover including position based detection, link quality based detection. This can be performed either centrally at the NCC or in a distributed manner by the RCSTs. The handover detection algorithm is outside the scope of this document, although the position-based detection is the preferred mechanism for train applications where it is highly

likely that a GPS receiver or other navigation system is installed. If the position-based approach is adopted, it is assumed that the NCC contains a database that includes all beam, gateway and satellite profiles and this same database will be passed from the NCC to the RCST at its logon phase or at regular intervals. The RCST will combine its position information and the beam profile information together with the pre-defined train route for handover detection. If a beam handover is required, the RCST sends a handover recommendation to the NCC. The handover recommendation will include a list of candidate beams. The recommendation message is encapsulated within the `Mobility_Control_Message` as defined in the beam handover procedure, which is carried in DVB-RCS SYNC burst.

8.3.2 Handover Decision

In the handover decision phase, the resource availability of the candidate beams, the SLA and QoS of the session to be handed over will be checked by the NCC. The handover decision will also take into account the mobility trajectory if it is known, such as the case of a train route or a flight path. Once a target beam is chosen, the NCC will check if the chosen target beam belongs to another gateway. If the target beam belongs to a new gateway, a gateway handover is decided. Normally it is assumed that the FL and RL have the same beam coverage and gateway configuration so that FL and RL handovers happen simultaneously.

Once a handover is decided, the NCC will update its SI tables, which include the TBTP, SCT, FCT and TCT. Signalling will be carried out between the NCC, the source and the target GWs for the preparation of handover. The NCC will send an SNMP Set-Request message to the target gateway for events synchronisation to ensure that the target GW gets ready for connection with the RCST. The updated SI tables, together with the routing update information of the RCST will be included in this message. The routing update information is generally implemented by sending the location change information to the broadcaster, which is generally handled by the location management scheme. Upon reception of the Set-Request signalling, the target GW will allocate bandwidth resources for the RCST according to the new burst time plan sent by the NCC. This will be followed by an acknowledgement Get-Response message sent from the target GW to the NCC. The NCC will then send a Set-Request message to the source GW, which includes the RCST identity and the SI tables. At the source GW, after receiving the NCC signalling, it will buffer the FL user traffic to be forwarded to the target GW during handover. The source GW updated its route mapping table and released resources used by the RCST. The source GW will then acknowledge the NCC by sending it a Get-Response message.

8.3.3 Handover Execution

Upon reception of the Get-Response message from the source GW, a gateway handover command is issued to the RCST from the NCC in a `Mobility Control Descriptor` carried in a `TIMu` message. Upon reception of the handover command, the RCST synchronises with the NCC and the target GW, retunes itself to the new beam and receives traffic from the new beam which comes through the new gateway. In the FW direction, the traffic to the RCST is redirected to the new gateway and then through the new FL to the moving-in RCST. On the RT direction, the RCST sends return traffic and signalling on the new RL and through the new RL to the new gateway. The gateway handover is complete when the RCST sends an `ACQ` message to the NCC after it finishes the retuning process and receives the `CMT` message from the NCC.

The signalling sequence flows as depicted in Figure 3.5, where the Functional Entity Actions (FEAs) are also shown. FEAs are internal actions of the functional entities in preparation for or in response to their associated signalling messages (see Table 11). In order to facilitate M&C message communication, DVB-RCS management is extended with newly defined SNMP messages in the format of `Trap`, `Set-Request` and `Get-Response`. The higher layer SNMP interaction (indicated by solid lines) and DVB FL/RL signalling (indicated by dotted lines) are synchronized and coordinated to accomplish the entire HO management.

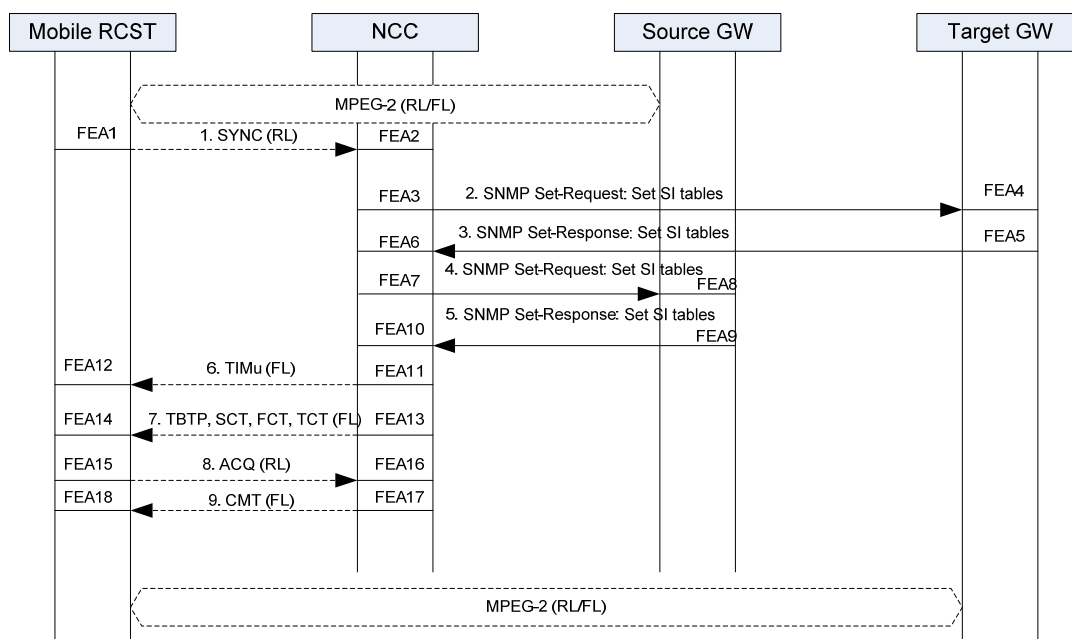


Figure 35: Gateway Handover Signalling Sequence

Table 11: Gateway Handover Functional Entity Actions

FEA Number	FEA description	Comments
FEA1	Periodically check the positioning/link quality information, triggering HO Recommendation on criteria agreements.	Refer to beam HO mechanisms [RD-3]
FEA2	Make HO decision, identifying it's HO type, i.e. beam HO, GW HO or SAT HO.	
FEA3	Set resource arrangement taking account to the new moved-in RCST.	
FEA4	Arrange resources for moving-in RCST.	
FEA5	Send resource allocation update response to the NCC.	
FEA6	Set resource arrangement taking account to the moved-out RCST.	
FEA7	Send Set-Request for moving-out RCST to forward the traffic.	
FEA8	Prepare to release resources for the moving-out RCST, forward the traffic	
FEA9	Send resource allocation update response to the NCC.	
FEA10	Receive resource allocation update response.	
FEA11	Issue HO CMD in source beam, indicating Target beam/GW/SAT.	
FEA12	Retune to target beam. Switch to new link.	Refer to beam HO mechanisms
FEA13	Issue general SI tables in target beam.	Refer to beam HO mechanisms
FEA14	Use new resources in target beam.	Refer to beam HO mechanisms
FEA15	Send ACQ in target beam.	Refer to beam HO mechanisms
FEA16	Receive ACQ in target beam, indicating RCST re-synchronisation.	Refer to beam HO mechanisms
FEA17	Send CMT to moved-in RCST in target beam, confirming the HO complete.	Refer to beam HO mechanisms
FEA18	Receive HO confirmation indicated by CMT.	Refer to beam HO mechanisms

9 Continuous Carrier Operation

The use of continuous carriers in the return link has been adopted as a simple and robust access mechanism for mobile DVB-RCS networks; in particular for RCST's with substantial traffic aggregation such as those serving trains, cruise ships and wide-body commercial aircrafts. Such RCST's may have a sufficient number of users behind to allow for an efficient use of the continuous carrier mode which has a better physical layer spectral efficiency (see Annex B).

Continuous carrier mode has advantages in other situations as well. For example, demand assignment mechanisms can break down in terrestrial-mobile applications that experience frequent channel blockages. This happens when the capacity requests and/or the assignment messages experience a high loss probability. Alternative access schemes for such problematic channels include the use of random access as well as continuous carriers. Continuous carriers have the advantage over random access that some measure of Quality-of-Service provisioning is possible, at least when the channel is not blocked.

The continuous carrier mode can be seen as an add-on to the classical DVB-RCS system, as illustrated in Figure 36

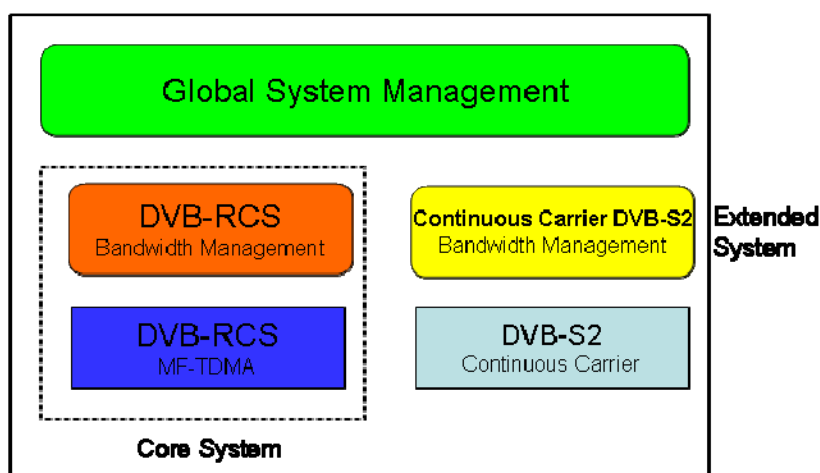


Figure 36: DVB-RCS system with DVB-S2 return link in continuous mode

The following sections assess the system architecture and signalling implications for the operation of a DVB-RCS network that employs both continuous carriers and MF-TDMA.

9.1 Architecture and Modes of Operation

In any given network, it is unlikely that all terminals operate in continuous carrier mode all the time. The provisions in the normative document are therefore designed to support a hybrid architecture that retains all the characteristics of a classical MF-TDMA DVB-RCS network, while adding continuous carrier operation as an overlay. This approach has several additional advantages, including:

- The forward link is unchanged, except for the addition of some signalling elements needed to control the continuous carrier mode.
- RCSTs can maintain the existing NCR synchronisation scheme, relaxing the requirement for high-stability, high-cost reference oscillators otherwise needed for terminals in continuous carrier systems.
- A forward signalling path for controlling the RCST operation exists at all times. This simplifies management.

In order to ensure the maximum flexibility for RCST operation while retaining backwards compatibility, continuous carrier operation is defined in terms of a number of functional states that complement the existing RCST state diagram defined in clause 7.1 of the standard. The steady-state mode corresponding to MF-TDMA operation is called "Fine Sync" mode. The additional mode of operation is called "Continuous". Therefore, there are four modes of operation:

- (a) Fine Sync: As defined for MF-TDMA operation; this is the existing MF-TDMA operational mode.

- (b) Continuous: Operation with a continuous carrier only.
- (c) Continuous + Fine Sync (Optional): Simultaneous operation of continuous carrier and MF-TDMA.
- (d) Off/Standby: As defined for MF-TDMA operation; the terminal is not operating.

Mode (c) is optional and it requires two transmitters in the RCST. The possible transitions between these four states, and the events that trigger these transitions, are defined in detail in clause 10.1 of the standard. The basic trigger mechanisms are the assignment and release of continuous carriers, and DVB-RCS logon and logoff events.

Any RCST must be able to log on in mode (a), corresponding to conventional MF-TDMA DVB-RCS. Once logged on, requests and/or commands can be used to assign a continuous carrier to the RCST. Depending on the terminal capabilities, such an assignment may be accompanied by a log-off. If the DVB-RCS session is maintained, the RCST moves to mode (c). If the RCS session is cancelled, it moves to mode (b).

A terminal in mode (b) is not logged on in the real sense of DVB-RCS, since it does not operate a regular return link with SYNC bursts etc. However, the terminal is still logged on in the sense that it exchanges traffic and signalling with the NCC. Return link signalling required in this mode can be sent using DULM on the continuous carrier, as discussed below. The NCC may need some specific functionality to be able to accept and process this signalling.

Any terminal-specific configuration parameters required for operation of the continuous carrier can be sent through the forward link signalling (Unicast TIM).

In some systems, primarily very small ones, it may be appropriate that the RCST starts directly in continuous mode on a pre-assigned carrier, without going through a DVB-RCS log-on process. To facilitate this, while maintaining the flexibility to change the operational mode later, the standard includes requests and commands to assign a carrier semi-permanently to an RCST. At the next "logon", the RCST will immediately start using this carrier, rather than going through the RCS logon process. There are similarly requests/commands to revoke this "permanent" carrier assignment, so that the RCST will attempt its next log on in normal RCS mode. Depending on the implementation, it may also be possible to configure these parameters directly in the terminal, so that an RCST can start directly in continuous mode, without ever going into RCS mode.

9.2 Bandwidth Management

Continuous carrier based systems can be operated with different degrees of sophistication, chosen in accordance with the size of the network and the activity level of individual terminals. At one extreme, each terminal can be permanently assigned to a carrier. This is obviously the simplest scenario in terms of management, but it may be very inefficient if the terminal is inactive for extended periods. Given that signalling paths exist in the DVB-RCS network, it is relatively easy to implement a simple, carrier-switched multiple access scheme, in which continuous carriers are dynamically assigned to RCST's. The primary justification for such a dynamic assignment is the usage profile of the mobile terminals: trains and aircraft, for example, routinely spend extended periods out of service.

The continuous carrier assignment can be done in response to terminal requests. In this case, the scheme becomes a simple demand-assignment scheme. Once the basics of such a mechanism are in place, it becomes simple to extend it to cover also the adaptation of the continuous carrier characteristics, for example to accommodate fluctuations in traffic demand or to change the level of protection to adapt to changing channel conditions.

A carrier-based demand assignment scheme is much less responsive and flexible than the full MF-TDMA scheme; it only really applies at the session/connection level. Once the demand variability reaches a certain degree, it will become advantageous to switch to MF-TDMA mode. Such decisions, as well as any admission control functions required for the carrier-switched demand assignment.

The set of continuous carriers supported by the NCC is signalled to the RCST's using the broadcast signalling in a manner analogous to that used for MF-TDMA time slots. Carriers are assigned, modified and revoked by signalling between the NCC and the RCST. The RCST can request carriers by type, allowing the NCC to manage "pools" of equivalent carriers.

9.3 Signalling

Control of the system is implemented by exchange of signalling messages between the RCST and NCC. In the RCS system architecture, a forward link signalling path always exists. The definition of the characteristics of the available continuous carriers and the assignment/revocation process are handled within the existing signalling framework, i.e.,

using the SCT, FCT, TCT and TBTP. The specific interpretation of these tables necessary for continuous carrier operation is described in the corresponding clauses of the normative document and is summarised below.

The existence of a return link signalling path depends on the mode of operation. If the RCST is in mode (a) or (c), the MF-TDMA return link provides a signalling path. DULM has been chosen for the return link signalling associated with continuous carrier operation. Usually, such signalling will be relatively infrequent. The potential delay associated with the DULM method, caused by the need to request and be granted capacity before a message can be issued, is therefore likely to be acceptable. Since the signalling frequency requirement is low, any signalling mechanism permanently assigned within the MF-TDMA system (e.g., in the SAC field of SYNC or traffic bursts) are likely to be bandwidth inefficient.

RCST's operating in mode (b) do not necessarily have an obvious return link MAC layer signalling path. This may be the case if the the continuous carrier traffic is dedicated to real-time applications with tight delay and jitter constraints. In such case, an arbitrary injection of signalling messages in the return link stream can be disruptive. In other situations this may not be a problem, for example where the continuous carrier aggregates best-effort traffic. The methods used to solve this trade-off are system dependent. Here, two examples are described:

- RCST's capable of operating in mode (c) can logon in MF-TDMA mode without disrupting the continuous carrier and transmit the signalling there.
- RCST's not capable of mode (c) operation can make some provision for signalling within the continuous carrier, for example by setting aside a small amount of bandwidth for this purpose (e.g. assign one MAC layer packet out of every N , which is permanently dedicated to DULM messages).

This aspect need not be standardised, since the management of the bandwidth within the continuous carrier is system-dependent. The NCC must however be able to accept signalling messages from the RCST through either of these channels.

9.3.1 Forward Link Signalling

9.3.1.1 Carrier Description

Continuous carriers are defined in the forward link broadcast signalling through the same hierarchy of tables as MF-TDMA time slots. A continuous carrier is fundamentally identified in the TCT as a "time slot" with zero duration. This is not a valid value for genuine MF-TDMA time slots, so this signalling method does not interfere with normal system operation or with terminals that are unaware of the continuous carriers.

When the zero-duration option is chosen, certain physical layer parameters are interpreted slightly differently than they are for MF-TDMA time slots. The details are provided in clause 8.5.5.4 of the normative document.

The basic transmission scheme is DVB-S2, but the signalling provides "user space" for defining other air interfaces. Also, a number of parameters in the TCT entry are irrelevant for continuous carriers. These include `burst_start_offset`, `inner_code_type`, `inner_code_ordering`, `outer_coding`, `Route_ID_flag`, `ACM_flag`, `SAC_length`, `request_flag`, `M_and_C_flag`, `Group_ID_flag`, `Logon_ID_flag`, `capacity-requests_number`, `New_permutation`, `P0`, `P1`, `P2`, `P3`, `preamble_length` and `preamble_symbol`.

The following fields in the Time Composition Table are relevant for continuous carrier mode :

Table 12: TCT table use for continuous carrier mode

TCT	Continuous Carrier
Timeslot_composition_section(){	X
SI_private_section_header	X
timeslot_loop_count	X
For(i=0;i <= timeslot_loop_count;i++){	X
timeslot_id	X
symbol_rate	X
timeslot_duration	X ⁽¹⁾
burst_start_offset	
inner_code_type	
inner_code_ordering	
outer_coding	
inner_code_puncturing	X
modulation	X
baseband_shaping	X
timeslot_payload_type	X
Route_ID_flag	
ACM_flag	
MOB-flag	
SAC_length	
request_flag	
M_and_C_flag	
Group_ID_flag	
Logon_ID_flag	
capacity_requests_number	
New_permutation	
If((Inner_code_type == 1) and (New_permutation == 1)) {	
P0	
P1	
P2	
P3	
}	
preamble_length	
for (j=0;j<preamble_length;j++){	
preamble_symbol	
}	
while (!bytealigned){	
stuffing_bit }	X
}	
CRC_32	X
}	

- (1) The field « timeslot_duration » shall be equal to 0. This field identifies the continuous carrier mode and allows the enumeration of the timeslots.

The following fields in the Frame Composition table are relevant for continuous carrier mode:

Table 13: FCT table use for continuous carrier mode

FCT	Continuous Carrier
Frame_composition_section(){	X
SI_private_section_header	X
frame_ID_loop_count	X
for(i=0;i<=frame_ID_loop_count;i++){	X
frame_id	X
frame_duration	X ⁽¹⁾
total_timeslot_count	X
start_timeslot_number	X
timeslot_loop_count	X
for(j=0;j<=timeslot_loop_count;j++){	X
timeslot_frequency_offset	X
timeslot_time_offset	X ⁽²⁾
timeslot_id	X
repeat_count	X ⁽³⁾
}	X
}	X
CRC_32	X
}	X

Instantiation of the continuous carrier "time slot" in the FCT is done as for MF-TDMA time slots, with the observations that

- 1) The frame_duration parameter is used only to identify frames; it does not indicate any physical property of the continuous carriers defined in the frame;
- 2) The timeslot_time_offset parameter is used only to define the order of time slots within the frame as per the normal counting rules and hence to identify the time slot for assignment purposes. It does not indicate any physical property of the continuous carrier;
- 3) The repeat_count must be set to 0 (1 instance) for the continuous carrier "time slot";

Details are given in clause 8.5.5.3 of the standard.

Instantiation of frames containing continuous carriers is done in the SCT in the same basic manner as for any other frame. It is noted however that the frame_start_time parameter is used only to define the order of frames within the superframe as per the normal counting rules and hence to identify the frame for assignment purposes.

A number of parameters in an SCT entry are irrelevant for continuous carrier aspects of the frame. These include superframe_start_time_base, superframe_start_time_ext and superframe_duration. Details are given in clause 8.5.5.2 of the standard. The following fields in the Superframe Composition table are relevant for continuous carrier mode:

Table 14: SCT table use for continuous carrier mode

Syntax	Continuous Carrier
Superframe_composition_section(){	X
SI_private_section_header	X
superframe_loop_count	X
for(i=0;i<=superframe_loop_count;i++){	X
superframe_id	X
Large_Timing_Uncertainty_Flag	X
uplink_polarization	
superframe_start_time_base	
superframe_start_time_ext	
superframe_duration	X
superframe_centre_frequency	
superframe_counter	X
frame_loop_count	X
for(j=0;j<=frame_loop_count;j++){	X
frame_id	X
frame_start_time	X ⁽¹⁾
frame_centre_frequency_offset	X
}	X
}	X
CRC_32	X
}	X

(1) The field "frame_start_time" is only used for the enumeration of the frames.

9.3.1.2 Carrier Assignment and Release

Assignment and revocation of continuous carriers is carried out using the TBTP, in the same manner as for MF-TDMA time slots. The only difference is the interpretation of the assignment_type parameter, which is modified for continuous carrier "time slots" in order to reflect the selection of operations applicable to this type of transmission. Clause 10.4.5 of the standard provides detailed specifications of the RCST's behaviour. It should be noted that assignment and release of carriers may cause the RCST to change its operational mode.

9.3.1.3 Other Forward Link Signalling

Forward link signalling other than the carrier definition and assignment/release described above is generally either irrelevant for the continuous carrier operation or applicable to the entire RCST. Therefore, in most cases, no special provisions are required in order to distinguish between the MF-TDMA and continuous carrier air interfaces of the RCST. There are two exceptions to this: the Logon Initialize Descriptor and the Return Interaction Path Descriptor contain definitions of the VPI/VCI or PID to be used for traffic and signalling. In order to allow signalling of these parameters to be handled within the existing mechanisms, without imposing a need for distinct MAC addresses for the two air interfaces, previously reserved bits are used to allow distinction between the continuous carrier and MF-TDMA air interfaces for these two descriptors. Details are provided in clauses 8.5.5.10.4 and 8.5.5.10.17 of the normative document.

9.3.2 Return Link Signalling

As described above, DULM has been selected as the signalling method of choice for return link messages relevant to continuous carrier operation. A single information element is used for this purpose; this is defined in clause 10.4.4 of the normative document. This information element can carry all the required messages from the RCST to the NCC, including

- Temporary carrier assignment / release requests
- Permanent carrier assignment / release requests
- Acknowledgement of assignment/release messages, including accept/reject of assignment

9.4 Physical Layer

9.4.1 Synchronisation

System synchronisation is based on NCR distribution in the same manner as for MF-TDMA operation. Depending on system requirements, this open-loop synchronisation can be augmented by closed-loop control. However, RCST's operating in mode (b) do not have a logon ID and group ID assigned, which is necessary in order to use the CMT. It is therefore recommended to use the Correction Control Descriptor in the unicast TIM for closed-loop control of continuous carriers. It is further noted that burst timing control is not applicable to continuous carriers. Power control is addressed in section 9.4.4.

9.4.2 Encapsulation

The stream format of the continuous carrier is defined in clause 10.4.3 of the normative document as a single, packetised generic stream. ATM is mandatory; MPEG encapsulation must be supported if the RCST supports this in MF-TDMA mode. It should be noted that, when MPEG is used, the stream is a sequence of MPEG packets and therefore somewhat reminiscent of a Transport Stream. However, it lacks the basic signalling used for transport streams and is therefore still characterised as a generic stream.

Generic stream encapsulation may be supported in the future.

9.4.3 Modulation and Coding

The outer BCH codes adopted for each of the three inner LDPC codes are specified in Table C.2 of [i.1], reported hereafter for the sake of clarity. For the (4096,3072) LDPC code, the outer code is a (3072,3000) BCH code obtained by shortening of a (4095,4023) 6-errors correcting BCH code. For the (4096, 2048) and for the (4096, 1024), the outer codes are extended BCH codes with respective parameters (2048,1992) and (1024, 993). The first one is obtained by the 1-bit extension of a (2047,1992) 5-errors correcting BCH code, the second one is obtained by the 1-bit extension of a (1023,993) 3-errors correcting BCH code. The extension is performed by computing the additional parity bit as modulo-2 sum of all the codeword bits generated by the BCH encoder. The 1-bit extension increases the error correction capability of the code: for the (1023,993) the minimum distance is 7, while for its extended version the minimum distance is 8. Similarly, for the (2043,1992) BCH code the minimum distance is 11, while its extended version possesses minimum distance 12. The additional bit in these two cases can be either used for enhancing the outer error correction, or to increase the error detection capabilities of the outer code (i.e., sequences with odd Hamming weight at the output of the BCH decoder will be detected as erroneous).

Table 15: BCH code specification for very short FECFRAME

Outer code for the (4096,3072) LDPC	
Shortened BCH	(3072, 3000, t=6) from shortening of a (4095,4023) BCH
g(x) coefficients (g ₀ ...g ₇₂)	1011101011010010100111001100010011001100100010111101000100111010010001001
Outer code for the (4096,2048) LDPC	
Extended BCH	(2048,1992,t=5)
g(x) coefficients (g ₀ ...g ₅₅)	1011001101111001100010010000000001000010101010010101011
Outer code for the (4096,1024) LDPC	
Extended BCH	(1024,993, t=3)
g(x) coefficients (g ₀ ...g ₃₀)	1100100010001000100101010000101

All provisions reported in [i.2] for the BCH code which are not in contradiction with the above apply also to the very short FECFRAME..

9.4.4 Adaptive Operation

The provisions in [i.1] are intended to allow three types of adaptation to signal variations. This does not prevent the use of other techniques that can be implemented using the available signalling mechanisms. The three types of adaptivity foreseen are:

- RCST transmit power control
- Variable Coding and Modulation (VCM) under control of the NCC
- VCM with distributed control

These are described in the following sub-sections.

9.4.4.1 Transmit Power Control

Transmit power control can be implemented in essentially the same manner as for MF-TDMA. For the same reasons as described in section 9.4.1, it is recommended to use the TIM for correction messages. Furthermore, in order to facilitate the use of power control in combination with VCM, it is recommended to use the E_s/N_0 reporting option of the correction message.

9.4.4.2 VCM Controlled by the NCC

Variable Coding and Modulation (VCM) under the complete control of the NCC can be implemented using the carrier assignment and release mechanisms described in section 9.3.1.2. As also noted in clause 10.4.1.1 of the standard, such VCM operation can be performed "in place", i.e., effectively changing the modulation/coding parameters of a given carrier, if the NCC receiver allows this. To achieve this "in-place" operation, the various modulation/coding combinations are defined in the broadcast signalling as different "time slot" types, which are subsequently instantiated at the same carrier frequency in the FCT. It is the responsibility of the NCC to ensure that only one such "time slot" is assigned at any time.

The NCC will typically base decisions to change the modulation and coding on the signal quality measured on the received continuous carrier.

9.4.4.3 VCM with Distributed Control

As an alternative to the NCC-controlled VCM operation, it is possible to distribute parts of the control to the RCST. To facilitate this, the NCC can broadcast a list of modulation/coding combinations supported in its receiver. This list is contained in the Return Transmission Modes (RTM) descriptor, which is defined in clause 8.5.5.10.21 of the normative text. The list is organised per superframe; this allows management of a bank of receivers with different capabilities. The list may additionally contain threshold values, defining the minimum E_s/N_0 required at the NCC for reception of each modulation/coding combination.

RCST's that support this option must indicate this in the CSC burst when logging on as MF-TDMA terminals. They can subsequently use any of the listed modes at any time, without signalling changes. RCST's will typically use the signal quality (E_s/N_0) reported in the TIM as a basis for the decision to change.

RCST's that do not support the distributed VCM technique must use the modulation/coding parameters given in the TCT and implied by the assignment received in the TBTP. The same is true if no RTM descriptor is broadcast.

9.4.5 Spectrum Spreading

Return link continuous carriers may require the use of spectrum spreading for the same reasons as MF-TDMA transmissions; i.e., to reduce the spectral density, in particular off-axis. The spreading technique to be used is derived from that defined for DVB-S2 forward link transmissions and is described in clause 10.4.2 of the normative document. Details of the spreading parameters are included as part of the modulation/coding definition in the TCT entry for the carrier.

9.4.5.1 Return link continuous carrier code synchronization

Frame/code acquisition in the continuous mode return link is the first operation to be performed, i.e. before carrier and timing estimation. In fact, in the presence of direct sequence spread spectrum it is difficult to perform parameter estimation with sufficient accuracy before code acquisition. In the following, results on the code synchronization performances are shown taking into account the following assumptions:

- Symbol Rate (information symbol rate after coding & modulation) = 1 Msps
- Spreading factor (SF) = 1, 2,3,4, 8
- Chip rate = Symbol Rate *SF
- Modulation: QPSK
- Framing:
 - Short FEC frame (16200 coded symbols)
 - Data payload length: $16200/2 = 8100$ QPSK symbols
 - Unique Word length: 26 symbols, DVB pattern
 - Pilot: 11 fields of 36 symbols each
 - Frame Length = $SF*(8100 + 11 * 36 + 26 + 64) = SF*8586$ [chips]
- Operating points
 - $E_s/N_0 = 0.7\text{dB}$, $r = 1/2$; $E_s/N_0 = -1\text{dB}$, $r = 1/3$
 - $E_c/N_0 = E_s/N_0 - 10 \log_{10}(SF)$
- Propagation conditions:
 - LOS AWGN
 - Frequency uncertainty = 3.0 kHz (oscillator mismatch + Doppler)
- Timing recovery
 - No timing recovery prior to frame synchronization
 - Two hypotheses per chip are considered, worst case fractional timing delay of $0.25T_c$
 - Timing frequency error= 100 ppm
- Phase recovery
 - Phase noise: the same as for the FL scenario

9.4.5.1.1 Phase noise sensitivity assessment

In Figure 37, it can be seen that the spectrum spreading technique is insensitive to phase noise, which is therefore neglected in the analysis hereafter.

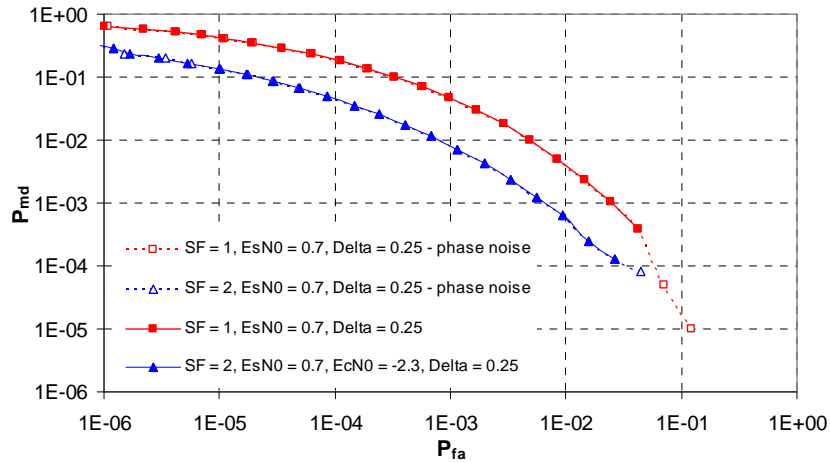


Figure 37 : Performance evaluation in the presence of phase noise

9.4.5.1.2 ROC performance in AWGN

In Figure 38, analytical and simulated ROCs are presented in AWGN with $E_s/N_0 = 0.7$ dB and chip time misalignment $\delta = 0.25$, in the exemplary scenarios with SF = 1, 2, 8. Notably, the analytical curves well validate the simulation results. This conclusion is further extendable to all possible SF values, allowing in the following to consider only simulation results. By comparing the different SF, it clearly emerges that the introduction of DS spreading improves ROC performance because the interference introduced by the unknown information data (self-noise) during the Start of Frame (SoF) search procedure is attenuated in this case.

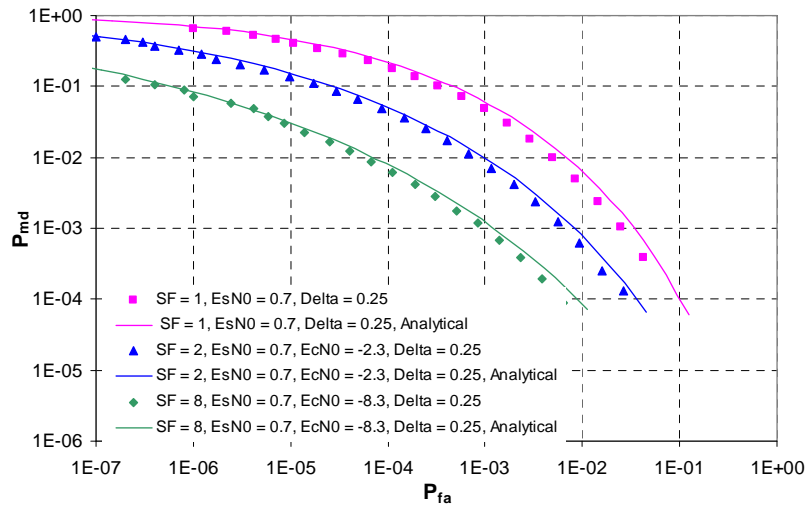


Figure 38: Simulated and analytical ROC performance at $E_s/N_0 = 0.7$ dB, with non ideal sampling ($\delta = 0.25$) considering spreading factors SF = 1, 2, 8.

By considering all possible SF values, the simulated ROC curves at $E_s/N_0 = 0.7$ dB reported in Figure 39 are obtained, where the same considerations with increasing SF still hold. Similarly, the performance at $E_s/N_0 = -1$ dB are reported in Figure 40.

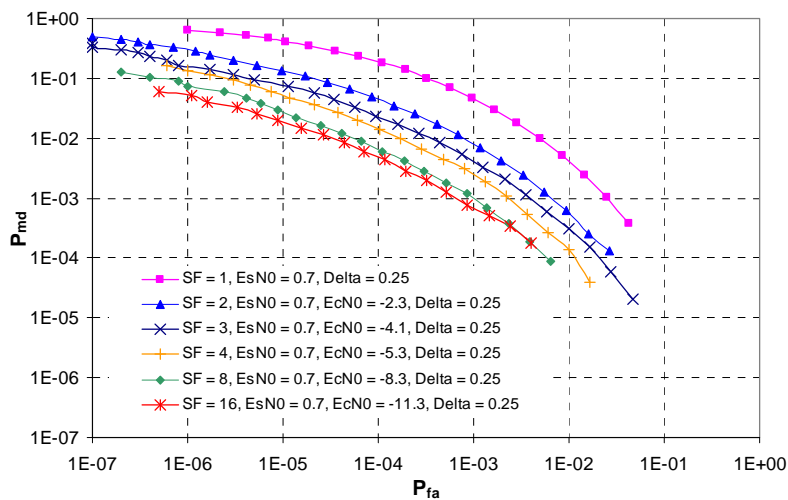


Figure 39: Simulated ROC performance at $E_s/N_0 = 0.7$ dB, with non ideal sampling ($\delta = 0.25$) considering spreading factors SF = 1, 2, 3, 4, 8, 16.

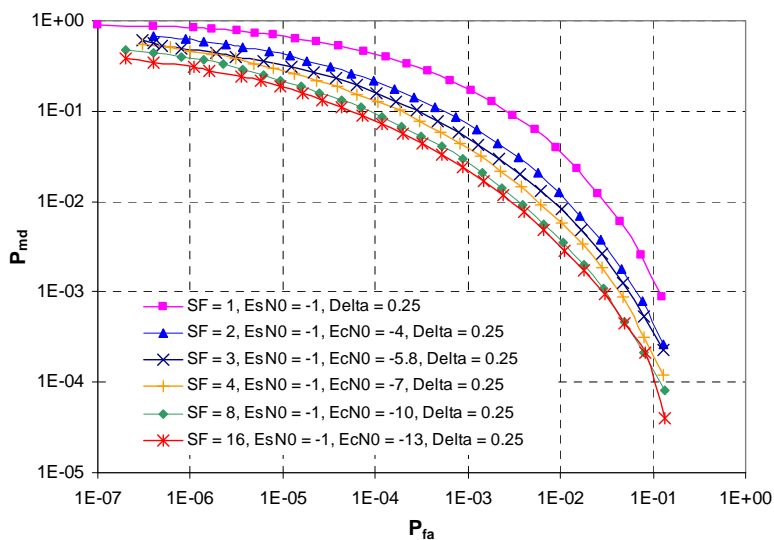


Figure 40: Simulated ROC performance at $E_s/N_0 = -1$ dB, with non ideal sampling ($\delta = 0.25$) considering spreading factors SF = 1, 2, 3, 4, 8, 16.

9.4.5.1.3 Mean Acquisition Time performance in AWGN

In Figure 41 and Figure 42, the mean acquisition time (MAT) is reported vs. the false alarm probability, considering a single dwell serial search procedure 0, with two hypotheses per symbol to contrast the chip timing uncertainty. In particular, the worst case condition for the sampling error is assumed, considering a symbol/chip timing misalignment $\delta = 0.25$. The procedure terminates when the correct alignment has been detected. In case of false alarms, the procedure restarts after a penalty time $T_p = 2 T_F$ (non-absorbing false alarm), being T_F the frame duration.

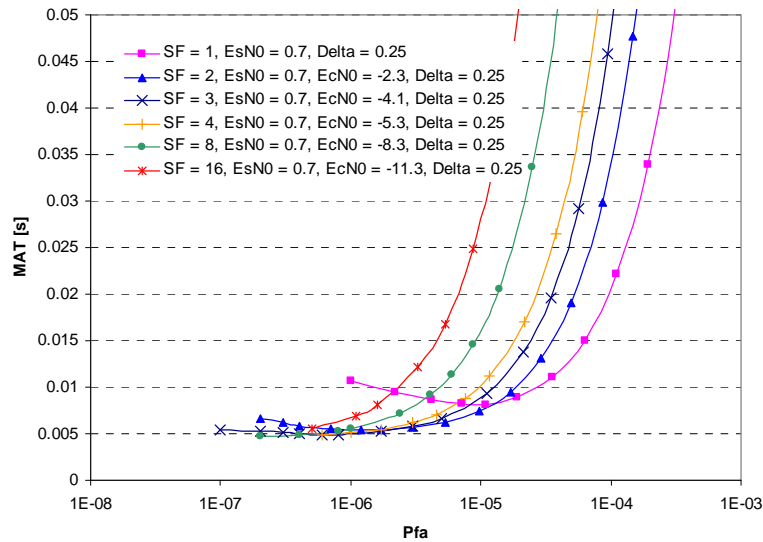


Figure 41: Mean Acquisition Time performance in AWGN at $E_s/N_0 = 0.7$ dB, with non ideal sampling ($\delta=0.25$) considering spreading factors SF = 1, 2, 3, 4, 8, 16.

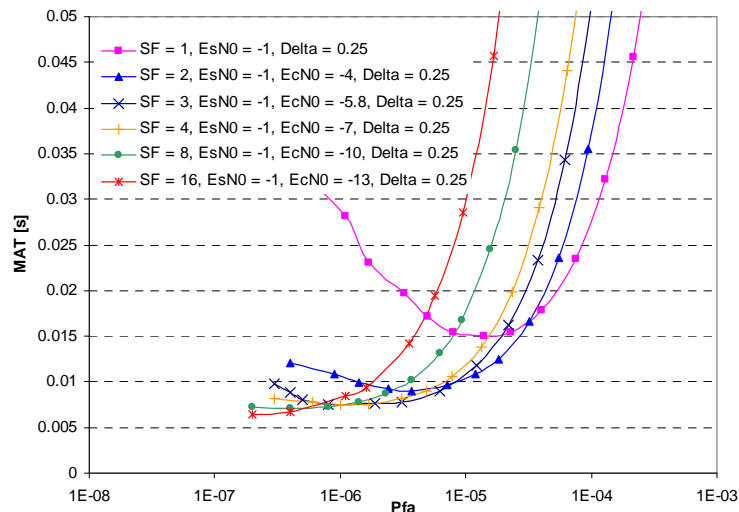


Figure 42: Mean Acquisition Time performance in AWGN at $E_s/N_0 = -1$ dB, with non ideal sampling ($\delta=0.25$) considering spreading factors SF = 1, 2, 3, 4, 8, 16.

The MAT performance confirms the results shown by ROC, i.e. performance improves by increasing the spreading factor. The best performance is achieved in correspondence of the minimum points of the MAT curves, which are summarized in Table 16. In any case, the worst case performance, which is 15 ms for SF = 1 and $E_s/N_0 = -1$ dB appears to be satisfactory.

Table 16: Minimum MAT in AWGN at $E_s/N_0 = 0.7$ and -1 dB, with non ideal sampling ($\delta = 0.25$) considering spreading factors SF = 1, 2, 3, 4, 8, 16.

E_s/N_0 [dB]	SF	E_c/N_0 [dB]	Pfa	Pmd	MAT [s]
0.7	1	0.7	1.9E-05	0.34	0.0082
	2	-2.3	1.7E-06	0.24	0.0054
	3	-4.1	8.0E-07	0.17	0.0049
	4	-5.3	6.0E-07	0.16	0.0049
	8	-8.3	2.0E-07	0.12	0.0047
	16	-11.3	5.0E-07	0.06	0.0055
-1	1	-1.0	1.4E-05	0.64	0.0150
	2	-4.0	3.7E-06	0.50	0.0090
	3	-5.8	8.0E-07	0.48	0.0075
	4	-7.0	1.0E-06	0.46	0.0075
	8	-10.0	4.0E-07	0.44	0.0071
	16	-13.0	2.0E-07	0.38	0.0064

Finally, Figure 43 shows the performance in terms of MAT for a smaller signal to noise ratio, $E_s/N_0 = -3$ dB, and SF = 2. The resulting MAT equal to 0.024s confirms that the requirements are largely satisfied, also in a scenario like this.

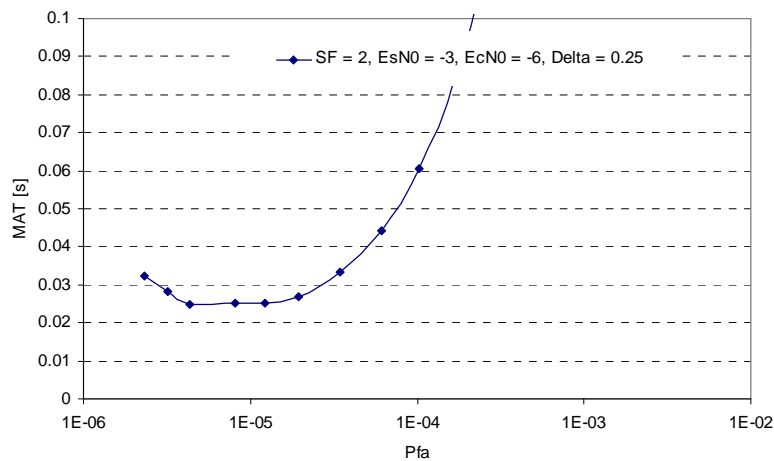


Figure 43: MAT in AWGN at $E_s/N_0 = -3$ dB with SF=2 and non ideal sampling

9.4.5.1.4 Performance in Rice Fading Channels

To conclude the performance analysis, in Figure 44 and Figure 45 ROC and MAT are respectively reported in the presence of Rice fading channel with Rice factor $K = 17.4$ dB and $E_s/N_0 = 0$ dB. The reported results are obtained adopting a semi-analytical approach that is valid for a channel coherence time larger than the SoF duration, but shorter than the frame duration. The proposed approach allows to achieve code/frame synchronization on average in 5ms for SF = 4 and 10 ms for SF = 1.

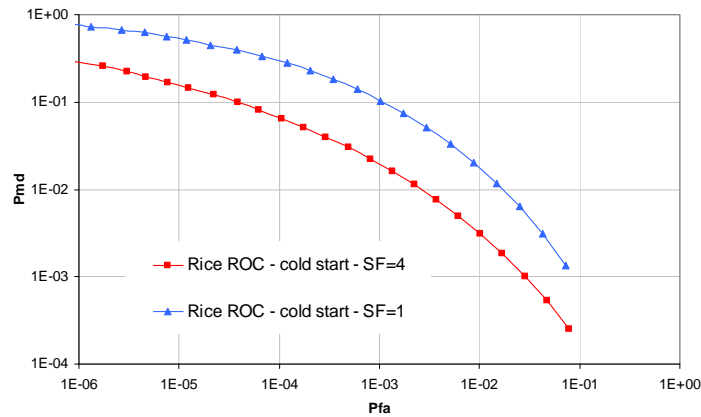


Figure 44: ROC performance in the presence of Rice fading channel, $K=17.4$, $E_s/N_0 = 0$ dB

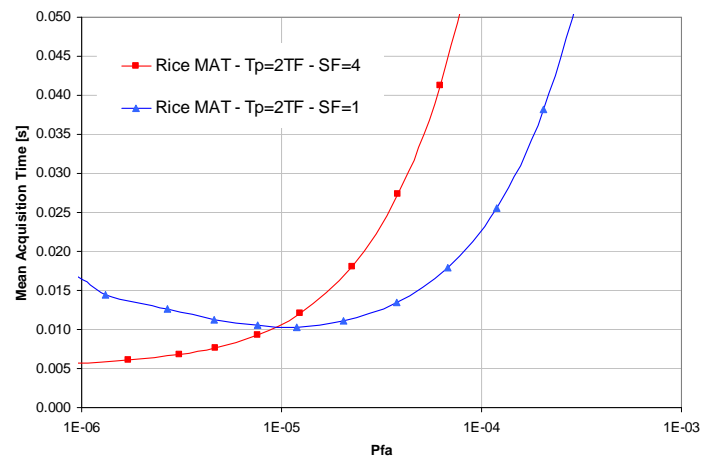


Figure 45: MAT performance in the presence of Rice fading channel, $K=17.4$, $E_s/N_0 = 0$ dB. Two values of penalty time are considered for comparison

9.4.6 Phase Noise impact

This section analyses the performance of the return link continuous carrier mode against the phase noise.

9.4.6.1 Simulation conditions

The DVB-S2 physical layer standard is assumed in the return link of a mobile RCS terminal with “high user traffic aggregation” with a baud rate of 1 Msps and FECFRAME length of 64800 symbols. Lower baud rates are most likely less interesting as the application of DVB-S2 SCPC solution pertains mainly to high rate systems. Both QPSK and 8PSK modulations are considered. Pilot blocks are inserted in the XFECFRAME as per the DVB-S2 standard, i.e. 36 QPSK pilot symbols every 1440 XFECFRAME symbols. These pilot symbols are meant to be used to help the carrier recovery circuits. However, DVB-S2 systems not employing pilots may also exist (pilot symbols are optional in DVB-S2) and would certainly be more prone to performance degradation due to phase noise.

In the computer simulations the satellite channel is considered ideal (AWGN) with the addition of phase noise.

The DVB-S2 demodulator is a typical proprietary design and constitutes a key aspect of the receiver terminal. A possible implementation has been proposed in [i.26] and is depicted in Figure 46:.

The demodulator of Figure 1 includes several synchronization sub-systems which cover coarse and fine carrier phase/frequency and timing recovery assuming typical initial synchronization errors. However, in the context of this analysis, carrier frequency offset and timing are assumed known so that only the phase noise and the thermal noise affect the receiver's performance.

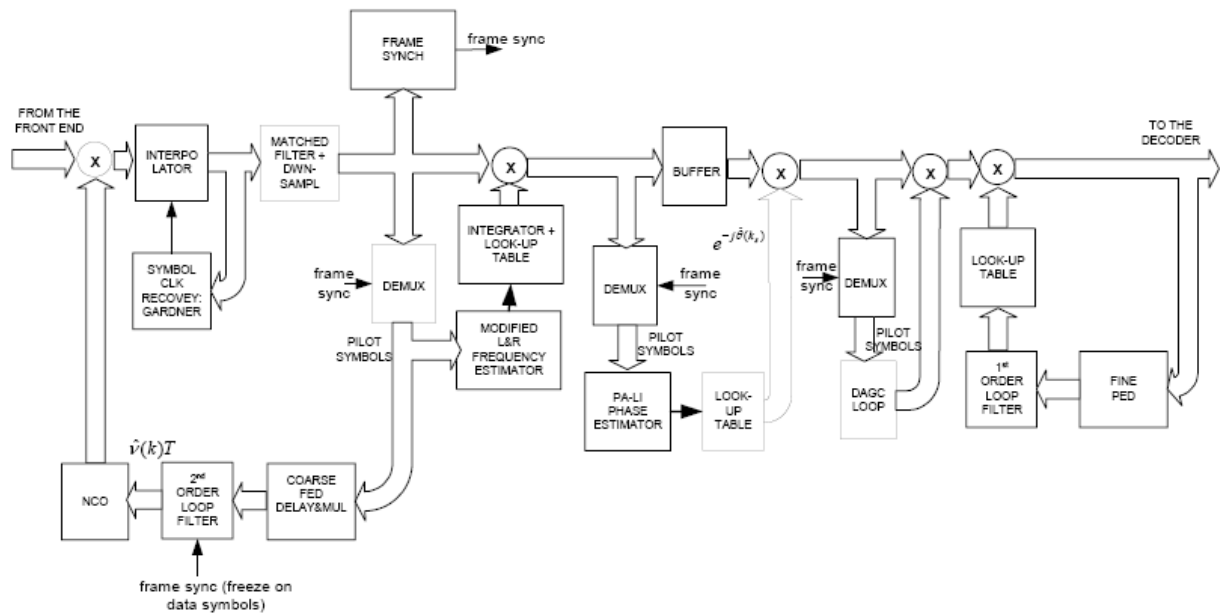


Figure 46: DVB-S2 demodulator according to the design in [i.26]

9.4.6.2.1 Phase noise generation and synchronization circuits

A phase noise process is generated to model the instabilities of the return link upconverter block according to the [i.3], for symbol rates greater than 128 Ksps. Figure 47 reports the power spectral density (PSD) of the reference mask and the one of the actual synthesized phase noise process.

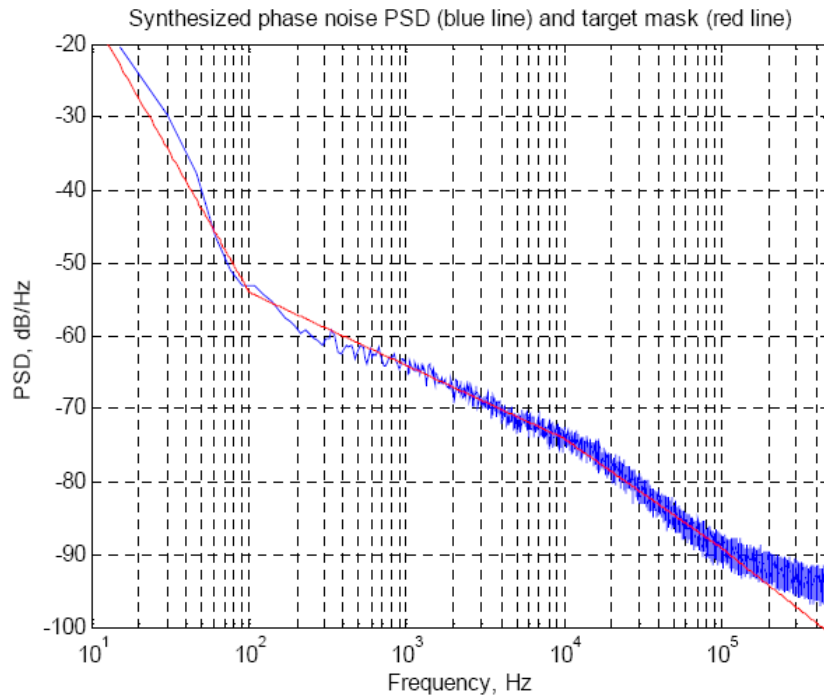


Figure 47: Phase noise Power Spectral Density: RCS model and actual synthesized phase noise PSD.

To counteract the phase noise impact, a carrier phase recovery circuit is considered based on the design proposed in [i.26]. It consists of pilot-aided phase estimator coupled with a linear interpolator. The phase is estimated on each pilot block and then a linear phase interpolator estimates the carrier phase over the symbols between two consecutive pilot fields. Figure 3 depicts a block diagram of such phase estimator.

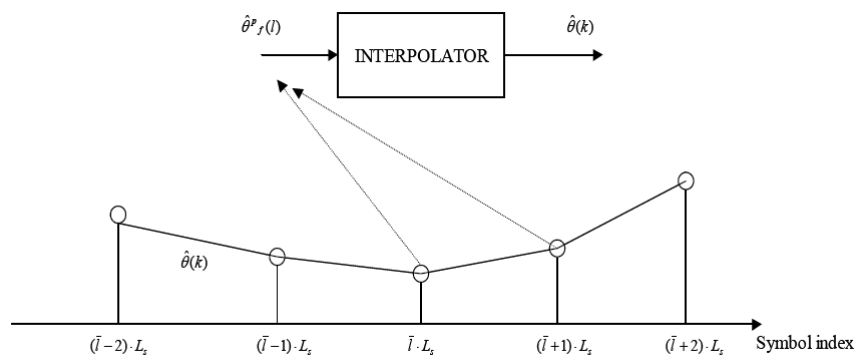


Figure 48: Phase linear interpolator. Circles represent the phase estimated on the pilot blocks whereas the lines are the carrier phase value estimated through linear interpolation.

9.4.6.2 Performance results

Figure 49 shows the BER performance of the a return link based on DVB-S2 with 1 Msps symbol rate and with and without the phase noise, for three MODCODS, QPSK $\frac{1}{2}$, QPSK $\frac{3}{4}$ and 8PSK $\frac{2}{3}$.

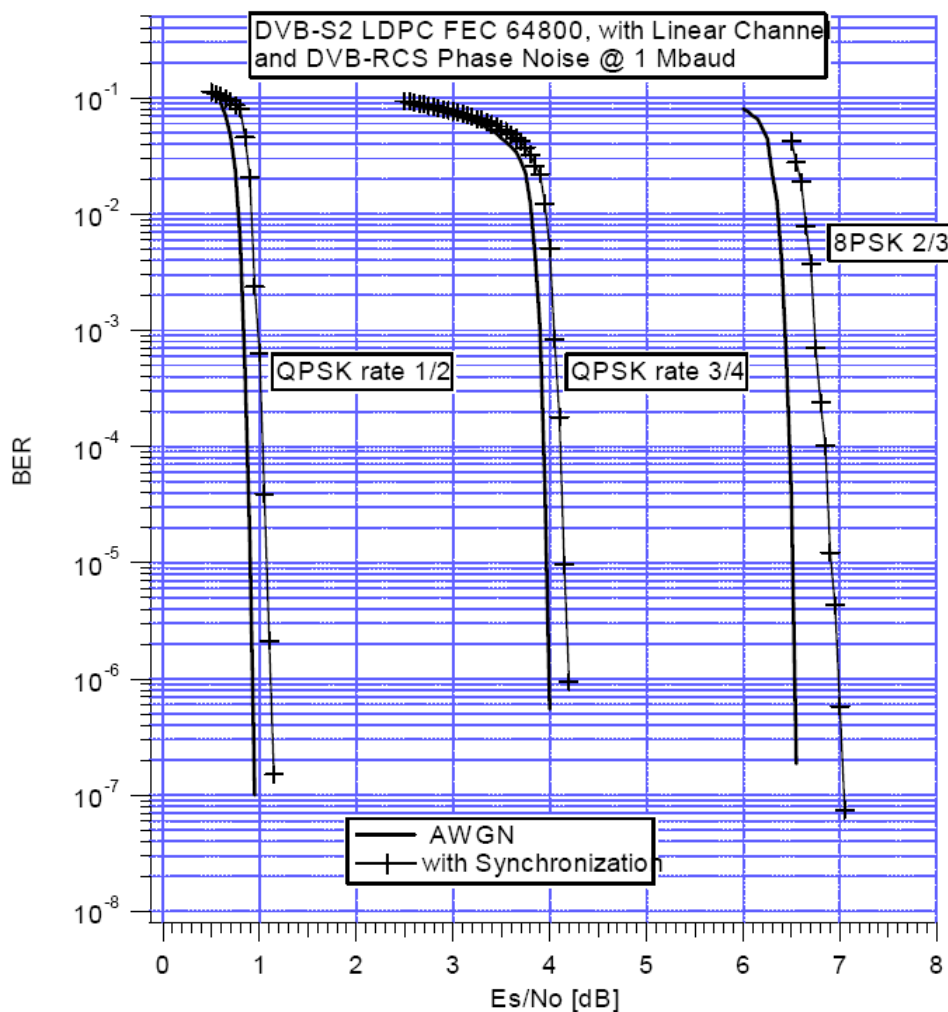


Figure 49: DVB-S2 BER performance in the presence of a typical DVB-RCS phase noise

9.4.6.1 Conclusion

From the previous results, it can be concluded that the impact of the phase noise on the performance of the continuous carrier in the range of symbol rate of interest is quite limited.

10 System and performance requirements

10.1 QoS requirements for user traffic

the following table shows some basic services and the basic QoS features for each service as general performance parameters, independent of the context and technology.

Table 17: target QoS per Service

Service	Symmetry	Typical Downlink Capacity	Typical Uplink Capacity	Max Delay* [2] [3]	Jitter Tolerance [3] [4]	Packet Loss Target [4]
Web browsing	Mostly one-way	50-200 kbps	A few kbps	< 2-4 seconds	N/A	Zero
File download	Mostly one-way	50 kbps – 1 Mbps	A few kbps	Several seconds acceptable	N/A	Zero
E-mail	Mostly one-way	10-100 kbps	10-100 kbps	Several Seconds acceptable	N/A	Zero
Instant Messaging	Mostly symmetric	250 bytes/message	250 bytes/message	< 2 seconds	N/A	Zero
Two-party VoIP	Symmetric unless VAD used	6 - 30 kbps	6 - 30 kbps	< 400 ms	< 75 ms	< 3%
N-party VoIP	Symmetric or asymmetric	(N-1)*6 – 30 kbps	6 - 30 kbps	< 400 ms	< 75 ms	< 3%
N-party Videoconference	Symmetric or asymmetric	(N-1)*50 – 460 kbps	50 – 400 kbps	< 400 ms	< 75 ms	< 1%
P2P Networks	Mostly asymmetric	10-200 kbps	10-50 kbps	Several Seconds acceptable	N/A	Zero
Video streaming	Mostly one-way	16-384 kbps	A few kbps	< 10 seconds	< 1ms	< 1%
Audio streaming	Mostly one-way	16-128 kbps	A few kbps	< 10 seconds	< 1 ms	< 1%
Interactive gaming	Mostly Symmetrical	8-128 kbps	8-128 kbps	400 ms	Small	Zero
Real time remote control applications	Mostly one-way	4 – 64 kbps	A few kbps	400 ms	< 75 ms	Zero

10.2 Analysis and recommendations for Signalling QoS

The signalling system in DVB-RCS is designed for channels with relatively low probability of loss or errors in both forward and return link. Mobile satellite channels can substantially increase the probability of loss of signalling elements. In particular, the railway propagation environment is characterised by a Rice channel with amplitude variations (typically 2 dB in Ku band), with some interruptions due to short obstacles, power arches and tunnels.

Non-line-of sight channels can substantially increase the probability of loss of signalling elements. Acceptable loss ratios and patterns are system dependent and can be set depending on equipment performance, tolerances and acceptable capacity losses. This section summarises the major impacts of imperfect signalling delivery. These issues should be considered when designing or configuring a system to operate with specific equipment and in a specific environment.

10.2.1 QoS requirement for the forward link signalling

- NCR distribution: The RCST must cease transmission when the variation of the local NCR replica is such that its transmission frequency and timing exceed the system's tolerances. For conventional fixed systems, a typical maximum time that the RCST can "free-wheel" is about 6 seconds. This period can be extended by using a high-quality local oscillator in the RCST.
- Broadcast signalling (e.g., PMT, SCT, FCT, TCT, SPT, bTIM): In fixed systems, the repetition interval of this signalling is typically of the order of several seconds. The main impact of loss of this signalling is an increase in the

log-on time corresponding to one repetition interval. Depending on whether the system implements "warm-start" features, the repetition period can be adjusted within the limits given in the normative document.

- Unicast signalling (e.g., uTIM, TBTP, CMT): Loss of a unicast TIM may result in failure of the RCST to behave in the expected manner. Most systems will already have mechanisms in place to monitor the RCST's behaviour and take corrective action as required, for example in the form of re-transmission of the TIM. Such mechanisms should certainly be applied to mobile systems.

Loss of the TBTP is largely a performance issue; the RCST will not receive capacity assignments. Applications using return link bandwidth will suffer accordingly, and the system utilisation will be reduced since the capacity described in the lost TBTP will essentially be wasted. The actual impact on performance will depend on the duration of the TBTP loss and the TBTP refresh rate.

Loss of a number of consecutive CMT's is considered a failure of the synchronisation maintenance procedure and causes the RCST to log off. A typical setting for fixed systems is to log the RCST off after three consecutive losses. The number is primarily determined by the tolerable time and frequency drifts, which are corrected by the messages contained in the TIM. For mobile systems, a suitable number of allowed losses before log-off is determined by the equipment stability and factors such as the maximum terminal speed, which in turn determines Doppler shift and timing drift rate.

10.2.2 QoS requirement for the return link signalling

- CSC bursts: Loss of a CSC burst will result in a longer log-on time, but will not normally have any other detrimental effects.
- ACQ bursts: Loss of a number of consecutive ACQ's is considered a failure of the coarse synchronisation procedure and causes the RCST to go to the off/standby state, where the CSC transmission is attempted anew. This will result in a longer log-on time, but will not normally have any other detrimental effects. The tolerated number of consecutive losses can be set according to the expected channel characteristics.
- SYNC bursts: Loss of a number of consecutive SYNC's is considered a failure of the fine synchronisation procedure or of the synchronisation maintenance and causes the RCST be logged off. It must then logon anew, either through a "cold start" or a "warm start", depending on its capabilities. A typical setting for fixed systems is to log the RCST off after three consecutive SYNC losses. The number is primarily determined by the tolerable time and frequency drifts, which may be corrected based on measurements on the SYNC bursts. For mobile systems, a suitable number of allowed losses before log-off is determined by the equipment stability and factors such as the maximum terminal speed, which in turn determines Doppler shift and timing drift rate. The loss of SYNC bursts can also have a performance impact, since
 - The NCC does not receive the capacity requests they may carry and
 - The power, timing and frequency measurements based on the SYNC burst are interrupted

Applications using return link bandwidth will suffer accordingly. Also, the request queues in the RCST and NCC can become de-synchronised by the loss of SYNC bursts. Most systems will already have mechanisms in place to remedy this, for example by the periodic use of AVBDC. Such mechanisms should certainly be applied to mobile systems.

Interruptions of the measurements can cause the failure of return path rain fade mitigation schemes during rapid fading events.

- In-band signalling: Loss of a the signalling carried in traffic bursts can have performance impacts in the same manner as the loss of SYNC bursts described above.

11 Simulation results and performance

11.1 Simulation scenarios

In the following section, performance results are reported based on simulations on selected channels. The selected channel models are representative of LOS situations: AWGN and Correlated Ricean (with $K > 17$ dB). Doppler spectrum impact is modelled according to the following section.

11.1.1 Channel model: Doppler Spectrum

Multipath propagation in the mobile channel causes a fast fading phenomena superimposed to the short fading effects originated by the trellises, power arches and bridges or tunnels as seen in section 4.2. This phenomenon can be characterised in the frequency domain by the Doppler Power Spectral Density (PSD), also referred to as Doppler Spectrum. Jakes PSD model is widely accepted to characterise this Doppler spectrum, however it is based on the assumption of omnidirectional antenna in both ends. In the Railroad Satellite channel, a highly directional terminal antenna in the train is considered and therefore the Doppler PSD function will be different from the Jakes spectrum.

The Doppler power spectrum received on the directional antenna is defined by the intensity of the received multipath components with their Doppler shift. We assumed that the azimuth Direction of Arrival (DOA) of multipath components is uniformly distributed over $(0, 2\pi]$. Only part of the multipath components will be received by the train terminal antenna due to the directional geometry. The Doppler shift components varies between $f_d \cos(\alpha - \phi)$ and $f_d \cos(\alpha + \phi)$, where α is the angle between the direction of movement and the direct Line-Of-Sight (LOS) component and 2ϕ is the beamwidth of the directional antenna, as depicted in the figure below.

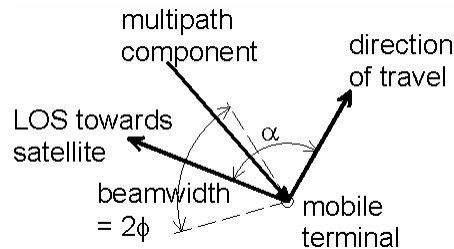


Figure 50: Multipath Geometry

For a detailed derivation of the Doppler PSD for directional antennas refer to [i.27] and [i.28]. Here we restrict to present the equation for sake of completeness without providing further details:

- if $0 \leq \alpha \leq \phi$

$$S_1(f) = \begin{cases} \frac{A}{f_d \sqrt{1 - \left(\frac{f}{f_d}\right)^2}} & \text{if } f_d \cos(\phi + \alpha) < f < f_d \cos(\phi - \alpha) \\ \frac{2A}{f_d \sqrt{1 - \left(\frac{f}{f_d}\right)^2}} & \text{if } f_d \cos(\phi - \alpha) \leq f < f_d \\ 0 & \text{otherwise} \end{cases}$$

- if $\pi - \phi \leq \alpha \leq \pi$

$$S_1(f) = \begin{cases} \frac{A}{f_d \sqrt{1 - \left(\frac{f}{f_d}\right)^2}} & \text{if } -f_d \cos(\phi - \alpha) < f < -f_d \cos(\phi + \alpha) \\ \frac{2A}{f_d \sqrt{1 - \left(\frac{f}{f_d}\right)^2}} & \text{if } -f_d < f \leq -f_d \cos(\phi - \alpha) \\ 0 & \text{otherwise} \end{cases}$$

- if $\phi < \alpha < \pi - \phi$

$$S_1(f) = \begin{cases} \frac{A}{f_d \sqrt{1 - \left(\frac{f}{f_d}\right)^2}} & \text{if } f_d \cos(\phi + \alpha) < f < f_d \cos(\phi - \alpha) \\ 0 & \text{otherwise} \end{cases}$$

where $f_d \approx vf_0/c$ denotes the maximum Doppler shift (v : train velocity, f_0 : carrier frequency, c : speed of light).

11.2 Performance in LOS channels

11.2.1 Forward link PER performance

Physical layer performance has been carried out considering the complete DVB-S2 transmit-receive chain in a railway environment. LOS channel condition (Rice factor $K=17.4$ dB) is assumed. Two different train speeds have been considered: 30 and 300 km/h. Simulations have been carried out under the hypothesis of a receiving antenna with high directivity. In other words, the Doppler spectrum is reduced with respect to the Jakes model to take into account the decreased multi-path power captured by the receiving antenna (see section 11.1.1). The carrier phase noise is generated according to the phase noise mask reported in the DVB-S2 implementation guidelines [i.29] and the non-linear distortion introduced by the HPA has been kept into consideration according to the data reported in section 5.1.1.1. The predistortion technique applied in the following is based on a fractional approach (post matched filter) described in [i.30]

Table 18 – HPA related parameters

Modulation	IBO (dB)	OBO (dB)
QPSK	0.5	0.33
8-PSK	1	0.43
16-APSK	2	1.08

For the performance analysis purposes, the digital receiver architecture depicted in Figure 51 has been considered. It is worthwhile noting that since this analysis is focused on the data detection performance of the receiver, the initial frequency acquisition and frame synchronization operations are assumed to be successfully accomplished, (see section 5.1). Hence, the white block of Figure 51 has not been considered in the software simulations for this analysis.

In the following, a brief description of the different sub-subsystems is reported in order to ease the understanding of the performance results reported in this section. The carrier frequency coarse correction is firstly performed to allow matched filtering with minimal inter-symbol interference re-growth; then the clock recovery for timing adjustment is performed by a digital interpolator. A demultiplexer is used to separate pilots from data symbols carried in the PLFRAME. The pilot symbol stream is used by the following four sub-systems: the noise level estimator, the digital automatic gain and angle control (AGAC), the block in charge of tracking the residual frequency offset and carrier phase and, finally, the coarse frequency acquisition loop (not performed in hot start). On the other path, the data symbols feed the hard/soft demodulator. The demodulator provides the hard decisions on data symbols as a feed-back for carrier frequency and phase tracking, and computes the soft initial A Posteriori Probability (APP) on the received information bits. Finally, the APPs are de-interleaved and provided to the LDPC-BCH decoder. From the point of view of the sequential order of operations, during initial acquisition, the first operating sub-system is the clock recovery; since it can operate in the presence of large carrier frequency errors. For example, the Gardner timing detector [i.31] exhibits good estimation performance also in the presence of rather high-carrier frequency mismatch. Then, after the frame synchronization algorithm, the pilot symbols can be used to initiate the frequency recovery loop. Upon reaching the steady state, the fine channel tracking operations is run. In particular, the adopted noise level estimation algorithm is derived from the maximum likelihood (ML) theory [i.32]. The data-aided (DA) version is considered, since the estimator operates on the PLHEADER (90 known symbols: 26 from the SOF field and 64 from the MODCOD information). The digital AGAC sub-system [i.33] is a feed-forward algorithm based on an ML approach which exploits the presence of the pilot symbols by estimate the channel coefficients as follows:

$$\hat{c}_k = \frac{\sum_{i=0}^{P-1} d_i^* r_i^{(p)}}{\sum_{i=0}^{P-1} |d_i|^2}$$

where P is the number of pilot symbols, d_i is the known pilot symbol, and finally $r_i^{(p)}$ is the received pilot symbol.

Each fading estimate is then linearly interpolated between consecutive pilot fields to better tracking the channel propagation fluctuations. This solution is supported by the fact that the channel coherence time is always longer (90 μ s; at least) than the time distance between two consecutive pilot slots (50 μ s at 27.5 Msp/s). Finally, the AGAC angle estimate initializes the block in charge of tracking the residual frequency offset and carrier phase-noise fluctuations. For this purpose, a well-known second-order loop filter is implemented.

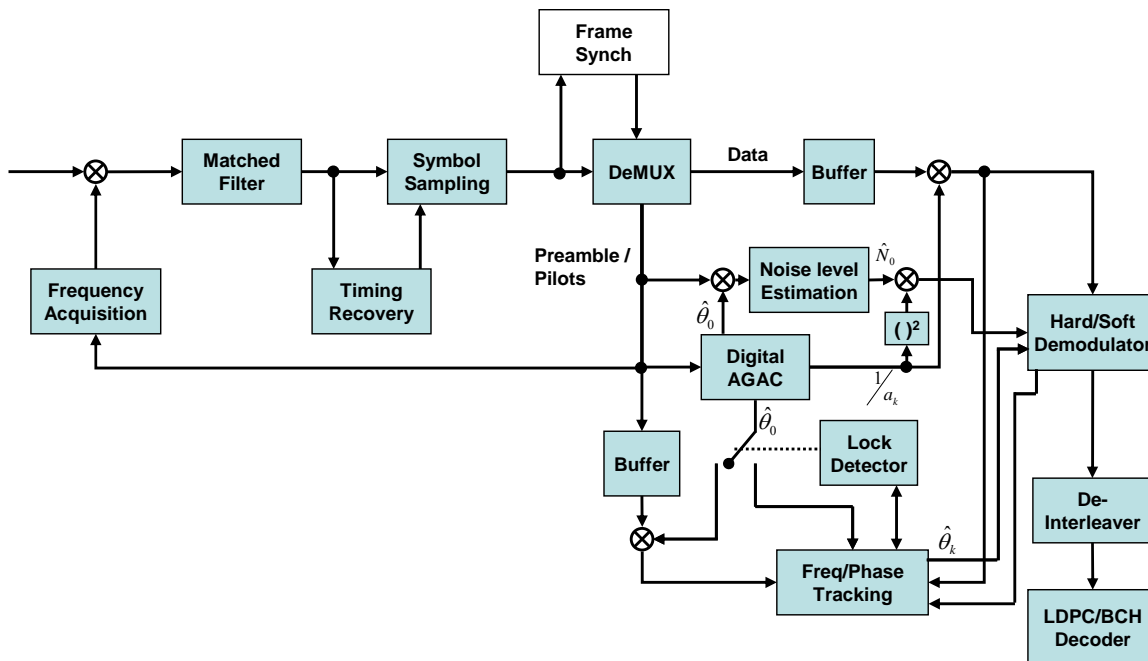


Figure 51: Block diagram of the digital receiver

Figure 52 and Figure 53 report packet error performance measured on the DVB-S2 DATAFIELD (see Figure 3) without spreading, for 30 and 300km/h, respectively, confirming that in LOS conditions, the detection performance is still satisfactory for both high and low train speed.

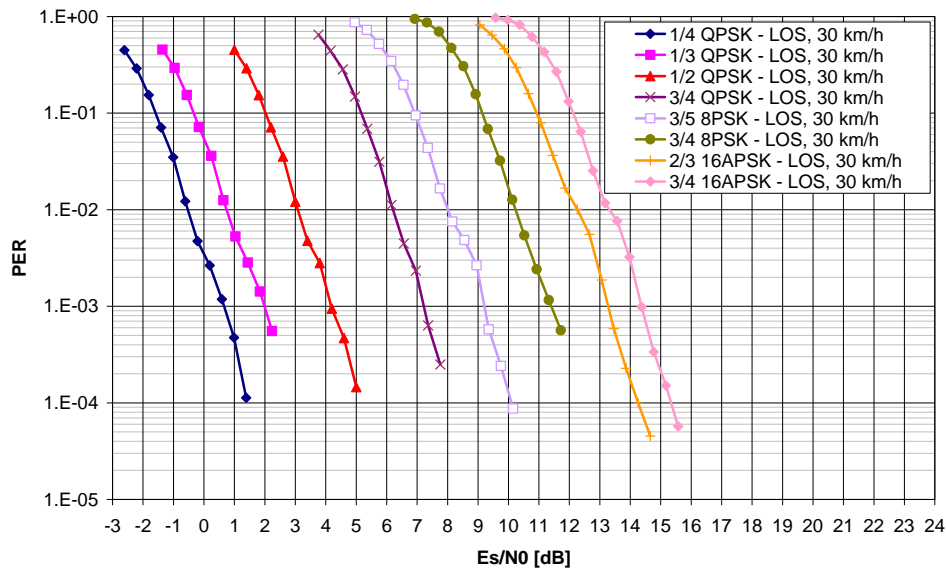


Figure 52: DVB-S2 performance for several MODCODs, in LOS channel condition, Ku band, no spreading, low train speed

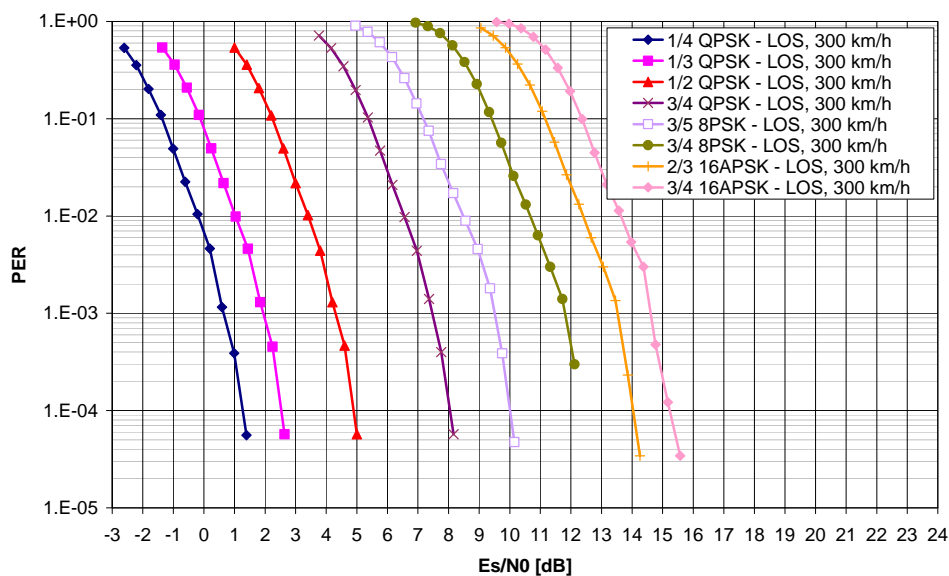


Figure 53: DVB-S2 performance for several MODCODs, in LOS channel condition, Ku band, no spreading, with and high train speed.

11.2.2 Forward link spectrum spreading performances

In the following, the impact on performance of the introduction of spread spectrum techniques is presented in terms of BER and PER. In particular, the following system parameters have been considered:

- MODCOD: 1/4-QPSK
- Chip rate = 27.5Mchip/sec;
- Symbol rate = Chip rate / Spreading factor
- Train speed = 300 km/h;
- Propagation channel: AWGN and correlated ricean channel with Rice factor = 17 dB

- Satellite HPA IBO = 0.5 dB
- No interference from adjacent satellites

First of all, the robustness of the DVB-S2 spread signal with respect to non-linear distortion is presented in Figure 54 with AWGN channel. The comparison between the spread and non-spread signals highlights that the signal spreading slightly increases the robustness to non-linear distortion.

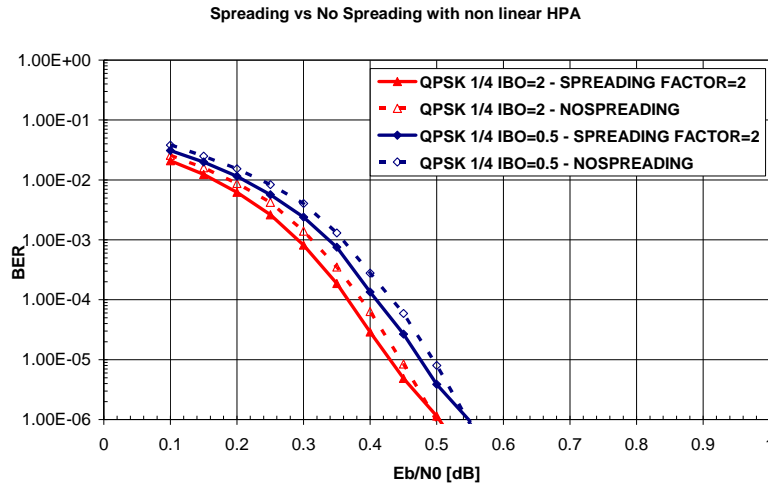


Figure 54: Comparison between spread and not spread signal in AWGN channel with the presence of non-linear HPA.

The terminal mobility impact on spread waveform performance is analysed in Figure 0 5. The same chip-rate has been considered, thus the transmission symbol rate is 13.75 Msps and 6.85 Msps for SF=2 and SF=4, respectively. As noted before, spectrum spreading introduces a slight gain with respect to the unspread signal. The real benefit of the introduction of the spreading factor is in term of link budget, showing a gain of 3 dB and 6 dB for spreading factor 2 and 4, respectively.

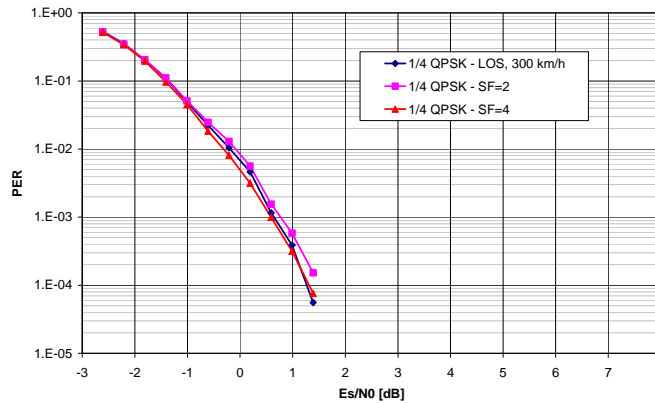


Figure 55: Comparison between spread and not spread signal in LoS channel with the presence of non-linear HPA

Annex A: Rate of Beam Roll-Off

A.1 Basics

This appendix derives the rate of change of gain experienced when a user moves in a satellite spot beam. Figure 56 shows a cross section of the main lobe of a typical beam. For simplicity, we assume that the beam has a Gaussian main lobe. This is a good approximation for simple circular and elliptical beams. For irregularly shaped beams, the roll-off near the edge — which is the region of most interest for the present investigation — can often be approximated by a Gaussian (parabolic in dB) shape.

In a Gaussian beam with 3-dB beam width B_3 , the gain loss $G(\Theta)$ at a position offset from the boresight by an angle Θ is given by

$$G(\Theta) = 12 \left(\frac{\Theta}{B_3} \right)^2 \quad (\text{A.1})$$

Let G_{eoc} be the contour corresponding to the loss that defines the edge of the beam. Typically, G_{eoc} is between 2 and 6 dB. From (A.1), the edge-of-coverage off-axis distance is

$$\Theta_{eoc} = \sqrt{\frac{G_{eoc}}{12}} B_3 \quad (\text{A.2})$$

In Figure 56, the useful beamwidth B_u is illustrated. This is of course given by $B_u = 2 \Theta_{eoc}$.

A.1.1 Rate of change

From (A.1), the variation of the gain with Θ is

$$G'(\Theta) = \frac{\partial G(\Theta)}{\partial \Theta} = \frac{24\Theta}{B_3^2} \quad (\text{A.3})$$

This is the slope in the radial direction. The maximum rate of change occurs at the edge of the beam:

$$G'(\Theta_{eoc}) = \frac{24\Theta_{eoc}}{B_3^2} = \frac{4\sqrt{3G_{eoc}}}{B_3} \quad (\text{A.4})$$

Note that these expressions are independent of the units used to express the off-axis distances.

A.1.2 Practical Example

The smallest spot beams produced by current geostationary satellites are of the order of 0.7° . Current limitations on attitude control accuracy are of the order of $\pm 0.05^\circ$. It is therefore unlikely that we will see spot beams much smaller than, say, 0.4° in the near or medium term. This is particularly the case in K_u -band, which is the main target for this investigation. At 11 GHz, a beam width of 0.4° requires an aperture diameter of at least 4.8m. Such a large aperture is difficult to launch; and the accuracy necessary at this frequency is difficult to achieve with inflatable or unfurlable apertures. We therefore use 0.4° as a worst-case example of the 3 dB beam width. From geostationary orbit, a 0.4° beam has a diameter of 250 km.

We also assume, as a worst case, that the beams are used to the 6 dB contour (This corresponds to a useful diameter B_u of ~350 km). From (A.4), the gain variation in the radial direction at the 6 dB contour is ~0.068 dB/km. For an aircraft flying at 1000 km/hr (0.278 km/s), this corresponds to a maximum slope of just under 0.02 dB/s. In other words; if a handover can be carried out in 10 seconds, a margin of 0.2 dB for gain variations is sufficient.

A more typical example of current Ku-band satellites has beams with 3 dB width of 1.5° , used to the 4 dB contour. In such a beam, the maximum slope is 0.015 dB/km, or 0.004 dB/s for the aircraft at 1000 km/hr.

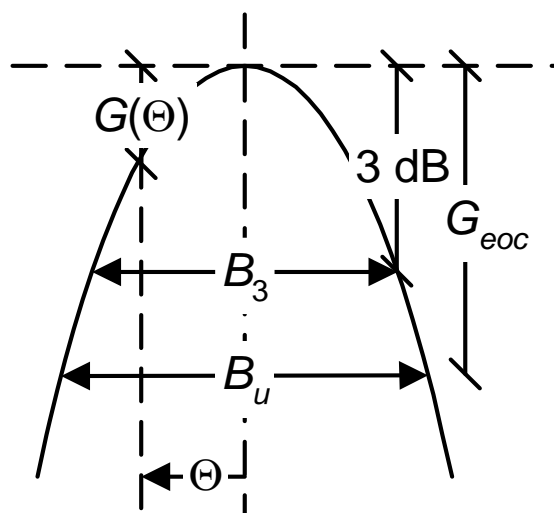


Figure 56: Cross-section of beam with Gaussian shape main lobe.

Annex B:

Continuous carrier mode vs. MF-TDMA mode comparison

B.1 Introduction

The following chapter aims at comparing the DVB-RCS MF-TDMA scheme with the new optional continuous carrier mode. It gives guidelines on the way to determine when the continuous carrier mode is more efficient compared to the standard MF-TDMA. It indicates on some concrete scenarios the needed bandwidth for both the cases.

B.2 Hypothesis

B.2.1 Satellite hypothesis

The assumptions for the satellite are the following :

- Continental Ku spot (5000 km)
- Channel Bandwidth 36 MHz
- Satellite G/T :
 - 0 dB/K (End of Coverage) for an Aircraft and Maritime mobile coverage up to 10 dB/K (Center of Coverage)

- 4 dB/K (End of Coverage) for a Train mobile coverage up to 10 dB/K (Center of Coverage)
- High Power Amplifier (HPA) mode :
 - Return link : Fixed Gain Mode (FGM) mode with Noise Power Ratio (NPR) of 18 dB and typically OBO=4.0 dB (linearized TWTA)

B.2.2 Terminal hypothesis

The assumptions for the satellite terminal are the following :

- Satellite terminal antenna size : range from 0.3 to 1.2 m (long axis size if elliptical)
- Antenna Tx efficiency of 58% (typical average value for satellite mobile terminal)
- Satellite terminal system temperature of 160 K (typical average value for satellite mobile terminal)
- Antenna radiation pattern based on recommendation ITU-R S.1428 (applicable for the Gateway and also applied for the mobile terminal in this analysis)
- HPA and feeder losses of 1.5 dB.

B.2.3 Regulation constraints

The regulatory constraints [i.19] [i.20][i.21] [i.22] [i.23][i.24][i.25] are taken into account in the system dimensioning and they will be considered for the preliminary and final system dimensioning.

They are applicable to the HUB, the satellite and to the satellite terminal with specific constraints for aircraft (AES : Aircraft Earth Station [i.19]) and Vessel (ESV : Earth Stations on board Vessel [i.21]). These constraints are defined in terms of limitation on emission at ground level and satellite level in terms of EIRP density.

The constraint for the HUB or satellite terminal are defined in terms of :

- Off-axis EIRP density towards adjacent systems (mobile, fixed and radio astronomic)
- Minimum elevation angle
- Minimum antenna diameter (only for Vessel (ESV))
- EIRP and EIRP density towards the horizon

And interference received at :

- Ground level : HUB or satellite terminal in terms of Off-axis EIRP density from adjacent satellite ground system
- Satellite level in terms of EIRP density from adjacent satellite

For each identified constraints per regulatory instance, the reference of the applicable rules has been used. Two distinctive scenarios in terms of regulatory constraints are analysed: ETSI (European) and FCC (North American).

To be noted that some identified constraints concerning the interference received coming from other terrestrial services can not be clearly defined yet. No specific hypotheses have been defined at this time of this analysis.

More over, the regulatory constraints defined and used for the system dimensioning are only known general limitations. Local limitations on emission are identified but as they are specific to the country and the type of interfering services no specific hypotheses have been taken into account.

B.2.4 System Scenario definition

For the comparison of the different systems, a fixed number of mobiles is considered and then the corresponding needed bandwidth is computed.

In the following, a pool of 100 mobiles in the system is considered.

The following 3 markets are studied :

- Aeronautical case : Antenna size of 60 cm and satellite G/T of 0 dB/K
- Railway case: Antenna size of 80 cm and satellite G/T of 4 dB/K
- Maritime case : Antenna size of 80 cm and satellite G/T of 0 dB/K

The FCC and ETSI regulation context will be addressed in the analysis.

The 2 possible return link systems are analysed and compared :

- Classical DVB-RCS system with MF-TDMA access. For FCC regulation compatibility, spectral spreading on the DVB-RCS waveform is considered. The dimensioning of the system is based on the mean data rate per mobile.
- DVB-S2 system with SCPC access method. The dimensioning of the system is based on the peak data rate per mobile.

B.2.5 Return link budget hypothesis

The general hypotheses for the return link are the following :

- The link budget is supposed to be in AWGN LOS condition for all mobile type.
- Symbol rate of up to 2 Msymb/s
- System margin of 1.5 dB (corresponding to RF impairments)
- Rain margin of 3 dB for Europe (worst case value in Europe for 99.8 % of availability) and 5 dB for USA (worst case value in USA for 99.8 % of availability)

For the return link, under rain conditions, the useful terminal to adjacent satellite path is affected by the same rain attenuation as nominal path.

B.2.6 Traffic scenarii

The considered traffic hypothesis are the following :

- A mean traffic of 7 kb/s per active end user (Web usage)
- The number of active users per mobile is around few tens of users.
- The number of active users per mobile is variable in time and supposed to be independent between mobiles.

Different hypothesis on the number of active users per mobile are considered. They are summarised in Table 19.

Table 19: Traffic profiles

Traffic Case	Mean Number of users	Peak Number of users	Typical capacity of the mobile	System usage (Load per mobile)
1	40	120	500 passengers	Low
2	60	120	500 passengers	Medium
3	100	120	500 passengers	High
4	20	60	250 passengers	Low
5	30	60	250 passengers	Medium
6	50	60	250 passengers	High

The corresponding mean and peak data rate per mobile is given in Table 20.

Table 20: Traffic Profiles data rate

Traffic Case	Mean data rate per mobile (kb/s)	Peak data rate per mobile (kb/s)	Peak/Mean data rate ratio
1	280	840	3
2	420	840	2
3	700	840	1.2
4	140	420	3
5	210	420	2
6	350	420	1.2

B.3 Return link overall efficiency

B.3.1 MF-TDMA

The global forward link spectral efficiency takes into account :

- The carrier filtering specified by the roll-off. The following analysis assumes 0.35 for the roll-off.
- The spectral spreading allowing compliance with the regulation constraints for small antennas. Spreading factor from 1 to 8 will be considered in the analysis.
- The MODCOD efficiency depending on the modulation (QPSK) and the coding rate (1/2, 2/3, ...).
- The frame efficiency taking into account the bursts guard times, the pre-ambls and the signalling bursts (CSC and SYNC). It also depends on the carrier symbol rate and the size of the bursts. A typical value is 85%.
- The IP packets to ATM packets encapsulation efficiency. This efficiency depends on the size of the IP packets. For the analysis, a mean IP packet size of 1400 bytes is considered. So, the encapsulation efficiency is around 85%.
- The DAMA efficiency. The MF-TDMA access scheme allows to have a very good statistical multiplexing. A typical value for the DAMA efficiency is 90%.

The overall efficiency is computed by :

$$\gamma_{Overall} = \frac{\gamma_{MODCOD} \cdot \gamma_{Framing} \cdot \gamma_{IP/ATM} \cdot \gamma_{DAMA}}{(1 + RollOff) \cdot SpreadingFactor}$$

Finally, taking into account all the efficiencies, the overall efficiency is given in Table 21.

Table 21: MF-TDMA overall efficiency

MODCOD	MODCOD Efficiency (Bit/Symb)	Spreading factor	Spectral Efficiency (Bit/s/Hz)	Es/N0 (dB) (PER = 10 ⁻⁵) & ATM cells
Spreaded QPSK 1/2	1.00	8.00	0.06	-5.4
Spreaded QPSK 1/2	1.00	6.00	0.08	-4.2
Spreaded QPSK 1/2	1.00	5.00	0.09	-3.4
Spreaded QPSK 1/2	1.00	4.00	0.11	-2.4
Spreaded QPSK 1/2	1.00	3.00	0.15	-1.2
Spreaded QPSK 1/2 or BPSK 1/2	1.00	2.00	0.23	0.6
Spreaded QPSK 2/3 or BPSK 2/3	1.33	2.00	0.30	2.4
QPSK 1/2	1.00	1.00	0.45	3.6
QPSK 2/3	1.33	1.00	0.61	5.4
QPSK 3/4	1.50	1.00	0.68	6.3
QPSK 4/5	1.60	1.00	0.73	7.3
QPSK 6/7	1.71	1.00	0.78	8.5

B.3.2 Continuous Carrier spectral efficiency

The global spectral efficiency for the continuous carrier mode in the return link takes into account :

- The carrier filtering specified by the roll-off. We will take 0.2 for the roll-off.
- The MODCOD efficiency depending on the modulation (QPSK, 8PSK, ...) the coding rate (1/4, 1/3, 1/2, ...), the pilots and the PLHeader.
- The MPEG-2 packets to BBFrame encapsulation efficiency. This efficiency depends on the data rate variations (Peak data rate over mean data rate ratio). These ratio depends on the number of users per terminal at a given time. For the analysis, a raw spectral efficiency is computed taking a value of 1 for the MPEG-2 to BBFrame encapsulation.
- The IP packets to MPEG 2 packets encapsulation efficiency. This efficiency depends on the size of the IP packets and the use (or not) of the section packing technique. For the analysis, a mean IP packet size of 1400 bytes is considered. Considering the use of the section packing option, the encapsulation efficiency is around 90%

The overall efficiency is computed by :

$$\gamma_{Overall} = \frac{\gamma_{MODCOD} \cdot \gamma_{MPEG/BBFrame} \cdot \gamma_{IP/MPEG}}{(1 + RollOff)}$$

Finally, taking into account all the efficiencies, the overall efficiency is given hereunder :

Table 22: DVB-RCS Continuous Carrier mode overall spectral efficiency

MODCOD	MODCOD Efficiency (Bit/Symb)	Overall Spectral Efficiency (Bit/s/Hz)	Es/N0 (dB)
QPSK 1/8	0.24	0.18	-5.00
QPSK 1/6	0.32	0.24	-3.90
QPSK 1/5	0.38	0.29	-2.57
QPSK 1/4	0.48	0.36	-1.6
QPSK 1/3	0.64	0.48	-1
QPSK 2/5	0.77	0.58	0
QPSK 1/2	0.97	0.72	1.3
QPSK 3/5	1.16	0.87	2.5
QPSK 2/3	1.29	0.97	3.4
QPSK 3/4	1.45	1.09	4.3
QPSK 4/5	1.55	1.16	5
QPSK 5/6	1.62	1.21	5.5
8PSK 3/5	1.74	1.30	6.2
8PSK 2/3	1.94	1.45	7
8PSK 3/4	2.18	1.63	8.3
16APSK 2/3	2.57	1.93	9.6
16APSK 3/4	2.90	2.17	10.7
16APSK 4/5	3.09	2.32	11.6
16APSK 5/6	3.22	2.42	12.1
16APSK 8/9	3.44	2.58	13.4
16APSK 9/10	3.48	2.61	13.6

B.3 Capacity Analysis

B.3.1 Methodology

In order to evaluate the return link capacity, the following methodology is considered:

- The link budget is constrained by the uplink path (Terminal to satellite), so the main parameters are the terminal EIRP and the satellite G/T. Another main constraint is the regulation limits. It defines the terminal maximum power spectral density toward the satellite depending on the terminal antenna size.
- Taking into account the adjacent system interferences power density, the achievable Es/N0 (Link budget) is computed at satellite level depending on the terminal antenna size and the satellite G/T.
- For the train and the maritime case, the rain margin for the achievable Es/N0 is considered
- After that, using the values in the overall spectral efficiencies, the MODCOD is determined and then the global spectral efficiency. Considering a transponder of 36 MHz, one can compute easily the useful aggregated data rate at IP level.
- Finally, the needed bandwidth for each case considering a fixed system size is computed. The maximum number of simultaneous mobiles to be considered will be 100. The assessment is done for the 3 markets (Aeronautical, Railway and Maritime), the 6 traffic load cases and the 2 regulation context (ETSI & FCC).

As far the adjacent satellite terminal interference is considered on the return link, the receive interference power density at the satellite level over thermal noise ratio is computed from the adjacent system terminal EIRP spectral density level and the useful satellite G/T in the direction of the adjacent system terminal. The I_o/N_o ratio gives a degradation of the E_s/N_0 which is used for the link budget.

B.3.2 Capacity comparison results

B.3.2.1 ETSI Context (European scenario)

The overall spectral efficiency for antenna size from 0.5 to 0.8 m in the ETSI context is shown in Figure 57.

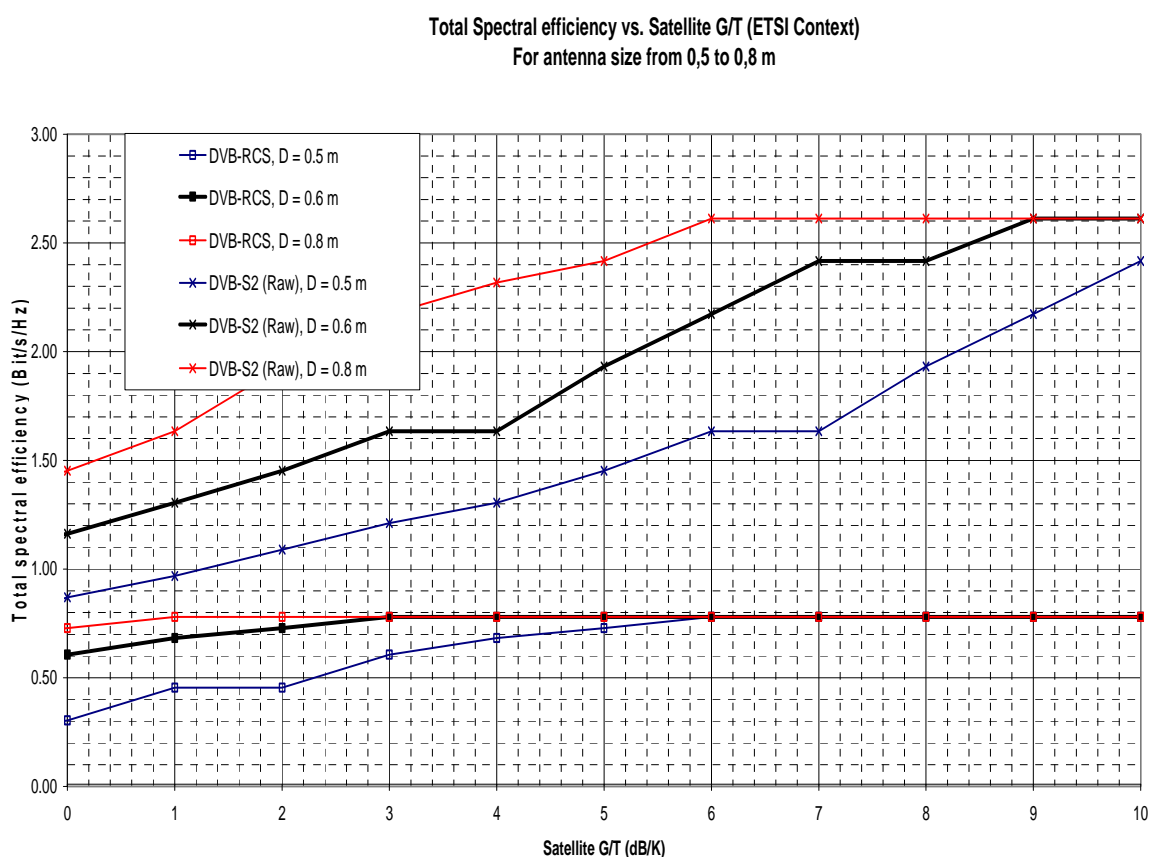


Figure 57: Spectral efficiency versus Satellite G/T (for MF-TDMA=DVB-RCS and Continuous carrier mode=DVB-S2) in the ETSI context

The DVB-S2 based continuous carrier mode shows the best potential capacity with a gain (up to 2.5 Bits/s/Hz) compared to the classical MF-TDMA DVB-RCS system. But the real capacity depends mainly on the traffic profile hypothesis. The load of the continuous carrier mode has to be higher than 40 % in order to be more efficient than the MF-TDMA.

The corresponding needed bandwidth for 100 mobiles is given in Table 23.

Table 23: Required Bandwidth (ETSI context)

	Traffic case	Mean data rate per mobile (kb/s)	Peak data rate per mobile (kb/s)	Needed bandwidth for 100 mobiles (MHz)	
				DVB-RCS	DVB-S2 SCPC
Aeronautical case	1	280	840	61.6	72.3
	2	420	840	92.3	72.3
	3	700	840	153.9	72.3
	4	140	420	30.8	36.1
	5	210	420	46.2	36.1
	6	350	420	76.9	36.1
Railway case	1	280	840	35.9	36.2
	2	420	840	53.9	36.2
	3	700	840	89.8	36.2
	4	140	420	18.0	18.1
	5	210	420	26.9	18.1
	6	350	420	44.9	18.1
Maritime case	1	280	840	38.5	57.9
	2	420	840	57.7	57.9
	3	700	840	96.2	57.9
	4	140	420	19.2	28.9
	5	210	420	28.9	28.9
	6	350	420	48.1	28.9

The use of continuous carrier in the return link can reduce significantly the needed bandwidth when compared with the classical DVB-RCS system. The reduction depends on the traffic profile. For high traffic load cases, the continuous carrier mode reduces the needed bandwidth by a factor 2. But in the case of low traffic loads, this mode can be less efficient than the MF-TDMA DVB-RCS system.

The aeronautical scenario is the most demanding scenario in terms of bandwidth. This is consistent with the fact that the antenna size is only 60 cm and the worst case satellite G/T is 0 dB/K. The train scenario is less bandwidth demanding. It needs only half of the bandwidth needed for the aeronautical scenario.

B.3.2.1 FCC Context (North American scenario)

The overall spectral efficiency for antenna size from 0.5 to 0.8 m in the ETSI context is in Figure 58.

Total Spectral efficiency vs. Satellite G/T (FCC Context)
For antenna size from 0,5 to 0,8 m

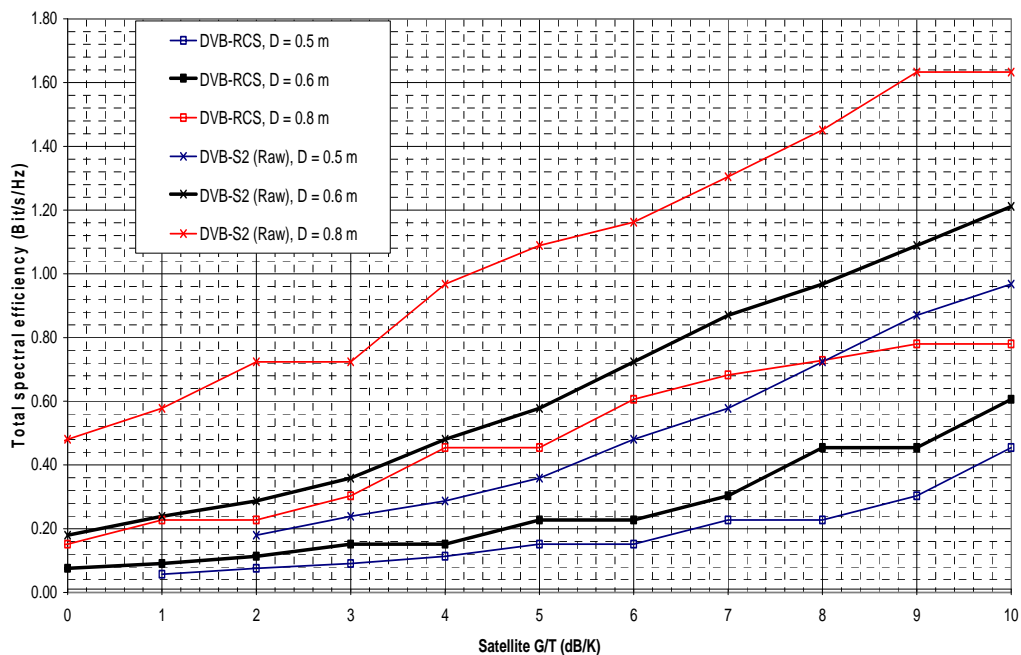


Figure 58: Spectral efficiency vs. satellite G/T (for MF-TDMA=DVB-RCS and Continuous carrier mode=DVB-S2) in the FCC context

The corresponding needed bandwidth for 100 mobiles is given in Table 24.

Table 24: Needed Bandwidth (FCC Case)

	Traffic case	Mean data rate per mobile (kb/s)	Peak data rate per mobile (kb/s)	Needed bandwidth for 100 mobiles (MHz)	
				DVB-RCS	DVB-S2 SCPC
Aeronautical case	1	280	840	369.3	468.1
	2	420	840	554.0	468.1
	3	700	840	923.3	468.1
	4	140	420	184.7	234.0
	5	210	420	277.0	234.0
	6	350	420	461.6	234.0
Railway case	1	280	840	92.3	86.8
	2	420	840	138.5	86.8
	3	700	840	230.8	86.8
	4	140	420	46.2	43.4
	5	210	420	69.2	43.4
	6	350	420	115.4	43.4
Maritime case	1	280	840	184.7	174.8
	2	420	840	277.0	174.8
	3	700	840	461.6	174.8
	4	140	420	92.3	87.4
	5	210	420	138.5	87.4
	6	350	420	230.8	87.4

The conclusions are similar to the ETSI case when we compare the different systems. The only difference is that the needed bandwidth is increased by a factor 3 up to 8 depending on the terminal. The corresponding satellite bandwidth cost will be also increased by the same factor.

B.4 Conclusions

The comparison between standard MF-TDMA and optional continuous carrier mode shows that the efficiency can be improved by a factor up to 2.5. But, the real efficiency will strongly depend on the traffic load. The carriers shall be dimensioned with the peak data rate of the mobile terminal but if the mean data rate (load) is lower than 40 % of the full load, the system is less efficient than the classical MF-TDMA one.

In conclusion, the continuous carrier mode is only well suited to systems with low burstness on the traffic profile.

History

Document history		
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