Technical Note

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Guidelines Document
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ANTARES

Communication Standard Implementation Guidelines Document

(DRL N°: D023)

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1. INTRODUCTION

1.1 Purpose

The Communication Standard Implementation Guidelines have been issued in the scope of WP2 of the ANTARES project and integrate the work performed by the ANTARES WP2 partners.

The main objective of the ANTARES project is the definition of a new Air Traffic Services (ATS) and Airline Operational Control (AOC) satellite Communications Standard.

1.2 Structure of the document

This document is structured as follows:

- Section 1 presents this introduction.
- Section 2 compiles the definition, symbols, acronyms and conventions used in the document.
- Section 3 reports the applicable and reference documents.
- Section 4 presents some scenarios of the communication system reference model and the aeronautical propagation channel description, and lists the system constraints derived from the definition of the Communication Standard.
- Section 5 presents the reference receiver and its performances.
- Section 6 describes guidelines for the network synchronization procedures.
- Section 7 presents guidelines for the ACM mechanism and procedure.
- Section 8 presents the guidelines for Radio Resource Management and ARQ.
- Section 9 reports the guidelines for network layer aspects.
- Section 10 presents the guidelines for control plane procedures.
- Section 11 contains Appendix A, which describes the aeronautical channel propagation channel.
- Section 12 contains Appendix B, which presents an estimation of L1-L2 overheads for the forward link.

2. DEFINITIONS, SYMBOLS, ABBREVIATIONS AND CONVENTIONS

2.1 Definitions

See definitions in [AD-02].

2.2 Symbols

Symbol	Definition
$Freq(t_0)$	Frequency offset estimation
$N_{nominal\_\#N}$	Nominal number of LDPC decoder iterations for MODCOD #N
$N_{reduced\_\#N}$	Reduced number of LDPC decoder iterations for MODCOD #N
P_d	Detection probability
P_{fa}	False alarm probability
P_{md}	Missed detection probability
P_{wd}	Wrong detection probability
$PER_{Nominal\_\#N}$	PER measurement at $N_{nominal}$ LDPC decoder iterations for MODCOD #N.
$PER_{Red\_\#N}$	PER measurement at $N_{reduced}$ LDPC decoder iterations for MODCOD #N.
$PER_{Thr\_Nominal}$	PER threshold at $N_{nominal}$ LDPC decoder iterations
$PER_{Thr\_Red\_\#N}$	PER threshold at $N_{reduced}$ LDPC decoder iterations for MODCOD #N.
t_0	Time offset estimation

Table 2-1: Symbols

Note: Counters (e.g., Packet Count and Fragment Count) are initialized to zero unless explicitly stipulated otherwise.

2.3 Acronyms list

Acronym	Definition
ACH	Auxiliary Channel
A-CDMA	Asynchronous Code Division Multiple Access
ACK	Acknowledgement
ACM	Adaptive Coding and Modulation
AD	Applicable Document
AF	Assured Forwarding/Address Format
AFDX	Avionics Full-Duplex Switched Ethernet
AGR	Air/Ground Router
ALSAP	Adaptation Layer Service Access Point

Acronym	Definition
AMS(R)S	Aeronautical Mobile Satellite (en Route) Service
AOC	Airline Operational Control
APSK	Amplitude and Phase-Shift Keying
AR	Airborne Router
ARQ	Automatic Repeat reQuest
ATM	Air Traffic Management, Asynchronous Transfer Mode
ATN	Aeronautical Telecommunication Network
ATN/IPS	ATN/Internet Protocol Suite
ATN/OSI	ATN/Open Systems Interconnection
ATS	Air Traffic Services
AVLC	Aviation VHF Link Control
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CC	Congestion Control
CCM	Constant Coding and Modulation
CDMA	Code Division Multiple Access
CFAR	Constant False Alarm Rate
CMU	Communications Management Unit
COCR	Communications Operation Concept and Requirements
CoS	Class of Service
CRC	Cyclic Redundancy Checksum
CS	Communication Standard
CW	Codeword
DA	Data Aided
DCH	Data Channel
DD	Decision Directed
D-DLL	Digital Delay Locked Loop
DFT	Discrete Fourier Transform
DL	Downlink
DS-SS	Direct Sequence Spread Spectrum
DW	Dataword
EATMN	European Air Traffic Management Network
EATMS	European Air Traffic Management System
ECAC	European Civil Aviation Conference
EDF	Earliest Deadline First
E-SSA	Enhanced Spread Spectrum ALOHA
ET	Expiration Time
FCH	Forward Channel
FEC	Forward Error Correction

Acronym	Definition
FF-DA	Feed-Forward Data-Aided (FF-DA)
FL	Forward Link
FLC	Forward Link Carrier
FMT	Fading Mitigation Techniques
FWD	Forward
G/G-R	Ground-Ground Router
GEO	Geostationary Orbit
GES	Ground Earth Station
GGR	Ground-Ground Router
GR	Ground Reflection
GS	Ground Segment
GSE	Ground Segment Elements
GSE	Generic Stream Encapsulation
HEO	Highly Elliptical Orbit
HO	Handover
IC	Interference Canceller
ID	Identifier
IDRP	Inter-Domain Routing Protocol
IP	Internet Protocol
IPS	Internet Protocol Suite
ISH	Intermediate System Hello
ISI	Inter Symbol Interference
ISO	International Organization Standardization
LCN	Local Communications Network
LDPC	Low Density Parity Check
LME	Link Management Entity
LOS	Line of Sight
LPDU	Link Protocol Data Unit
LS	Local Scatters
LSAP	Link Service Access Point
LSDU	Link Service Data Unit
MA	Multiple Access
MAC	Medium Access Control
MEO	Medium Earth Orbit
MF-TDMA	Multi Frequency Time Division Multiple Access
ML	Maximum Likelihood
MMSE	Minimum Mean Square Error
MODCOD	MODulation and CODing
MSE	Mean Square Error

Acronym	Definition
NCC	Network Control Centre
NCR	Network Clock Reference
NLMS	Normalized Least Mean Square
NMC	Network Management Centre
OSI	Open Systems Interconnection
OVSF	Orthogonal Variable Spreading Factor
PLR	Packet Loss Rate
PER	Packet Error Rate
PNPDU	Processed Network Data Unit
PPDU	Physical Protocol Data Unit
PSAP	Physical Service Access Point
PSDU	Physical Service Data Unit
PSK	Phase-Shift Keying
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RACH	Random Access Channel
RD	Reference Document
RL	Return Link
RLC	Return Link Carrier
RMSE	Root Mean Square Error
ROHC	RObust Header Compression
RMS	Root Mean Square
RRM	Radio Resource Management
RTN	Return
RTP	Real-Time Transport Protocol
RTT	Round-Trip Time
SIC	Successive Interference Canceller
SD	Soft Decision
SDL	Specification and Description Language
SDU	Service Data Unit
SF	Spreading Factor
SIC	Successive Interference Cancellation
SME	System Management Entity
SNDCF	Sub-Network Dependent Convergence Functions
SNIR	Signal to Noise plus Interference Ratio
SNR	Signal to Noise Ratio
SP	Strict Priority
SRD	System Requirements Document
SSA	Spread Spectrum Aloha

Acronym	Definition
STE	Satellite Translation Error
TBC	To Be Confirmed
TBD	To Be Defined
TBW	To Be Written
TC	Threshold Crossing
TCP	Transmission Control Protocol
TD95	Transit Delay 95th percentile
TDMA	Time Division Multiple Access
TNO	Technical Note
TX	Transmission
UL	Uplink
UT	User Terminal
VC	Virtual Circuit
VDL	VHF Data Link
VDR	VHF Digital Radio
VME	VHF Management Entity
XID	Exchange Identification

Table 2-2: Acronyms

3. APPLICABLE AND REFERENCE DOCUMENTS

3.1 Applicable documents

ID	Document Number	Title	Issue	Date
[AD-01]	Iris-B-OS-RSD-0002-ESA	Iris Phase 2.1 System Requirements Document	2.1	23/10/2012
[AD-02]	ANTAR-B1-CP-TNO-2006-IND	D018B Communication Standard Technical Specifications	1.1	16/09/2013

Table 3-1: Applicable documents

3.2 Reference documents

ID	Document Number	Title	Issue	Date
[RD-01]	COCRv2	Communications Operating Concept and Requirement for the Future Radio System	2.0	
[RD-02]		John G. Proakis, M. Salehi, “Digital Communications (Fifth Edition)”, McGraw Hill		2008
[RD-03]		S. Haykin, “Adaptive Filter Theory (Third Edition)”, Prentice Hill.		1996
[RD-04]		U. Mengali, A.N. D’Andrea, “Synchronization Techniques for Digital Receivers”, Kluwer Academics / Plenum Publishers.		1997
[RD-05]		A. Das, B.D. Rao, “SNR and Noise Variance Estimation for MIMO Systems”, IEEE Transactions on Signal Processing.		Aug. 2012
[RD-06]		O. del Río Herrero, R. de Gaudenzi, “A High Efficiency Scheme for Large-Scale Satellite Mobile Messaging Systems”, ICSSC 2009.		June 2009
[RD-07]		O. del Río Herrero, R. de Gaudenzi, “High Efficiency Satellite Multiple Access Scheme for Machine-to-Machine Communications”, Aerospace and Electronic Systems, IEEE Transactions, vol. 48, issue 4, pp. 2961-2989.		October 2012
[RD-08]		P. Patel, J. Holtzman, “Analysis of a DS/CDMA Successive Interference Cancellation Scheme Using Correlations”, IEEE Global Telecommunications Conference 1993, Globecom ’93.		Dec. 1993
[RD-09]		P.Patel, J. Holtzman, “Analysis of a Simple Successive Interference Cancellation Scheme Using Correlations”, IEEE Journal on Selected Areas in Communications, Volume 12, Issue 5, pp. 796-807.		June 1994

ID	Document Number	Title	Issue	Date
[RD-10]		Youngkwon Cho, Jae Hong Lee, "Analysis of an Adaptive SIC for Near-Far Resistant DS-CDMA", IEEE Transactions on Communications, vol. 46, no. 11.		Nov. 1998
[RD-11]		Hoa Tran, "A Method in Computing Successive Interference Canceller", Journal of Wireless Networking Communications.		2012
[RD-12]	ITU-R P.531-11	Ionospheric propagation data and prediction methods required for the design of satellite services and systems.		01/2012

Table 3-2: Reference documents

4. COMMUNICATION SYSTEM REFERENCEMODEL

4.1 Reference model

The general system reference model for the CS is documented in [AD-02]. That figure, repeated here for convenience, illustrates this general reference model.

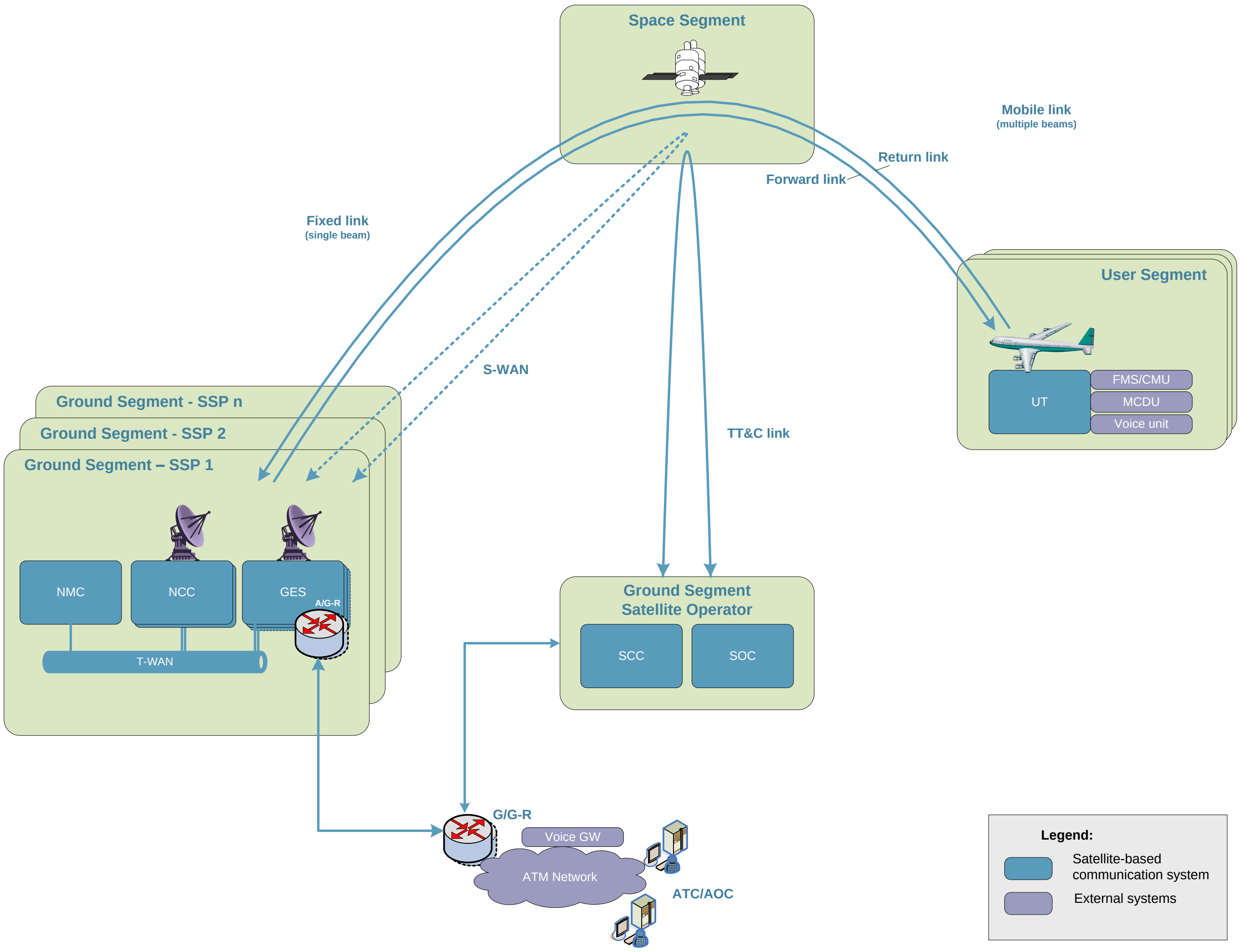


Figure 4-1: System reference model

Based on the general system reference model, the specific case of an HEO constellation considers that the relatively low traffic to be supported by this constellation, which provides service to the polar region, can be supported by a single GES. Therefore, for the HEO constellation, a single site is assumed to include the NMC, NCC and GES (the two latter elements assumed to be physically integrated as a Combined NCC-GES).

4.2 Reference orbital characteristics

In the case of the ECAC area, GEO satellites are located at a longitude that provides coverage using five mobile spot beams and a global beam. The global beam is intended to provide the required services over the entire Earth surface area visible from the GEO satellite's orbit. This area, also referred to as global coverage, is assumed to be covered by a single beam with a beamwidth of 17.3º and centre at the sub-satellite point.

Non-GEO satellite constellation, specifically MEO and HEO, is intended to provide service coverage over the North Polar area, defined as the area above 68 degrees latitude.

The considered HEO constellation is of Molniya type, with two satellites describing the following orbits:

Molniya Constellation Parameters	Molniya 1	Molniya 2
Semi-major axis	26553,1 Km	26553,1 Km
Eccentricity	0.74	0.74
Inclination	63.4 deg	63.4 deg
Argument of Perigee	270 deg	270 deg
Longitude of ascending node	348 deg	78 deg
True Anomaly	180 deg	0 deg
Period	12 h	12 h

Table 4-1: HEO constellation parameters

For the MEO case, the constellation has the following orbital characteristics:

Orbit Parameters	Value
Type of constellation	Walker (27/3/1)
Number of satellites	27
Orbital planes	3
Inclination	56°
Altitude	23222 km
Eccentricity	0
Period	14 h 5 min

Table 4-2: MEO constellation parameters

In both cases, only a single active GS, located in Tromsoe, Norway, is considered.

The feeder band frequency band for the HEO constellation is Ka, while C band is considered for the MEO case.

A single spot beam both in feeder and mobile links is considered for all cases. The mobile beam spot considered for the HEO case has a beamwidth of 17º (half-cone angle of 8.5º) while the mobile beam spot considered for the MEO case has a beamwidth of 20º (half-cone angle of 10º).

4.3 Frequency plan

An example of frequency plan in the case of a GEO satellite for the Forward Link is shown in Figure 4-3. Its main characteristics are:

- Mobile link with 5 beams.
- Frequency reuse of 3, meaning that the system uses three different frequencies assigned as follows:

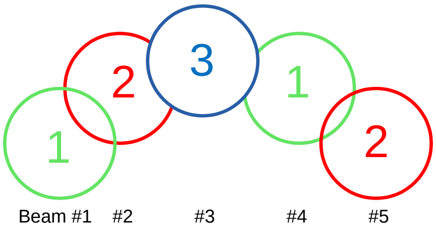


Figure 4-2: Frequency reuse for a 5 beam case

- The case depicted allocates a total of 18 carriers of 200 KHz distributed per beam as follows: 1 for beam 1, 2 for beam 2, 9 for beam 3, 4 for beam 4 and 2 for beam 5.

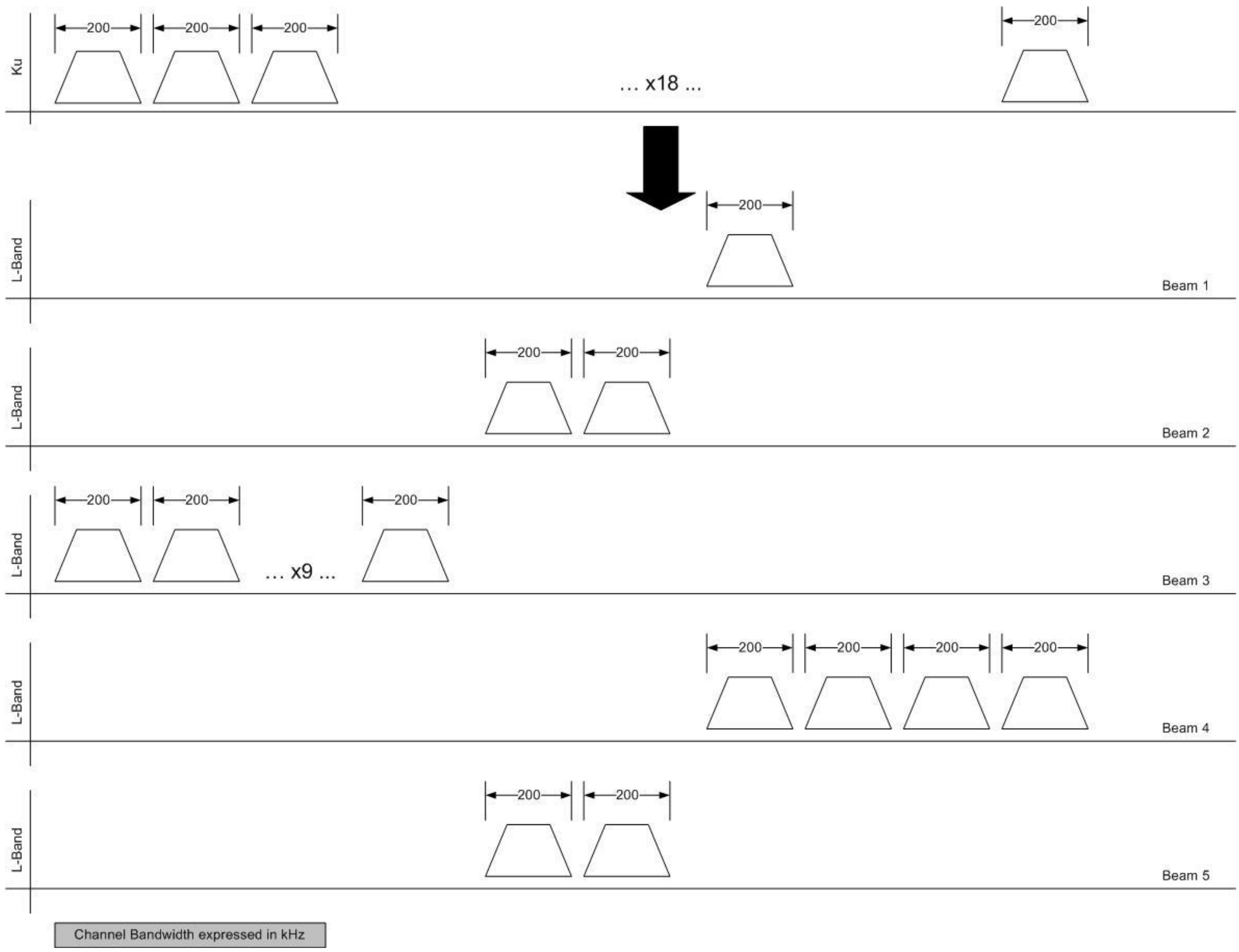


Figure 4-3: Forward link frequency plan

The frequency plan for the return link and medium capacity traffic is shown in the following figure. It shows the case of a total of 9 frequency channels in a full frequency reuse scheme allowed by the A-CDMA access scheme.

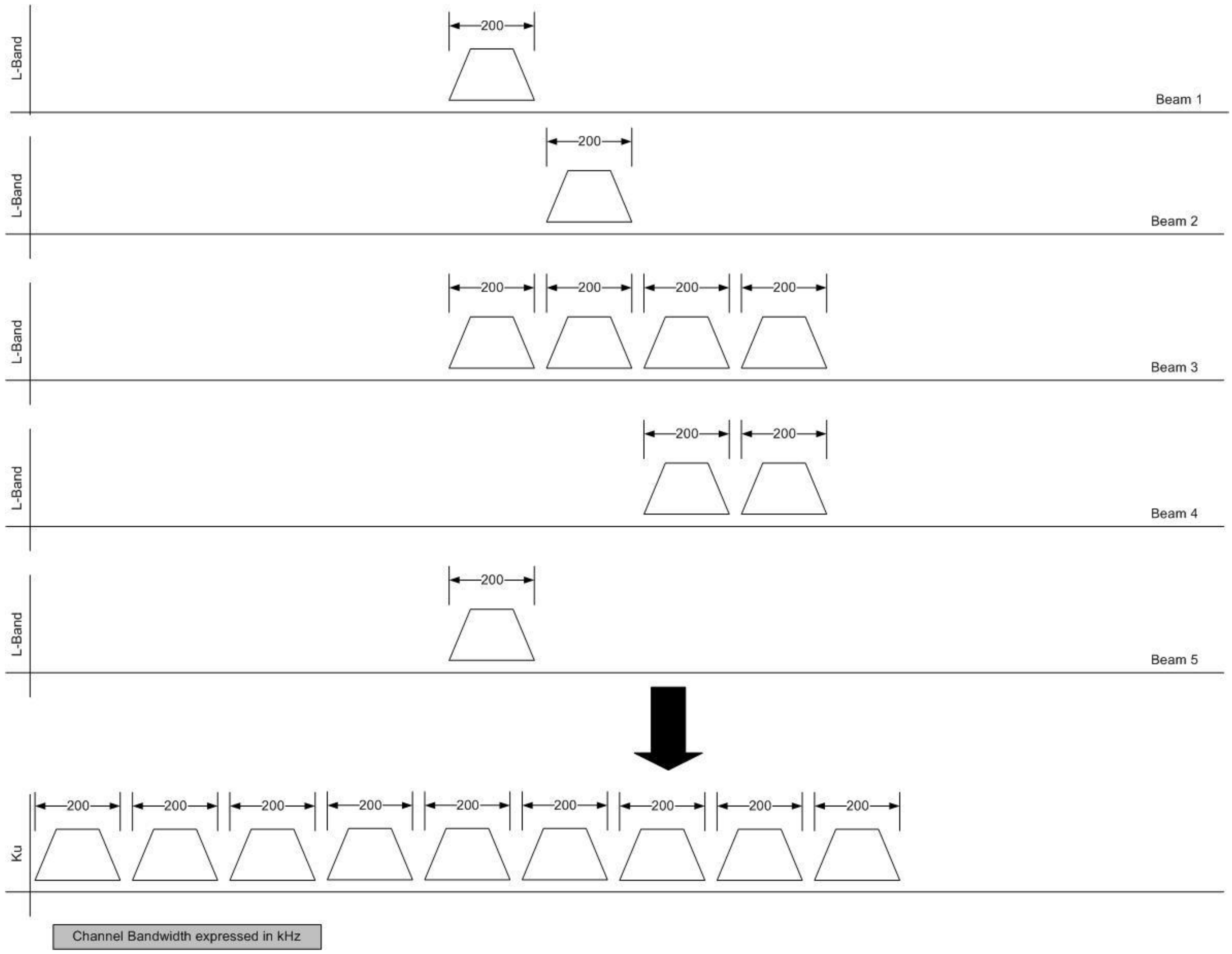


Figure 4-4: Return link frequency plan

4.4 Reference traffic profile

4.4.1 Introduction

A reference traffic profile, which is representative of the specific ATC/AOC communication profile, has been defined in order to drive the design of certain aspects of the CS. This reference is presented in more detail in the following sections, but its main characteristics are summarized hereafter:

- High number of active endpoints.
- Low data rate per endpoint (less than 256 kbps peak rate).
- Sporadic, transaction-type data exchanges.
- Transmission of a high percentage of rather short packets.

It should be noted that, although based on a specific European scenario, it is also considered representative enough to account for other cases, although a specific traffic profile is considered for the Atlantic Ocean coverage area low rate forward link, which is used only if no other carrier is present, and only for ADS-C service (voice not supported).

This profile has been used to support the following design activities:

- Information on aircraft routes and flight dynamics (yaw, pitch and roll angles) have provided inputs for aircraft visibility and ACM analysis.
- The traffic model has been a key input for the definition of the most adequate multiple access method. In particular, the efficiency of contention-based access methods is closely linked to the number of potential users contending for the same set of resources, their data rate and traffic burstiness (which impacts on collision probability), and the statistical distribution of data unit sizes to be transmitted.
- Traffic modelling has also provided inputs for the design of the control plane of the CS(for example, in terms of reference regarding expected logon and handover signalling overheads compared to the overall traffic).
- The selection of the most appropriate encapsulation scheme has taken into account traffic pattern and message size distribution characteristics in order to optimize overheads.

4.4.2 Traffic model

In order to derive a reference traffic profile for CS design activities, today's air traffic has been extrapolated to the target deployment time of the system on the basis of the Eurocontrol air traffic long-term forecasts at year 2010 (currently provided until year 2030) and Central Flow Management Unit (CFMU) data. The extrapolation took into account the effects of future developments in ATM (Single European Sky, great circle routes) and included scheduled and unscheduled flights. The data traffic model was developed on the basis of the air traffic models and the COCRv2 report [RD-01]. It includes ATS traffic in ORP, TMA and ENR domains and 30% of aircraft supporting AOC traffic for growth scenario A, which corresponds to an upper limit for the system dimensioning.

The ANTARES data traffic characterization was derived from the simulation results for representative coverage areas (areas of interest and/or beams). The traffic pattern was characterized by the overall PIAC (Peak Instantaneous Aircraft Count), geographical distribution of aircraft density, and the statistical characterization of the air-ground data traffic pattern.

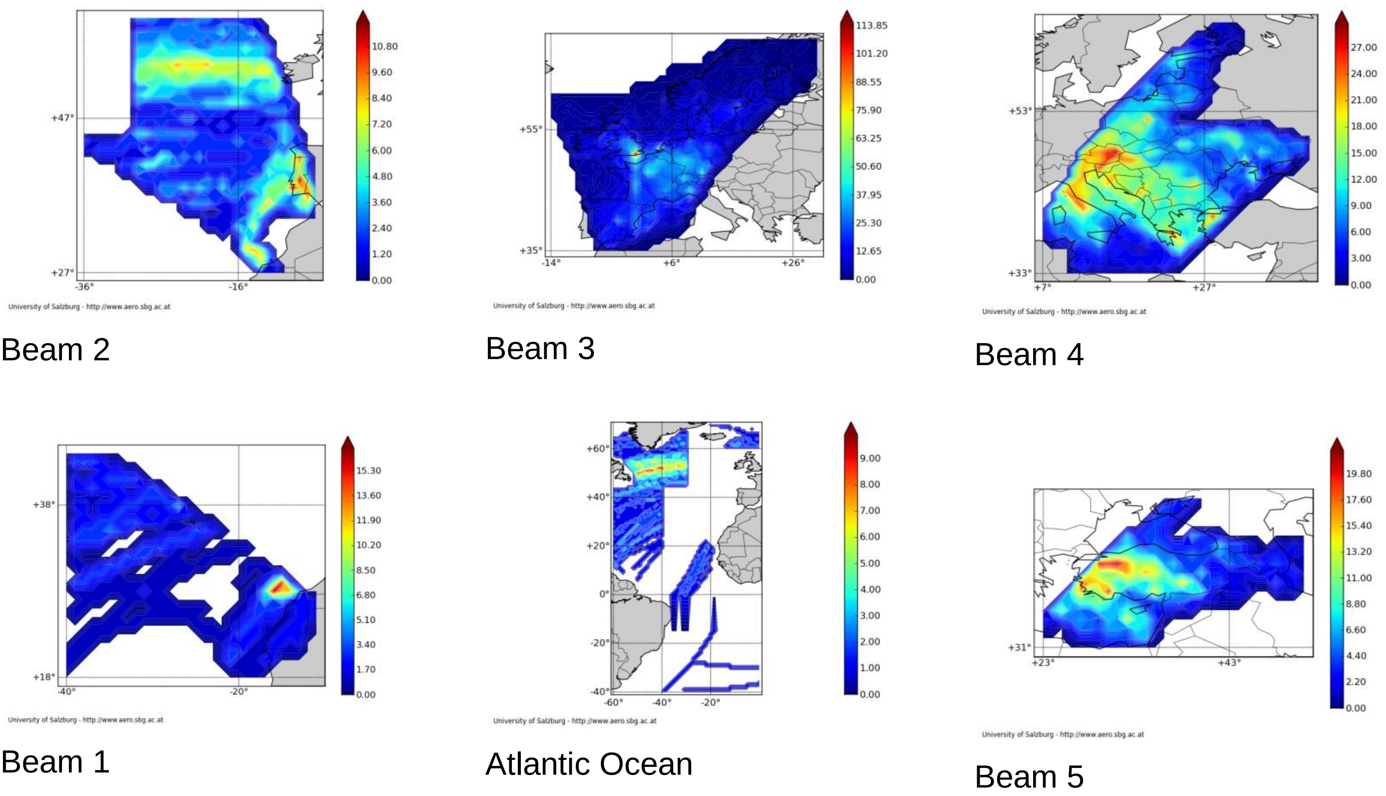


Figure 4-5: Considered coverage areas and aircraft density in ORP+ENR+TMA domain for year 2030, growth scenario A

4.4.3 Traffic pattern characterization

The ANTARES communications system has to support both data and voice Air/Ground communications for ATS and AOC services in the ECAC area, and just ADS-C data in the Atlantic Ocean area. Currently predominant voice communications will be progressively displaced by data-only communications and its use, while still required in ORP airspace, will be limited to contingency situations.

Voice services consist of short interactions lasting 14 seconds on average, whereas considered data transmissions consist of sporadic, transaction-type data exchanges, with varied message length and frequency. A voice call may consist of 3 to 5 exchanges of 2 transactions (one up and one down). Due to lack of specific information on voice calls duration, these assumptions are educated guesses agreed upon to perform the analysis.

In the case of emergency voice communications, the transactions will be very short, but there will be many more of them. In fact, an emergency can be as long as getting an aircraft to land. For an aircraft in the middle of the ocean this can last for an hour. This is very extreme and likely the adequacy of the communication system and protocols can be initially verified assuming a lower total duration lasting 14 seconds on average.

In the following, some meaningful statistics for CS design activities are presented for coverage areas shown in Figure 4-5, including:

– Number of active aircraft

- Handover rate
- Communication session duration
- Average data and message rate
- Overall bandwidth to be supported per beam coverage area

In any case, it is important to note that [RD-01] represents an effort to anticipate future traffic needs, but that information has to be taken with certain care as future services are not consolidated. In this sense, the CS is able to provide the sufficient flexibility to adapt to future changes.

According to [RD-01], forward link and return link data traffic composition differs greatly regarding latency requirements. Forward link traffic is formed by delay-sensitive traffic with demanding requirements (more than 65-80%) and less than 20-35% of more undemanding traffic (TD95<= 2.4 s). In the return link, traffic is sometimes composed, to a lesser degree, by delay sensitive traffic and more undemanding traffic represents around 14-50%.

Below is shown the composition of traffic in the return and forward links for the considered traffic profile in the different beams, regarding the sensitivity of traffic to delay (TD95 <= 2.4 s):

Scenario	% of packets with TD95 > 2.4 s	
	FL	RL
Beam 1	30.24 %	45.93 %
Beam 2	35.38 %	50.49 %
Beam 3	19.72 %	36.89 %
Beam 4	20.36 %	40.26 %
Beam 5	23.14 %	13.6 %
Atlantic Ocean	0 %	0 %

Table 4-3: Traffic composition regarding TD95

4.4.3.1 Number of active aircraft

Peak instantaneous number of active aircraft (PIAC) is highly variable depending on the considered coverage area. As shown in Table 4-4, beam 3 and beam 4 cover high-density areas and are therefore required to support significantly higher numbers of aircraft (in the order of thousands) than beam 1 covering parts of the Atlantic (in the order of one hundred).

Scenario	ORP PIAC	ENR/TMA PIAC	Avg. HO /sec[1]		95th percentile HO /sec[2]	
			in	out	in	out
Beam 1	53	70	0.021	0.018	0	0

[1] "Avg. HO/sec in" and "Avg. HO/sec out" values are for ORP/TMA/ENR domains.

[2] "Avg. HO/sec in" and "Avg. HO/sec out" values are for ORP/TMA/ENR domains.

Beam 2	280	171	0.090	0.086	1	1
Beam 3	28	3635	0.281	0.209	1	1
Beam 4	9	2081	0.295	0.269	1	1
Beam 5	16	480	0.104	0.094	1	1
Atlantic Ocean	452	0	0.059	0.049	1	0

Table 4-4: Air traffic volume (PIAC) and handover rate.

4.4.3.2 Handover rates

Statistics on handover rate consider the rate that aircraft enter or leave a certain beam and thus indicate the rate at which the system should perform a handover of the communication link.

As presented in Table 4-4, in terms of the average handover rate the high density beams (0.281-handovers/second) can also be clearly distinguished from the ORP beam (less than 0.1 handovers/second). Note, however, that the 95th percentile of the handover rate is always below 1.0 handover/second in all scenarios.

As an additional example, the figures on handover rate for the MEO case over ECAC SHOULD area (being almost the same as the North Polar area) are presented.

For this area, the following numbers of active aircraft with the following density are envisaged for the year 2030.

	min	mean	95thperc	99thperc	max	sdev
APT	0.00	0.00	0.00	0.00	0.00	0.00
ENR	1.00	24.22	33.00	36.00	36.00	6.18
ORP	0.26	32.14	39.00	41.00	42.00	4.81
TMA	0.00	10.20	14.00	16.00	17.00	2.70
all domains	1.26	66.56	73.00	75.00	77.00	6.19

Table 4-5: PIAC in ECAC SHOULD area

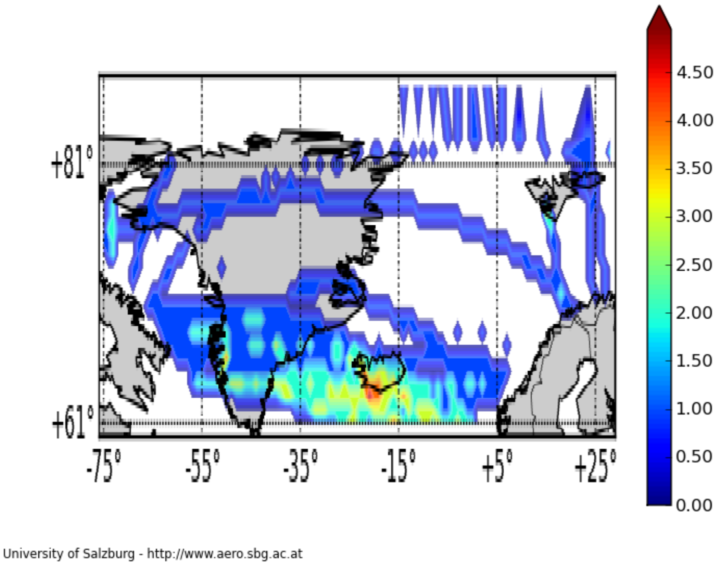


Figure 4-6: Aircraft density in ECAC SHOULD area

To obtain the HO rate, the highest impact in the overall handover rate will be derived considering a single TMA domain in the whole area.

Thus, a maximum of 17 active aircraft is considered on a single TMA domain (a 5 to 50 NM radius area surrounding the airport). If we consider aircraft being distributed uniformly in this area, and take into account the MEO coverage relative speed to the ground, the expected handover rate will be around 0.1 handover/second.

The contribution of ENR and ORP domains will be lower than the TMA domain.

4.4.3.3 Communication session duration

The data traffic pattern of all simulated aircraft was analyzed for communication session duration and communication session inter-arrival time. A session is defined as the time that at least one service is active in the aircraft (i.e., at least one message exchange takes place). That is, the inter-session times are the intervals where no data communication takes place (i.e., the aircraft has no open ATS or AOC dialogues).

Within the considered traffic profile (corresponding to all areas and ORP+ENR+TMA domains) the average inter-session times are typically in the order of 200 seconds. The 95th percentile of the inter-session time is below 1000 seconds in all cases for the ECAC area five beams, but it is 4755 seconds for the Atlantic Ocean area.

4.4.3.4 Average data and message rate

Table 4-6 displays the average number of unicast messages per second and bits per second generated by an aircraft during a session on each beam (multicast traffic is negligible in the provided traffic profile).

The ground-station transmits one message to each aircraft approximately every 20 seconds at an average data rate of 222 bit/second (forward link, FL). The aircraft replies with one message approximately every 16 seconds at an average data rate of 81 bit/second (reverse link, RL).

	ENR/TMA/ORP					
	FL			RL		
	Avg. msg/sec	Avg. bit/sec	99% bit/sec	Avg. msg/sec	Avg. bit/sec	99% bit/sec
Beam 1	0.0341	145.79	744.00	0.0424	55.72	856.00
Beam 2	0.0349	156.83	256.00	0.0453	46.01	824.00
Beam 3	0.0568	260.85	1544.00	0.0691	109.34	1184.00
Beam 4	0.0465	235.3	1040.00	0.0592	86.57	896.00
Beam 5	0.0466	226.1	1008.00	0.0597	76.19	856.00
Atlantic Ocean	0.0011	0.14	0.00	0.0011	8.53	0.00

Table 4-6: Unicast Data traffic per aircraft; ENR+TMA+ORP.

4.4.3.5 Overall throughput requirements

In addition to the analysis of the traffic pattern properties of single aircraft, the air traffic and data traffic simulations were also used to derive the overall bandwidth requirements for the indicated beam configurations. Table 4-7 displays the average data traffic load and the 95th percentile of the data traffic load for each beam. Beams covering the European high density areas can be clearly distinguished from the beams covering remote or oceanic areas.

	ENR/TMA/ORP			
	FL		RL	
	Avg. kbit/sec	95% kbit/sec	Avg. Kbit/sec	95% kbit/sec
Beam 1	11.461	174.568	4.380	20.282
Beam 2	54.210	196.904	15.902	39.177
Beam 3	877.479	1400.253	367.825	472.081
Beam 4	452.889	860.017	166.628	237.729
Beam 5	95.707	350.291	32.246	64.664
Atlantic Ocean	0.0492	0.256	3.074	16.000

Table 4-7: Aggregate unicast traffic load per beam coverage area in ENR/TMA/ORP

Spot beams covering high density areas, beam 3 and beam 4, require up to 1.5 megabits/second in the forward direction. Beam 1 covering parts of the Atlantic need not support more than 175 kbit/second of data traffic on the FL. On the RL a capacity of less than 0.5 megabit/second is generally sufficient. For the Atlantic Ocean coverage a capacity of less than 1 kbit/s is enough for the FL and of 16 kbit/s is enough for the RL. This is because just ADS-C service is considered in the Atlantic Ocean coverage, as already mentioned in sections 4.4.1 and 4.4.3.

4.4.4 Assumptions

It should be noted that the traffic profile used to derive the statistics described above assumes 77 bytes headers, which correspond to ATN/OSI headers. During the CS design (in particular for multiple access trade-off simulations), it has been assumed the use of IPS (no OSI). It is also assumed that TCP/IPv6 is used without header compression, except for VoIP (resulting in a 4 bytes compressed header). In order to do this, the 77 bytes overhead was subtracted to packet sizes specified in the traffic profile. Then, the TCP/IPv6 overhead was added, considering also segmentation introduced by the use of an MTU of 1280 bytes. The traffic profile included TL ACKs which size was set to model TCP/IPv6 ACKs of 16 bytes. Not included in the traffic profile, so in the reported statistics in the previous sections, ARQ protocol ACKs have been considered as designed in the CS during CS design activities.

4.5 Aeronautical propagation channel

The satellite communication link is divided in two parts,

- the Mobile Link, which is the communication link between the satellite and the aircraft (uplink and downlink) and
- the Fixed link, which is the communication link between the satellite and the Ground Segment elements (uplink and downlink).

The Mobile link operates at frequencies identified by ITU for Aeronautical Mobile Satellite (Route) Service (AMS(R)S), in agreement to Article 1, Section III, 1.33 of ITU Radio Regulations and allocated worldwide:

- 1545 to 1555 MHz for the mobile downlink (from satellite to User Terminal),
- 1646.5 to 1656.5 MHz for the mobile uplink (from User Terminal to satellite).

On the other hand, there is no regulation regarding which frequency has to be used for the fixed link. Typical frequency bands are Ku and Ka.

Since mobile and fixed links operate at different frequency bands, the propagation characteristics are the same. In addition, the mobile link is impacted by the multipath propagation.

In the following two sections, the propagation characteristics of the mobile and fixed link are reported presented.

4.5.1 Mobile link propagation channel

The mobile link is mainly characterised by the following propagation effects:

- Multipath; due to the fact that the aircraft is moving, it uses omni-directional antennas.
- Scintillation, which is important in the equatorial and polar regions.
- In addition, when considering rotary-wing aircraft, the signal is obstructed due to the rotor blades.

4.5.1.1 Multipath

The aeronautical multipath propagation channel is characterised by the combination of following components:

- A strong Line of Sight (LoS) component, present most of the time.
- Local scatters from the aircraft's fuselage.
- Multiple delayed reflections from the ground.

These three components are shown in the following figure.

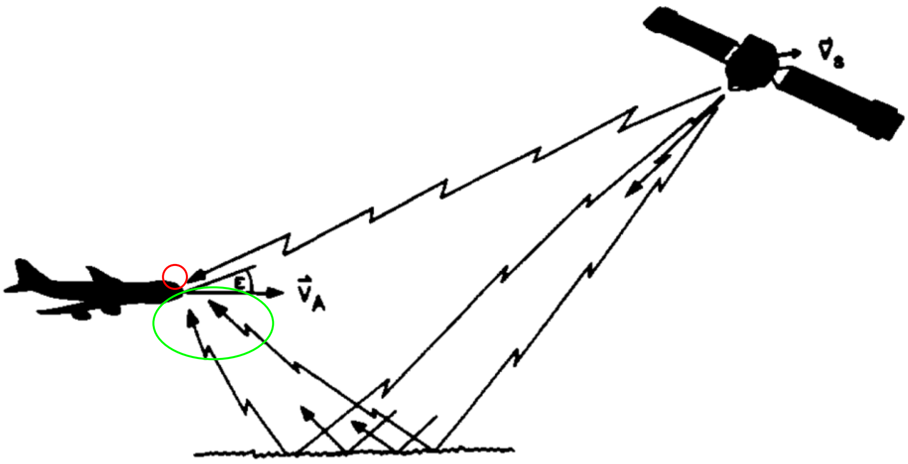


Figure 4-7: Illustration of the geometry of the aeronautical communication channel. Local scatters are illustrated with red, reflections from the ground with green.

4.5.1.1.1 Direct LOS

The LOS component suffers a Doppler shift due to the relative movement of the aircraft with respect to the satellite. This Doppler shift can be assumed to be

$$f_{DLOS} = \frac{v_{LOS}}{\lambda}$$

where λ denotes the wavelength and v_{LOS} is the speed of the aircraft in direction of the LOS component relative to the satellite.

4.5.1.1.2 Local scattering

The received signal might be deteriorated by local scatters on the hull of the aircraft. The characteristics of the local scatters are:

- The LOS signal amplitude is assumed to be Ricean due to local scatters from the hull of the aircraft.

- All echoes from the fuselage (or scatters) arrive within a very few nanoseconds with respect to the LOS component.
- All local scatters arrive with very similar incident angles.

The Rice factor for the model is 14 dB whereas the Doppler power spectrum density of the local scattering has a Gaussian shape with a standard deviation of 1 Hz.

4.5.1.1.3 Ground reflections

The reflections from the ground are modelled assuming:

- All the ground reflections arrive in a short time interval and hence are modelled by a single path.
- The amplitude of the multipath component is assumed to be Rayleigh distributed.

Three important parameters determine the ground reflections, which are:

- Direct signal to multipath power (C/M).
- Doppler power spectrum of the ground reflection.
- Delay of the specular component with respect to the LoS component.

4.5.1.1.3.1 Direct signal to Multipath (C/M)

The LOS signal power to reflected signal power from the ground ratio (C/M) follows the Fresnel reflection coefficient for specular reflection. The C/M values for reflection coefficient over sea water (worst case) are presented in the following figure. This figure shows that theoretical curves and measurement results match.

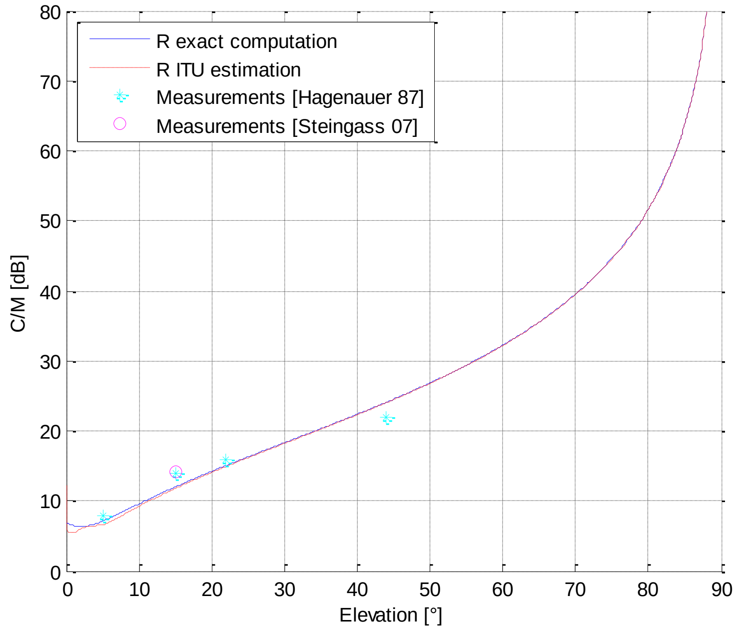


Figure 4-8: Theoretical curves for the reflection coefficient over the sea and measurement results

4.5.1.1.3.2 Doppler power spectrum

The Doppler power spectrum density of the ground reflections has a Gaussian shape with a standard deviation of 2.B_{rms}, where B_{rms} is the Doppler spread defined as

$$B_{rms} = \frac{4\alpha}{\lambda} \cdot \sqrt{(v_x \sin\varepsilon + v_z \cos\varepsilon)^2 + v_y^2 \sin^2\varepsilon}$$

Where α denotes the rms slope of the surface. The Doppler spectrum of the ground reflection depends on the elevation angle and on the aircraft speed. For an aircraft flying at 900 km/h, the Doppler spread is depicted in the following figure.

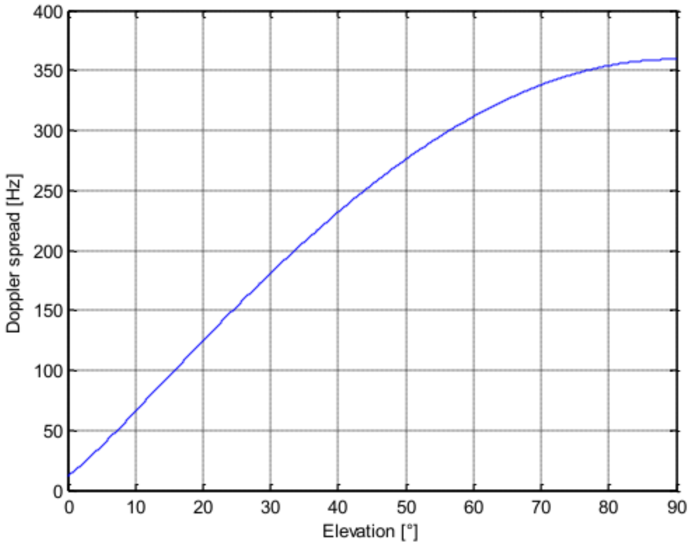


Figure 4-9: Doppler spread (as defined by Bello) vs. elevation for α = 0.07, vx = 900 km/h and λ = 19.5 cm

4.5.1.1.3.3 Delay of the specular component

The delay of the specular reflected component is determined based on geometrical considerations. It is given by the path difference of the reflected and the LOS path divided by the speed of light, c. This leads to the following expression:

$$\tau = \frac{2 \cdot H \cdot \sin\varepsilon}{c}$$

with H denoting the altitude of the aircraft above the ground.

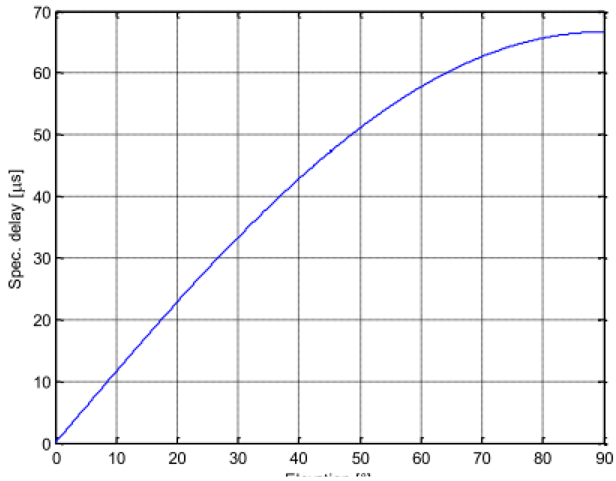


Figure 4-10: Delay of the specular reflected component for an altitude of 10 km

It should be noted that depending on the channel symbol rate and the delay of the specular component with respect to the LoS one, the propagation channel can turn into a frequency-selective channel.

4.5.1.2 Ionospheric Scintillation

Fast time variations in phase, amplitude, and angle of arrival of radio waves propagating through the ionosphere are known as ionospheric scintillations. They are created by random fluctuations of the medium's refractive index, due to heterogeneities inside the medium. Scintillations are strong in equatorial regions where they appear after sunset and may last a few hours. Particularly, they are related to the solar activity and the season. Also, very strong effects can be observed in the pole regions. At mid-latitudes, the scintillations are rather weak, except during conditions of ionospheric storms. In general, scintillation effects can be noticed for frequencies below 12 GHz, turning significant for frequencies below 3 GHz, thus also for L-band.

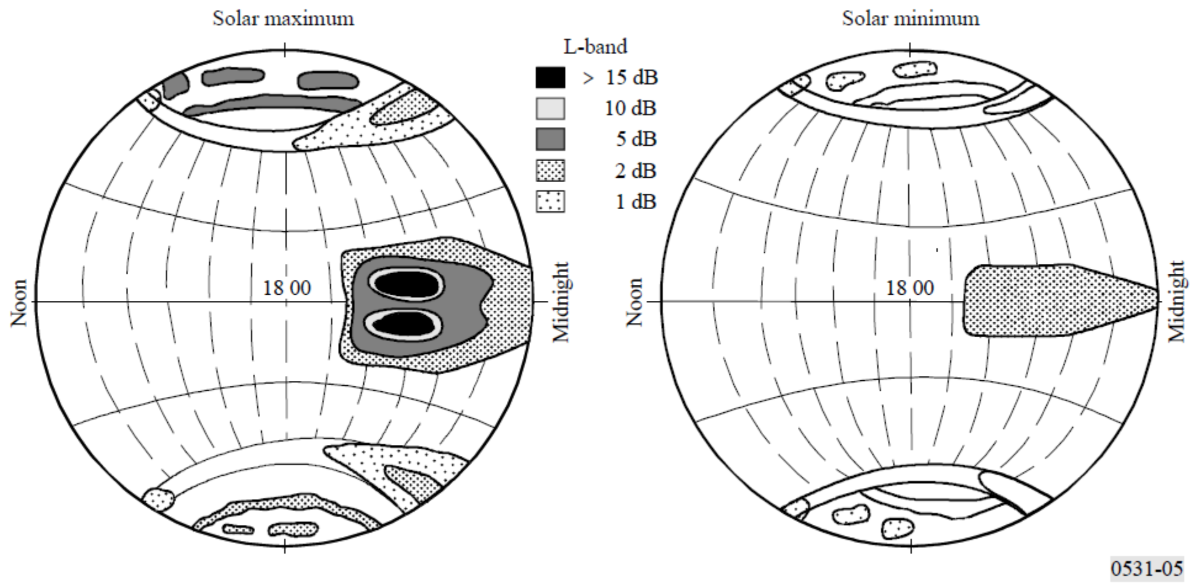


Figure 4-11: Depth of scintillation fading at L-band for solar maximum and minimum years from [RD-12].

The ITU recommendation in [RD-12]suggests using the global ionospheric scintillation model (GISM). It allows predicting the intensity fluctuations and the depth of amplitude fading, as well as the rms phase and angular deviations due to scintillation as a function of satellite and ground station locations, date, time and working frequency. An illustration of the fading depth due to scintillation from [RD-12] is given in Figure 4-11.

4.5.1.3 Rotorcraft

The effect of rotor blades is modelled by an on-off model: whenever the signal is obscured by the rotor blades, certain attenuation is applied on the signal. The attenuation due to the rotor blades is only applied to the LOS path (direct component and scatters), whereas ground reflections are assumed to be unaffected due to their different direction of arrival. The values used to model the effect of the rotor blades are the following ones:

– Attenuation (OFF state): 7 dB
– Fading periodicity: 54 ms
– Fading duration: 10.8 ms

Note that this the 20% of the time the channel state is OFF, i.e., the LoS and LS components are attenuated 7dB with respect to the GR component.

It is worth mentioning that the on/off model proposed is only an approximation. Due to diffraction at the edge of the blades the signal transition will be less sudden in reality. Further studies indicate that the position of the antenna, as well as the manoeuvring of the rotorcraft, play a big role on the signal characteristics.

4.5.2 Fixed link propagation channel

The channel impairments affecting the fixed link are different than the ones affecting the mobile link, mainly for two reasons:

- It uses higher frequency bands (Ku, Ka).
- There is no multipath propagation effect (not even in the case of considering non-GEO constellations as the GS elements use high directive antennas).

Then, the fixed link is characterised mainly by the following tropospheric propagation effects:

- Rain attenuation
- Atmospheric gases absorption

It is worth mentioning that apart from these two tropospheric propagation effects, there are other tropospheric propagation effects such as the clouds attenuation, the tropospheric scintillation, and the attenuation due to sand and dust storms, but they are less relevant in the frequency bands considered for the fixed link.

4.5.2.1 Rain attenuation

ITU regulations, in particular the ITU-R P.618-9, provide a general method to predict rain attenuation for long-term statistics. This regulation provides the maximum rain attenuation not exceeded for a given annual availability percentage, for frequencies up to 55 GHz. It also provides a similar worst month statistic, a diversity improvement factor and gain computation method, and some other general considerations about characteristics such as fading durations, rates and frequency correlation.

Rain attenuation depends on several parameters: frequency, site location (for rainfall statistics), relative position between the site and the satellite orbit position (elevation angle), polarization tilt angle, and availability considered.

4.5.2.2 Atmospheric gases absorption

The absorption caused by atmospheric gases is analyzed in the ITU-R P.676-7 regulation. The two major contributions to this phenomenon, in the frequency bands we are analyzing, are the water vapour attenuation and the oxygen attenuation.

Oxygen contribution is relatively constant, whereas water vapour density, and therefore its attenuation, varies both geographically and with time.

In the Ka-band (around 22 GHz) there is one peak in the in the atmospheric gases absorption. It is interesting to point out that the elevation has an important impact on the attenuation due to atmospheric gases.

4.6 System constraints derived from the Communication Standard

This section presents the list of constraints that the definition of the Communication Standard imposes to a system deployment. The constraints are grouped in the CS functions that originate them.

4.6.1 Handover

Handover detection

Since satellite and beam HO detection is proposed to be based on the ACM mechanism, the system constraints identified in section 4.6.3 are applicable as well if HO detection is implemented as proposed.

Guidelines for the handover detection can be found in section 10.1.1.

Bulk handover for non-GEO satellite

The GS is able to plan and detect, in the non-GEO satellite case, the overlapping period in which the current descending satellite coverage overlaps with the new ascending satellite coverage, based on SCC satellite information and GS location, to prepare in advance the synchronization to the new satellite.

4.6.2 Synchronisation aspects

The constraints presented in this section affect both forward and return link network synchronisation functionality.

Ground and User Segments

- The NCC reference clock long term instability is better than 0.01 ppm per year.
- The GES reference clock long term instability is better than 0.01 ppm per year.
- The UT reference clock long term instability is better than 1 ppm per year.
- UT and GS elements guarantees operation under significant UT movement:
  - UT speed up to Mach 2.5, i.e., up to 850 m/s at nominal atmospheric conditions at sea level.
  - UT acceleration up to 50 m/s^2.
  - UT angular velocity up to 3.33, 1.67 and 2 º/s (roll, pitch and yaw).
- The RRM function (scheduler) guarantees that all timeslots of all FLC carriers are filled with an FCH burst. If there is no FWD link traffic to be transmitted, dummy FCH bursts are inserted.

Space Segment

- The Satellite reference clock long term instability is better than 0.05 ppm over a period of 15 years.
- The ATM transceivers belonging to the same satellite use the same reference clock.
- If forward link network synchronisation procedures are implemented through feeder-to-feeder links, feeder-to-feeder and ATM transceivers belong to the same satellite.

- If forward link network synchronisation procedures are implemented through feeder-to-feeder links, the satellite feeder-to-feeder and ATM transceivers use the same reference clock.
- The maximum difference between the delays introduced by any satellite ATM transceivers is low enough not to impact the forward link time synchronisation:
  - Maximum delay difference lower than 1% of the symbol period in the forward link.
- If forward link network synchronisation procedures are implemented through feeder-to-feeder links, the maximum difference between the delays introduced by any satellite feeder-to-feeder and ATM transceivers is low enough in order not to impact the forward link time synchronisation:
  - Maximum delay difference lower than 1% of the symbol period in the forward link.
- If forward link network synchronisation procedures are implemented through the forward link carrier, GS elements are equipped with an L-band RF front-end (reception in L-band, apart from Ku, Ka or C-band, is required).

The previous constraints have been assumed in section 6.

4.6.3 Adaptive Coding and Modulation

As a consequence of the ACM functionality, the following constraints should be imposed on the system in order to guarantee proper ACM mechanism behaviour:

- The RRM function (FWD link scheduler) guarantees that all timeslots of all FLC carriers are filled with an FCH burst. If there is no FWD link traffic to be transmitted, dummy FCH bursts are inserted.
- The RRM function ensures that the next less robust MODCOD used by a UT in the FLC is transmitted in a regular way on the FLC (e.g., if the preferred MODCOD used by a UT is MODCOD #k, then the RRM should ensure that MODCOD #k+1 is transmitted in a regular fashion) in order to allow safe MODCOD upgrade. This means that dummy packets may need to be injected by the GS.
- In order to allow reliable PER measurements, the GS should ensure that at least, within the T_{obs} time, the number of packets required to perform a PER estimation are transmitted.
- The feeder link uplink power control should ensure that the power at satellite level is the same in all bursts no matter the GES transmitting them.

The previous constraints have been assumed in section 7.

4.6.4 Forward link waveform configurations

The FLC (Forward Link Carrier) supports two different symbol rates (note that each specified baud rate also has associated the corresponding waveform specification):

- Symbol rate of 160 kbaud
- Symbol rate of 16 kbaud (also called Low Rate Waveform)

These two FLC configurations are used as follows:

– FLC at 160 kbaud has been designed to support the ATS, AOC and voice service applications defined in [AD-01] and to fly across ENR, TMA and ORP aerospace domains. FLC at 160 kbaud can be used either in GEO, MEO and HEO constellations as defined in section 4.2.

– FLC at 16 kbaud has been designed to support ADS-C messages and to fly across ORP aerospace domains in constrained Link Budget. FLC at 16 kbaud is used only in GEO constellations as defined in section 4.2.

4.6.5 QoS

As explained in sections 8.1.1 for the FWD link and 8.2.4 for the RTN link, it is convenient that the L2 can determine the next QoS parameters associated with a given NPDU:

- Application type:
    - Voice
    - Data, then type of data application:
        - Low rate < 8 kbit/s
        - High rate >= 8kbit/s (only FLIPINT application in the profile shown in section 4.4)
- Application layer message size: in order to compute estimated transmission times for the scheduling process time margins computations, and also to map the application to a given burst according to known application messages sizes (see section 8.7.3 of [AD-02]).
- TD95: for LPDU scheduling prioritization.
- ET: to know whether to abort an ongoing NPDU transmission and for prioritization of LPDUs which have already exceeded the TD95 constraint of its NPDU.
- L2 ARQ use need: which depends on application messages length, continuity required and PER, but which usually is set as needed for all data transmissions and never used for voice and signalling.
- L2 CRC need: if integrity required is higher than 10^{-12}, assuming a PER of 10^{-3}.

The maximum integrity required by the applications considered in the profile described in section 4.4 is 5.10^{-10}, so L2 CRC would never be required. The L1 CRC provides integrity up to the mentioned level.

As explained in section 8.5.1.1 of [AD-02], the network adaptation layers (ATN/OSI and ATN/IPS) are in charge of determining the CoS of an incoming NPDU. The CoS information determined by network adaptation layer functions can be summarized on a single byte, which is mapped by the L2 to the mentioned QoS parameters. This mapping of CoS to QoS parameters would be configured by system network management procedures, according to the mechanism by which the QoS information is expected to be passed down from a higher layer on a given ATM network.

If the QoS parameters are not available, the L2 cannot group the incoming traffic into flows and give priority to traffic flows with the most stringent QoS requirements, resulting in a system that treats traffic without QoS parameters with a best effort approach (lowest priority). It is well known that a system that treats traffic with different QoS requirements, such as voice and data, but also among different data services with different QoS constraints such as low rate and high rate, would need more bandwidth in order to provide the same QoS than a more efficient system where the QoS information of traffic can be provided to L2.

4.6.6 GS Management Interface

In order to allow inter-system handover, at least the following parameters are available* for other inter-operable systems complying with the CS:

(*) The configuration of these parameters is usually a manual process done by the network operator inside their configuration procedures for network management from the NMC.

Parameter	Description	Type
SSP identifier	Identifies the Satellite Service provider.	Octet String
Initial system details	Includes the details necessary to acquire the initial System Information tables with signalling messages/carrier(s) for the system.	Octet String
Frequency bands	List of frequency bands used by the system.	Octet String
Coverage area	System coverage area description.	Octet String

Table 4-8: External system parameters

5. REFERENCE RECEIVER PERFORMANCES

5.1 Forward Link

5.1.1 Receiver model

This section is aimed at presenting the forward link reference receiver model. In addition, reference algorithms are described and their performances presented.

For the sake of completeness, the overall forward link physical layer receiver block diagram, already included in [AD-02], is shown in Figure 5-1.

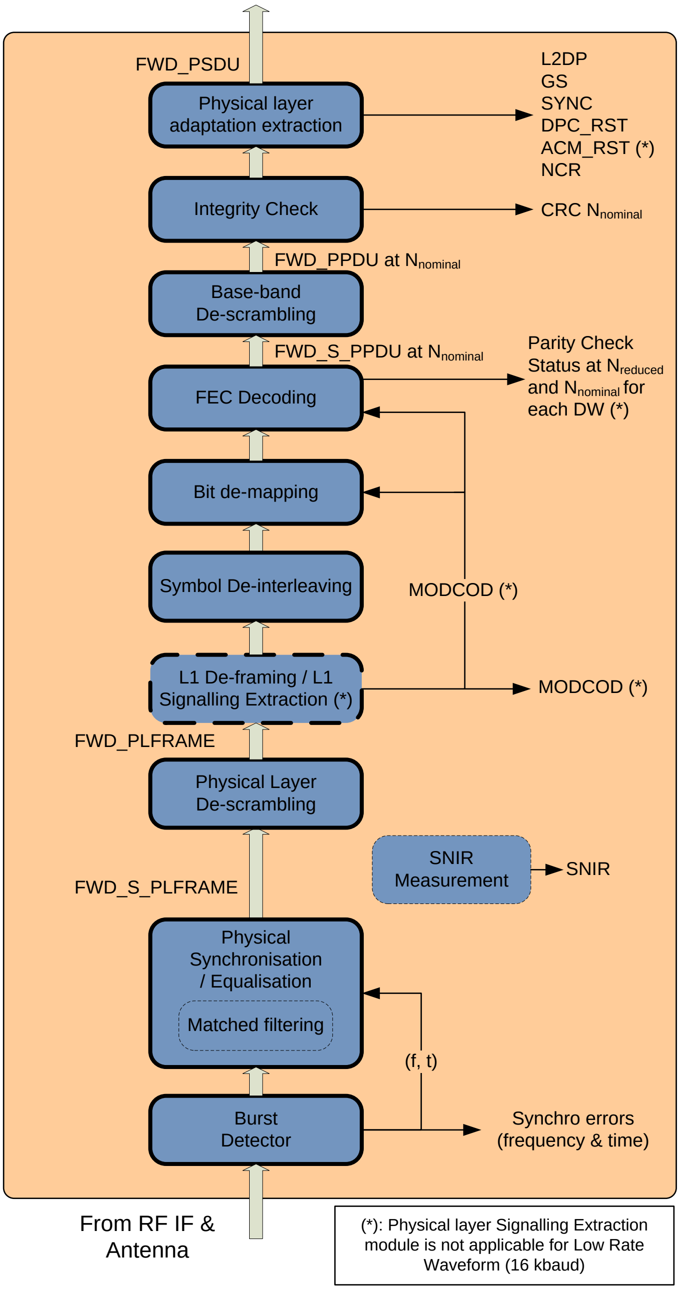


Figure 5-1: Forward link physical layer receiver block diagram (symbol rate of 160 kbaud)

It is worth mentioning that the physical layer receiver block diagram presented in the previous figure corresponds to a receiver of a symbol rate of 160 kbaud. Low rate waveform (16 kbaud) receiver does not implement the physical layer extraction module since ACM is not supported by carriers with a symbol rate of 16 kbaud.

5.1.2 LDPC decoder

The output of the IRA LDPC decoder is a decoded FWD_BBFRAME. The following MODCODs are defined for FLC at 160 kbaud:

MODCOD Id	MODCOD	N_{ldpc} (bits) [FWD_FECFRAME]	K_{ldpc} (bits) [FWD_BBFRAME]
MODCOD0	QPSK 1/4	6144 bits	1536 bits
MODCOD1	QPSK 1/3	6144 bits	2048 bits
MODCOD2	QPSK 1/2	6144 bits	3072 bits
MODCOD3	QPSK 2/3	6144 bits	4096 bits
MODCOD4	8-PSK 1/2	9216 bits	4608 bits
MODCOD5	8-PSK 2/3	9216 bits	6144 bits
MODCOD6	16-APSK 2/3	12288 bits	8192 bits

Table 5-1: FWD_BBFRAME (K_{ldpc}) and FWD_FECFRAME (N_{ldpc}) block sizes for FLC modulated at 160 kbaud.

FLC at low rate (16 kbaud) only supports MODCOD0 (QPSK 1/4) with the same parameters reported in the previous table (K_{ldpc} = 1536 bits and N_{ldpc} = 6144 bits). It is worth noting that the main difference between FCH transmitted at 160 kbaud and at 16 kbaud is the number of DW per burst (4 DW for FCH at 160 kbaud and 1 DW for FCH at 16 kbaud). As a result, the PER in AWGN presented in the following sections for MODCOD0 are also valid either for 160 kbaud or 16 kbaud.

5.1.2.1 Ideal performance results in AWGN channel

Figure 5-2 and Figure 5-3 show the Data Word Error Rate (DWER)[3] performance results in AWGN (DWER vs. Eb/N0 and DWER vs. EsN0) with ideal synchronization. These results have been obtained with:

- 50 iterations for QPSK 1/2, QPSK 2/3, 8-PSK 1/2, 8-PSK 2/3 and 16-APSK 2/3
- 80 iterations for QPSK 1/4 and QPSK 1/3

It is noted that DWER and PER are different concepts:

[3] Data Word is also referred as FWD_BBFRAME

- PER is measured per PPDU
- DWER is measured per data word (1 PPDU is composed of 4 data words)

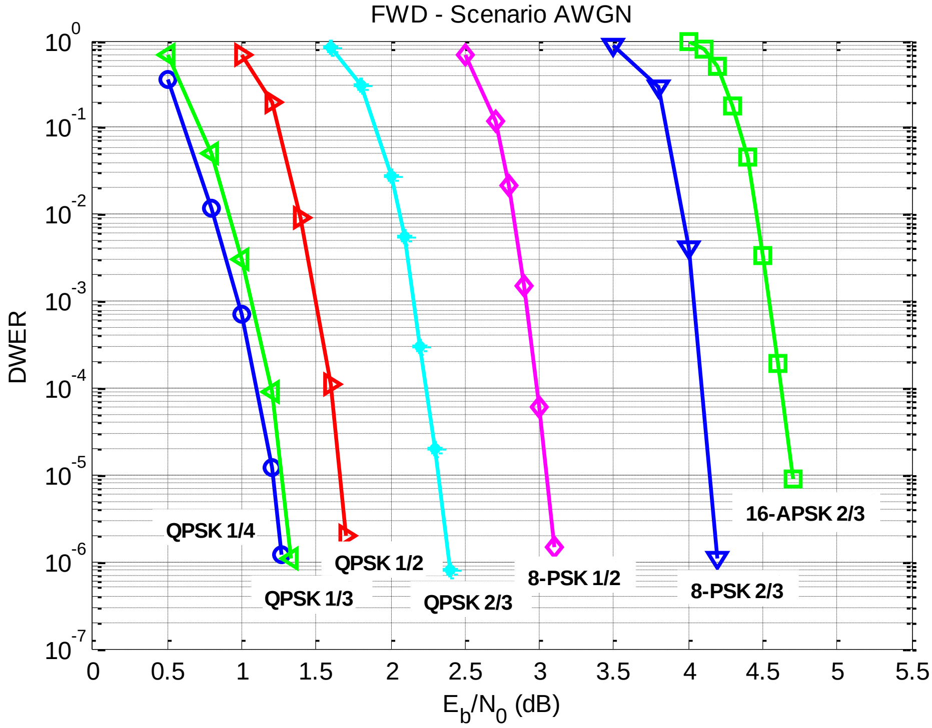


Figure 5-2: Forward link DWER vs. Eb/No for AWGN Channel with ideal synchronization

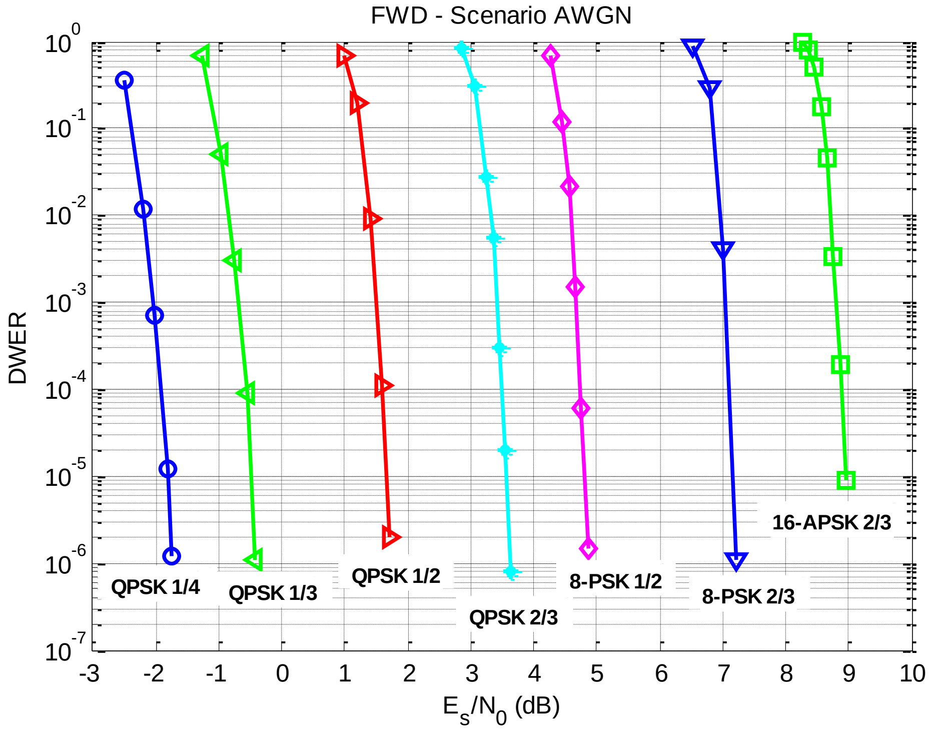


Figure 5-3: Forward link DWER vs. Es/No for AWGN Channel with ideal synchronization

5.1.2.2 Impact of LDPC decoder iterations on robust MODCODS

As stated in section 5.1.2.1, the nominal number of LDPC decoder iterations for all the MODCODs is set at 50 with the exception of the most robust MODCODs (i.e., QPSK 1/3 and QPSK 1/4), in which 80 iterations is used. The impact of using 50 iterations for these two MODCODs in the aeronautical propagation channel results in a marginal degradation around 0.2 dB, as shown in Figure 5-4 and Figure 5-5.

Note: additional PER curves in aeronautical environment are provided in section 5.1.3.2.2.

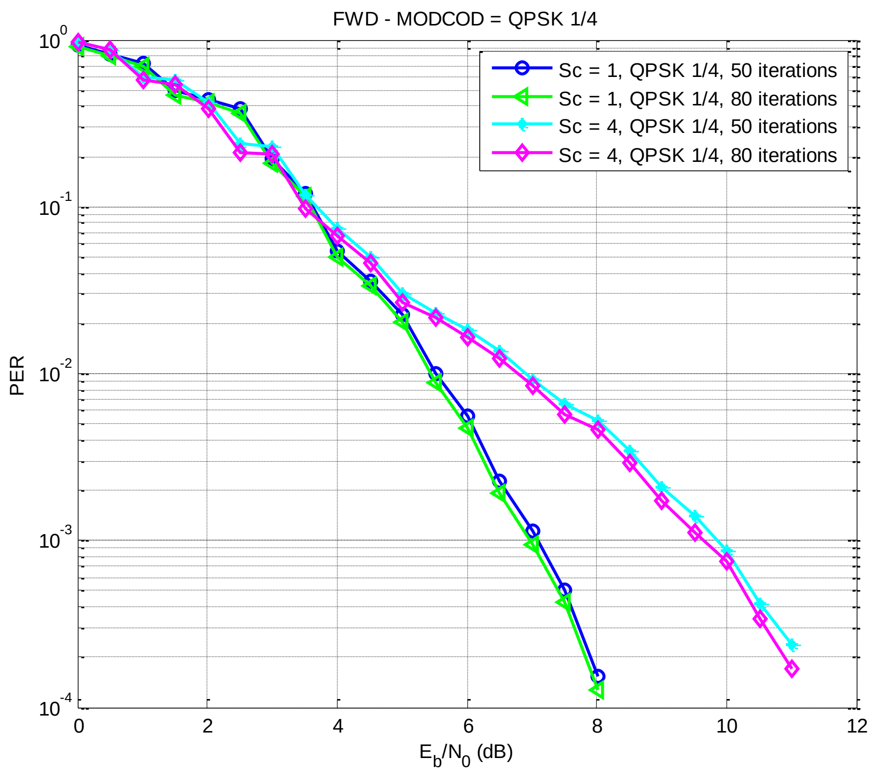


Figure 5-4: Impact of LDPC decoder iterations on the PER curves (QPSK 1/4)

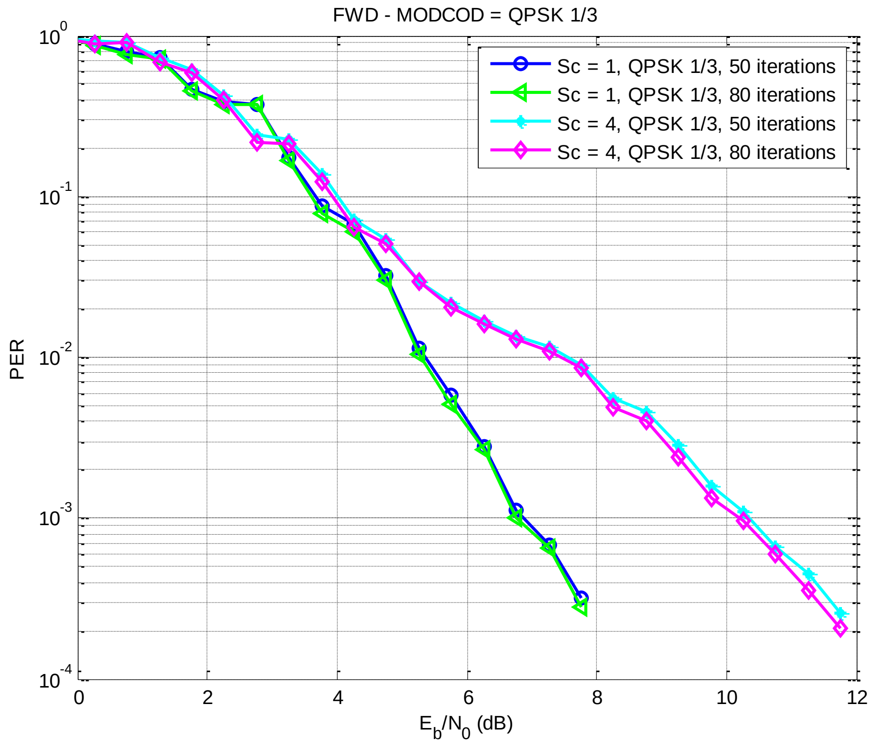


Figure 5-5: Impact of LDPC decoder iterations on the PER curves (QPSK 1/3)

5.1.3 Burst detection and synchronisation

A block diagram of the reference synchronisation algorithms is shown in Figure 5-6. It consists of two main blocks:

- *Burst detector/acquisition*: It is responsible for the burst detection by means of a preamble detector. Time and carrier frequency offsets are estimated and compensated for locally. The estimated time and frequency offsets are used to assist the UT transmitter Doppler pre-compensation mechanism (6.3.2.2) and the GES synchronisation closed-loop (6.2.1.3) implemented to synchronise their transmissions to the FLC. The preamble also allows the detection of the beginning of the burst payload. In addition, the preamble is also used in the equaliser training stage.
- *Burst demodulator*: it is in charge of demodulating the burst payload symbols by means of a synchronisation/equalisation scheme. The initial equaliser coefficients (from the equaliser training stage) are refined throughout the payload duration, taking advantage of the pilot symbols inserted within the burst payload. The burst demodulator produces the soft symbols to be decoded.

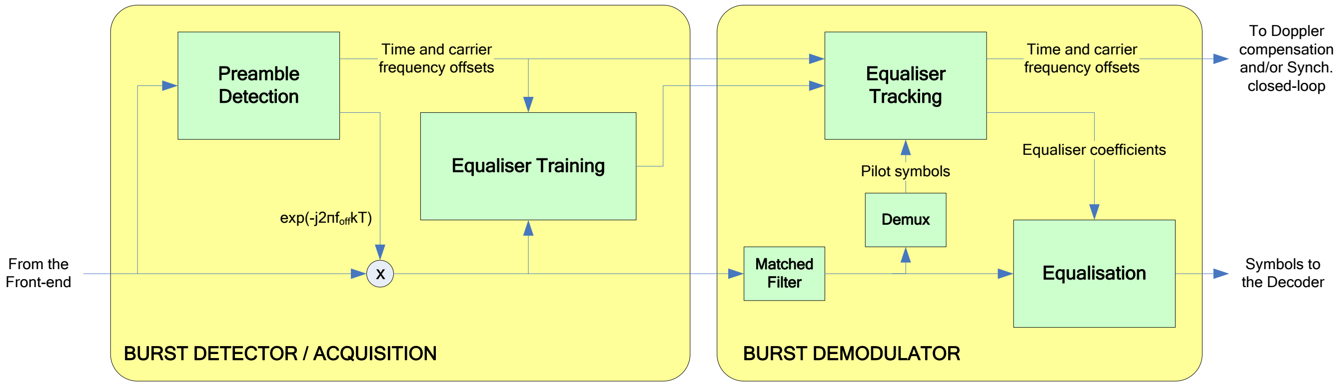


Figure 5-6: Block diagram of the burst detector and demodulator.

In the following sections, the aeronautical channel scenarios defined in “Appendix A: Aeronautical propagation channel” have been taken into account.

5.1.3.1 Burst Acquisition

In order to detect and acquire forward link bursts at the UT or GS elements, a preamble is inserted in the burst, which allows:

- The detection of the burst and the start of the burst payload.
- Estimation of the frequency offset caused by the Doppler Effect and clock errors in the forward link.
- Equaliser training to reach the best performances at the beginning of the payload.

5.1.3.1.1 Burst detector description

The block diagram of the reference burst detector algorithm is depicted in Figure 5-7.

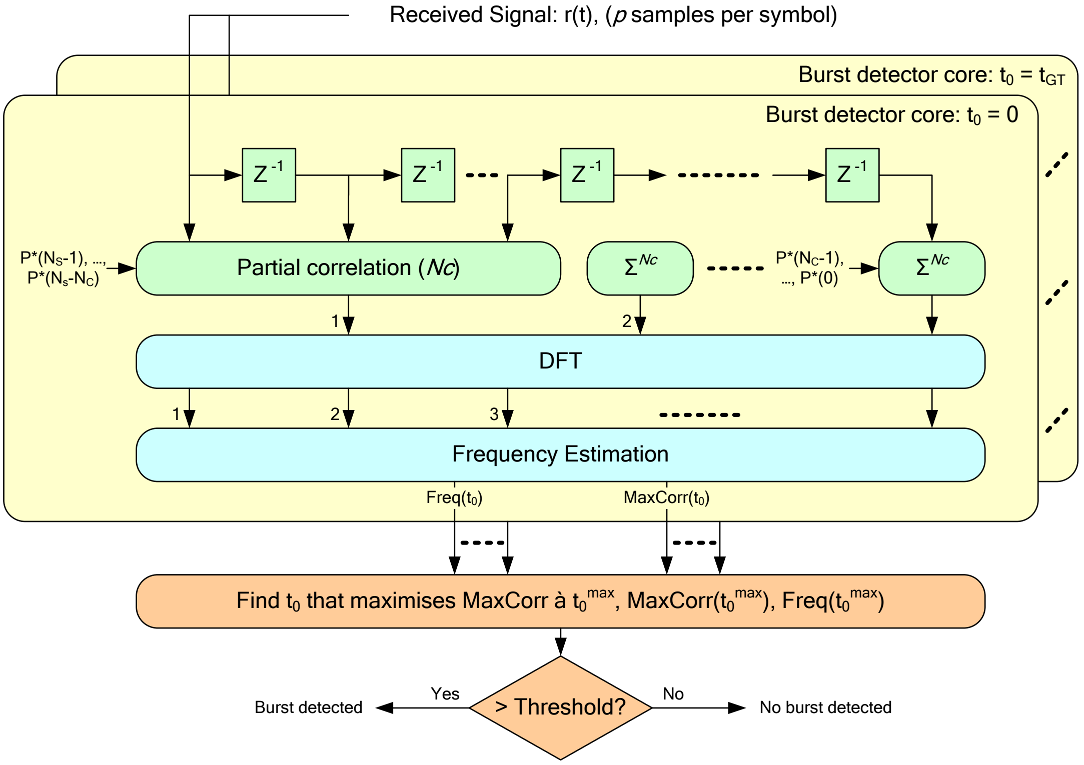


Figure 5-7: Forward link burst detector block diagram.

The burst detector is aimed at searching the time offset t_0 that maximises a correlation value *MaxCorr(t)*.

The core of the burst detector performs a full coherent integration of the received signal in two stages:

- 1st stage (in green): partial correlation (coherent integration) of *Nc* samples
  - $P(0..N_S)$ preamble sequence (preamble symbols after shaping).
  - *Nc* is adjusted according to the expected maximum input frequency error.
- 2nd stage (in blue): parallel coherent integration at several frequency offsets
  - Based on DFT.
  - The frequency with the highest correlation $MaxCorr(t_0)$ is selected.

Once the previous procedure is repeated for all time offsets within the guard time, the offset t_0^{max} that provides the highest correlation $MaxCorr(t_0^{max})$ is selected. Finally, a burst is assumed to be detected if such correlation is above a given threshold.

Thus, the previous algorithm provides the following outputs:

- Detection flag: after threshold comparison.
- t_0: time offset estimation.
- $Freq(t_0)$: frequency offset estimation.

In this document, the next concepts are defined and used as follows:

- False alarm probability (P_{fa}): probability of detecting a burst in a given time and frequency cell, i.e., the correlation being above the threshold, in the event of no transmission (no signal presence).
- Overall false alarm probability (*Overall* P_{fa} or P_{FA}): probability of detecting a burst in any time and frequency cells, i.e., the correlation being above the threshold, in the event of no transmission (no signal presence).
- Missed detection probability (P_{md}): probability of not detecting a burst, i.e., the correlation being below the threshold, in the event of transmission (signal presence).
- Detection probability (P_d): 1 - P_{md}
- Wrong detection probability (P_{wd}): probability of not detecting a burst at the right time and frequency cell. The wrong detection probability is only evaluated for those bursts which have been detected (missed detections are excluded).

  The following ranges are used to decide if the burst is detected at the right time and frequency cell:

  - Timing: ±0.3125 symbol periods
  - Normalised frequency: ±3.75e-3 Hz/baud

P_{fa} and *Overall* P_{fa} are defined in the absence of signal whereas P_{md}, P_d and P_{wd} are defined in the presence of signal.

The time offset (t_{max}) at which the matched filter output should be sampled is computed as follows:

$$\begin{aligned} r[n,t] &= \left|c_1[n]\cdot g(t)+c_2[n]\cdot g(t-\tau)\right| \\ r[n,t_{\max}] &= \max(r[n,t]) \end{aligned},$$

where:

- n denotes the n-th received symbol;
- $c_1[n]$ and $c_2[n]$ are the taps of the LOS+LS and GR channel components(see Appendix A: Aeronautical propagation channel), respectively;
- $g(t)$ is the raised cosine pulse with roll-off factor 0.2;
- τ is the delay of the GR component with respect to the LOS+LS component.

<u>Burst detector approach:</u>

A MAX search approach is adopted for the forward link burst detector:

- Time slots with signal on them are assumed to be known beforehand. Therefore, the burst detector threshold is set to zero. Note that, even if the time slots with signal is unknown, the false alarm event on those with no signal will not cause any packet losses.
- By definition P_{fa} = 1, P_{md} = 0, P_d = 1. Thus, the only relevant probability is P_{wd}.

5.1.3.1.2 Burst detector performances

Forward link burst detector performances are presented in Table 5-2 for the FCH preamble length, i.e., $N_{FWD\_PREAMBLE}$ = 100 symbols for carriers of 160 kbaud and $N_{FWD\_PREAMBLE}$ = 160 symbols for carriers of 16 kbaud (Low Rate Waveform specification). Performances have been obtained from bursts affected by time and frequency errors uniformly distributed within the maximum range specified in [AD-02]:

- Time error: ± G_T / 2, with G_T the forward link guard time.
- Frequency acquisition range:
  - ±37 kHz for a symbol rate of 160 kbaud and
  - ±7 kHz for a symbol rate of 16 kbaud.
- Frequency drift (Doppler rate): ±350 Hz/s.

Scenario Id	Aero Scenario	Rs [kbaud]	Es/No [dB]	Pwd
FWD-1.1	1	160	-0.5	<< 1E-4
FWD-3.1	3	160	1	2.00E-05
FWD-3.2	3	16	2	<< 1E-4
FWD-4.1	4	160	2	1.15E-04
FWD-4.2	4	160	4.25	<< 1E-4
FWD-5.1	5	160	5	4.00E-05
Note: the characteristics of each Aero Scenario are reported in Appendix A.				

Table 5-2: Forward link burst detector performances

The above performances show that wrong detection probabilities are at least one order of magnitude better than the PER (see section 5.1.3.2.2).

Forward link burst detector/acquisition timing and frequency synchronisation errors reach the following performances for all the scenarios above:

- RMS time estimation error: σ_T< 1/16 symbol periods.
- RMS frequency estimation error: σ_F/R_s< $7.5\cdot10^{-4}$ Hz/baud, R_s being the symbol rate.

5.1.3.2 Burst demodulation

5.1.3.2.1 Burst demodulation description

The forward link burst payload is composed of data and pilot symbols, the latter used to assist coherent demodulation/equalisation at the receiver. Pilots in the payload are structured--

- In blocks of L=24 consecutive pilot-symbols with a distance of P+L=24+224=248 symbols between blocks in carriers with a symbol rate of 160 kbaud, and
- in blocks of L=6 consecutive pilot-symbols with a distance of P+L=6+54=60 symbols between blocks in carriers with a symbol rate of 16 kbaud.

The reference demodulator is based on a joint synchronization/equalization technique (equalizer for short).

A linear time-variant filter (equalizer) of variable length operating at 2 samples per symbol (Nyquist rate) is introduced to adjust the received signal timing, compensate partially for the carrier synchronization errors (residual frequency offset and frequency drift), and mitigate the intersymbol interference (ISI) introduced by the aeronautical channel in some scenarios (e.g., scenario 4). The equalizer coefficients are adjusted, minimizing the mean square error (MSE) between the output of the filter and the reference symbols coming from the preamble (initial design) and from the payload pilot symbols (during tracking) [RD-02]. It is found that, by minimizing that MSE, we are indirectly maximizing the strobes SNIR at the FEC decoder input.

The equalizer is first trained using the known preamble symbols and then a normalized least mean square (NLMS) algorithm [RD-03] is run to adapt the equalizer coefficients (in the MMSE sense) to the temporal evolution of the channel impulse response. Because the optimal equalizer length depends on the scenario and changes in time during the burst, the pilot symbols are exploited to monitor and switch the equalizer length along every received burst under a MMSE criterion. Fine frequency offset maximum-likelihood synchronization [RD-04] based on the equalized preamble data is also incorporated to improve carrier synchronization in highly dispersive scenarios (e.g., scenario 4). The flow diagram of the FWD link burst demodulator is depicted in Figure 5-8.

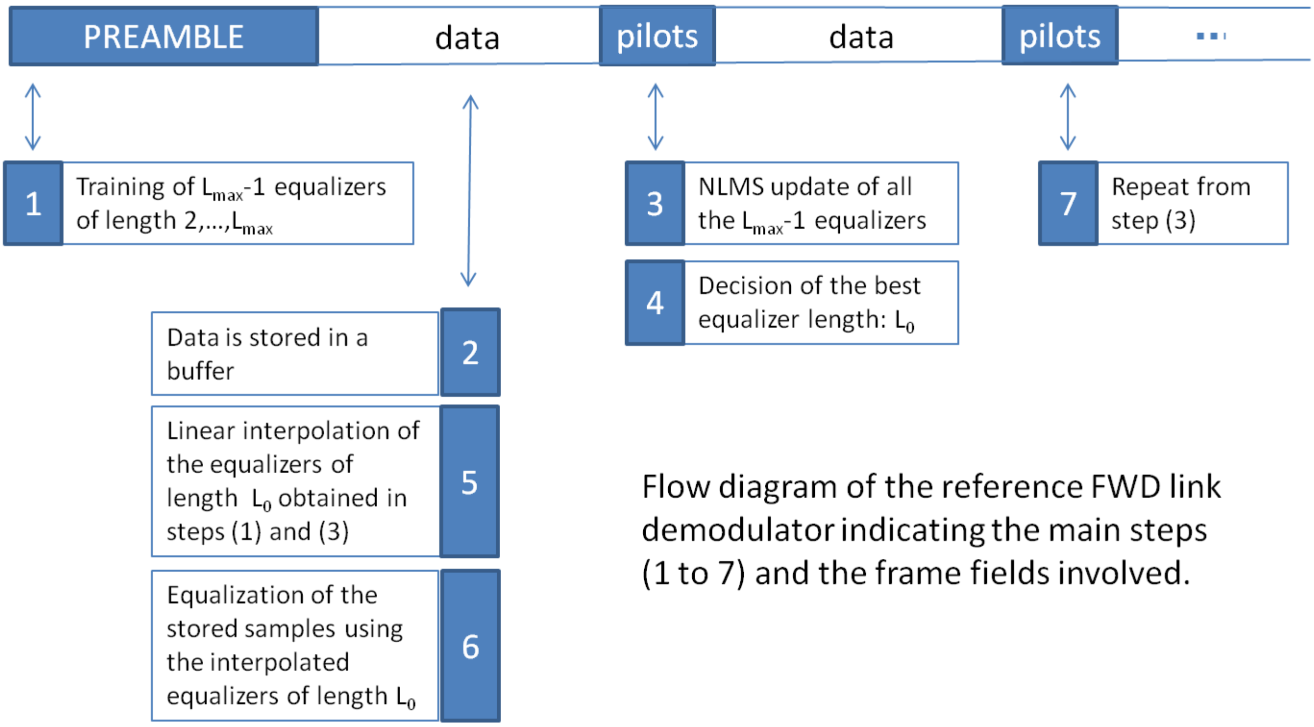


Figure 5-8: Flow diagram of the FWD link burst demodulator

The LDPC decoder is provided with the instantaneous SNIR metric, which is estimated using a SNIR tracker that processes both the payload pilot and data symbols. The reference scheme operates at symbol rate and corresponds to the optimal maximum likelihood estimator in case of uncorrelated noise plus interference. Closed-form data-aided estimates are obtained from the pilot symbols whereas an iterative fixed-point method provides non-data-aided estimates from the random data symbols by exploiting the knowledge of the transmitted constellation (e.g., QPSK, 8-PSK, 16-APSK). The reference scheme is the adaptation of [RD-05] to the single-antenna case operating at the matched-filter output. Linear smoothing is applied to the above raw reduced-rate SNIR estimates in order to decrease the estimator variance while tracking correctly the SNIR temporal evolution.

5.1.3.2.2 Burst demodulation performances

The performances of the joint synchronisation/equalisation scheme are presented in terms of PER. Performances have been obtained under the effect of the residual burst detector carrier frequency and timing errors, which are emulated as follows:

- Residual frequency error: frequency error emulated with a zero-mean normal distribution with σ_F/R_S = 7.5e-4 Hz/baud, R_s being the symbol rate.
- Residual timing errors: time error emulated with a zero-mean normal distribution with σ_T = 1/16 symbol periods.

In addition, a frequency drift (Doppler rate) uniformly distributed within ±350 Hz/s has been considered. PER performances are reported in the following figures.

No transmitter and receiver phase noise masks have been used to obtain the following performances because their impact has been proven to be negligible.

Performances for symbol rate of 160 kbaud have been obtained with the reference burst demodulator presented in 5.1.3.2.1 configured as follows:

- NMLS forgetting factor: μ = 0.85
- Maximum equalizer length: L_{max} = 15

It is worth mentioning that if no equalisation is performed, PER curves have a floor at high Es/No. For instance, a floor above $2{\cdot}10^{-3}$ and $2{\cdot}10^{-2}$ is obtained in Scenario 4 with MODCOD QPSK 1/4 and QPSK 1/3 respectively, even if ideal synchronisation is assumed.

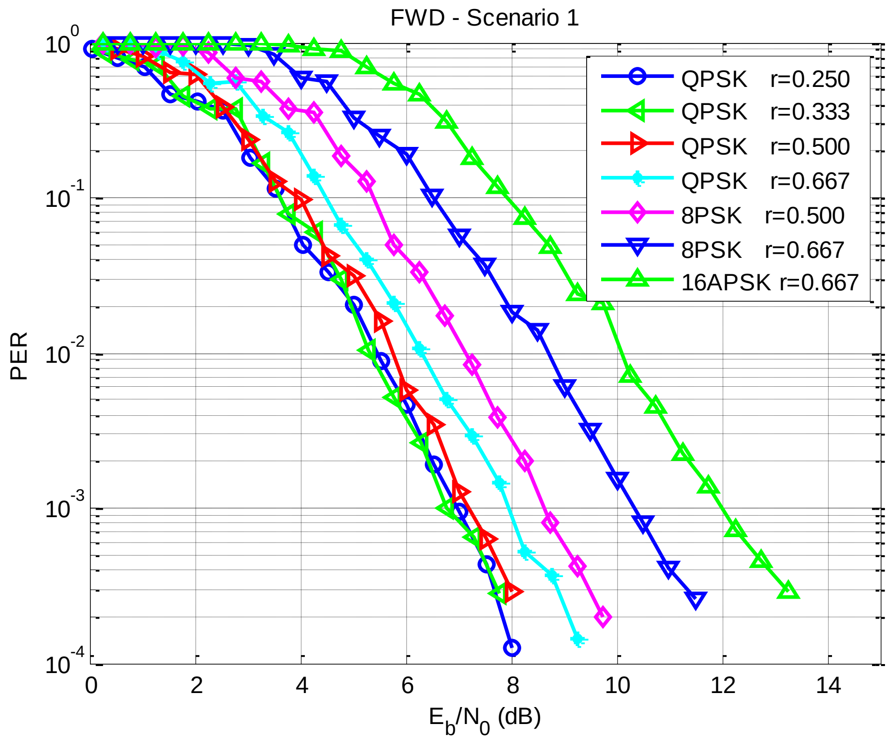


Figure 5-9: Forward link PER vs. Eb/No for Channel Scenario 1 and symbol rate of 160 kbaud.

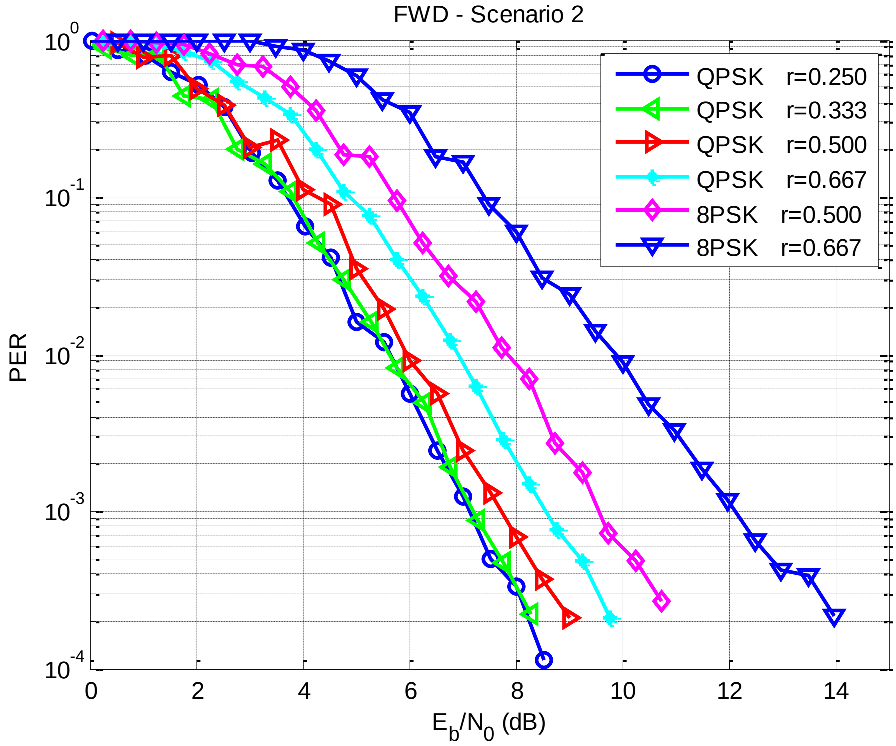


Figure 5-10: Forward link PER vs. Eb/No for Channel Scenario 2 and symbol rate of 160 kbaud.

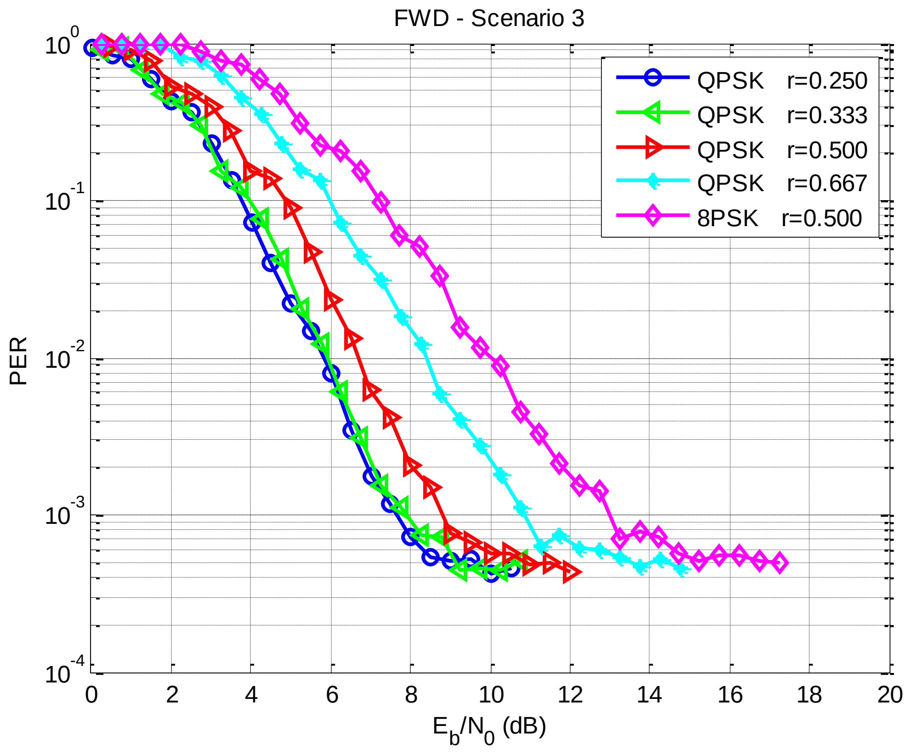


Figure 5-11: Forward link PER vs. Eb/No for Channel Scenario 3 and symbol rate of 160 kbaud.

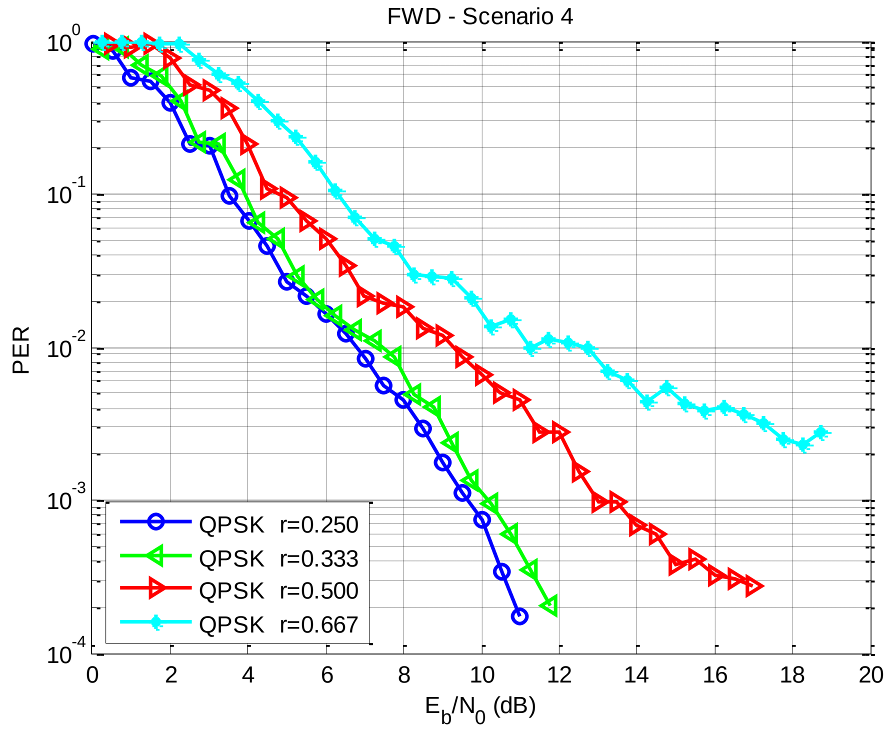


Figure 5-12: Forward link PER vs. Eb/No for Channel Scenario 4 and symbol rate of 160 kbaud.

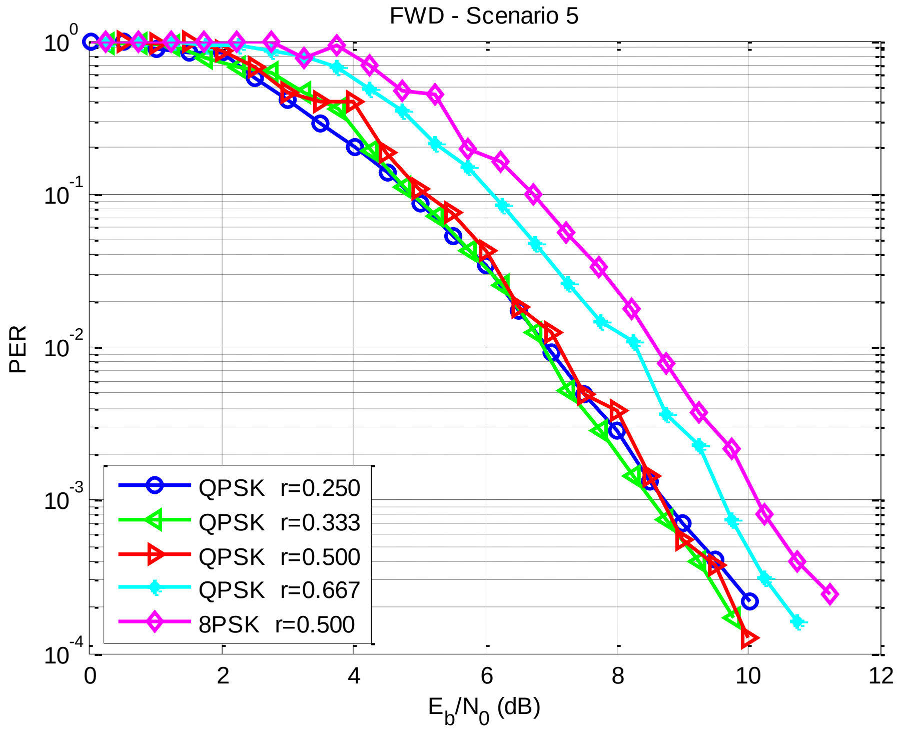


Figure 5-13: Forward link PER vs. Eb/No for Channel Scenario 5 and symbol rate of 160 kbaud.

Performances for low rate waveform (16 kbaud) have been obtained taking into account the same residual burst detector carrier frequency and timing errors and frequency drift as for the symbol rate of 160 kbaud and with the reference burst demodulator presented in 5.1.3.2.1 configured as follows:

- NMLS forgetting factor: $\mu = 0.85$
- Maximum equalizer length: $L_{max} = 2$

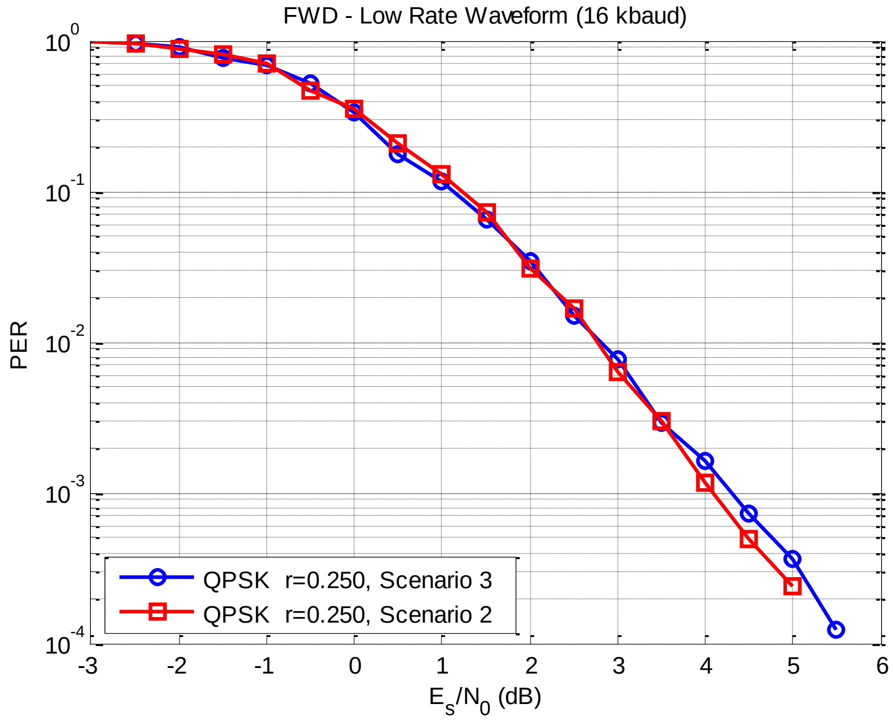


Figure 5-14: Forward link PER vs. Eb/No for Scenario 2 and 3 and Low Rate Waveform (16 kbaud)

5.1.4 Physical layer signalling (L1 header)

L1 header carries the necessary information to configure the receiver in order to demodulate and decode the received burst. The Physical layer signalling (also called L1 header) is only inserted in the FCH bursts transmitted at a symbol rate of 160 kbaud, in which ACM is supported. Low rate carriers do not support ACM, and in consequence, L1 header is not inserted.

The following figure illustrates the FCH burst structure for a symbol rate of 160 kbaud.

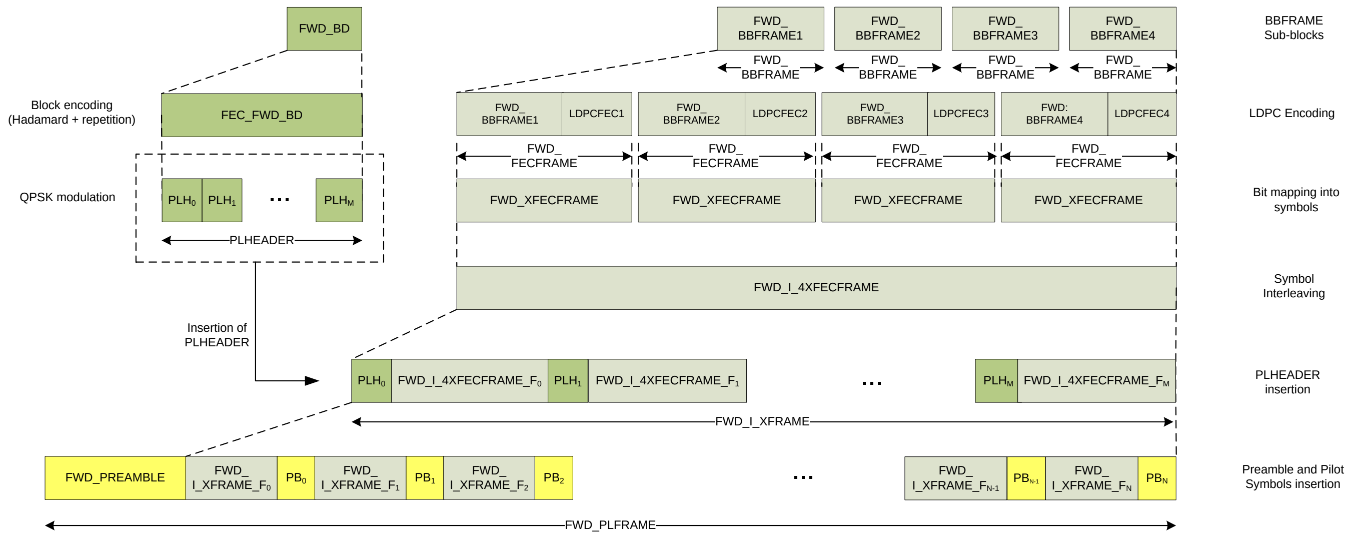


Figure 5-15: FCH burst structure for a symbol rate of 160 kbaud

The PLHEADER is a 128-bit Code Word QPSK modulated, i.e., 64 QPSK symbols, which conveys 4 bits of information (FWD_DD field). The 4 bits carried by the FWD_DD field inform the receiver about the modulation and coding scheme of the Data Payload. The 128-bit Code Word is the result of concatenating a Hadamard code [8, 4, 4] and a repetition code with a repetition factor 16.

5.1.4.1 Physical layer signalling decoding

The PLHEADER can be easily decoded in Soft Decision (SD) as follows:

- First, the repetition code is exploited by adding the 16 repetitions of the same bit, resulting in a Code Word (*w*) of size 8.

- Compute $S = w * H_8$, or explicitly, $S_i = \sum_{j=0}^{7} w_j H_{8\,ji}$ for $0 \le i < 8$

  where the Hadamard matrix H_8 is defined recursively with

$$H_1 = (1) \qquad H_{2n} = \begin{pmatrix} H_n & H_n \\ H_n & -H_n \end{pmatrix}$$

- Let 'i' represent the index of the S_i having the largest absolute value. The Data Word (FWD_DD) is then:
  - *FWD_DD* $= i_{bin}$ if $S_i > 0$
  - *FWD_DD* $= i_{bin} \oplus 1000_{bin}$ if $S_i < 0$

5.1.4.2 Physical layer signalling performances

The performances of the L1 header coding scheme are presented in terms of PER. Figure 5-16 shows the theoretical PER of an ideal receiver in AWGN channel (ideal receiver with perfect synchronization) using SD decoding.

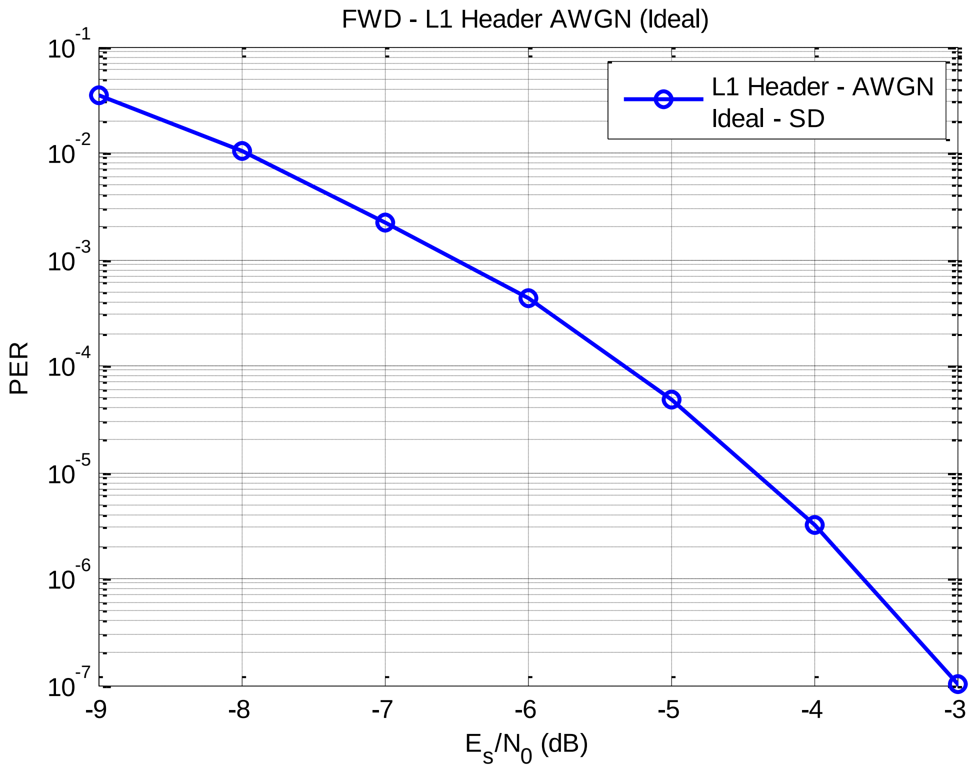


Figure 5-16: L1 header PER vs. Es/No in AWGN Channel with perfect synchronization

L1 header PER performances with joint synchronization/equalization scheme is presented in Figure 5-17 and Figure 5-18 for aeronautical Scenario #1 and #4 respectively. L1 header performances have been obtained under the same conditions and with the same burst demodulator configuration reported in section 5.1.3.2.2.

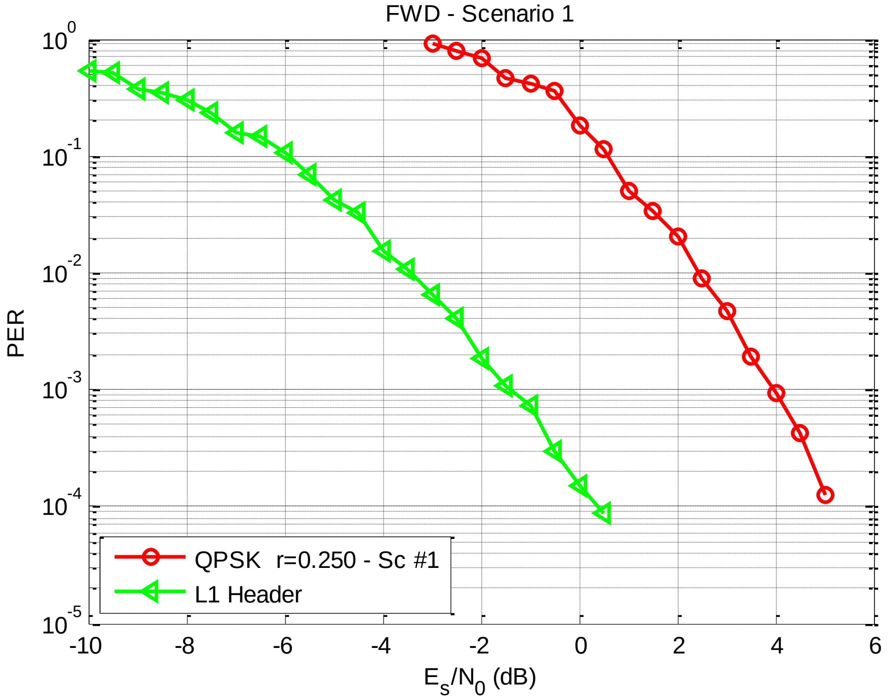


Figure 5-17: PER vs. Es/N0 results for MODCOD0 (QPSK 1/4) and L1 header for Scenario 1 and symbol rate of 160 kbaud

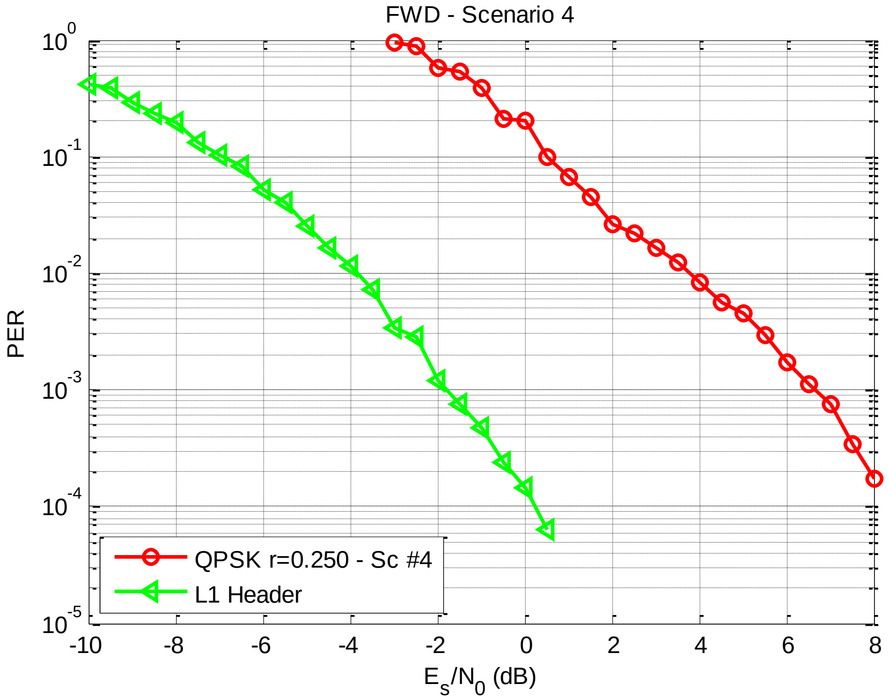


Figure 5-18: PER vs. Es/N0 results for MODCOD0 (QPSK 1/4) and L1 header for Scenario 4 and symbol rate of 160 kbaud

5.2 Return Link

5.2.1 Receiver model

This section is aimed at presenting the return link reference receiver model. Reference algorithms are described and their performances presented.

For the sake of completeness, the overall return link physical layer receiver block diagram, already included in [AD-02], is shown in Figure 5-19.

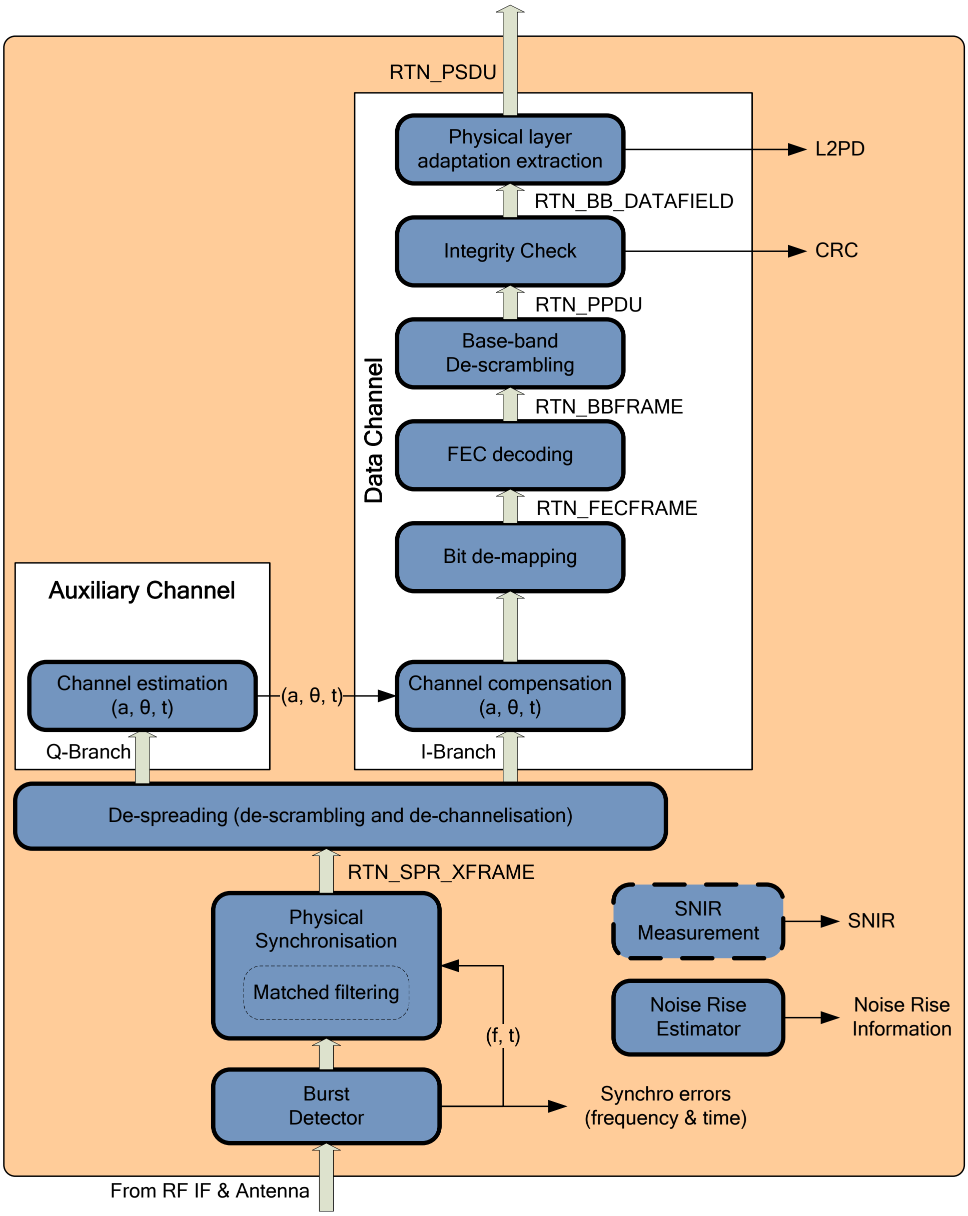


Figure 5-19: Return link physical layer receiver block diagram

The previous block diagram corresponds to a regular SSA receiver and is also applicable to an E-SSA receiver in which Successive Interference Cancellation (SIC) is implemented (see [RD-06] to [RD-11]).The reference SIC is presented in section5.2.4.

5.2.2 TCC decoder

TCC performance results in AWGN with ideal synchronization are depicted in the following figures. These results have been obtained with a maximum of 10 decoder iterations with the Log-MAP algorithm.

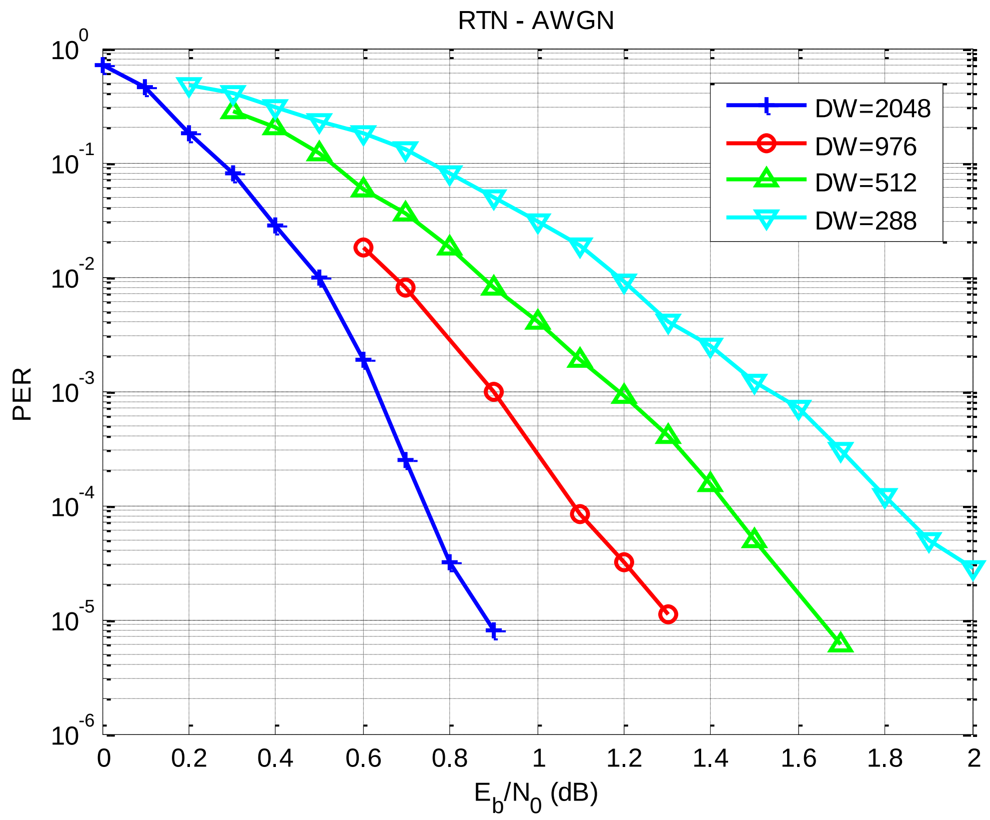


Figure 5-20: Return link PER vs. Eb/No for AWGN channel with ideal synchronization

5.2.3 Burst detection and synchronisation

A block diagram of the reference synchronisation algorithms is shown in Figure 5-21. It consists of two main blocks:

- *Burst detector/acquisition*: It is responsible for the burst detection by means of a preamble detector. Time and carrier frequency offsets are estimated and compensated for locally. The preamble also allows the detection of the beginning of the burst payload. The preamble is also used in the synchronisation tracker acquisition stage.
- *Burst demodulator*: It is in charge of demodulating the burst payload symbols by means of a synchronisation tracker. Time and carrier phase tracking algorithms take advantage of the pilot symbols inserted within the burst payload. The payload demodulator produces the soft symbols to be decoded.

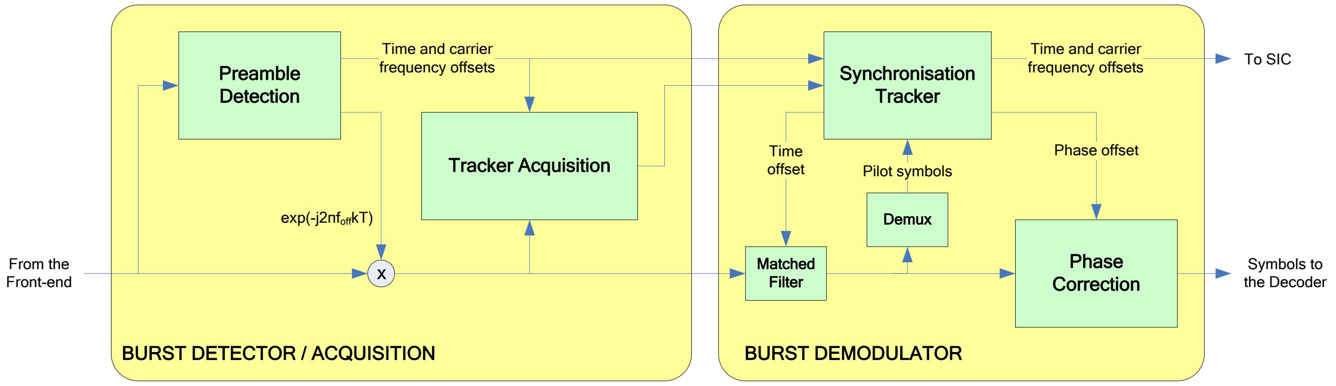


Figure 5-21: Block diagram of the burst detector and demodulator

In the following sections, the aeronautical channel scenarios defined in “Appendix A: Aeronautical propagation channel” have been taken into account.

5.2.3.1 *Burst acquisition*

In order to detect and acquire return link bursts at the GS, a preamble is inserted in the burst, which allows:

- The detection of the burst and the start of the burst payload.
- Estimation of the residual frequency offset that remains after the Doppler pre-compensation process implemented by the UT in the return uplink.
- Synchronisation tracker acquisition to reach the best performances at the beginning of the payload.

5.2.3.1.1 *Burst detector description*

The block diagram of the reference burst detector algorithm is depicted in Figure 5-22.

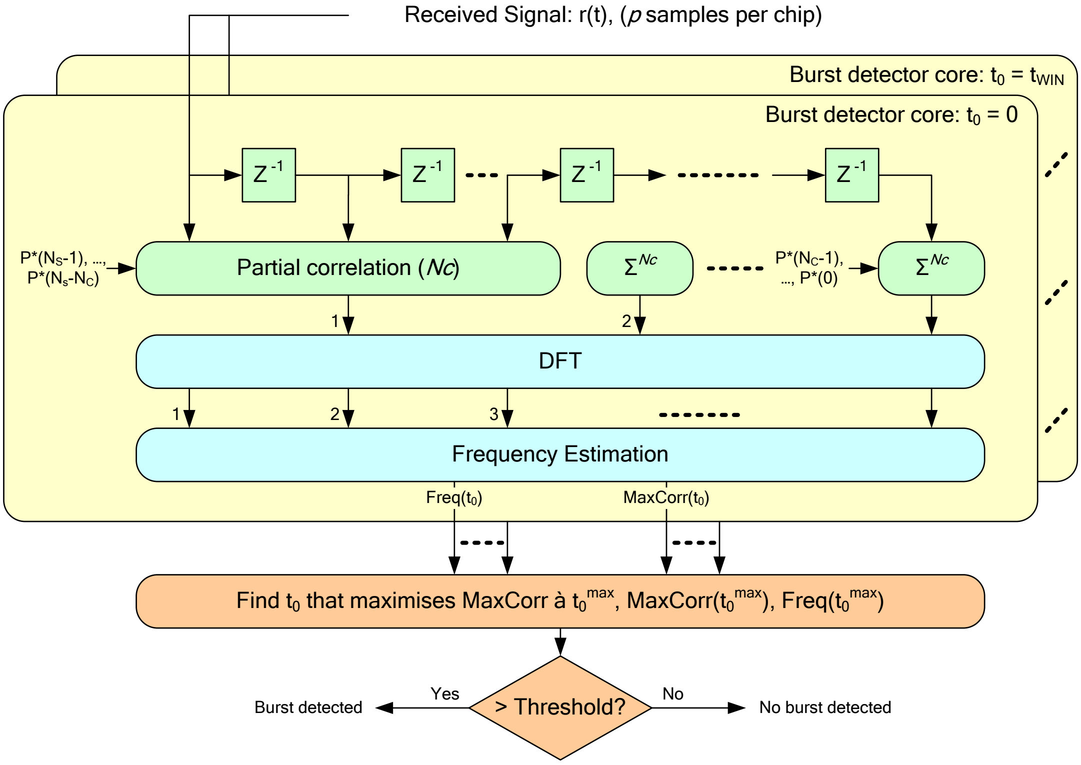


Figure 5-22: Return link burst detector block diagram

The burst detector is aimed at searching for the time offset t_0 that maximises a correlation value *MaxCorr(t)*.

The core of the burst detector performs a full coherent integration of the received signal in two stages:

- 1st stage (in green): partial correlation (coherent integration) of *Nc* samples.
  - $P(0..N_S)$ preamble sequence (preamble chips after shaping).
  - *Nc* is adjusted according to the expected maximum input frequency error.
- 2nd stage (in blue): parallel coherent integration at several frequency offsets.
  - Based on DFT.
  - The frequency with the highest correlation $MaxCorr(t_0)$ is selected.

Once the previous procedure is repeated for all time offsets within the observation window of the SIC, the offset t_0^{max} that provides the highest correlation $MaxCorr(t_0^{max})$ is selected. Finally, a burst is assumed to be detected if such correlation is above a given threshold.

Thus, the previous algorithm provides the following outputs:

- Detection flag: after threshold comparison
- t_0: time offset estimation
- $Freq(t_0)$: frequency offset estimation

Those definitions presented in section 5.1.3.1 are applicable.

Burst detector approach

A Threshold Crossing (TC) approach is adopted for the return link burst detector:

- The signal presence is unknown. For this reason, a threshold is properly set to reach a certain false alarm probability: Constant False Alarm Rate (CFAR) approach. Note that, excessive false alarm events may impact significantly on Successive Interference Canceller performances.

5.2.3.1.2 Burst detector performances

Return link burst detector performances are presented in Table 5-3 for the RACH preamble length, i.e., $N_{RTN\_PREAMBLE}$ = 128 symbols. Performances have been obtained from bursts affected by frequency errors uniformly distributed within the maximum range specified in [AD-02]:

- Frequency acquisition range: ±5.8 kHz.
- Frequency drift (residual Doppler rate after UT transmitter Doppler pre-compensation): ±50 Hz/s.

The burst detector threshold has been set to reach a false alarm probability of 10^{-6}.

Scenario Id	Aero Scenario	Rs [kbaud]	Es/No [dB]	Pmd or Pwd
RTN1.1x	1	10	0.23	1.30E-5
RTN1.2x	1	40	-0.77	5.00E-5
RTN2.1x	2	10	0.23	1.30E-5
RTN2.2x	2	40	-0.77	4.70E-5
RTN3.1x	3	10	0.23	1.50E-5
RTN3.2x	3	40	-0.52	6.20E-5
RTN4.1x	4	10	0.23	3.90E-4
RTN4.2x	4	40	-0.27	3.30E-4
RTN5.1x	5	10	5.23	4.00E-6
RTN5.2x	5	40	4.73	1.00E-4
Note: the characteristics of each Aero Scenario is reported in Appendix A.				

Table 5-3: Return link burst detector performances.

The above performances show that missed-/wrong-detection probabilities are at least one order of magnitude better than the PER (see section 5.2.3.2.2).

Return link burst detector/acquisition timing and frequency synchronisation errors reach the following performances for all the scenarios above:

- RMS time estimation error: σ_T< 1/16 chip periods.
- RMS frequency estimation error: σ_F/R_s< $7.5 \cdot 10^{-4}$ Hz/baud, R_s being the symbol rate.

5.2.3.2 Burst demodulation

5.2.3.2.1 Burst demodulation description

The return link A-CDMA payload is composed of two orthogonal BPSK channels transmitted simultaneously using different OVSF codes:

- DCH (Data CHannel): channel containing de-encoded data.
- ACH (Auxiliary CHannel): channel consisting of a pre-defined pilot symbol sequence. Its only purpose is to assist the return link receiver to allow coherent demodulation.

The same Spreading Factor (SF) is used for both DCH and ACH channels. The spread DCH and ACH channels are then scrambled using a complex scrambling sequence. The ACH channel is transmitted with a level 10 dB below the DCH channel.

The reference burst demodulator consists of a carrier phase estimator performed on the ACH channel.

The reference carrier phase estimator is a Feed-Forward Data-Aided (FF-DA) scheme with a sliding window filter of *N* symbols. A Maximum Likelihood (ML) phase estimator [RD-04] is used for carrier phase recovery[4]:

$$\hat{\theta} = \arg\left(\sum_{k=0}^{N-1} c_k^* x(k)\right),$$

where *x(k)* is the output of the matched filter and c_k are the known ACH symbols. Note that a delay of (N-1)/2 symbols is applied in order to remove the estimator bias.

Since the time drift during the burst duration is negligible (thanks to the transmitter Doppler pre-compensation implemented by the UT and the Feeder link Doppler pre-compensation implemented by the GS elements), no time tracking is required.

5.2.3.2.2 Burst demodulation performances

The performances of the burst payload demodulator are presented in terms of PER. Thus, in order to obtain the payload demodulator performances in terms of PER, the soft symbols recovered by the burst demodulator are input to the Turbo Decoder.

Performances have been obtained under the effect of the residual burst detector carrier frequency and timing errors, which are emulated as follows:

- Residual frequency error: frequency error emulated with a zero-mean normal distribution with σ_F/R_S = 7.5e-4 Hz/baud, R_s being the symbol rate.
- Residual timing errors: time error emulated with a zero-mean normal distribution with σ_T = 1/16 chip periods.

In addition, a frequency drift (Doppler rate) uniformly distributed within ±50 Hz/s has been considered.

[4] The frequency offset estimated by the burst detector is compensated for before initiating the carrier phase recovery process.

The following transmitter and receiver phase noise masks have been used.

GES phase noise

The reference carrier phase noise mask generated by a GES in downlink is listed here:

- -65 dBc/Hz at 100Hz
- -75 dBc/Hz at 1 kHz
- -85 dBc/Hz at 10 kHz
- -95 dBc/Hz at 100 kHz

UT phase noise

The carrier phase noise mask (random fluctuation) generated by the UT in uplink (TX phase noise) is listed here:

- -55 dBc/Hz at 100Hz
- -73 dBc/Hz at 1kHz
- -75 dBc/Hz at 10kHz
- -90 dBc/Hz at 100kHz
- -90 dBc/Hz at 1MHz

The reference discrete phase noise spectral components generated by the UT in uplink are listed below with reference to the indicated frequency offset:

- -40 dBc for 100Hz to 3kHz
- -50 dBc for 3kHz to 10kHz
- -60 for 10kHz to 100kHz

The integrated sum of all discrete and continuous spectral components (double sideband) generated by the UT in uplink between 10 Hz and 100 kHz from the carrier are below -24 dBc.

PER performances are reported in the following figures.

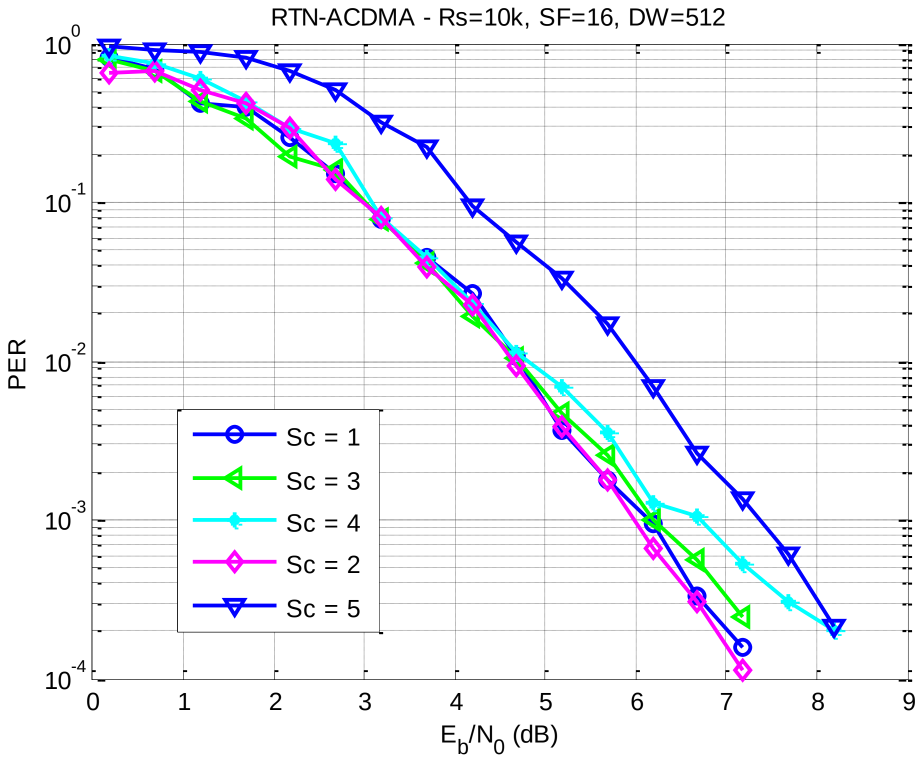


Figure 5-23: Return link PER vs. Eb/No for RACH_CR$_{160}$_SF$_{16}$_DB$_{512}$

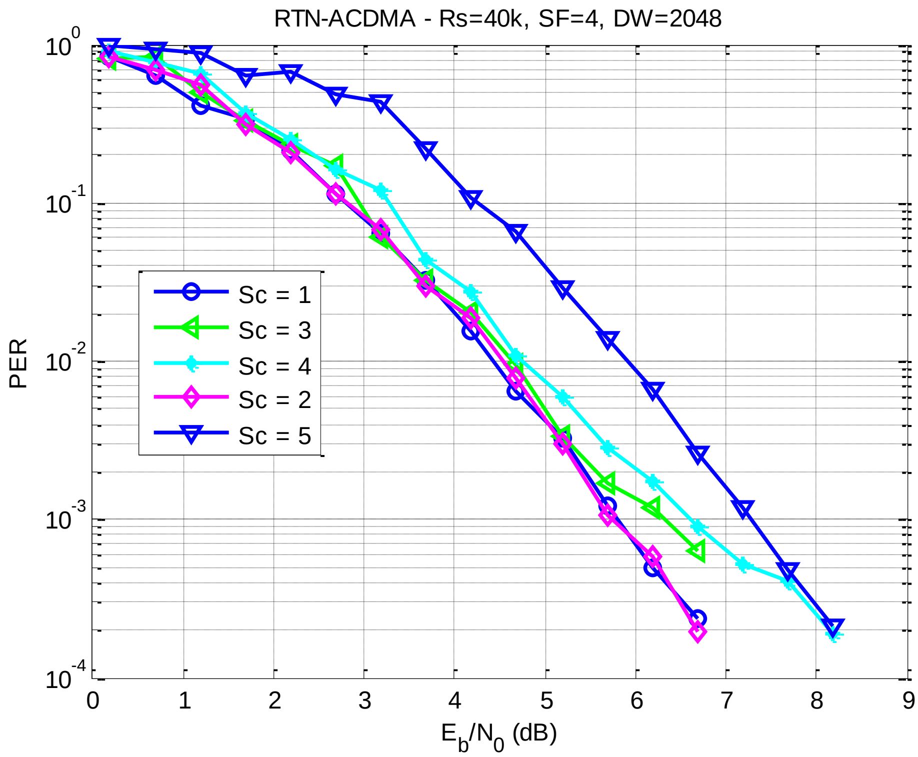


Figure 5-24: Return link PER vs. Eb/No for RACH_CR$_{160}$_SF$_4$_DB$_{2048}$

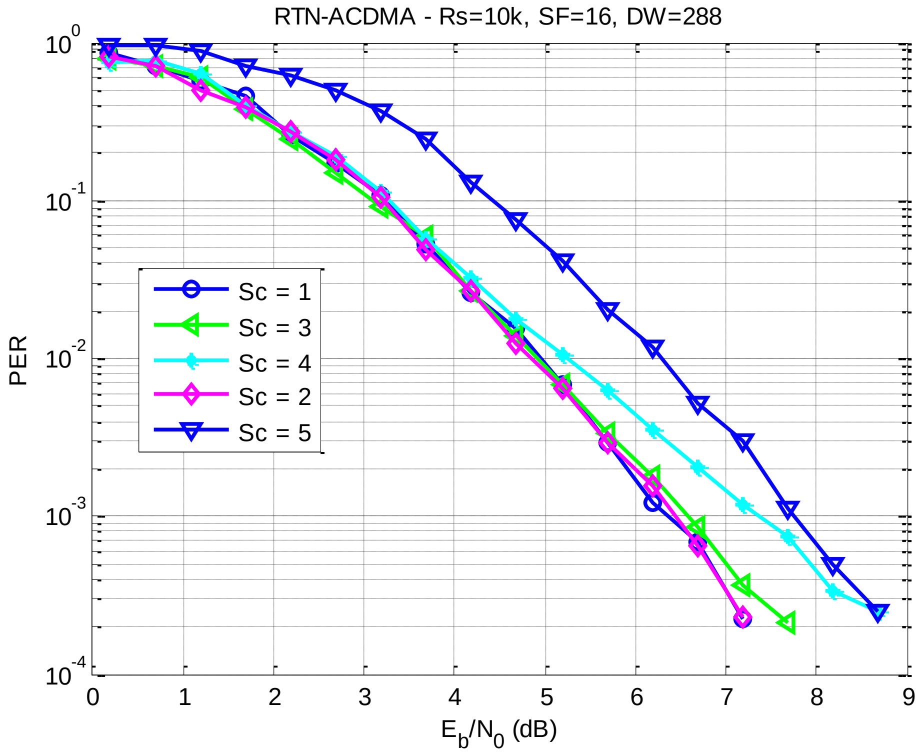


Figure 5-25: Return link PER vs. Eb/No for RACH_CR$_{160}$_SF$_{16}$_DB$_{288}$

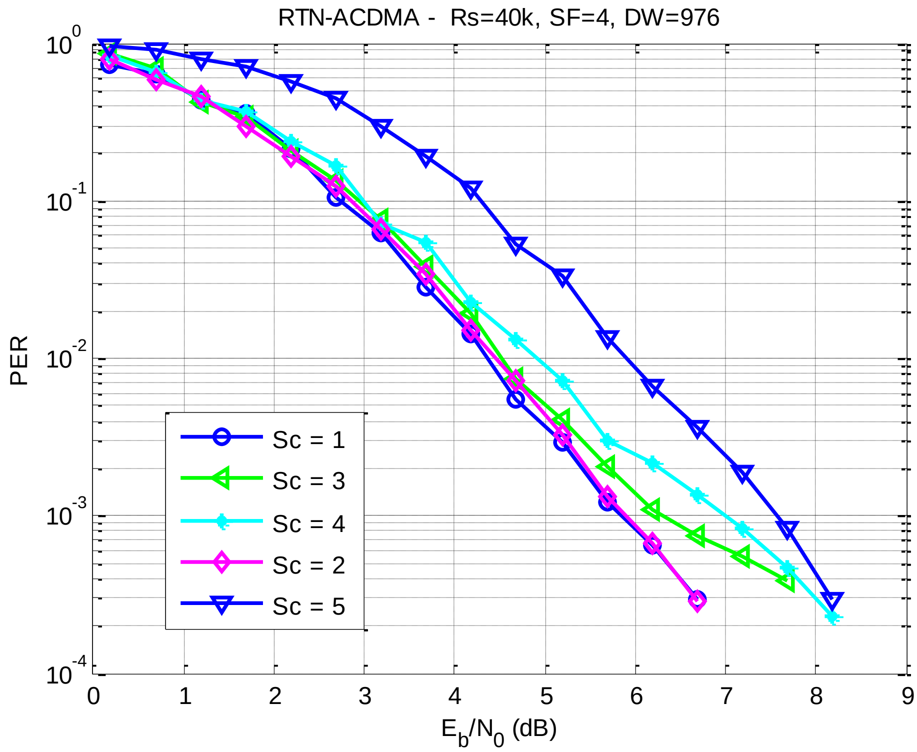


Figure 5-26: Return link PER vs. Eb/No for RACH_CR$_{160}$_SF$_{4}$_DB$_{976}$

5.2.4 *Successive Interference Canceller*

5.2.4.1 Reference SIC functional architecture

This section presents the reference Successive Interference Canceller (SIC) functional architecture, which is based on the work presented in [RD-06] and [RD-07].

5.2.4.1.1 Introduction

The received signal $r_a(\tau)$, once band-pass filtered and converted to baseband, is sampled at $F_{SAMP} = p \cdot F_{CHIP}$, where F_{CHIP} is the chip rate and p the oversampling factor. The resulting digitised signal $r(t)$ is the input to the IC.

The IC will perform successive burst detection, demodulation, decoding, reconstruction and cancellation of each burst contained in the digitised signal $r(t)$. The signal $r(t)$ is processed in overlapping time windows. For each time window, the previous steps are iteratively implemented by the IC core module, whose functional architecture is presented in section 5.2.4.1.3.

In this section, the following terminology has been adopted:

- $r(t)$: IC input signal sampled at $F_{SAMP} = p \cdot F_{CHIP}$
- T_{BURST} : burst duration in seconds
- W: sliding window length in seconds
- N_W: sliding window length in number of burst duration ($W = N_W \cdot T_{BURST}$)
- M: sliding window length in number of window steps ΔW
- $\Delta W = W/M$: sliding window step size in seconds
- w: sliding window shift index (outer loop), ($w \geq 0$)
- N_{max}: maximum number of IC iterations (inner loop) on any given window
- n: current IC iteration counter (inner loop), $1 \leq n \leq N_{max}$
- C_n^w: number of burst detector preamble correlation local maximums above the burst detector threshold on a given window w and IC iteration n
- $r^w(t^w)$: w-th window signal ($w \geq 0$)
- $r_n^w(t^w)$: w-th window signal before the n-th iteration of the IC core module
    - By definition $r_1^w(t^w) \triangleq r^w(t^w)$
- $r_{n+1}^w(t^w)$: w-th window signal after the n-th iteration of the IC core module

5.2.4.1.2 Overall IC algorithm

The overall IC algorithm is described hereafter.

The algorithm is composed of 2 nested loops:

- The outer loop is responsible for processing the received signal $r(t)$ on a window-by-window basis, using sliding windows of length W and step ΔW. This loop is identified by the index w.
- The inner loop is in charge of detecting, demodulating, decoding, reconstructing and cancelling the bursts of the current sliding window w. This loop is identified by the index

n. The functionalities implemented in each iteration of the inner loop are those implemented by the IC core module presented in section 5.2.4.1.3.

At each window step w, $(\mathrm{w} \geq 0)$, the following burst reception and interference cancellation process takes place (outer loop):

1. The baseband signal samples corresponding to the current window step w, i.e., $\mathrm{r}^{\mathrm{w}}(\mathrm{t}^{\mathrm{w}})$, are stored in the IC detector memory (see Figure 5-27).

   The first signal segment to be processed $(\mathrm{w}=0)$ is
   - $\mathrm{r}^0(\mathrm{t}^0) = \mathrm{r}(\mathrm{t});\ \ 0 \leq \mathrm{t}^0 \leq \mathrm{W};\ \ 0 \leq \mathrm{t} \leq \mathrm{W}$

   For the w-th window step $(\mathrm{w} > 0)$, the signal to be processed is
   - $\mathrm{r}^{\mathrm{w}}(\mathrm{t}^{\mathrm{w}}) = \left[\mathrm{r}^{\mathrm{w}-1}_{\mathrm{N}_{\max}+1}(\mathrm{t}^{\mathrm{w}-1}),\ \mathrm{r}(\mathrm{t})\right];\ \ 0 \leq \mathrm{t}^{\mathrm{w}} \leq \mathrm{W}$

   where
   - $\mathrm{r}^{\mathrm{w}-1}_{\mathrm{N}_{\max}+1}(\mathrm{t}^{\mathrm{w}-1})$ is the output of the IC core module after processing the signal of the $(w-1)$-th window (after $\mathrm{N}_{\max}$ canceller iterations).
   - $\Delta\mathrm{W} \leq \mathrm{t}^{\mathrm{w}-1} \leq \mathrm{W}$
   - $\mathrm{W} + (\mathrm{w}-1)\cdot\Delta\mathrm{W} \leq \mathrm{t} \leq \mathrm{W} + \mathrm{w}\cdot\Delta\mathrm{W}$
2. The signal $\mathrm{r}^{\mathrm{w}}(\mathrm{t}^{\mathrm{w}})$, $0 \leq \mathrm{t}^{\mathrm{w}} \leq \mathrm{W}$, is input to the IC core module, which implements the following inner loop (IC iteration counter n, $1 \leq \mathrm{n} \leq \mathrm{N}_{\max}$):
   a. The burst detector will search for bursts (preamble presence) within the time range $[0,\ \mathrm{W} - \mathrm{T}_{\mathrm{BURST}}]$, obtaining the preamble correlation function.

      For each correlation function local maximum (in E_s/N_t descending order) that is above the burst detector threshold, the following steps are implemented:
      i. The time and frequency offsets estimated by the burst detector are compensated for and the resulting signal input to the payload demodulator.
      ii. The payload is demodulated and decoded.
      iii. If the packet is correct after CRC check, then:
         1. Fine data-aided channel estimation over the whole recovered burst is performed.
         2. The burst is then reconstructed at baseband for the following cancellation step.
         3. Finally, the reconstructed burst is cancelled from the input signal.

      If no local maximum above the burst detector threshold is found, the inner loop is completed and the observation window is advanced by $\Delta\mathrm{W}$.
   b. Repeat step 2.a until $\mathrm{N}_{\max}$ iterations are performed. When the limit is reached, advance the observation window by $\Delta\mathrm{W}$.

The IC sliding window management is depicted in Figure 5-27.

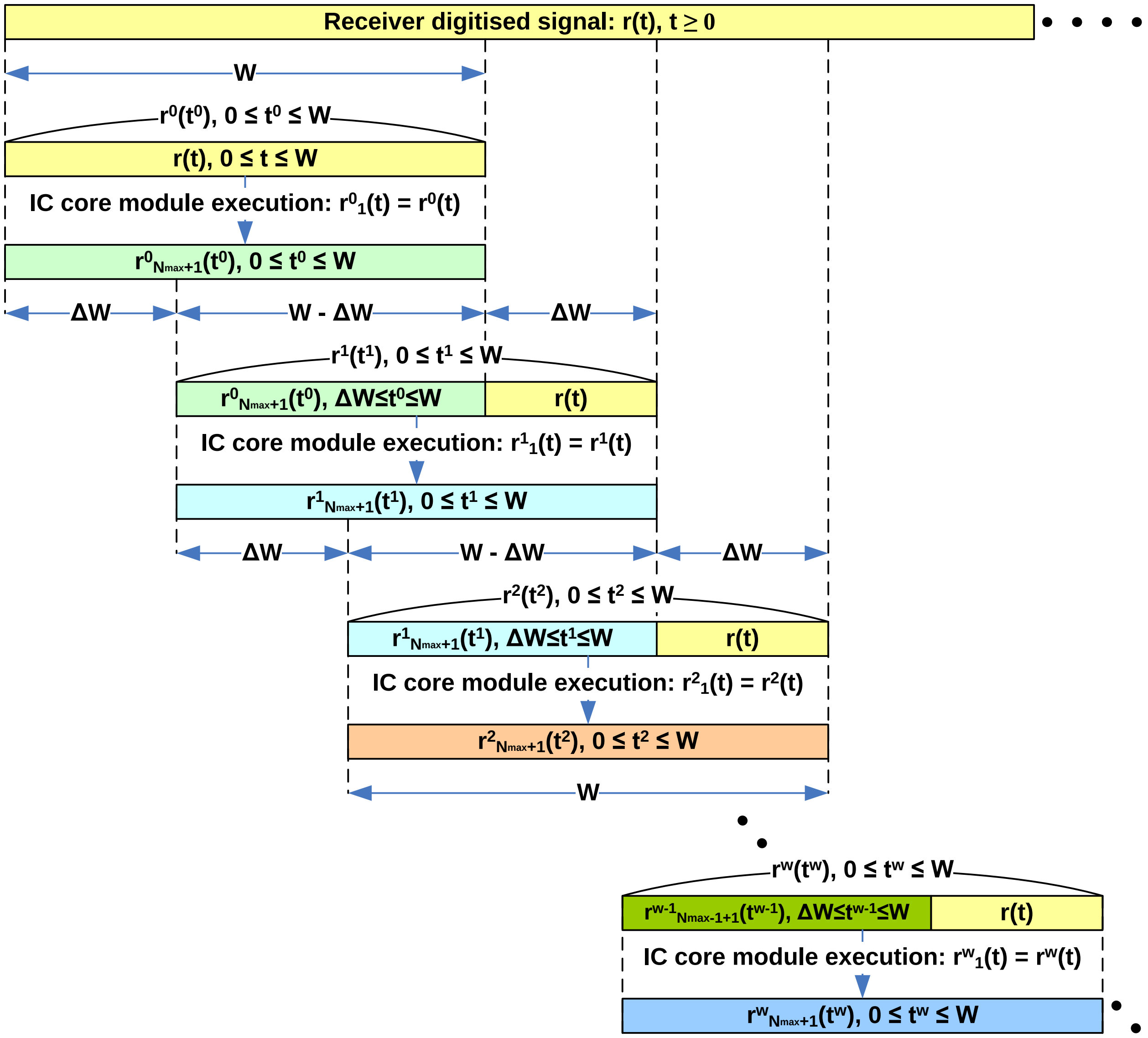


Figure 5-27: IC sliding window management

The following IC parameters are proposed:

- N_W: sliding window length in number of burst duration
  - $N_W = 3$.
- M: sliding window length in number of window steps ΔW, ($\Delta W = W/M$)
  - $M = 3, \Delta W = W/3$.
- N_{max}: maximum number of IC iterations (inner loop) on any given window
  - N_{max} from 2 to 4

5.2.4.1.3 IC core functional architecture

As explained in the previous section, the IC will perform successive burst detection, demodulation, decoding, reconstruction and cancellation of each burst contained in the received signal $r(t)$ on a window-by-window basis.

The functionalities that need to be implemented in each IC iteration are depicted in Figure 5-28, where n identifies the n-th inner loop iteration (n≥1).

This section describes the procedures that must be implemented by the IC core module to process the signal $r^w(t^w)$ of a given sliding window w. For the sake of clarity, and given that the actual window under process is not relevant, the superindex w has been removed from the signals identified in Figure 5-28. Thus, $r_n(t)$ is used instead of $r_n^w(t^w)$. In particular, $r_1(t) = r_1^w(t^w) = r^w(t^w)$. Note that the signal output $r_{n+1}(t)$ serves as input for the following IC iteration.

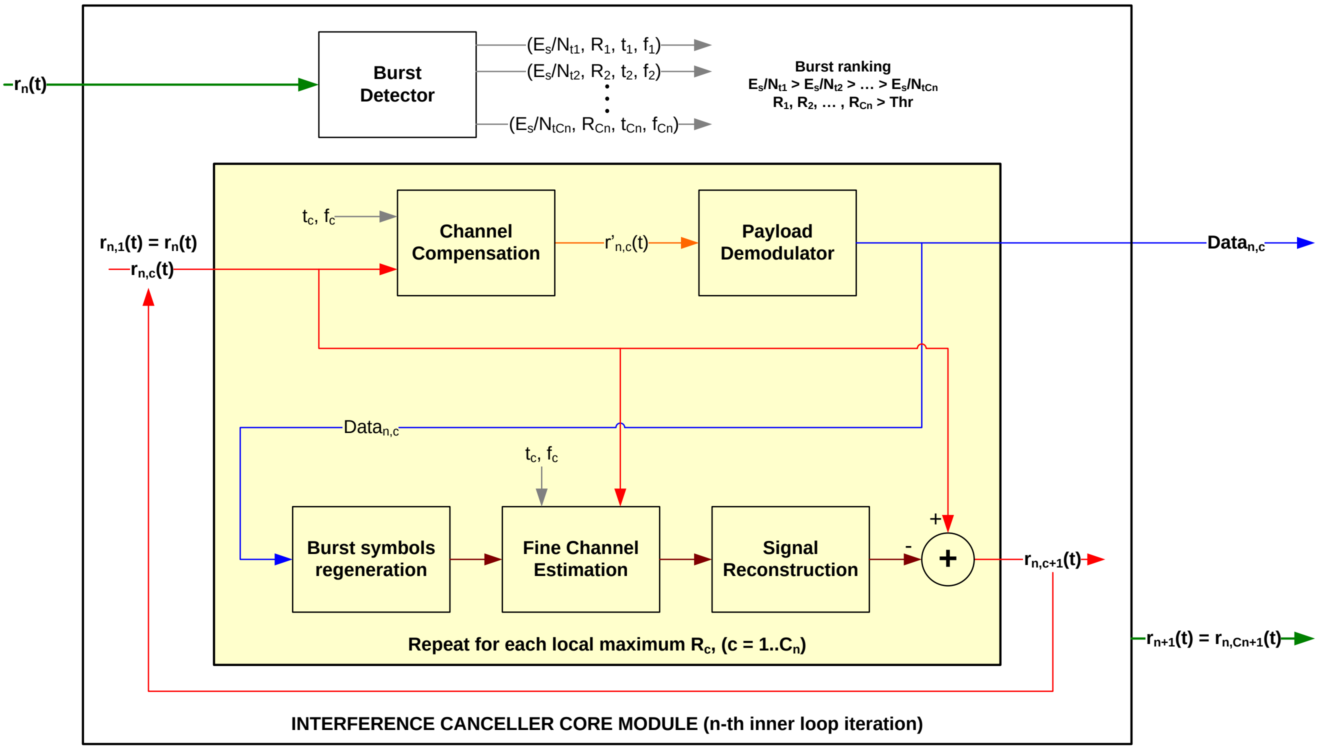


Figure 5-28: IC core module functional architecture (n-th iteration)

A brief description of each IC core module is presented below.

The <u>burst detector</u> is in charge of detecting bursts in the input signal $r_n(t)$ (by means of the detection of a preamble composed of a sequence of known symbols), which is affected by channel impairments, thermal noise (AWGN) and interference. Interference is commonly driven by that generated by the system itself. The burst detector is also responsible for performing a coarse estimation of the time and frequency offsets affecting each of the C_n detected bursts: t_c, f_c, c=1..C_n.

The detected bursts (those with preamble correlation R_c above the burst detector threshold) are then processed (by the modules in the yellow box) in E_s/N_t descending order. The first signal to be processed is $r_{n,1}(t)$ and by definition $r_{n,1}(t) \triangleq r_n(t)$.

The channel compensation is responsible for compensating for the coarse time and frequency offsets estimated by the burst detector. The resulting signal $r'_{n,c}(t)$ is delivered to the following module, i.e., payload demodulator.

The payload demodulator is in charge of demodulating and decoding the burst payload in order to recover the data. Demodulation takes advantage of the auxiliary channel, which is composed of a sequence of known symbols synchronously transmitted with the data channel and with much lower power (10 dB below). Note that if the CRC inserted in the recovered data packet is wrong, the process of the c-th burst is aborted and the process of (c+1)-th burst (the next in the ranking) is initiated with $r_{n,c+1}(t) = r_{n,c}(t)$.

The burst symbols regeneration module is responsible for generating the actual burst symbols/chips from the already known preamble and auxiliary channel sequences and the data that has just been recovered. Note that data must be encoded, interleaved and spread before adding it to the auxiliary channel and applying the complex scrambling that will provide the burst chips.

The fine channel estimation module is in charge of estimating the time, carrier phase and amplitude of the burst signal required for its reconstruction. Estimation algorithms take advantage of data-aided schemes (on preamble and data and auxiliary channels) to obtain fine parameters estimation. It is important to remark that the synchronisation offsets estimated by the burst detector and payload demodulator are not accurate enough for signal reconstruction (estimation was only aimed at detecting a burst and recovering its data).

The signal reconstruction module is responsible for regenerating the burst signal using the signal parameters estimated by the previous module. The resulting signal is subtracted from the original signal $r_{n,c}(t)$ to obtain the signal $r_{n,c+1}(t)$, which would be the input for the cancellation of the next burst in the ranking.

After processing all C_n bursts in the ranking, the signal $r_{n,Cn+1}(t)$ is obtained, which is the IC core module output signal for the n-th iteration $r_{n+1}(t)$, i.e., $r_{n+1}(t) \triangleq r_{n,Cn+1}(t)$.

5.2.4.2 Fine channel estimation

Fine time, carrier phase and amplitude estimation are required before burst signal reconstruction and cancellation. Estimation algorithms take advantage of data-aided schemes (on preamble and data and auxiliary channels) to obtain fine parameters estimation.

5.2.4.2.1 Non-coherent Digital Delay Lock Loop for time estimation

In order to obtain a fine time estimation, a non-coherent Early-Late (E-L) discriminator is proposed for the Digital Delay Locked Loop (D-DLL). The non-coherent version is proposed to make the delay and the carrier phase estimations independent. The discriminator output is modelled as

$$e_m = \frac{\left|\sum_{k=1}^{K} r_{k-\Delta/2}\right|^2 - \left|\sum_{k=1}^{K} r_{k+\Delta/2}\right|^2}{K^2}$$

where *K* represents the coherent integration time (before applying the squared absolute operator) expressed in symbol times.

The early-late correlator spacing Δ is assumed equal to half of a chip, so the early and late local signal replicas are anticipated and delayed by $\Delta/2 = T_c/4$.

Assuming a constant delay to be estimated, the equation for updating the loop is represented by $\hat{\tau}_{m+1} = \hat{\tau}_m + \gamma e_m$, i.e., a first order filtering equation. The delay estimate $\hat{\tau}_m$ is updated every integration time ($T_{up} = T_{\text{int}} = KT_s$), holding its value constant for next *K*-1 symbol times ($\hat{\varepsilon}_{k+1} = \hat{\varepsilon}_k$). Considering that $\hat{\varepsilon}_m = \tau - \hat{\tau}_m$, the updating equation for the loop can be rewritten in terms of the timing error $\hat{\varepsilon}_{m+1} = \hat{\varepsilon}_m - \gamma e_m$, linking the values of the current and the next estimation error.

The iterative equations depend on the *"step-size"* γ, which determines the amount of the correction, and so it has an impact on the speed of the algorithm to reach the steady state and the final accuracy reached in that state.

This parameter is designed taking into account: the equivalent noise bandwidth of the loop B_n, the integration/updating time and the slope of the S-curve evaluated for a zero timing error. The expression for the step-size is

$$\gamma = \frac{1}{S'(0)} \cdot \frac{4B_n T_{\text{int}}}{1 + 2B_n T_{\text{int}}}$$

in which can be seen the relation of the step-size with respect to the $B_n T_{\text{int}}$ product, keeping constant the signal and thus the S-curve shape.

The loop bandwidth is selected following the $B_n = 1/\left(2T_{eq}\right)$, where T_{eq} is the estimation time of an equivalent "open loop" timing estimator.

The proposed coherent integration time is $K=N_{int}$ = 64 symbols ($T_{int} = N_{int}/R_s$, R_s being the symbol rate).

The proposed estimation time is $T_{eq} = N_{step}/ R_s$, R_s being the symbol rate and N_{step} = 256 symbols.

5.2.4.2.2 Joint amplitude and phase estimation

To perform joint fine amplitude and phase estimation, the use of an averaging window algorithm is proposed, derived from the ML estimation theory.

The estimation algorithm is described as follows:

$$\hat{B}_k = \frac{\sum_{m=k-N+1}^{k} r_m}{N}$$

$$\hat{A}_k = \mathrm{abs}(\hat{B}_k), \quad \hat{\vartheta}_k = \arg(\hat{B}_k)$$

in which:

- $\hat{B}_k$ is the estimator of the complex channel coefficient;
- N is the length of the averaging window, in terms of symbol times;
- $\hat{A}_k$ is the amplitude estimator, derived as the amplitude of $\hat{B}_k$;
- $\hat{\vartheta}_k$ is the phase estimator, derived as the phase of $\hat{B}_k$.

The proposed averaging window length is N = 128 and 96 symbols for SF 4 and 16.

5.2.4.3 SIC performances

SIC performances have been obtained with bursts affected by frequency errors uniformly distributed within the maximum range:

- Frequency acquisition range: ±5.8 kHz.
- Frequency drift (residual Doppler rate after UT transmitter compensation): ±50 Hz/s.

The transmitter and receiver phase noise masks defined in section 5.2.3.2.2 have been used.

A log-normal bursts power distribution at the GSE receiver and Poissonian traffic has been assumed.

The reference SIC presented in section 5.2.4.1 has been assessed. The SIC configuration used is summarised below:

- $\mathrm{N_W}$: sliding window length in number of burst duration
  - $\mathrm{N_W} = 3.$
- M: sliding window length in number of window steps $\Delta\mathrm{W}$, ($\Delta\mathrm{W} = \mathrm{W}/\mathrm{M}$)
  - $\mathrm{M} = 3,\ \Delta\mathrm{W} = \mathrm{W}/3.$
- $\mathrm{N_{max}}$: maximum number of IC iterations (inner loop) on any given window
  - $\mathrm{N_{max}}$ = 3.
- Fine channel estimation
  - DLL coherent integration time: N_{int} = 64 symbols.
  - DLL equivalent “open-loop” estimation time N_{step} = 256 symbols.
  - Complex amplitude averaging window: N_{avg} = 128 and 96 symbols for SF 4 and 16, respectively.

Overall SIC performances are presented below in terms of:

- Overall SIC: Packet Loss Rate vs. offered traffic
- Signal reconstruction and cancellation: IC efficiency
  - $\eta_{IC} = 1 - \frac{P_{res}}{P_{burst}}$, where P_{burst} is the power of the burst to be reconstructed and cancelled and P_{res} the residual signal power after cancellation.

Results are presented for the aeronautical scenario #1 and for a mixed scenario. The mixed scenario corresponds to a scenario in which 98% of the bursts are received through an aeronautical scenario #1 and the rest from an aeronautical scenario #4.

The GSE receiver Es/No mean and standard deviation considered are properly indicated in the figures legend.

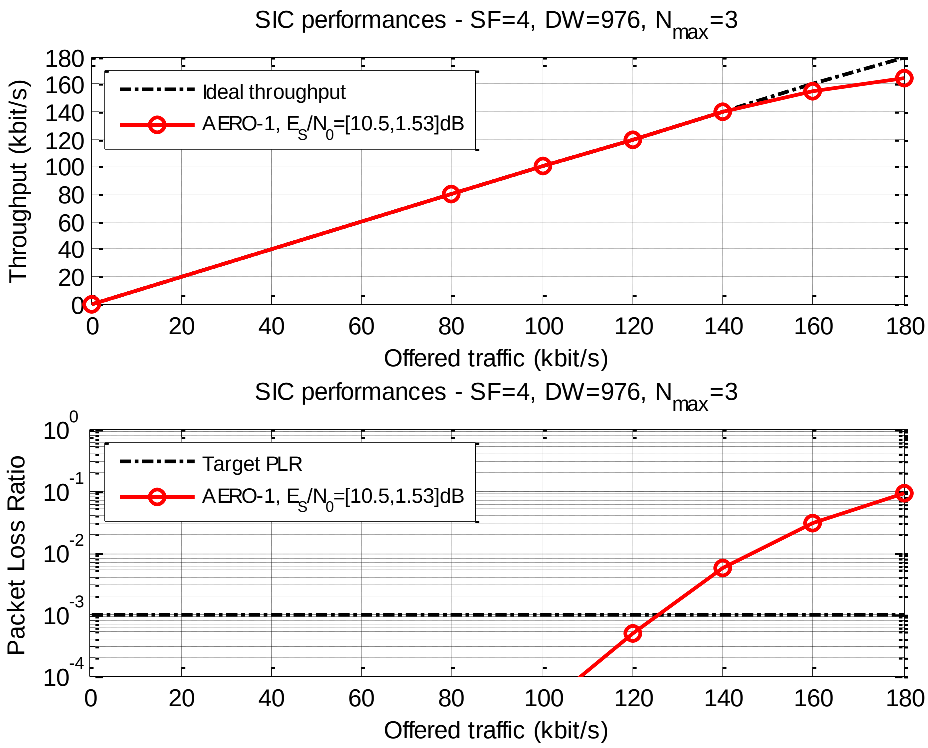


Figure 5-29: SIC performances for RACH_CR160_SF4_DB976. Scenario: 1

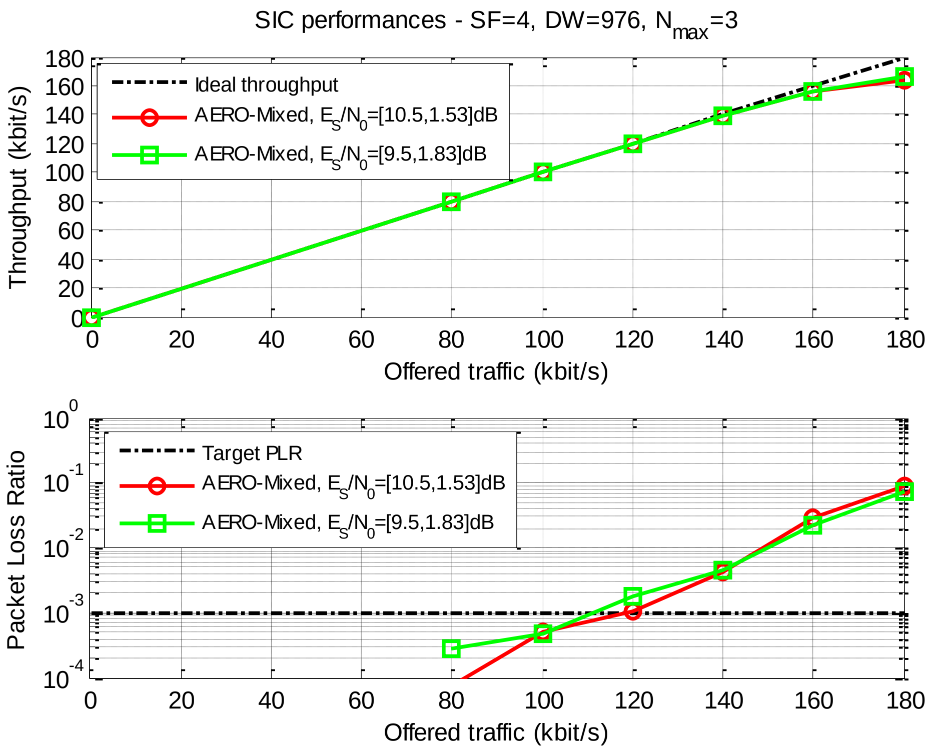


Figure 5-30: SIC performances for RACH_CR160_SF4_DB976. Scenario: Mixed

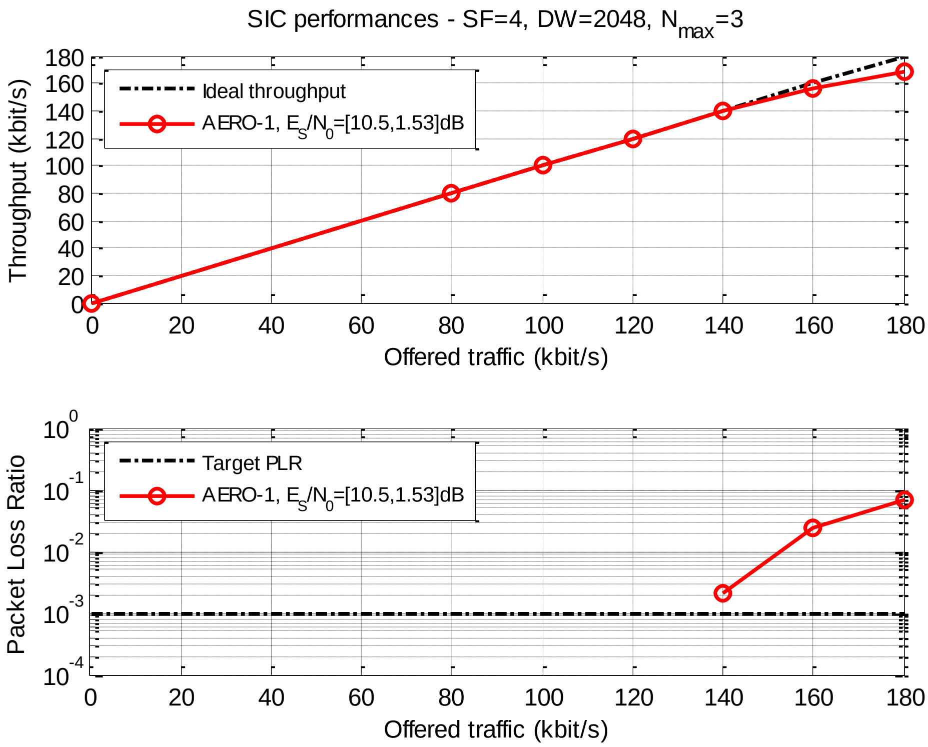


Figure 5-31: SIC performances for RACH_CR160_SF4_DB2048. Scenario: 1

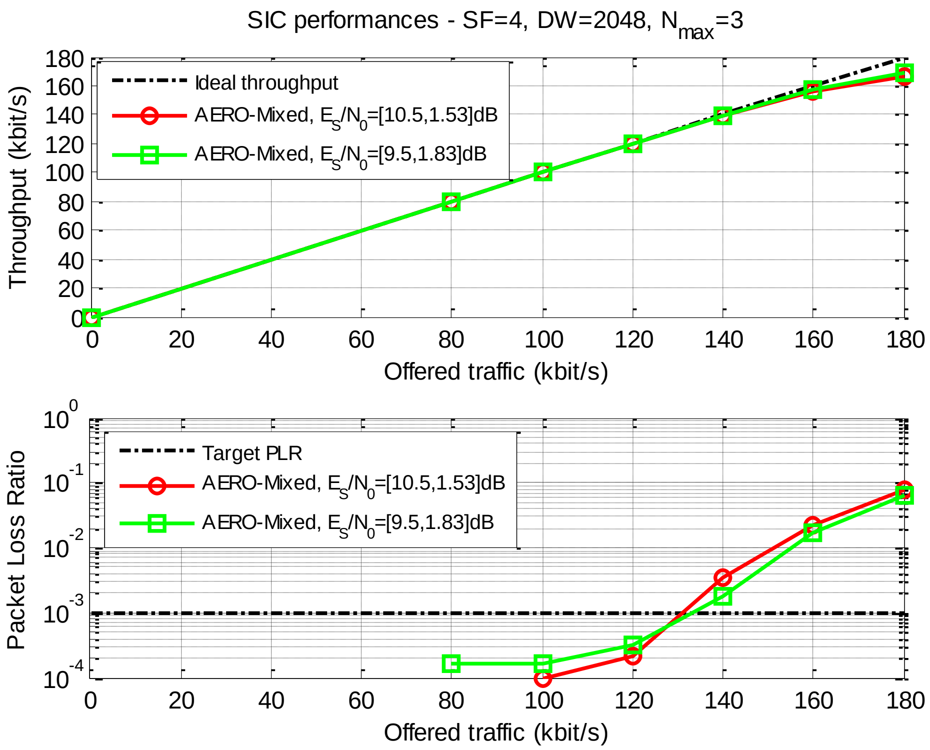


Figure 5-32: SIC performances for RACH_CR160_SF4_DB2048. Scenario: Mixed

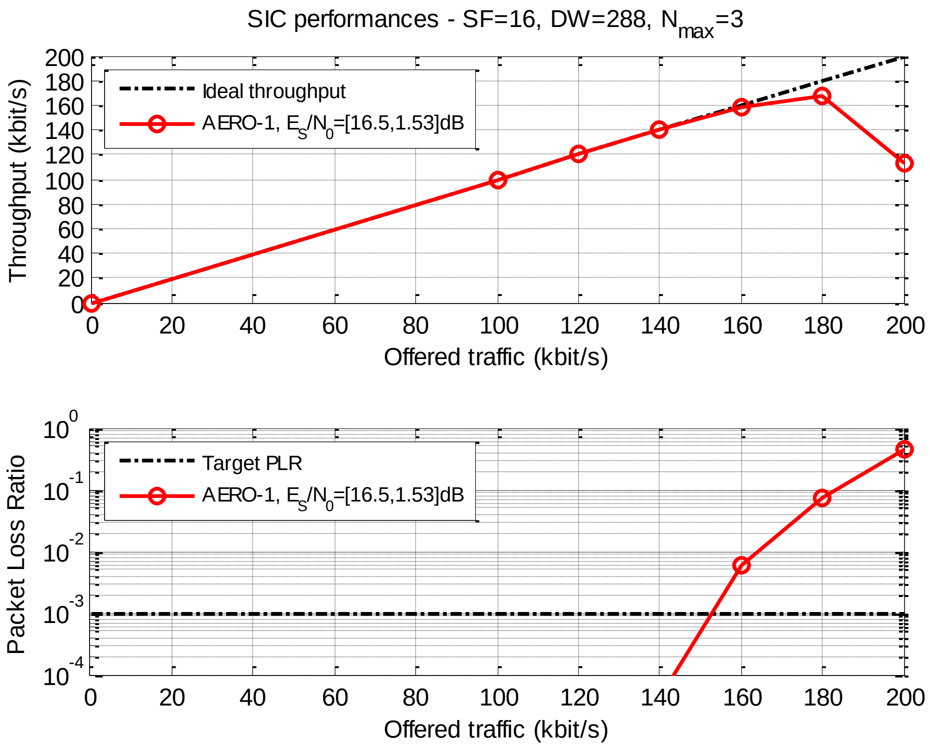


Figure 5-33: SIC performances for RACH_CR160_SF16_DB288. Scenario: 1

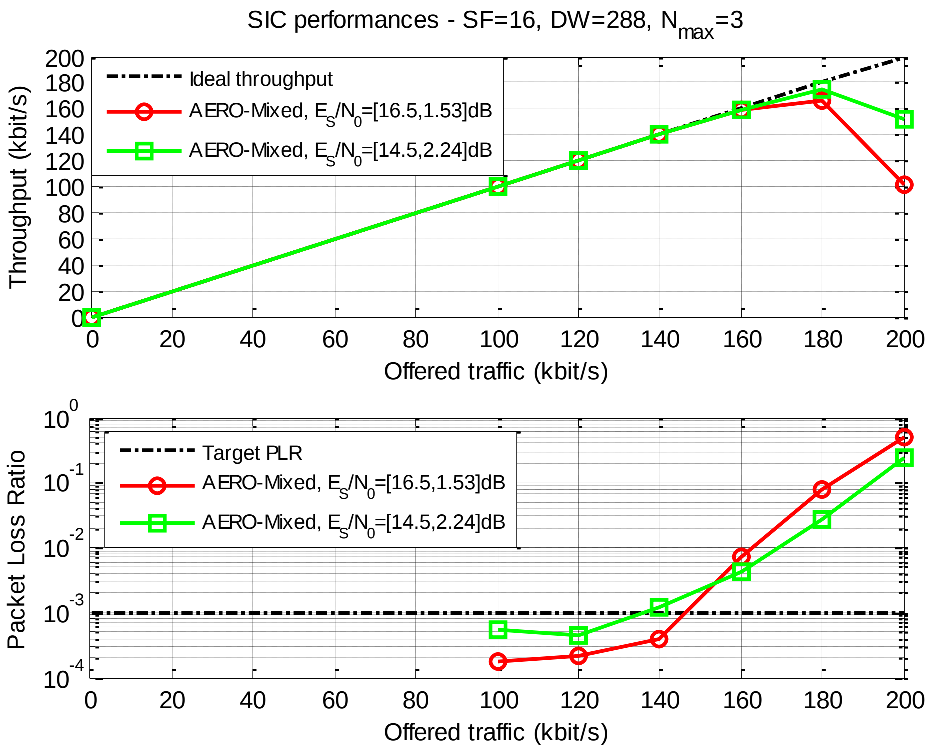


Figure 5-34: SIC performances for RACH_CR160_SF16_DB288. Scenario: Mixed

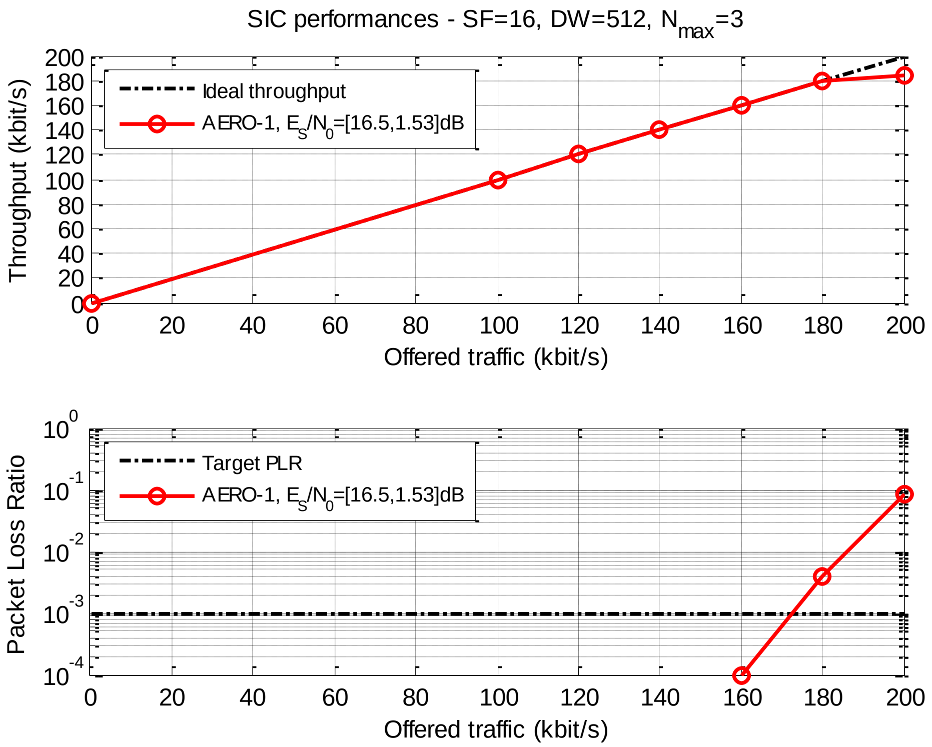


Figure 5-35: SIC performances for RACH_CR160_SF16_DB512. Scenario: 1

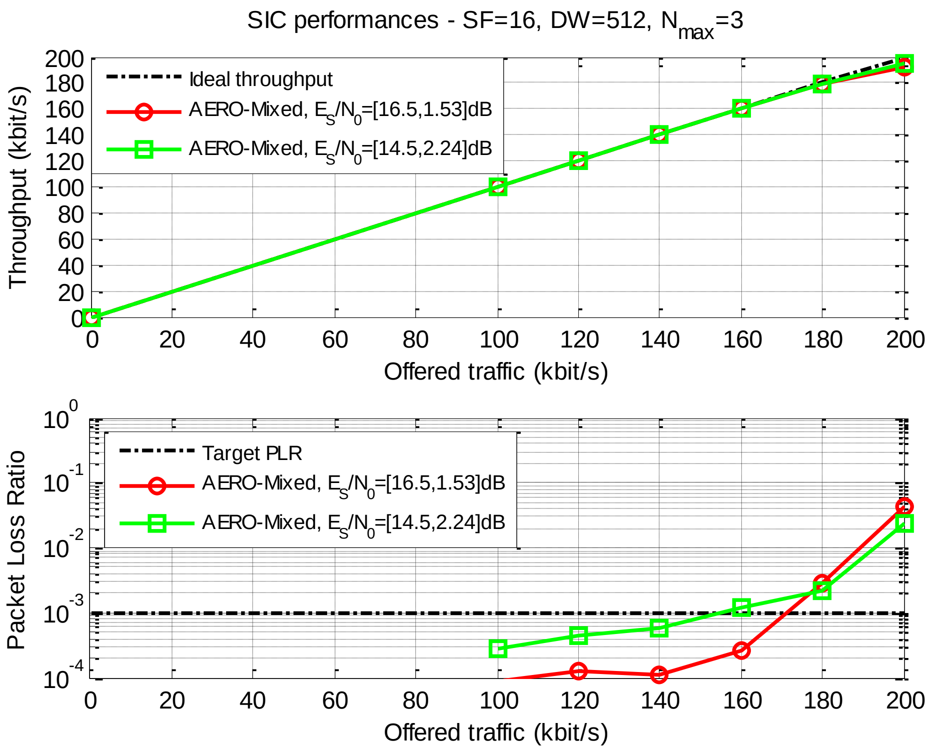


Figure 5-36: SIC performances for RACH_CR160_SF16_DB512. Scenario: Mixed

The IC efficiencies associated with the working points that provide a PLR of 10^{-3} are presented in the following table. A log-normal Es/No distribution at the GSE receiver with mean μ = 10.5 and 16.5 dB for SF 4 and 16 and standard deviation σ = 1.53 dB have been assumed.

RACH Configuration ID	Average SIC Efficiency (PLR = 10^{-3})	
	Aero Scenario #1	Aero Mixed Scenario
RACH_CR$_{160}$_SF$_4$_DB$_{976}$	95.6 %	95.5 %
RACH_CR$_{160}$_SF$_4$_DB$_{2048}$	95.5 %	95.5 %
RACH_CR$_{160}$_SF$_{16}$_DB$_{288}$	94.7 %	94.7 %
RACH_CR$_{160}$_SF$_{16}$_DB$_{512}$	94.6 %	94.6 %

Table 5-4: Average SIC efficiency

5.2.5 Noise Rise Estimation

The Multiple Access Interference (MAI) generated by RTN link A-CMDA bursts is modelled as an increment in the thermal noise, named Noise Rise (*NR*).

The Noise Rise is defined as follows:

$$NR = \frac{N+I}{N} = \frac{No+Io}{No} = 1+\sum_{i=1}^{Nc}\left.\frac{Ec}{No}\right|_i = 1+\sum_{i=1}^{Nc}\left.\frac{Es}{No}\right|_i \frac{1}{SF_i},$$

where *Ec/No* and *Es/No* are the chip and symbol energy to noise spectral density ratio, *Nc* the number of A-CDMA bursts overlapping in time and frequency, and *SF* the spreading factor.

The noise rise must be estimated for each RTN link carrier (RLC), i.e., frequency band, to assist the RTN link congestion control.

5.2.5.1 Noise rise estimator description

The reference noise rise estimator is depicted in Figure 5-37. It is composed of two branches. The upper one is devoted to the estimation of N_0 ($\widehat{N}_0$), while the lower one estimates the term $(N_0 + I_0)$, i.e., $\left(\widehat{N}_0 + \widehat{I}_0\right)$. The algorithm processes one second of the received signal and delivers the estimated noise rise ($\widehat{NR}$) at such rate.

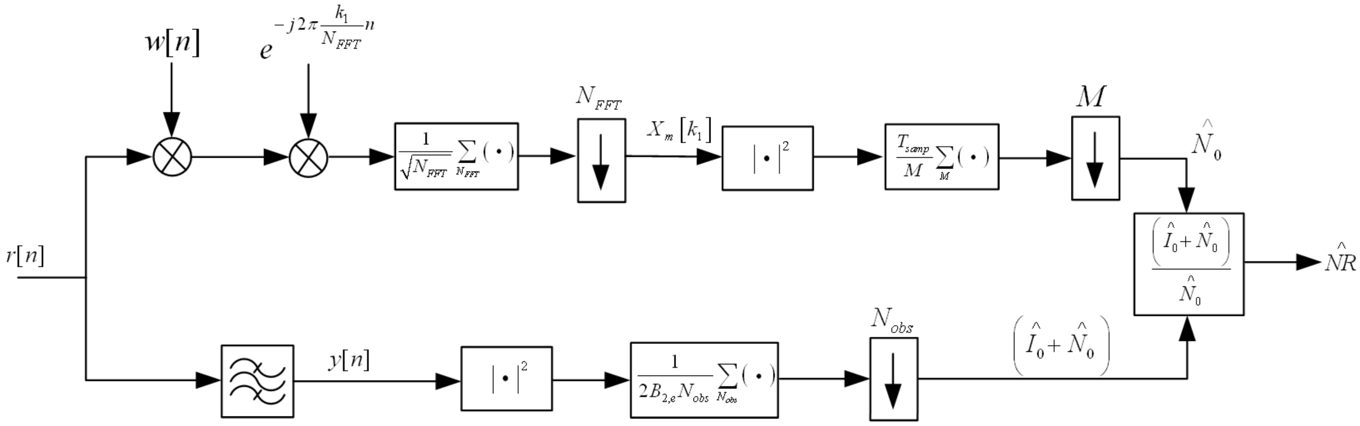


Figure 5-37: Block scheme of the noise rise estimation algorithm

5.2.5.1.1 N_0 estimation

On the upper branch the N_0 level is estimated on a single frequency bin of the DFT (Discrete Fourier Transform), instead of a group of bins. The whole FFT computation is not necessary, because an averaged periodogram is evaluated only on a specified frequency bin centred around 100 kHz (selected by k_1 in Figure 5-37).

Keeping constant the length of the observed signal $r[n]$ at 1 second (i.e., $N_{obs} = 640000$ samples considering a sampling rate of 4 samples per chip), it is split into $M = N_{obs}/N_{FFT}$ pieces, each one labelled as $x_m[n]$. For each m-th piece, the DFT coefficient located at the frequency bin k_1 is computed:

$$X_m[k_1] = \frac{1}{\sqrt{N_{FFT}}} \sum_{n=0}^{N_{FFT}-1} x_m[n] w[n] e^{-j2\pi\frac{k_1}{N_{FFT}}n}$$

where $w[n]$ is a Hamming window function.

After computing all the $X_m[k_1]$ values, we obtain a sequence of complex samples having zero-mean and variance N_0. Thus, the N_0 estimate $\widehat{N}_0$ is found computing the variance on the $X_m[k_1]$ sequence:

$$\hat{N}_0 = \frac{T_{samp}}{M} \sum_{m=0}^{M-1} \left| X_m[k_1] \right|^2$$

Since the N_0 level changes slowly with time, the N_0 estimates $\hat{N}_0$ can be conveniently smoothed on a longer observation time through this simple operation:

$$\hat{N}_{0,L} = \alpha_0 \hat{N}_{0,(L-1)} + \alpha_1 \hat{N}_0$$

where the $\hat{N}_{0,(L-1)}$ is the previous smoothed estimate (i.e., computed on the previous 1 second of signal).

The previous algorithm can be further improved by taking advantage of the symmetric frequency bin, i.e., bin around -100 kHz.

5.2.5.1.2 (N_0 + I_0) estimation

Regarding the estimation of *(N_0 + I_0)*, it is done by the use of a low pass filter with bandwidth below $B_2 = \left(\frac{1-\alpha}{2 \cdot T_c} - |f_D|_{max}\right)$. This bandwidth is selected in order to assure that the filtered spectral density of any received bursts, even of those with the highest Doppler shifts ($|f_D|_{max}$), is flat. The power at the filter output will be:

$$P_y = 2B_{2,e}I_0 + 2B_{2,e}N_0$$

where $B_{2,e}$ is the noise equivalent band of the particular filter in use, and P_y is computed on the N_{obs} available samples:

$$P_y = \frac{1}{N_{obs}} \sum_{n=0}^{N_{obs}-1} |y[n]|^2$$

The desired estimate is then computed as:

$$\left(\hat{I}_0 + \hat{N}_0\right) = \frac{P_y}{2B_{2,e}}$$

Four low pass filters are shown in Figure 5-38 to implement the *(N_0 + I_0)* estimation. Performances reported in 5.2.5.2 are presented for these filters. IIR filters are recommended in terms of performances and complexity.

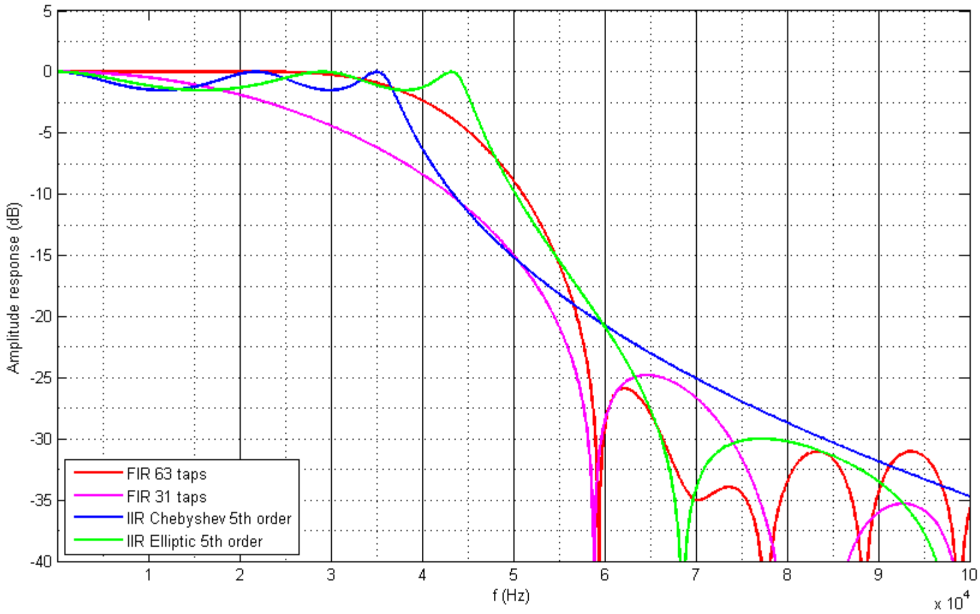


Figure 5-38: Low pass filter for B_2 = 60 kHz

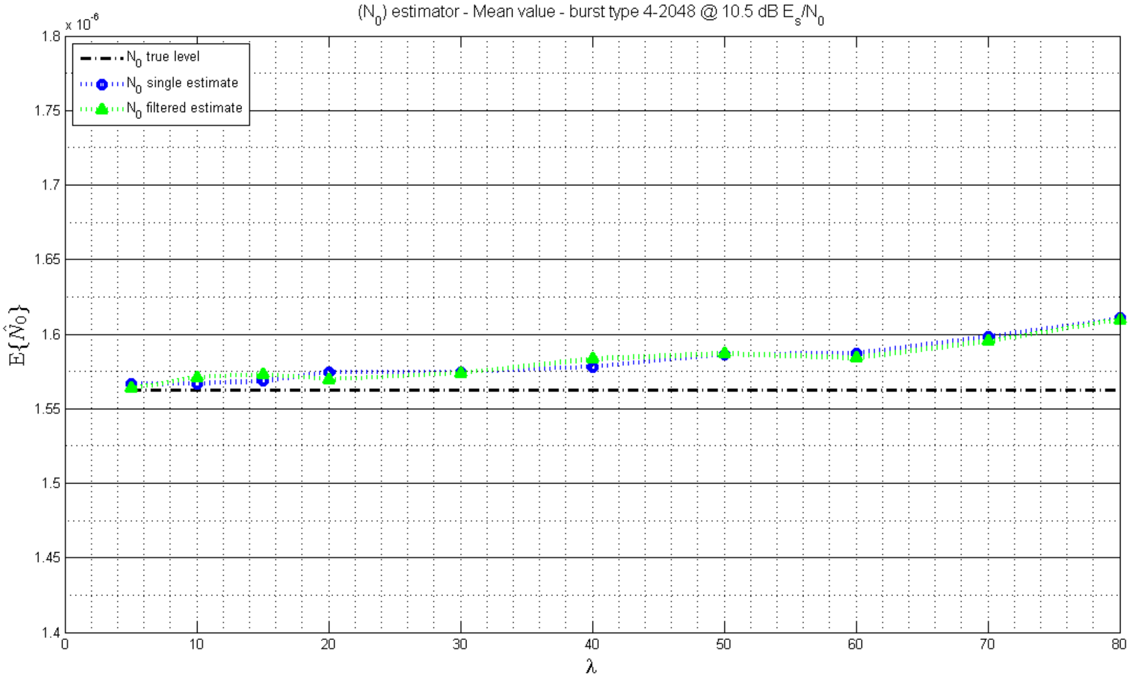
5.2.5.2 Noise rise estimator performances

The noise rise estimator performances are presented below. They have been obtained with the following configuration:

- Observation window: 1 second
- Input signal sampling rate: ${}^{4}/{}_{T_c}$
- Equivalent FFT length: $N_{FFT} = 1024$
- Frequency bin: $k_1 = 159$
- N_0 estimates smoothing: $\alpha_0 = 0.9, \alpha_1 = 0.1$
- Low pass filters presented in Figure 5-38
- Traffic generated with bursts of type $RACH\_CR_{160}\_SF_4\_DB_{2048}$ with E_s/N_t = 10.5 dB

The mean and normalised RMSE of the N_0, (N_0+I_0) and *NR* estimates are shown below for different traffic loads. The x-axis λ corresponds to the traffic intensity in terms of bursts arrivals per second.

Results for N_0 and *NR* estimates are presented with and without theN_0 smoothing, although smoothing must be enabled in nominal operation.



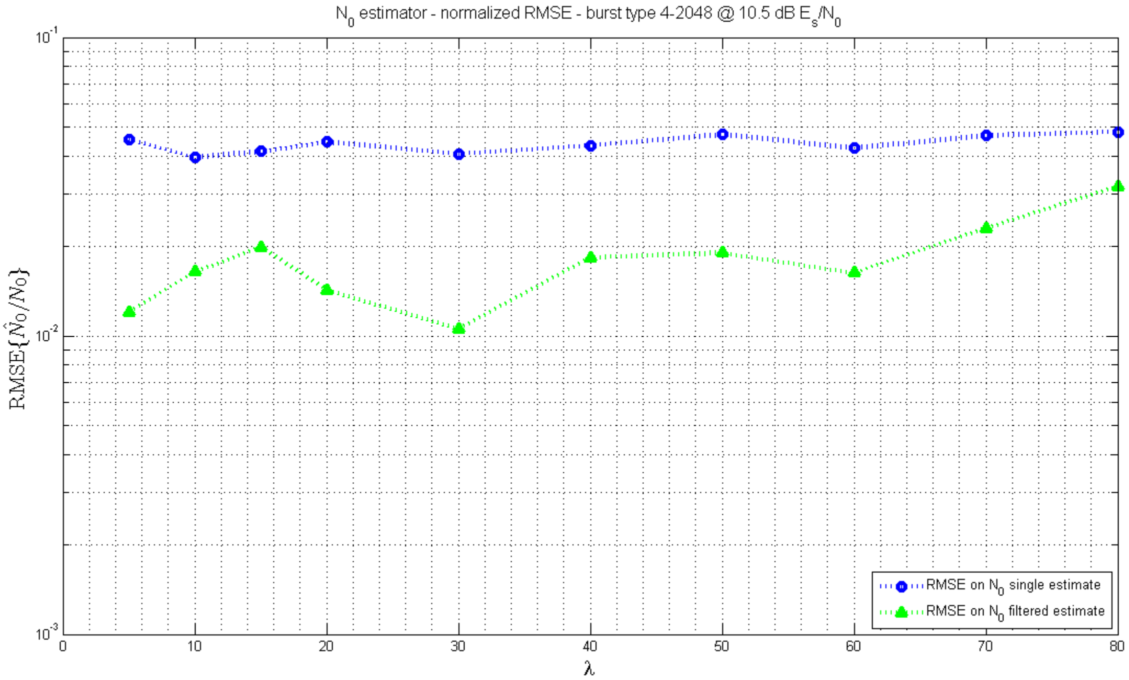
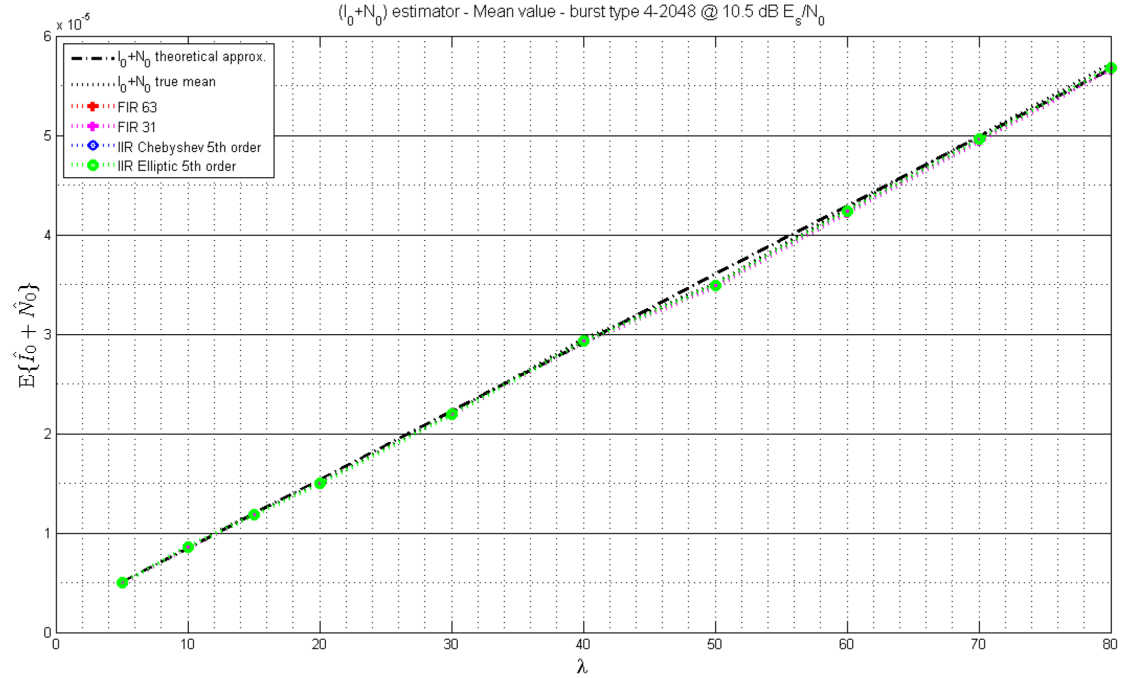


Figure 5-39: No estimation



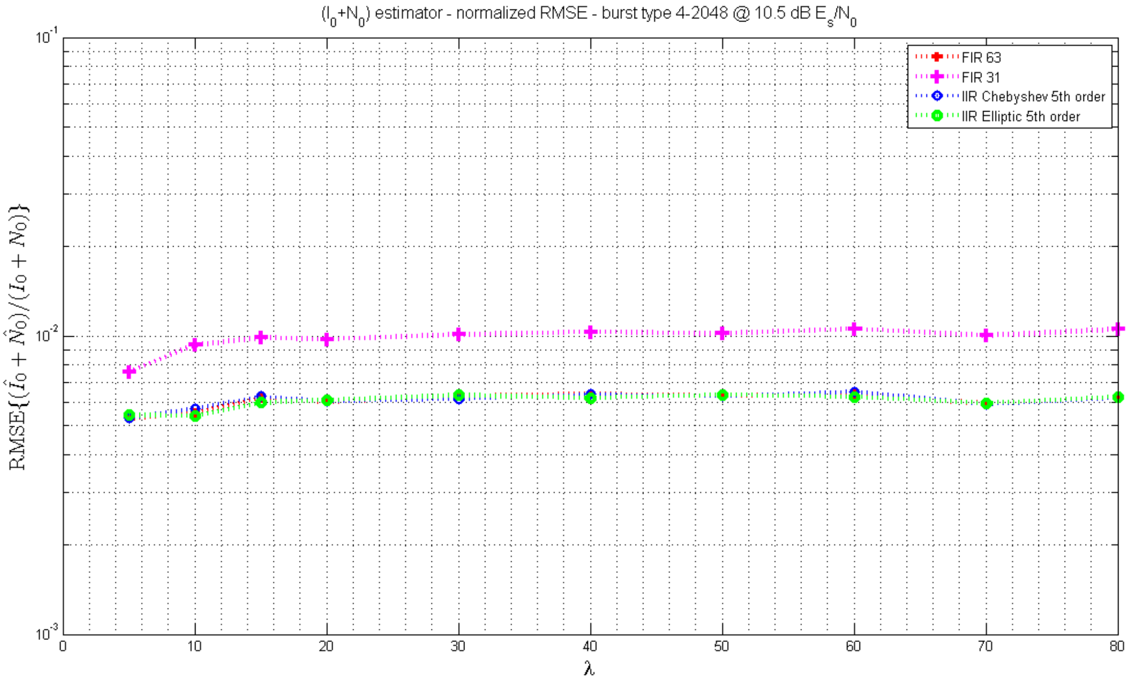


Figure 5-40: (Io+No) estimation

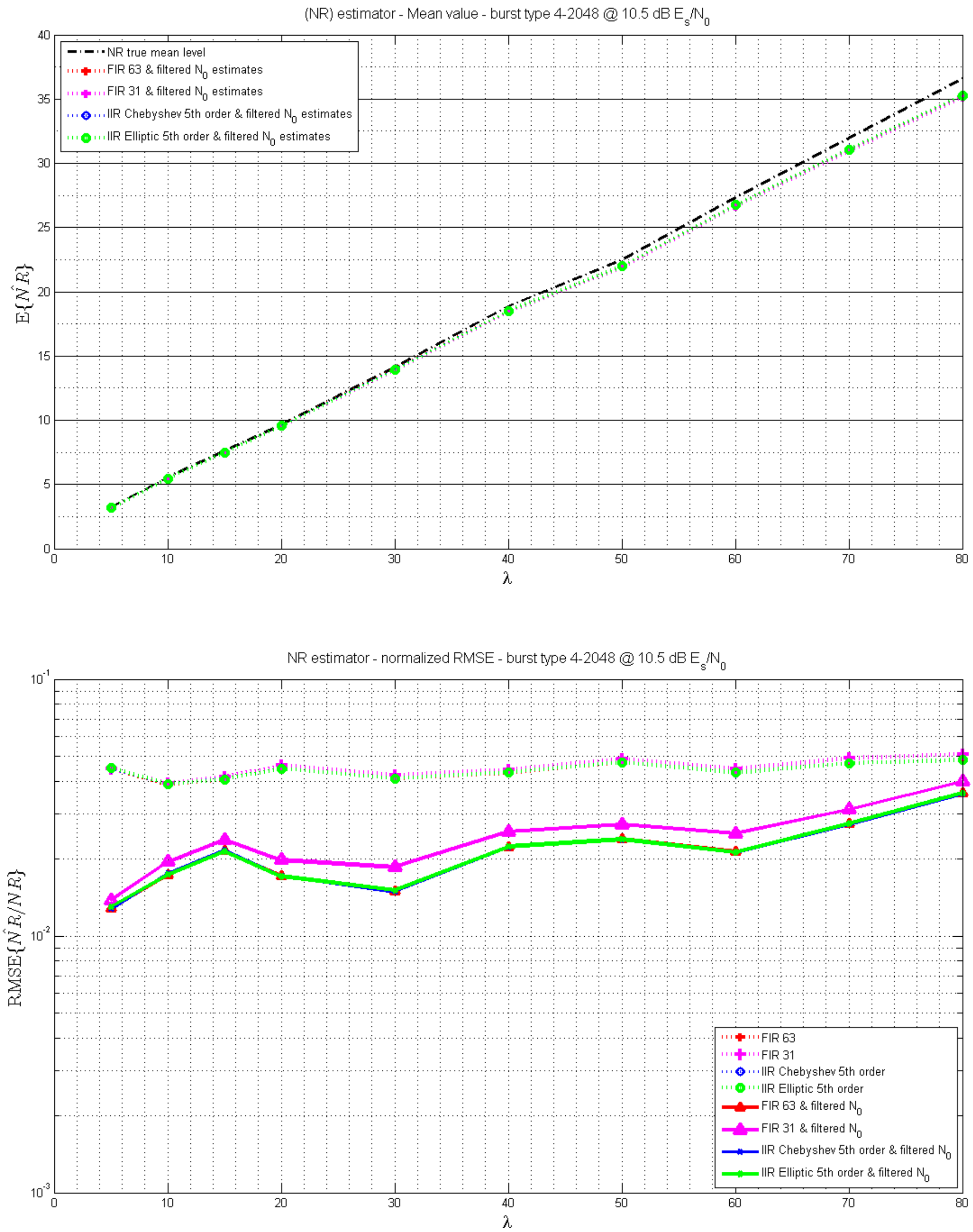


Figure 5-41: NR estimation

6. NETWORK SYNCHRONISATION

6.1 Frequency and time error budget

This section presents the synchronisation error budget for both forward and return links.

Table 6-1 compiles the values assumed for the computation of the frequency and timing error budget. In general, worst case values, i.e. highest frequencies, clock instabilities, speeds, accelerations and so on, have been selected in order to perform a conservative approach to the synchronisation error budget computation.

It is worth mentioning that GEO, HEO and MEO constellations have been taken into account for fixed-wing aircrafts. In addition, for GEO constellations, ECAC area (5 user beams) and Visual Earth coverage (global beam[5]) scenarios have been considered. Since rotary-wing aircrafts experience lower speeds and accelerations, they have only been taken into account for a GEO constellation in the ECAC area.

Input parameters	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
GS uplink frequency	31	31	31	6.175	31	GHz
GS downlink frequency	20	20	20	3.95	20	GHz
UT uplink frequency	1.65	1.65	1.65	1.65	1.65	GHz
UT downlink frequency	1.55	1.55	1.55	1.55	1.55	GHz
FWD link symbol frequency	160	16	160	160	160	kbaud
RTN link chip frequency	160	160	160	160	160	kchip/s
RTN link symbol frequency	10, 40	10, 40	10, 40	10, 40	10, 40	kbaud
UT local oscillator long term stability	1	1	1	1	1	ppm/year
GS local oscillator long term stability	0.01	0.01	0.01	0.01	0.01	ppm/year
Satellite local oscillator long term stability	0.05	0.05	0.05	0.05	0.05	ppm over 15 years
UT speed	850	850	850	850	87.5	m/s
GS speed	0	0	0	0	0	m/s
Satellite speed	3	3	5600	3090	3	m/s
UT acceleration - linear movement	50	50	50	50	2.5	m/s^2
GS acceleration	0	0	0	0	0	m/s^2
Satellite acceleration	0	0	1.023	0.43	0	m/s^2
UT roll - angular velocity	3.33	3.33	3.33	3.33	3.33	º/s
UT pitch - angular velocity	1.67	1.67	1.67	1.67	0.5	º/s
UT yaw - angular velocity	2	2	2	2	3.33	º/s

Table 6-1: Synchronisation errors sources

The resulting forward and return link synchronisation errors are presented in Table 6-2 and Table 6-3 whereas Table 6-4, Table 6-5, Table 6-6 and Table 6-7 show the contribution of each source of error to the final synchronisation errors. It must be noted that frequency and timing errors have been computed as the addition of the maximum absolute values of all contributions involved. This corresponds to a worst case situation.

[5] The use of the FWD link low rate waveform (16 kbaud) has been assumed for the Visual Earth coverage.

FWD link error budget (GS-UT)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
Aggregated frequency offset	8049.7	8049.7	615324	85802	4110.1	Hz
Aggregated frequency drift	326.5	326.5	437.5	337.6	29.5	Hz/s
Aggregated time drift	3.863	3.863	41.177	24.443	1.322	us/s
Aggregated time drift variation	0.211	0.211	0.217	0.214	0.019	us/s^2

Table 6-2: Forward link error budget

RTN link error budget (UT-GS)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
Aggregated frequency offset	7659.0	7659.0	411576	64160	3465.3	Hz
Aggregated frequency drift	347.6	347.6	421.4	355.6	31.4	Hz/s
Aggregated time drift	3.863	3.863	41.177	24.443	1.322	us/s
Aggregated time drift variation	0.211	0.211	0.217	0.214	0.019	us/s^2

Table 6-3: Return link error budget

FWD link frequency errors (GS-UT)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
GS clock instability	310	310	310	61.75	310	Hz
Doppler uplink caused by the GS	0	0	0	0	0	Hz
Doppler uplink caused by the satellite	310	310	578666.7	63602.5	310	Hz
Satellite translation error	1472.5	1472.5	1472.5	231.25	1472.5	Hz
Doppler downlink caused by the satellite	15.5	15.5	28933.3	15965	15.5	Hz
Doppler downlink caused by the UT	4391.7	4391.7	4391.7	4391.7	452.1	Hz
UT clock instability	1550	1550	1550	1550	1550	Hz
Aggregated frequency offset	**8049.7**	**8049.7**	**615324**	**85802**	**4110.1**	**Hz**
Frequency drift (UT acceleration)	326.5	326.5	326.5	326.5	29.5	Hz/s
Frequency drift (Satellite acceleration - UL)	0.0	0.0	105.7	8.9	0.0	Hz/s
Frequency drift (Satellite acceleration - DL)	0.0	0.0	5.3	2.2	0.0	Hz/s
Frequency drift (GS acceleration)	0.0	0.0	0.0	0.0	0.0	Hz/s
Aggregated frequency drift	**326.5**	**326.5**	**437.5**	**337.6**	**29.5**	**Hz/s**

Table 6-4: Forward link frequency error budget

RTN link frequency errors (UT-GS)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
GS clock instability	200	200	200	39.5	200	Hz
Doppler downlink caused by the GS	0	0	0	0	0	Hz
Doppler downlink caused by the satellite	200	200	373333.3	40685	200	Hz
Satellite translation error	917.5	917.5	917.5	115	917.5	Hz
Doppler uplink caused by the satellite	16.5	16.5	30800	16995	16.5	Hz
Doppler uplink caused by the UT	4675.0	4675.0	4675.0	4675.0	481.3	Hz
UT clock instability	1650	1650	1650	1650	1650	Hz
Aggregated frequency offset	**7659.0**	**7659.0**	**411576**	**64160**	**3465.3**	**Hz**
Frequency drift (UT acceleration)	347.6	347.6	347.6	347.6	31.4	Hz/s
Frequency drift (Satellite acceleration - UL)	0.0	0.0	5.6	2.4	0.0	Hz/s
Frequency drift (Satellite acceleration - DL)	0.0	0.0	68.2	5.7	0.0	Hz/s
Frequency drift (GS acceleration)	0.0	0.0	0.0	0.0	0.0	Hz/s
Aggregated frequency drift	**347.6**	**347.6**	**421.4**	**355.6**	**31.4**	**Hz/s**

Table 6-5: Return link frequency error budget

FWD links timing errors (GS-UT)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
GS clock instability	0.01	0.01	0.01	0.01	0.01	us/s
Doppler uplink caused by the GS	0	0	0	0	0	us/s
Doppler uplink caused by the satellite	0.01	0.01	18.667	10.3	0.01	us/s
Doppler downlink caused by the satellite	0.01	0.01	18.667	10.3	0.01	us/s
Doppler downlink caused by the UT	2.833	2.833	2.833	2.833	0.292	us/s
UT clock instability	1	1	1	1	1	us/s
Aggregated time drift	**3.863**	**3.863**	**41.177**	**24.443**	**1.322**	**us/s**
Time drift variation (UT acceleration)	0.211	0.211	0.211	0.211	0.019	us/s^2
Time drift variation (Satellite acceleration - UL)	0.000	0.000	0.003	0.001	0.000	us/s^2
Time drift variation (Satellite acceleration - DL)	0.000	0.000	0.003	0.001	0.000	us/s^2
Time drift variation (GS acceleration)	0.000	0.000	0.000	0.000	0.000	us/s^2
Aggregated time drift variation	**0.211**	**0.211**	**0.217**	**0.214**	**0.019**	**us/s^2**

Table 6-6: Forward link timing error budget

RTN links timing errors (UT-GS)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
GS clock instability	0.01	0.01	0.01	0.01	0.01	us/s
Doppler downlink caused by the GS	0	0	0	0	0	us/s
Doppler downlink caused by the satellite	0.01	0.01	18.667	10.3	0.01	us/s
Doppler uplink caused by the satellite	0.01	0.01	18.667	10.3	0.01	us/s
Doppler uplink caused by the UT	2.833	2.833	2.833	2.833	0.292	us/s
UT clock instability	1	1	1	1	1	us/s
Aggregated time drift	**3.863**	**3.863**	**41.177**	**24.443**	**1.322**	**us/s**
Time drift variation (UT acceleration)	0.211	0.211	0.211	0.211	0.019	us/s^2
Time drift variation (Satellite acceleration - UL)	0.000	0.000	0.003	0.001	0.000	us/s^2
Time drift variation (Satellite acceleration - DL)	0.000	0.000	0.003	0.001	0.000	us/s^2
Time drift variation (GS acceleration)	0.000	0.000	0.000	0.000	0.000	us/s^2
Aggregated time drift variation	**0.211**	**0.211**	**0.217**	**0.214**	**0.019**	**us/s^2**

Table 6-7: Return link timing error budget

In order to minimise the high return link synchronisation errors, the implementation of a user link Doppler pre-compensation mechanism is required. Furthermore, a feeder link Doppler pre-compensation mechanism, in both forward and return links, is also required for non-GEO constellations where satellite motion is significant. Such procedures are defined in the following sections. The residual errors after implementing these procedures are presented hereafter.

Residual errorsafter compensation

The maximum residual errors after compensation are presented in Table 6-8 and Table 6-9. The following assumptions have been made for the calculations:

- Implementation of the Feeder Link Compensation procedures defined in section 6.2.1 and 6.3.1 is assumed.
- Implementation of the User Link Doppler pre-compensation defined in section 6.3.2 is assumed.
- Synchronisation error sources defined in Table 6-1.

- Satellite speed is estimated by the GSE from the satellite ephemerides with an accuracy of 10^{-3}.

FWD link error budget (GS-UT)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
Aggregated frequency offset	6572.3	6572.3	36676	22167.8	2632.7	Hz
Aggregated frequency drift	326.5	326.5	331.9	328.8	29.5	Hz/s
Aggregated time drift	3.853	3.853	22.529	14.154	1.312	us/s
Aggregated time drift variation	0.211	0.211	0.214	0.212	0.019	us/s^2

Table 6-8: Residual forward link errors after compensation

RTN link error budget (UT-GS)	GEO - ECAC / Fixed-wing	GEO - Global / Fixed-wing	HEO / Fixed-wing	MEO / Fixed-wing	GEO - ECAC / Rotary-wing	Units
Aggregated frequency offset	4498.9	4411.6	5766.3	3798.7	4463.8	Hz
Aggregated frequency drift	46.8	23.4	46.9	46.8	22.4	Hz/s
Aggregated time drift	2.500	2.447	3.247	2.262	2.479	us/s
Aggregated time drift variation	0.028	0.014	0.028	0.028	0.014	us/s^2

Table 6-9: Residual return link errors after compensation.

6.2 Forward Link

6.2.1 Ground Segment Synchronization

Forward link network synchronisation procedures are aimed at providing the mechanisms required to guarantee that transmissions in the forward link (FLC carriers) are properly synchronised, which implies that:

- Transmissions to a given frequency (carrier) do not overlap in time at the satellite front-end.
- Simultaneous transmissions (to different carriers) do not overlap in frequency at the satellite front-end.

To achieve the previous objectives, guard times and guard bands are defined to face time and frequency uncertainties.

To synchronise GS transmissions in the forward link, the distribution of a common reference is required from a central element (NCC) to all the other GS elements. Absolute time reference is distributed to GS elements by means of NCR messages.

Thus, GS elements must recover the absolute time reference before being able to transmit to the satellite.

6.2.1.1 Forward link network synchronisation procedures

There are two alternatives to implement the forward link network synchronisation:

- Through FLC carrier, or
- By means of feeder-to-feeder links.

The procedures presented below are valid for any of the previous options, which provide equivalent performances.

Forward link network synchronisation includes three main procedures:

- The NCC distributes time and frequency reference to all GS elements.
  - Absolute time reference is distributed by means of NCR messages generated as described in 6.2.1.2.3.
  - Frequency reference is distributed implicitly through the bursts transmitted by the NCC.
- Feeder link compensation
  - It is actually part of both forward and return link network synchronisation and is fully implemented by GS elements.
  - It is aimed at (i) compensating for the Feeder Link Doppler Effect, including both time and frequency, and (ii) compensating for the satellite translation error (STE) in both forward and return links.
- GS elements synchronisation
  - Time and frequency reference recovery by means of the reception of bursts transmitted by the NCC. Absolute time reference is recovered from the NCR messages as described in 6.2.1.3.2.2.
  - GS elements compensate their transmitters by means of the recovered time reference (open loop) and starts transmitting bursts. The round trip delay is estimated from the GSE and satellite positions and pre-compensated.
  - GS elements receive their own transmissions and compute frequency, time and power errors with respect to the received NCC bursts.
  - GS elements apply synchronisation (frequency and time) and power corrections (if required).
  - Synchronisation maintenance: last steps are repeated continuously while the GS elements are attached to the network.

Each of the processes performed by the GS elements is detailed in the following sections.

6.2.1.2 NCC synchronisation processes

A flow chart of NCC synchronisation procedures is shown in the following figure, where forward link network synchronisation through the Forward Link Carrier has been assumed. The actual synchronisation procedures are explained afterwards.

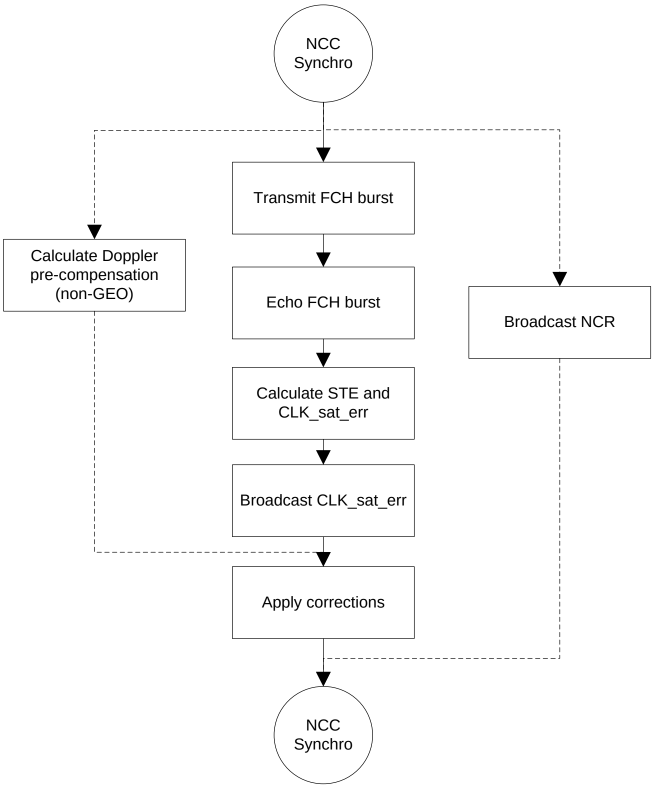


Figure 6-1: NCC synchronisation procedure

6.2.1.2.1 Feeder link Doppler pre-compensation

This procedure is required to compensate for satellite movement, being critical for non-GEO constellations and optional for GEO constellations.

The Feeder Link Compensation mechanism is fully implemented by GS elements and is aimed at:

- Compensating for the feeder uplink Doppler Effect, including both frequency and timing.
- Compensating for the feeder downlink Doppler Effect, including both frequency and timing.

The Feeder Link Compensation is implemented by all GS elements and consists in:

- Computation of Doppler and Doppler rate from
  - GS element location
  - Satellite location and speed from satellite ephemerides
  - Nominal carrier and symbol/chip frequencies
- Open-loop compensation of the feeder link Doppler and Doppler rate
  - Forward uplink: pre-compensation of the transmitter carrier and symbol frequencies and time reference
  - Return downlink: pre-compensation of the receiver carrier and chip frequencies

6.2.1.2.2 Satellite translation error (STE) compensation

6.2.1.2.2.1 STE compensation procedure

The satellite translation errors must be compensated for by all GS elements in both forward and return links with the following procedure:

- The NCC is responsible for the estimation of the satellite clock errors (CLK_{SAT}^{EST}) by means of the procedure defined in section 6.2.1.2.2.2.
- The NCC distributes to the rest of GS elements the estimated satellite clock errors.
- All GS elements (including their own NCC) compute the satellite translation error affecting their forward link transmissions and return link receptions and compensate for such errors on the transmitter and receiver carrier frequencies.
  - $STE^{FL} = CLK_{SAT}^{EST} \cdot \left(F_{GS}^{UL} - F_{UT}^{DL}\right)$
  - $STE^{RL} = CLK_{SAT}^{EST} \cdot \left(F_{UT}^{UL} - F_{GS}^{DL}\right)$

6.2.1.2.2.2 STE estimation

The NCC can estimate the satellite clock errors by measuring the satellite translation error affecting its transmissions.

A possible method for satellite translation error estimation is presented below. It is based on the reception of the forward link carrier by the NCC (reception of its own transmission). The forward link carrier frequency offset measured at the NCC receiver would be:

$$F_E^{NCC} = \frac{v_{SAT}}{c} \cdot \left(F_{NCC}^{UL} + F_{NCC}^{DL}\right) + CLK_{NCC} \cdot \left(F_{NCC}^{UL} - F_{NCC}^{DL}\right) + STE^{FL},$$

$$STE^{FL} = CLK_{SAT} \cdot \left(F_{NCC}^{UL} - F_{NCC}^{DL}\right),$$

where *STE*^{FL} is the forward link satellite translation error.

Since the Doppler Effect has been pre-compensated for in non-GEO satellites, as explained in 6.2.1.2.1, and since the effect for GEO satellites is negligible, the satellite translation error can be approximated by the frequency offset detected by the NCC from the reception of its own transmissions as follows:

$$STE_{EST}^{FL} \approx F_E^{NCC}, \qquad CLK_{SAT}^{EST} \approx \frac{F_E^{NCC}}{F_{NCC}^{UL} - F_{NCC}^{DL}}.$$

The process is performed permanently, obtaining the frequency offset detected from all forward link NCC transmissions.

6.2.1.2.3 Network time reference (NCR) distribution

The NCC broadcasts the NCR counter to provide GS elements with the absolute time reference. The NCR counter must be inserted regularly in forward link bursts as explained below.

As defined in [AD-02], the NCR specification for absolute time distribution to GS elements is shown below.

Parameters	Value	Units
NCR frequency	27	MHz
NCR counter length	40	Bits
Minimum NCR message rate	0.5	Message/s

Table 6-10: NCR specification

The NCR counter must be broadcast by the NCC to all GS elements through the FLC carrier or feeder-to-feeder links (depending on the FWD network synchronisation approach adopted). Although not strictly required, the usage of the most robust MODCOD for the bursts containing the NCR counter is recommended.

When operating in non-GEO constellations, the Feeder link Doppler pre-compensation mechanism defined in 6.2.1.2.1 must be also applied to the distributed NCR in order to compensate for the uplink Doppler Effect and minimise the timing errors of the locally recovered NCR in the rest of the GS elements. Such compensation is optional in GEO constellations.

6.2.1.3 GES synchronisation processes

The flow charts of GES initial synchronisation and GES synchronisation maintenance stages are shown in the following figures, where forward link network synchronisation through the Forward Link Carrier has been assumed. The actual synchronisation processes involved in each stage are explained afterwards.

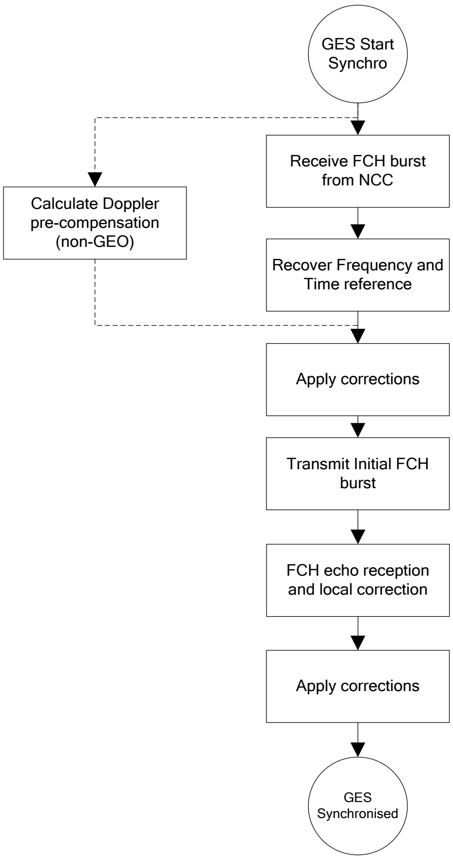


Figure 6-2: GES initial synchronisation procedure.

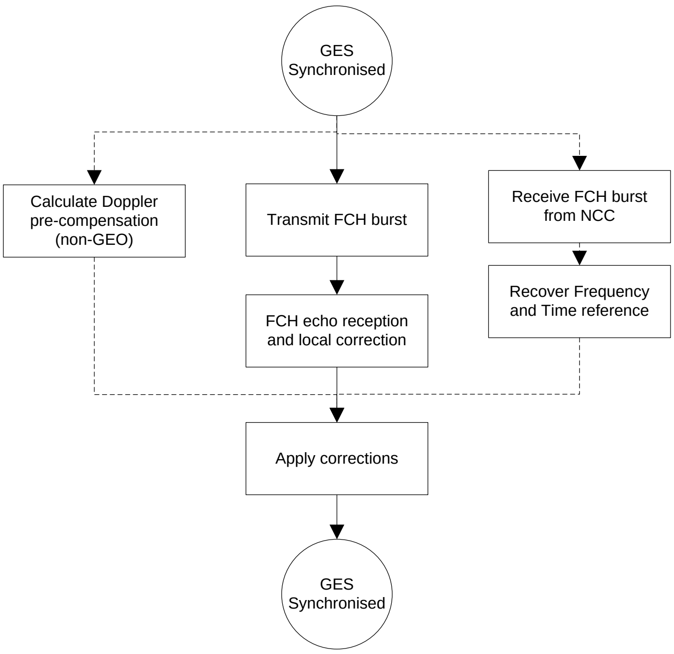


Figure 6-3: GES synchronisation maintenance procedure

6.2.1.3.1 Feeder link Doppler pre-compensation

The procedure defined in 6.2.1.2.1 must be implemented.

6.2.1.3.2 Time and frequency reference recovery

6.2.1.3.2.1 Reference recovery procedure

The GES receives forward link bursts transmitted by the NCC (used as a reference) and obtains the following physical information:

- Carrier frequency offsets: used as a frequency reference.
- Burst time estimation: time estimation from the forward link bursts containing an NCR counter value are used as input (along with the NCR counter received) to the NCR loop responsible for locally recovering the NCR to be used as time reference (see 6.2.1.3.2.2).
- Frame synchronisation by identification of starting timeslot.

  The NCR counter associated with the beginning of the first timeslot of a given frame will be distributed by the NCC through the management plane (along with the number of timeslots per frame). Since the NCR counter is an absolute non-ambiguous time reference within 11.3 hours, this information can be distributed at a very low rate. Note that the beginning of any timeslot of any frame can be derived from the knowledge of the beginning of a given frame.

Other network information, such as System Information Tables, resource allocation or the satellite clock error information, is made available at the GES before performing initial transmission.

This process is performed every time that a forward link burst from the NCC is received.

6.2.1.3.2.2 NCR recovery

Absolute time synchronisation is based on the recovery of the network time reference from the NCR time tags transmitted by the NCC through the forward link. GS elements decode these messages and locally reconstruct the network reference clock.

NCR time tags are inserted within the Extended FWD_DD header of the FCH bursts. The following figure gives the principle of the NCR synchronization (use of the FLC for the forward link synchronisation is assumed).

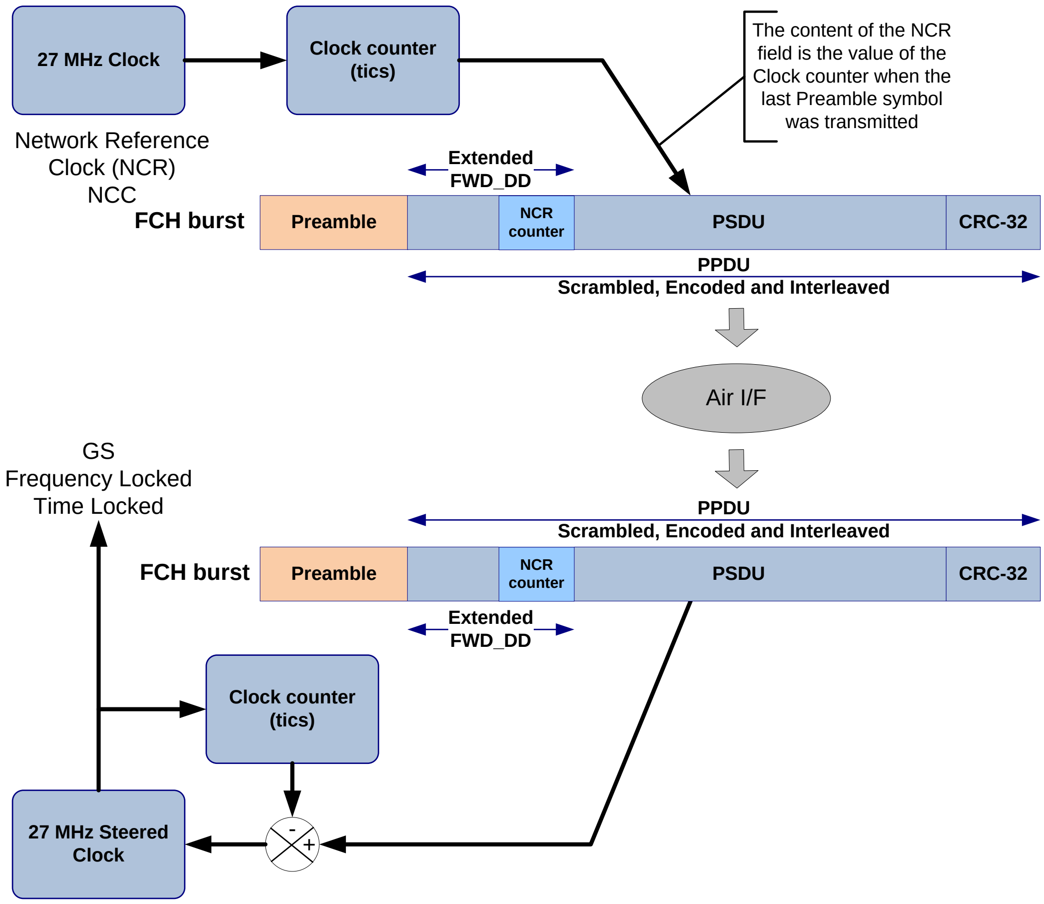


Figure 6-4: NCR distribution and GS elements recovery

The NCC (Top of the figure) uses a high quality clock reference (long term clock instability better than 0.01 ppm per year). This clock reference is followed by a counter that counts the number of clock reference periods, which is periodically inserted into the FLC carrier (within the extended FWD_DD of FCH bursts).

GS elements estimate the time of arrival of the FCH bursts containing the NCR information by means of demodulating the corresponding FCH bursts and, thanks to an internal loop (NCR loop), it locks its local clock with the NCC one. The locally recovered NCR is used by the GS elements as an absolute time reference.

6.2.1.3.3 Burst transmission

A GES cannot transmit the first forward link burst on the assigned timeslot before:

- Locally recovering the NCR and the frequency reference (see previous section).
- Obtaining the theoretical round trip delay (based on GES and satellite locations) and pre-adjusting the local absolute time reference with such a delay.
- Obtaining the satellite clock error estimated and distributed by the NCC, and computing the STE affecting its transmissions.

Once the previous information is available, the GES transmits the initial forward link burst to one of its pre-assigned timeslots:

– Using the locally recovered absolute time reference (derived from the NCR and after compensating for the expected round trip delay);

– Compensating for the transmitter carrier frequency with the estimated carrier frequency offset from NCC bursts reception. Note that, by doing this, both the NCC-to-GES clock errors and the STE are automatically compensated for.

  In non-GEO constellations, the transmitter carrier frequency is compensated for with the computed STE.

6.2.1.3.4 FWD link echo reception

The GES receives its own transmitted forward link bursts and estimates the carrier frequency and timing errors of the transmitted burst with respect to the reference transmitted by the NCC. Such synchronisation errors are used to correct its transmitters. By this means, the NCC-to-GES clock errors are compensated for and their effect on subsequent transmissions is thus negligible.

Furthermore, the transmitter power level is adjusted as well so that NCC and GES bursts are received with the same Es/No.

Note that after completing this procedure on the first forward link burst transmitted by a GES, the GES switches to the GES synchronisation maintenance stage.

While in synchronisation maintenance stage, the GES must continuously repeat the previous steps, i.e., FWD link echo loop.

If forward link network synchronisation is implemented through the FLC carrier, traffic (FCH) bursts transmitted to UTs located in the same L-band beam as the GES can be reused to estimate synchronisation and power errors.

A GES must interrupt its transmissions and go back to the GES initial synchronisation procedure if any of the following events occur:

- When no transmissions from the NCC are received for NCC_loss_timeout seconds. The recommended value is NCC_loss_timeout = 2.
- When GES_n_loss consecutive GES transmissions are not received. The recommended value is GES_n_loss = 2.
- When the time and frequency reference is not properly recovered.

6.2.1.4 Bulk handover

In a bulk handover the NCC must be pre-synchronised to a new satellite before performing the switch, transmitting simultaneously to both satellites and compensating for the differences in time and frequency offsets in the new one with respect to the old one. The NCC estimates and distributes the new satellite clock error to the GES. All GS elements coordinate the exact timeslot in which the switch is performed.

Several strategies can be adopted to implement the previous procedure, such as for example:

- Pre-synchronisation to the new satellite through a feeder-to-feeder link.
- Pre-synchronisation to the new satellite through a FLC carrier different than those used by the old satellite.

- Pre-synchronisation to the new satellite using one of the FLC carriers of the old satellite

By implementing either of the first two strategies, operation through the old satellite is not affected at all.

By contrast, if the last approach is adopted, the NCC transmits a dummy (SYNC) burst to the shared FLC carrier over the new satellite to perform the pre-synchronisation process. For this specific transmission, an extra guard time must be allocated in order to guarantee that the burst does not collide with adjacent time slots when received in any point of the coverage area due to the difference in the satellites RTT (see figure below).

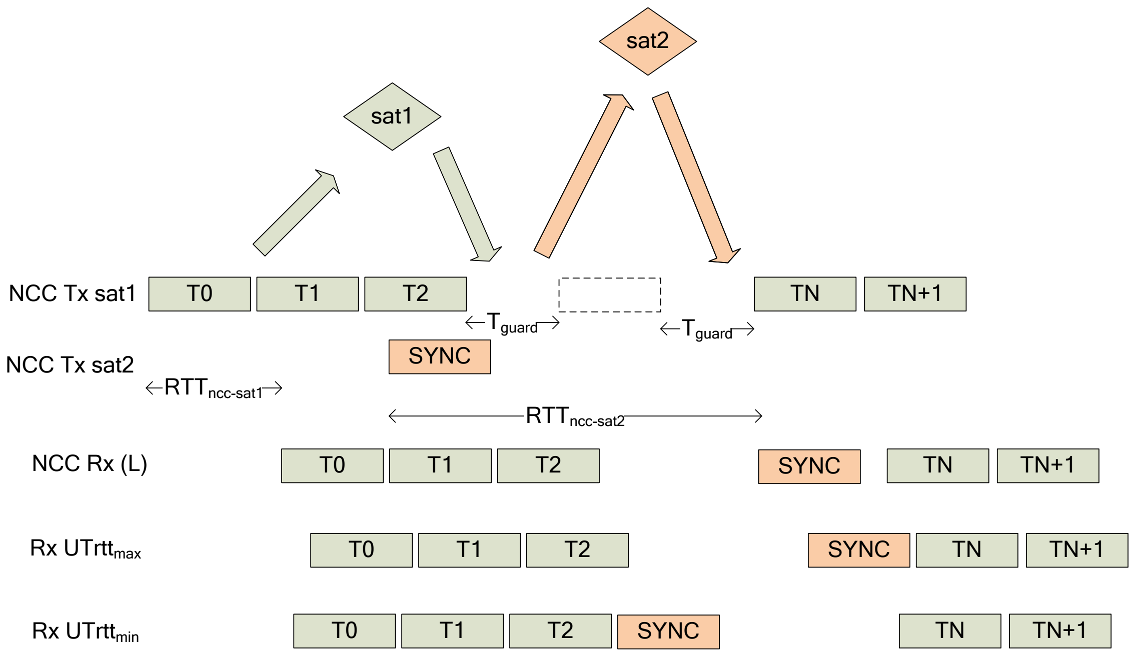


Figure 6-5: NCC pre-synchronisation to the new satellite (FLC carrier shared by both satellites)

The maximum required guard time can be approximated by twice the maximum distance between two points of the coverage area (worst case for an elevation of 0º). Some example scenarios:

- For HEO/MEO considering the coverage area of latitudes over 68 deg (Polar area), the worst case will be 32 ms.
- For MEO (Galileo-like orbits) and spot beam coverage, considering a beam spot width of 20º (half cone of 10º), which provides coverage to an area of 87.2º on the Earth surface, the guard time will be 58 ms.

For convenience, and in order not to impact the frame structure of the shared FLC carrier, the guard time should be a multiple of a timeslot. Based on the above calculations, a single timeslot will suffice.

Regardless of the GS approach adopted, upon a bulk HO the GS must signal (by means of the DPC_RST toggle bit of the FWD_DD of FCH bursts) the need to reset the Doppler pre-compensation mechanism to all UTs. This process is required because Doppler Effects depend on the relative motion between the satellite and the UT.

If GS elements have the capability of simultaneously receiving RTN link bursts from the two feeder link downlinks associated with the two satellites, after executing a bulk HO, GS elements start receiving from the new satellite and keep on receiving from the old satellite for several seconds.

Otherwise, GS elements switch the RTN link reception from the old satellite to the new one twice the average RTT after the execution of the bulk HO.

6.3 Return Link

The adopted MA scheme in the return link, A-CDMA, does not require that UTs implement elaborate synchronisation procedures before transmitting. Actually, no absolute time synchronisation is required at all and quite simple processes are needed to limit carrier and chip frequency errors.

Thus, the return link network synchronization must guarantee that UTs are always ready to transmit with limited frequency (both carrier and chip) errors. Synchronisation is maintained by a UT transmitter Doppler pre-compensation mechanism.

6.3.1 Ground Segment Synchronization

GS elements must implement the following network synchronisation procedures:

- GS elements distribute a frequency reference to all UTs through the forward link carrier
  - The frequency reference is distributed implicitly through all FLC carriers.
- Feeder Link Compensation
  - It is actually part of both forward and return link network synchronisation and is fully implemented by GS elements.
  - It is aimed at (i) compensating for the Feeder Link Doppler Effect (see 6.2.1.2.1), including both time and frequency and (ii) compensating for the satellite translation error in both forward and return links (see 6.2.1.2.2).

6.3.2 User Terminal Synchronization

The return link UT synchronisation procedure is composed of the following processes:

- UT forward link carrier reception
- UT transmitter Doppler pre-compensation

6.3.2.1 UT Forward Link Carrier Reception

This process involves the following steps:

- GS elements transmit the FLC carrier multiplexing network control and user data information in an MF-TDMA basis. There is not a system carrier specifically devoted to carrying network control information.

- The UT knows beforehand the information required to receive one or more FLC carriers (carrier frequency, symbol rate, coding scheme, etc.) and starts to tune the different frequencies, looking for available carriers.
- Reception of FLC carriers involves the following steps (note that this process is performed permanently by the UT, until it logs out of the network):
  - FLC physical synchronisation: the physical FLC synchronisation process entails the detection, synchronisation and decoding of FCH burst.
  - Network frequency reference recovery: it is based on the carrier frequency offsets estimated by the UT receiver and used to assist the UT transmitter Doppler pre-compensation mechanism in order to compensate for the Doppler and Doppler rate affecting UT transmissions in the return uplink.
  - Extraction of MF-TDMA data, i.e., system information tables.

6.3.2.2 UT transmitter Doppler pre-compensation

The UT transmitter Doppler pre-compensation mechanism is aimed at compensating for the Doppler (shift) and Doppler rate (drift) in the return uplink. It is fully implemented by UTs and includes the following steps:

- Upon reception of an FCH burst, the UT implements the following process:
  - It computes open loop transmitter carrier and chip frequencies corrections (both offsets and drifts) based on the FLC carrier frequency offsets estimated by the receiver.

    Transmitter carrier and chip frequencies corrections can be estimated with the following mechanism:

    - Input:
      - Carrier frequency offsets estimated from FCH burst reception. This process is valid even if FCH bursts are originated by different GS elements because their transitions are synchronous.
    - Output:
      - Transmitter carrier frequency corrections (offset and drift).
      - Transmitter chip frequency corrections (offset and drift).
    - Procedure:
      - Forward downlink Doppler shift and drift estimation from the carrier frequency offsets of FCH bursts.

        Receiver frequency offsets can be low-pass filtered (for instance, through averaging using a sliding window) and the result used for frequency drift estimation. The estimated frequency drifts and the averaged frequency offsets can be used to predict the forward downlink carrier frequency offset variation through the time, i.e., $F_{OFF}^{FWD}(t)$.

- Return uplink Doppler shift and drift (Doppler variation through the time) estimation from the output of the previous step, as follows:

$$Doppler_{UPLINK}^{RTN}(t) = 10^6 \cdot \left(\frac{F_{OFF}^{FWD}(t)}{F_{CARR}^{FWD}}\right) [ppm]$$

  with F_{CARR}^{FWD} the nominal forward downlink carrier frequency.

- Transmitter carrier and chip frequencies corrections computation including offsets and drifts, i.e., offsets variation through the time, as follows:

$$F_{CARR}^{TX}(t) = -\left(\frac{Doppler_{UPLINK}^{RTN}(t)}{10^6}\right) \cdot F_{CARR}^{RTN}$$

$$F_{CHIP}^{TX}(t) = -\left(\frac{Doppler_{UPLINK}^{RTN}(t)}{10^6}\right) \cdot F_{CHIP}^{RTN}$$

  F_{CARR}^{RTN} and F_{CHIP}^{RTN} being the nominal return uplink carrier frequency and return link chip frequency, respectively.

  - The UT must discard the following FCH bursts for the previous process:
    - FCH bursts not properly received (wrong CRC).
    - FCH bursts identified (with the SYNC field of the FWD_DD descriptor) by the GS as inappropriate for UT transmitter Doppler pre-compensation because they are likely to be affected by high[6] carrier frequency errors.

– The UT must continuously apply the carrier and chip frequency corrections to its transmitter.

  In particular, the UT should keep correcting the transmitter carrier and chip frequencies with the drifts computed while a RACH burst is being transmitted so that the residual frequency drift is minimised.

– The UT must reset the UT Doppler pre-compensation mechanism if the DPC_RST bit of the FWD_DD of the incoming FCH burst has changed its value (DPC_RST is used in toggle mode).

– The UT must interrupt its transmissions if any of the following abnormal events occurs:
  - The FLC is lost, i.e., no FCH bursts are received within FLC_loss_timeout seconds. The FLC_loss_timeout parameter is broadcast to the UT through the system tables. The recommended value is FLC_loss_timeout = 5.
  - The Doppler pre-compensation mechanism is not able to estimate return uplink Doppler shift and drift.

  The UT can resume transmissions once the nominal operation is reached.

[6] Higher than ±10 Hz.

The UT Synchronisation process is depicted in the following figure.

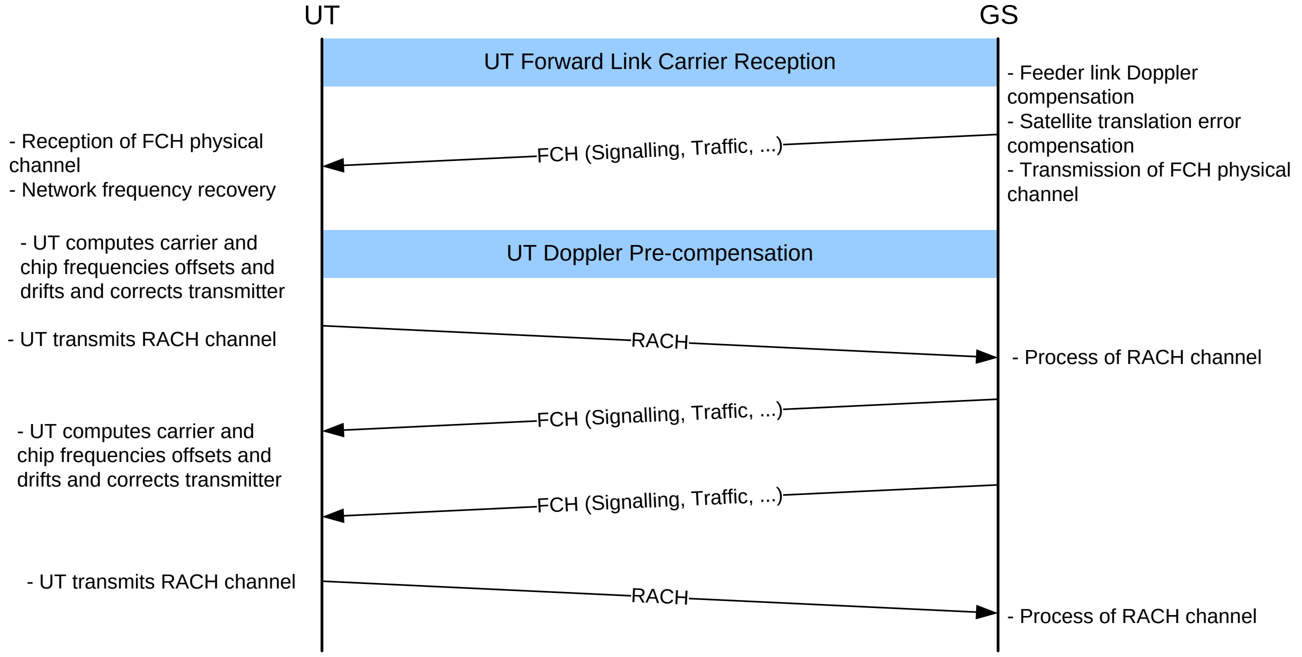


Figure 6-6: UT Synchronisation Process

During satellite HO execution process, in which non-simultaneous transmission on two satellite carriers may occur, two independent UT synchronisation processes are executed in parallel, one for each of the satellite carriers.

On a burst per burst basis, the UT adjusts the transmitter according to Doppler corrections of the corresponding satellite carrier. Thus, the UT has at least the capacity to receive two FLC carriers simultaneously, and estimate the corresponding transmitter carrier and chip corrections to pre-compensate for Doppler effects.

7. ADAPTIVE CODING AND MODULATION

Adaptive Coding and Modulation (ACM) is a Fading Mitigation Technique (FMT) which allows adaptation of the physical layer to the propagation channel variations to maximize the system spectral efficiency. The adoption of ACM techniques, in contraposition of Constant Coding and Modulation (CCM) techniques, in which only one MODCOD is used to communicate with a user terminal, results in an optimization of the radio resources usage.

In order to exploit the advantage of the ACM, it is deemed necessary to devise efficient and reliable techniques adapted to mobile environments for accurately estimating the link quality and adapting the transmission parameters in order to obtain the maximum spectral efficiency while keeping the target performance requirements (PER).

The main hurdle in implementing ACM in the aeronautical environment (and in general in any mobile environment) is that there is not a univocal relation between the Link Quality (BER/PER) and the estimated SNIR. The consequence is that the ACM mechanism cannot rely only on SNIR measurements; other indicators of the propagation channel conditions are required.

In the following sections, two possible ACM mechanisms suited to the aeronautical environment are presented. These two mechanisms are not exclusive and they can work in parallel since each one is intended to cope with different channel variations:

- Slow ACM mechanism is intended to adapt the channel coding and modulation to slow variations of the propagation channel.
- Fast ACM mechanism is designed to react against fast variation of the propagation channel (e.g., aircraft banking).

7.1 Slow ACM mechanism

As a difference of ACM control loops for static environments, which are typically based on SNIR measurements, the proposed ACM mechanism relies on the observation of the performance of the LDPC decoder (PER).

The key point of the proposed slow ACM mechanism relies on the behavior of the iterative LDPC decoder, which needs a certain number of iterations (typically around 50 although for some code rates 80 iterations could be required) to assure the decoding convergence. Assuming a target PER = 10^{-3}, at least 10^{5} code-words are needed for a reliable PER estimation, which makes the estimation unfeasible in terms of ACM loop response time. Furthermore, the longer the ACM response time, the higher the required ACM margin.

This issue can be solved if it is possible to determine a PER threshold ($PER_{Thr\_Nred\_\#N}$) at a reduced number of LDPC decoder iterations ($N_{reduced\_\#N}$) which can be used as an indication of the PER at $N_{nominal\_\#N}$ LDPC decoder iterations ($PER_{Nominal\_\#N}$). The advantages of measuring the PER at $N_{reduced\_\#N}$ LDPC decoder iterations and comparing it with the correspondent $PER_{Thr\_Nred\_\#N}$ are that:

- It is quickly measurable as $PER_{Thr\_Nred\_\#N} << PER_{Thr\_Nominal}$.
- It allows anticipating the risk of packet loss at nominal LDPC decoder iterations.

In this section, the following nomenclature is used:

- LDPC decoder iterations

  - $N_{nominal\_\#N}$: Nominal number of LDPC decoder iterations for MODCOD #N (usually 50 iterations are needed).
  - $N_{reduced\_\#N}$: Reduced number of LDPC decoder iterations for MODCOD #N.
- PER measurements
  - $PER_{Thr\_Nominal}$: PER threshold at $N_{nominal}$ LDPC decoder iterations (i.e., target PER, which is common to all the MODCODs).
  - $PER_{Nominal\_\#N}$: PER measurement at $N_{nominal}$ LDPC decoder iterations for MODCOD #N.
  - $PER_{Thr\_Red\_\#N}$: PER threshold at $N_{reduced}$ LDPC decoder iterations for MODCOD #N.
  - $PER_{Nred\_\#N}$: PER measurement at $N_{reduced}$ LDPC decoder iterations for MODCOD #N.
- MODCOD numbering: MODCODs are numbered in function of the spectral efficiency, from the least efficient to the most efficient. Thus MODCOD #k-1 is less efficient than MODCOD #k.

The slow ACM consists of two basic mechanisms: MODCOD downgrade and MODCOD upgrade. Both mechanisms are based on the concept of measuring PER at $N_{reduced}$ LDPC decoder iterations for the MODCODs present in the system.

- Mechanism for MODCOD downgrade (Slow ACM mechanism):

  In order to trigger a MODCOD downgrade, the UT continuously estimates the $PER_{Nred\_\#N}$, for its preferred MODCOD[7] and the less efficient MODCODs available in its assigned FLC carrier. Once the value crosses a given PER threshold ($PER_{Thr\_Nred\_\#N}$), the UT requests that further packets be sent using a more robust MODCOD available on the forward link. Note: PER measurement at reduced number of LDPC decoder iterations for MODCND #k ($PER_{Nred\_\#N}$) is based on a sliding window as reported in section 7.1.2.

- Mechanism for MODCOD upgrade (Slow ACM mechanism) :

  If the UT preferred MODCOD is MODCOD #k, then to trigger a MODCOD upgrade, the UT continuously monitors the $PER_{Nred\_\#N}$ for the preferred MODCOD and for the next less robust system MODCODs available[8] on the forward link (i.e., $PER_{Nred\_\#k+1}$, $PER_{Nred\_\#k+2}$, *etc*).

  The UT will request further packets be sent with the less robust MODCOD for which the PER threshold at reduced LDPC decoder iterations (i.e., $PER_{Thr\_Nred\_\#k+1}$, $PER_{Thr\_Nred\_\#k+2}$, *etc*) is crossed.

  Note: A more conservative MODCOD upgrade mechanism can used in which only upgrade to the next less robust MODCOD is allowed (progressive upgrade). In this case,

[7]Preferred MODCOD it refers to the most efficient MODCOD, in terms of spectral efficiency, the UT is able to receive while keeping the target PER.

[8] At least, the system should assure that the next less robust MODCOD (i.e., k+1) should be present in the FLC assigned to the UT.

the UT only needs to measure the $PER_{Nred\_\#N}$ for the preferred MODCOD and for the next less robust system MODCOD on the forward link (i.e., $PER_{Nred\_\#k+1}$).

It is noted that the pair [$N_{reduced\_\#N}$,$PER_{Thr\_Red\_\#N}$] are unique for a given MODCOD over the coverage due to the lack of knowledge of the UT position.

The methodology used for the definition of these values will be further detailed in the next section.

7.1.1 PER thresholds ($PER_{Thr\_Red\_\#N}$) and $N_{reduced\_\#N}$ LDPC Decoder Iterations determination

The proposed slow ACM mechanism depends on the performance of the LDPC decoder at $N_{reduced\_\#N}$ iterations. This means that the decoder has to be characterized. This section is intended to present the procedure to determine the following key parameters:

– $PER_{Thr\_Nred\_\#N}$ (PER threshold at $N_{reduced}$ LDPC decoder iterations for MODCOD #N)

– $N_{reduced}$ (Reduced number of LDPC decoder iterations for MODCOD #N)

So, for each MODCOD supported by the system, it has to be obtained the pair [$PER_{Thr\_Nred\_\#N}$, $N_{reduced}$]. Both parameters are obtained from the analysis of the PER curves in the predefined propagation scenarios.

$PER_{Thr\_Nred\_\#N}$ should be selected in such a manner that:

– should be reliably measurable in a short time period (few seconds) and

– should be able to anticipate a degradation of the PER after $N_{nominal}$ LDPC decoder iterations and prevent any packet loss.

It is worth noting that the selection of $PER_{Thr\_Nred\_\#N}$ impacts the system efficiency and the ACM loop response time. It is recommended to take $PER_{Thr\_Nred\_\#N}$ around 10^{-1} ($PER_{Nreduced\_recomended}$) for all the MODCODs as it allows a reliable PER estimation with a short monitoring period.

The procedure recommended for determining the pairs [$PER_{Thr\_Nred\_\#N}$, $N_{reduced\_\#N}$] for each MODCOD is the following:

1. For each MODCOD and for each propagation scenario, identify the $Es/N0_{target\_PER}$ at which the target PER ($PER_{Thr\_Nominal}$), e.g., 10^{-3}, is achieved at $N_{nominal}$ LDPC decoder iterations.
2. For each MODCOD and propagation scenario, identify the $N_{reduced\_\#N}$ LDPC decoder iteration at which the PER at $Es/N0_{target\_PER}$ dB is around the $PER_{Nreduced\_recomended}$. Note that for a given MODCOD, [$PER_{Thr\_Nred\_\#N}$, $N_{reduced:\#N}$] will be likely different for each propagation scenario.
3. For a given MODCOD, the pair [$PER_{Thr\_Nred\_\#N}$, $N_{reduced:\#N}$] has to be unique. Then, it is recommended that for a given MODCOD take the worst case pair [$PER_{Nreduced}$, $N_{reduced}$] (i.e., the propagation scenario which has higher $PER_{Nreduced}$).

As a matter of example, the following figure shows QPSK 1/2 (MODCOD #2) PER performances for different propagation scenarios and for $N_{nominal}$ (50 iterations) and for $N_{reduced\_\#N}$ (2 iterations) LDPC decoder iterations. In particular,

– the solid lines show the *PER* at $N_{reduced\_\#N}$ (2 iterations) LDPC decoder iterations ($PER_{Nreduced}$) for the Es/N0 at which the arget PER ($PER_{Thr\_Nominal}$ = 10^{-3}) is achieved at $N_{nominal}$ LDPC decoder iterations (red line corresponds to Scenario #1, green line to Scenario #2, yellow line to Scenario #3 and orange line to Scenario #4).

– The threshold ($PER_{Thr\_Nred\_\#N}$) for QPSK 1/2 is selected as the lowest PER value at $N_{reduced\_\#N}$ (2 iterations) LDPC decoder iterations, which in this case corresponds to Scenario #4. The selected value is $2.5 \cdot 10^{-1}$. Note that by selecting the threshold in such a way, the target PER at Nominal LDPC decoder iterations is guaranteed for all the propagation scenarios (the dashed lines show the operating point for each scenario taking into account the selected $PER_{Thr\_Nred\_\#N}$ value).

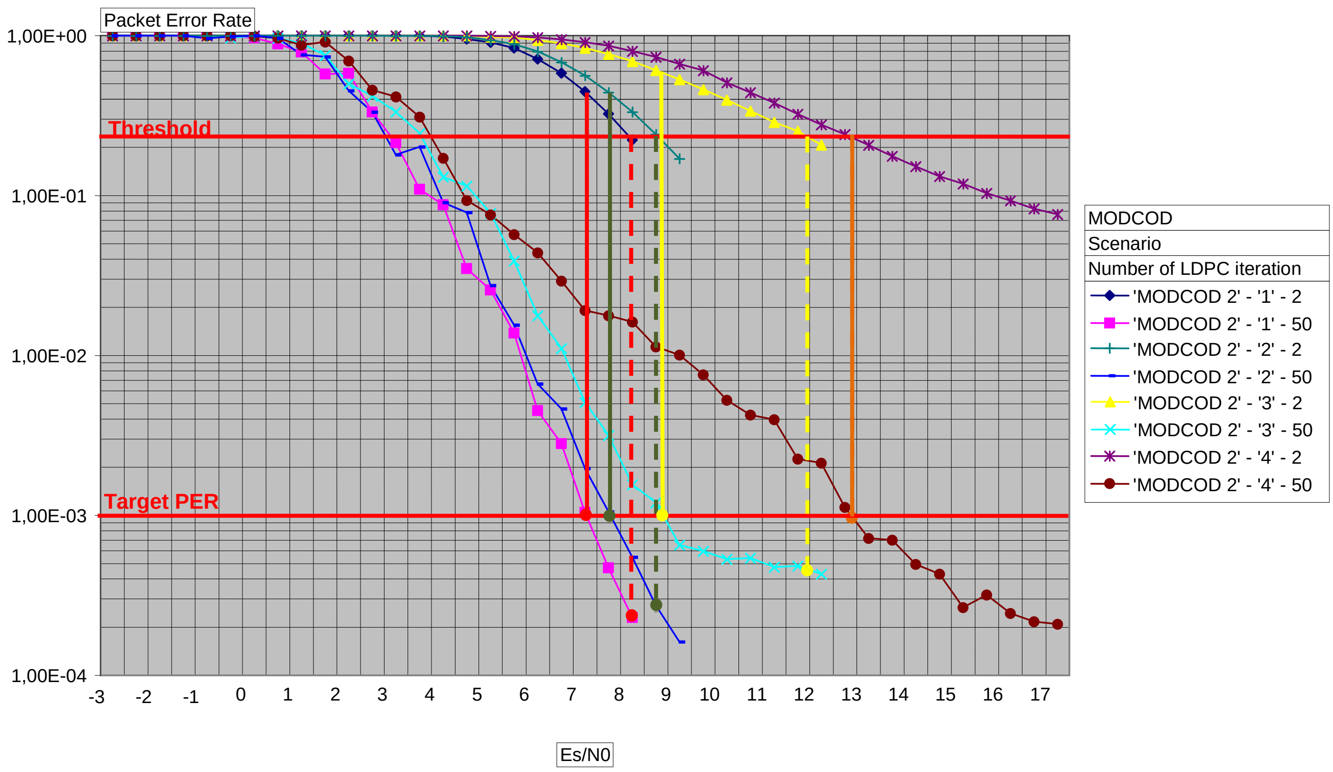


Figure 7-1: QPSK 1/2 PER performance at $N_{nominal}$ LDPC decoder iterations (50) and a $N_{reduced}$ LDPC decoder iterations (2)

The relevant results of the previous example are summarised in the following table. Assuming $PER_{Thr\_Nominal}$= 10^{-3}, the parameters to be taken into account for the slow ACM mechanism are:

– $N_{reduced\_\#2}$: 2 iterations

– $PER_{Thr\_Nred\_\#2}$: 0.25

Propagation scenario	$N_{nominal}$	$PER_{Thr\_Nominal}$ (Target PER)	$Es/N0_{target\_PER}$	$N_{reduced\_\#N}$	$PER_{Thr\_Nred\_\#N}$	EsN0 at $PER_{Thr\_Nred\_\#N}$
Scenario #1	50	10^{-3}	7.25 dB	2	$\sim 2.5 \cdot 10^{-1}$	8.25 dB
Scenario #2	50	10^{-3}	7.75 dB	2	$\sim 2.5 \cdot 10^{-1}$	8.75 dB
Scenario #3	50	10^{-3}	9 dB	2	$\sim 2.5 \cdot 10^{-1}$	12
Scenario #4	50	10^{-3}	12.75 dB	2	$\sim 2.5 \cdot 10^{-1}$	12.75 dB

Table 7-1: Summary results for QPSK ½ (MODCOD #2)

As for any MODCOD the pair [$PER_{Nreduced}$, $N_{reduced}$] has to be the same whatever the propagation condition is. An ACM margin is implicitly assumed. It is important to note that these thresholds can vary according to the final UT implementation (i.e., the thresholds are UT implementation dependent).

7.1.2 Link Quality Estimator

The link quality estimator is based on measuring the PER at $N_{reduced\_\#N}$ LDPC decoder iterations for all the MODCODs present in the FLC assigned to the UT. PER measurement at $N_{reduced\_\#N}$ LDPC decoder iterations could be implemented as a sliding window.

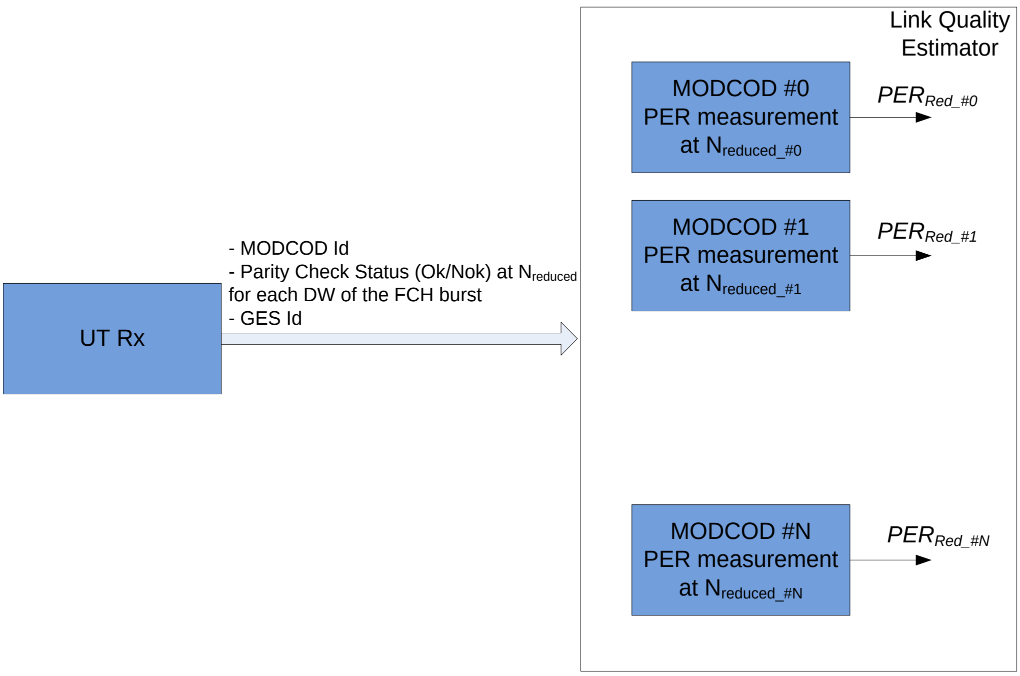


Figure 7-2: ACM link quality estimator

The performance of the FWD link uplink power may impact the $PER_{Red\_\#N}$ measurement since, in order to assure reliable measurements using bursts coming from different GS entities, the uplink power should ensure that the power at satellite level is the same in all the time-slots whatever the GES. If this cannot be assured, it is recommended that the UT use only bursts coming from its assigned GES to compute the $PER_{Red\_\#N}$.

As the slow ACM mechanism is based on PER measurements, the link quality estimator has to take into account the observation period in order to assure reliable $PER_{Red\_\#N}$ measurements. PER measurement should only take into account FCH bursts received during the last T_{obs} seconds. Taking into account the aeronautical channel, a T_{obs} = 120 seconds can be used.

7.1.3 MODCOD Selector

Based on the Link Quality outputs, the MODCOD selector estimates at any time the optimal physical layer configuration. Taking into account the nature of the ATM communications (safety communications), the link availability should prevail in front of the spectral efficiency optimization. In this sense, for the implementation of the MODCOD selector (MODCOD upgrade and MODCOD downgrade) it is recommended that these rules be followed:

- MODCOD upgrade: upgrade to any more efficient MODCOD, for which the $PER_{Red\_\#N}$ is measurable and the threshold is crossed, is allowed. A more conservative alternative is to perform the MODCOD upgrade in a progressive way, i.e., only upgrade to the next less robust MODCOD.
- MODCOD downgrade: in order to prevent packet loss, downgrade to any of the less efficient MODCODs is allowed.

The following figure shows a generic ACM state machine for the MODCOD Selector mechanism.

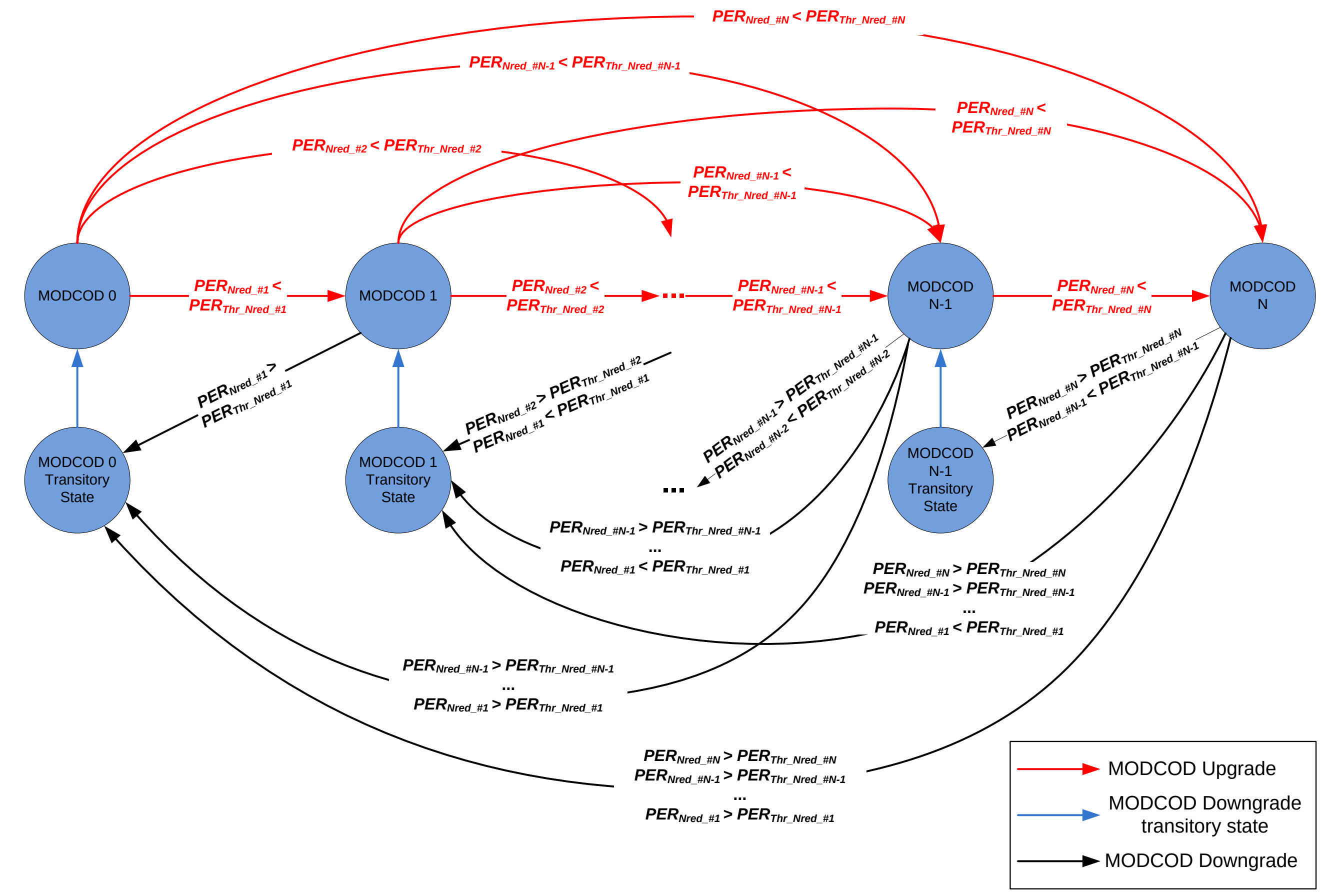


Figure 7-3: Generic MODCOD selector state machine

The proposed mechanism allows selection of the optimal MODCOD at any instant anticipating the packet error degradation after $N_{nominal}$ LDPC decoder iterations and preventing packet losses. Nevertheless, due to the link quality estimator, the MODCOD selector may often ping-pong between two adjacent MODCODs resulting in an increment of the signalling on the RTN link. In order to prevent or mitigate the ping-pong effect (rapid switch back and forth between adjacent MODCODs) a hysteresis mechanism could be implemented: each time a switch to a more robust MODCOD is decided, a timer is started in order to stabilize the loop. This hysteresis is represented in Figure 7-3 with the “MODCODN Transitory” states.

In the “MODCOD N Transitory” state:

- MODCOD upgrades are forbidden (in order to stabilize the loop and avoid ping-pong between 2 adjacent MODCODs).
- Even though not included in the MODCOD selector state machine, a MODCOD downgrade is allowed.

It is recommended that a timer be set to stabilize the loop to T_{obs} time.

7.2 Fast ACM mechanism

The fast ACM mechanism is envisaged as a safety mechanism able to react against fast and serious degradation of the propagation channel.

In contrast to the slow ACM mechanism, which is used either for MODCOD upgrade and downgrade, the fast ACM mechanism is only used for MODCOD downgrade. As it is a safety mechanism, the fast ACM mechanism is used to downgrade the MODCOD to the most robust one.

– Mechanism for MODCOD downgrade (fast ACM mechanism):

  In order to trigger the fast ACM MODCOD downgrade, the UT continuously monitors the decoder status after the nominal number of LDPC decoder iterations for all the packets transmitted using the preferred MODCOD and any of the less efficient MODCODs (i.e., more robust MODCODs) on its assigned FLC.

  In the event that a certain number of consecutive DW are erroneously decoded on the preferred MODCOD or on the less efficient MODCODs, then the UT has to trigger the fast ACM MODCOD downgrade and request that further bursts be sent using the most robust MODCOD. The algorithm has been satisfactorily tested, setting the number of consecutive DW in error to 2.

It is worth noting that the fast ACM must prevail in front of the slow ACM and, obviously, the state of the slow ACM MODCOD state machine (see section 7.1.3) has to be updated accordingly (i.e., moved to MODCOD0 transitory state).

8. RADIO RESOURCE MANAGEMENT

8.1 Forward Link Radio Resource Management

The management of the forward link resources, namely carriers and time-slots, is centralized at the NMC (Network Management Centre) entity. This entity uses standard network management interfaces, for example SNMPv3 (Simple Network Management Protocol with security extensions, RFC3410), to apply the radio resource assignments.

The FL radio resource allocation is based on capacity polices that are generated at the NMC and disseminated towards GES (Gateway Earth Station) and/or NCC (Network Control Centre) entities under its domain using management procedures. The CS does not impose a specific solution.

Radio resource management policies observe the following rules derived from system design:

- Only one on-board receiver is used in nominal mode for the reception of both system signalling and traffic. The other is kept for back-up and handover detection and execution purposes.
- Different GES and NCC entities can simultaneously transmit in the same carrier in different time-slots. In this case, the time-slots are assigned to each GES and NCC entity statically, following a predefined configured pattern that could be repeated periodically or not (see Figure 8-1, which shows assignments in two consecutive frames of a carrier, horizontal dimension is time and vertical dimension in frequency). The group of time-slots that form the periodic assignments to different GSE sharing a carrier is a frame (term defined in [AD-02]). In the event of no carrier sharing there is no need to define frame structure on a carrier, and all time-slots are available to the assigned Ground Segment Element (GSE)

GES1	GES2	GES3	NCC1	NCC2	GES1	GES2	GES3	NCC1	NCC2

Figure 8-1: Example of frame allocation between GSE

- Frame structures are periodic assignments of time-slots defined for sharing a carrier between GSE, although the time-slots assigned can vary from one frame to the next
- User terminals (UT) are not aware of frame structure. Hence, no frequency hopping can be envisaged in the radio resource assignment in forward link to UT. A handover must be executed for a UT to change the carrier it is receiving.

8.1.1 Forward Link Reference Scheduler

This section presents a scheduler discipline for the traffic sent through the forward link. The principal objective of the section is to provide a reference for the implementation of the FL scheduler, for both system signalling and user data traffic.

The scheduler takes PNPDU (Processed Network Protocol Data Unit) units from LSAP (Link Service Access Point) and delivers a multiplex of LPDU (Link Protocol Data Unit) units to PSAP (Physical Service Access Point) for the physical layer to transmit them. In between, the scheduler applies priority polices, fragments and encapsulates PNPDU into LPDUs, provides ARQ support, and selects the burst's MODCOD. Throughout this section the terms PNPDU and NPDU are used interchangeably as they both refer to an IP datagram.

8.1.1.1 Scheduling policies and CoS

The reference scheduler provides four generic levels of priorities as shown below, from the highest to the lowest priority:

Priority	Type
Highest	*System signalling. Includes ARQ ACKs to information sent through the RL*
Higher	*Voice traffic*
High	*Data traffic (re-transmitted)*
Normal	*Data traffic (transmission)*

Table 8-1: Forward Link Reference Scheduler Priorities

For the scheduler to apply an EDF scheduling policy to traffic of normal priority, a deadline must be derived for each incoming NPDU. At the time the NPDU is taken from LSAP, some information that could be used for such a purpose (e.g., IPv6:TrafficClass or TCP/UDP ports) may be no longer accessible to the scheduler just inspecting the PNPDU contents, due to the header compression applied by the network adaption layer. Hence, this information should be identified and extracted, the latest at ALSAP (Adaptation Layer Service Access Point), as indicated in section 4.6.5, and attached to the datagram handler[9]:

- TD95 or latency
- Expiration time

When this information is available, the reference scheduler uses it to derive the information required to drive the scheduling policies as indicated below. Otherwise, the scheduler fallbacks to a best effort discipline as indicated in 8.1.1.1.1.

- The TD95 (latency) parameter is selected as the initial EDF target time. The reference scheduler attempts to make available at the receiver the NPDUs within the TD95 value.
- In the event that the scheduler fails to fulfil the latency requirement (i.e., transmission errors) the scheduler switches the EDF target time to the expiration time using the process shown hereafter, if the information is available.
    - Best effort categories, namely DG-A, DG-J, DG-K & DG-L, set their target time to *(TD95 * 4) * 0.75*. Notice that, in [AD-01], these best effort categories do not define an expiration time value; hence the value is derived from its TD95 requirement.

[9] For instance, the Linux Kernel defines the `sk_buff` structure that is used to convey, describe and manage network datagrams all across the kernel. For example, this structure is used by `iptables` and `tc` (Queue Disciplines) to add marks to the associated datagrams, without requiring the modification of the datagram itself, with the purpose of applying QoS policies. Even more, the structure has been specially defined to allow the datagram to flow through modules that modify the datagram itself, such as header compression.

- Remain categories set their target time to ET * 0.50.

  The scaling factors, 0.75 and 0.50, provide some enhanced priority to the continuity requirement which is much more demanding than the latency one. If this information is not available, a 0.50 factor is recommended.

- If eventually, the scheduler fails to make the NDPU available at the receiver within the expiration time (without considering the scaling factor which is just used for priority boosting), the NPDU allocated resources are released.

The scheduling process just shown is presented in form of pseudo-code hereafter:

```
Function RefScheduler::AllocateResources () {
  foreach (slot) {
    RefScheduler::DescheduleExpiredTraffic ();
    RefScheduler::UpdateTargetTime         ();
    RefScheduler::SortTraffic              ();

    // Burst payload construction as described in 8.1.1.2
    RefScheduler::AllocateSystemSignalling (Policy::SP);
    if (slot.more_room () == true)
      RefScheduler::AllocateVoiceTraffic (Policy::SP);
    if (slot.more_room () == true)
      RefScheduler::AllocateDataTraffic_ReTx (Policy::EDF);
    if (slot.more_room () == true)
      RefScheduler::AllocateDataTraffic_Tx (Policy::EDF);
  } // slot
}

Function RefScheduler::DescheduleExpired ()
{
  foreach (NPDU) {
    // Check if the scheduler failed to satisfy ET QoS constraint
    if (NPDU.IsETViolated ()) {
      NPDU.Drop ();     // Dequeue and drop the NPDU and its allocated resources
    }
  }
}

Function RefScheduler::UpdateTargetTime ()
{
```

```
  foreach (NPDU) {
    if (NPDU.TargetTime == NONE) {
NPDU.TargetTime = NPDU.TD95;
      continue;
    }

    if (NPDU.TargetTime == NPDU.TD95 &&NPDU.IsLatencyViolated ()) {
      if (NPDU.Cos == {DG-A, DG-J, DG-K, DG-L})   /* Best Effort */
NPDU.TargetTime = 0.75 * (NPDU.TD95 * 4);
      else
NPDU.TargetTime = 0.50 * (NPDU.ET);
    }  // endif
  } // NPDU
}

Function RefScheduler::SortTraffic ()
{
  // This functions sorts NPDUs based on the value of NPDU.TargetTime
}
```

For the case of applications in which messages span more than one NPDU (because their size is higher than the MTU), a procedure can be devised to compute the remaining ET and TD95 considering previously transmitted NPDUs from the same message. Another option (fallback) is to apply the application layer message ET and TD95 to all NPDUs.

Finally, it is necessary to mention that the GES must keep a register of the TD95 of NPDUs latencies and the number of NPDUs successfully transmitted vs. arrived (continuity estimator), per application, in order to measure its compliance with required application QoS. This can be done on an SNMP MIB or log text files following any format. This is not standardized.

8.1.1.1.1 Scheduling fallback: Best Effort

In the case the scheduler has no CoS information associated with the NPDU, it fallbacks to a best effort mechanism. The scheduler tries to maximize the throughput of the transmitted data although it cannot apply individual CoS constraints to the NPDUs.

Due to safety issues, it is recommended to treat the unknown traffic with “best effort” QoS, i.e., assuming infinite ET and the maximum TD95 of potentially recognizable applications configured.

In the event that the QoS parameters applicable to an incoming NPDU cannot be determined, first an alarm (trap) should be generated towards the NMC, indicating a probable misconfiguration of packet scheduling that must be corrected. This is not standardized.

Nonetheless, it is expected that VoIP traffic could still be identified, i.e., seeking for RTP traffic, so the scheduler can still use the strict priority policy for this kind of traffic. The same statement also applies to system signalling that can be even more easily identified as it consists of system-internal information.

8.1.1.1.2 Voice Scheduling

In comparison to other services, the voice service is constrained by real time constraints and is handled by the scheduler with a different approach than for service application data. In particular, the voice service requires a policy that guaranties as much as possible a low end to end delay and jitter. This objective is achieved by using a high and strict priority scheduling policy in which incoming VoIP datagrams are scheduled as they arrive.

As the slot duration in the forward link, 86.35 ms, is higher than the VoIP cadence, 20 ms, the scheduler buffers4 or 5 VoIP datagrams and map them at once, each one with its own L2 encapsulation headers, to the burst. The number of VoIP datagrams to buffer depends on the frame structure. The maximum number of VoIP datagrams that are aggregated to a burst depends on the frame structure. For example, in the case of a frame consisting of 2 slots assigned to different GSE, the scheduler needs to buffer up to 9 VoIP datagrams.

Further optimization of the CS to better cope with voice service may be necessary when final specification of the end-to-end voice service is available.

8.1.1.1.2.1 Voice latency considerations

The scheduler should drop VoIP packets exceeding the buffering depth to prevent a monotonic increase of the voice latency. This is a normal QoS policy to apply to real time services such as voice as most vocoders (i.e., G729) provide error concealment mechanisms that are able to strongly mitigate the effect of losses in the channel. By contrast, there are no mechanisms counteracting a monotonic increase of the delay.

To preserve the quality and integrity of the voice service as much as possible, the scheduler needs to be selective when dropping the VoIP datagrams. The scheduler should avoid dropping header-compressed VoIP datagrams that are needed by the header decompressor to refresh or synchronize compression contexts. The selective drop mechanism could be based on the size of the compressed headers; the scheduler can drop PNPDUs whose compressed headers size is equal or below 6 bytes.

8.1.1.2 Scheduling burst payload construction

The forward link scheduler works in a per slot basis, building a multiplex of LPDUs to be forwarded to the physical layer when the slot is ready to be transmitted. For each slot and carrier pairs the scheduler produces the next outcomes:

- A PPDU payload consisting of a multiplex of LPDUs addressed to one or several UTs.
- A payload handler which contains additional information which is required by the physical layer to build the burst:
    - PPDU payload length which is copied into the DLF field of the physical forward link data descriptor (FWD_DD).
    - The MODCOD used to transmit the burst.

- The NCR flag which instructs the physical layer whether or not to include the NCR information in the burst. The scheduler takes into consideration the reduction of the PPDU payload size by 40 bits when the NCR is inserted.

8.1.1.2.1 Encapsulation

The FWD link layer encapsulator (L2 module) first has to determine whether or not the incoming NPDU must be fragmented, in order to first set the S and E flags (Start and End bits) accordingly, as described in section 8.6.1.2 of [AD-02].

The AF field allows reusing the addresses specified on the previous LPDU in the same PPDU, e.g., when sending data and signalling LPDUs on the same PPDU to the same terminal.

The C flag indicates the presence of CRC at LPDUs. As there is already a CRC at physical layer level, this option is not needed, except if the application requires an integrity higher than 1e-12. The reassembly of fragmented LPDUs can be based on the packet count field when CRC is not used, as described in section 8.1.1.2.1.2.

In the FWD link, the L2 encapsulator can receive from the network adaptation layer the NPDU to encapsulate with the next QoS parameters attached (out of band), determined by the network adaptation layer for the NPDU: TD95, ET, ARQ need, L2 CRC need. The QoS parameters can be deduced by the network adaptation layer from incoming NPDU DSCP field, transport layer protocol or ports, or by other mechanism, agreed at system level and configured in the GS element. If the QoS parameters are not determined, the NPDU is treated following a best effort policy.

The GES selects and reuses FID values from the pool of 16 available FID values. Notice that applications with the same QoS parameters can be assigned the same FID (belong to the same flow), as the objective of the FID field is to allow the pre-emption of the transmission of packets with different QoS requirements. As there is no ARQ flag in the FWD link encapsulation in the case of fragmentation, in the system tables it is indicated which FWD link FIDs are to be used with traffic requiring ARQ and which are for traffic not requiring ARQ. Usually, just two FID values are reserved for traffic that does not need ARQ, one for voice and another for signalling.

8.1.1.2.1.1 Fragmentation

The fragmentation of incoming NPDUs is performed at run time while the burst payload is being constructed with the objective of:

- Reducing the burst's padding insertion as much as possible, by generating fragments whose size depends on the burst's instant free room rather than in its capacity.
- Providing a smooth integration within an ACM system. In practice, the actual capacity of the burst, which depends on the MODCOD, is not known until the time of the burst transmission.

In the event that the NPDU to be transmitted requires ARQ support, the link layer fragmentation process or encapsulator prevents the generation of LPDUs whose size (including headers) is larger than the PPDU payload of the most robust MODCOD.

The ARQ process is executed at LPDU level, i.e., after the fragmentation and before the reassembly of packet upon reception. It is assumed that before reassembly in the receiver, ARQ reorders the received fragments correctly. Re-assembled NPDUs will be delivered to the receiver network adaptation layer in the order of reception of the last fragment of the NPDUs.

The FID field, together with the Packet Count and the Fragment Count fields or the CRC-32 field, provides the possibility for parallel fragmentation and reassembly of several NPDUs.

8.1.1.2.1.1.1 General process

To perform fragmentation and reassembly, which is required when the NPDU does not fit in the available resource offered in a single physical layer payload, the following rules apply:

1) All fragments carrying data of the same NPDU have the same Flow ID.

2) The first fragment of an NPDU with a given Flow ID has S equal 1 and E bit equal 0.

3) The fragments, which are neither the first nor the last NPDU fragment, have S bit equal 0 and E bit equal 0.

4) The last fragment of a NPDU with a given Flow ID has S bit equal 0 and E bit equal 1.

5) All fragments of an NPDU with the same Flow ID must be transmitted in order.

Whenever fragmentation is to be applied, assuming that the physical layer offers n_1 bytes of resource in its payload (n_1 being lower than the NPDU total length), the encapsulation process takes the first (n_1-L2_header_length) bytes of the NPDU and performs the following:

- If CRC-32 is used for this NPDU, it computes CRC value using a systematic 32-bit CRC encoder on all NPDU data (the same CRC-32 algorithm specified for the physical layer).
- Forms a fragment with S bit set to value "1", E bit set to value "0".
- Appends the Address Format field with the corresponding value depending on addressing requirement of the message (see AF field values described in previous section and in section 8.6.1.2 of [AD-02]).
- Sets the Length field to the value (n_1-L2_header_length). L2_header_length takes into account the 6 mandatory bytes of the first fragment header + 2, 3 or 4 optional bytes for the addressing field (in consistence with the Address Format field).
- Appends the appropriate Protocol Type field (see PT field values in section 8.6.1.2 of [AD-02]).
- Sets the Total Length field to the number of bytes of the un-fragmented NPDU, adding the 4 bytes of CRC-32 field (if present).
- Set C flag to 1 if CRC-32 is used for this NPDU.
- Sets Flow ID to a value associated with the class of service of the corresponding NPDU.
- Appends the value of the packet counter to the Packet Count field of the NPDU and the value of the fragment counter to the Fragment Count field (see reassembly description below for description of the value to be appended).
- Appends the 2, 3 or 4 bytes of the Source/Dest address field if used. The presence and content of this field depend on the AF field value (see AF field description in previous section and in section 8.6.1.2 of [AD-02]).

- Inserts the first (n1-L2_header_length) bytes from the NPDU data in the L2 Payload.
- Places this fragment into the current L1 frame as first fragment or directly after any other fragment already present in the L1 frame.

If the resource offered next by the physical layer (n_x) does not permit to transmit the remaining bytes of the NPDU, the encapsulation process performs the following steps:

- Forms a fragment with S bit set to value "0", E bit set to value "0".
- Appends the Address Format field with the corresponding value depending on addressing requirement of the message (see AF field values described in previous section and in section 8.6.1.2 of [AD-02]).
- Sets the Length field to the value (n_x-L2_header_length). L2_header_length takes into account the 4 mandatory bytes of the first fragment header + 2, 3 or 4 optional bytes for the addressing field (in consistence with the Address Format field).
- Sets C flag to 1 if CRC-32 is used for this NPDU.
- Sets Flow ID to a value associated with the class of service of the corresponding NPDU.
- Appends the value of the packet counter to the Packet Count field of the NPDU and the value of the fragment counter to the Fragment Count field (see reassembly description below for description of the value to be appended).
- Appends the 2, 3 or 4 bytes of the Source/Dest address field if used. The presence and content of this field depend on the AF field value (see AF field description in previous section and in section 8.6.1.2 of [AD-02]).
- Inserts the following (n_x-L2_header_length) bytes from the NPDU data in the L2 Payload.
- Places this fragment into the current L1 frame as first fragment or directly after any other fragment already present in the L1 frame.

The previous steps are repeated by the encapsulation process for all fragments that are not the last fragment (and not the first), i.e., until the remaining quantity of bytes of the NPDU to be transmitted is larger than the resource offered by the physical layer.

When the resource offered by the physical layer is equal to or larger than the remaining quantity of bytes of the NPDU, the encapsulation process performs the following steps:

- Forms a fragment with S bit set to value "0", E bit set to value "1".
- Appends the Address Format field with the corresponding value depending on addressing requirement of the message (see AF field values described in previous section and in section 8.6.1.2 of [AD-02]).
- Sets the Length field to the value (n_{x+}L2_trailer_length-L2_header_length). L2_header_length takes into account the 2 mandatory bytes of the intermediate fragment header + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (in consistence with the Address Format field). L2_trailer_is 4 bytes if CRC-32 field is used, 0 byte otherwise.
- Sets C flag to 1 if CRC-32 is used for this NPDU.

- Sets Flow ID to a value associated with the class of service of the corresponding NPDU.
- Appends the value of the packet counter to the Packet Count field of the NPDU and the value of the fragment counter to the Fragment Count field (see reassembly description below for description of the value to be appended).
- Appends the 2, 3 or 4 bytes of the Source/Dest address field if used. The presence and content of this field depend on the AF field value (see AF field description in previous section and in section 8.6.1.2 of [AD-02]).
- Inserts the final (n_x-L2_header_length) bytes from the NPDU data in the L2 Payload.
- Appends CRC-32 computed in first step if C flag is set to 1. If Packet Count field is used its value is computed as follows: The sender has a variable counter of Packet Count for every Flow ID. The current value of the counter for the corresponding Flow ID is appended to the Packet Count value. Then the counter is incremented by 1 unit. When the counter reaches 255, the next increment cycles back to 0.
- Places this fragment into the current L1 frame as first fragment or directly after any other fragment already present in the L1 frame.
- If there is no other signalling/data to be sent by the GES or if the remaining resource in the L1 payload is shorter than (L2_header_length+1) bytes then padding bits (0x00)are inserted in the remaining bytes of the L1 payload. L2_header_length takes into account the 4 mandatory header bytes if a first fragment needs to be inserted or the 2 mandatory header bytes if an intermediate fragment needs to be inserted or the 2 mandatory header + 4 optional CRC trailer if a last fragment need to be inserted + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (consistent with the Address Format to be used).

8.1.1.2.1.2 Reassembly

8.1.1.2.1.2.1 Using CRC-32 field

When the last fragment of an NPDU is received (i.e., when S bit is 0 and E bit is 1), the following steps are executed:

- If there is no first fragment having the same Flow ID, all fragments having this specific Flow ID are thrown away.
- If there is a first fragment having the same Flow ID, a first check consists of verifying that the Total Length indicated in the first fragment corresponds to the sum of the Length fields of all the fragments since the last first fragment having the same Flow ID. If this is not the case, all fragments since the last start fragment are thrown away.
- If the previous step shows that the Total Length corresponds to the sum of the Lengths of all the fragments since the last first fragment having the same Flow ID, the CRC-32 field is used to check the integrity of the re-assembled NPDU. If the CRC detects corruption, all fragments since the last start fragment are thrown away; otherwise it is passed to the network adaptation layer.

8.1.1.2.1.2.2 Using packet and fragment count fields

The following steps describe the method when using the Flow ID, the Fragment Count and the Packet Count fields:

- The sender and the receiver need to initialize their respective counters for every Flow ID to the same value (to the value 0). This has to be done at logon and during the handover process to perform a seamless handover for the encapsulation process.
- The sender has a packet counter for every Flow ID and a fragment counter for every packet.

  When emitting every fragment:

  - The fragmentation process appends the value of the packet counter to the Packet Count field of the L2 header and the value of the fragment counter to the Fragment Count field.
  - The fragment counter is incremented by 1 unit.
  - When emitting an end fragment the fragmentation process increments the packet counter and re-initialize the fragment counter to 0. When the packet counter reaches 15, the next increment cycles back to 0.
- The receiver has a packet counter as well for every Flow ID. This counter contains the Packet Count field expected in the next L2 packet. When the receiver gets the last fragment of a packet, it does the following:
  - If there is no first fragment having the same Flow ID, all fragments having this specific Flow ID are thrown away.
  - If there is a first fragment having the same Flow ID, the reassembly process checks that the Total Length indicated in the first fragment corresponds to the sum of the Length fields of all the fragments since the last first fragment having the same Flow ID. If this is not the case, all fragments since the last start fragment are thrown away.
  - It compares the Packet Count field value of all fragments received since last first fragment in this Flow ID to the packet counter of the given Flow ID. If they differ, all the fragments since the last start fragment are thrown away.
  - It checks that Fragment Count fields of every fragment of the concerned Flow ID are consistent since the reception of the last start fragment, meaning that no fragments are missing. If this is not the case, all fragments from the last start fragment are thrown away. If Fragment Count fields are consistent, the packet is passed to the corresponding layer.
  - It then increments the packet counter for the corresponding Flow ID to the value of Packet Count +1 with a cycle every 15.

8.1.1.2.2 MODCOD selection

The scheduler dynamically adapts the burst's MODCOD while allocating LPDUs to remain physical payload using the mechanism explained hereafter:

- The scheduler has knowledge of the MODCOD assigned to each of the UTs assigned to it. This information is obtained from the ACM algorithm as explained in Section 7.
- The scheduler initially sets the burst's MODCOD to the MODCOD required for the 1st mapped LPDU.
- For successive LPDUs the scheduler compares the MODCOD required by the LPDU and the one already assigned to the burst:
    - The burst's MODCOD is conserved whenever the LPDU requires the same, or a more efficient MODCOD.

    Otherwise, the LPDU requires a more robust MODCOD.

    - The burst's MODCOD is conserved if after the MODCOD change there is no room for the new LPDU. In this case, the LPDU scheduling attempt is postponed to the next assigned slot

    Otherwise,

    - The burst's MODCOD is changed to the one required by the new LPDU.

The process finalizes when there is no more data to schedule or when there is no more free room in the burst's payload.

8.1.1.2.2.1 MODCOD selection: ACM Dummy packets

As indicated in section 4.6.3, the GS scheduler of the system must observe the constraints observed by the ACM functionality.

It is recommended that the MODCOD selection process be used as much as possible so as to satisfy the ACM constraints while reducing the insertion of dummy FCH bursts.

The scheduler creates dummy FCH bursts by generating bursts containing a dummy LPDU with the GS ID as source address and the reserved L2 dummy address as the destination address. The receiver uses the destination address of the received LPDU to identify that it belongs to a dummy LPDU and:

- Extracts the GS ID from the source address field and includes it in the information sent to the ACM module.
- Discards the LPDU payload.

8.1.1.2.2.2 MODCOD selection: NCR insertion

As indicated in section 6.2.1.2.3, the scheduler of the NCC broadcasts the NCR to GSEs. The scheduler controls the insertion of the NCR using the parameters recommended in section 6.2.1.2.3 by. In the event that the NCR needs to be broadcast, the scheduler reduces the available PPDU payload by the size of the NCR before performing the nominal MODCOD selection procedure shown in section 8.1.1.2.2 and section 8.1.1.2.2.1.

Then, the scheduler indicates to the physical layer that the NCR has to be inserted.

8.1.1.2.3 ARQ Procedure recommendations

The reference scheduler provides support to ARQ procedure using information located in the L2 headers, which is the information fully available for each LPDU at the receiver.

The ARQ process in the FL is described by the SDL and walkthrough specified in [AD-02], section 8.6.1.1.2, which provides a generic approach to the process in which some of the functions are left to implementation decisions. The present section proposes an implementation.

8.1.1.2.3.1 Transmitter side

The transmitter generates the ARQ blocks as indicated below in the case of multi-fragment NPDUs. That is, those NPDUs that are fragmented in several LPDUs:

- The transmitter groups *k* consecutive LPDUS belonging to the same NPDU and sets the P/F bit for the last LPDU of the group to request the receiver to generate an ARQ-ACK for it. Notice that the last generated ARQ block can contain less than k LPDUs depending on the number of LPDUs in which the NPDU is eventually fragmented. *It is recommended that k=5 be used.*

In the case of single-fragment NPDUs, the receiver always generates ARQ-ACK.

As indicated in the ARQ protocol, there is only one retransmission entity per NPDU that is associated with the active block. The retransmission timer is started at the time the slot is actually transmitted and *its recommended value is 1.3 seconds.*

The reference scheduler tries to retransmit the whole *active_block* in the very first available slot upon expiration of the retransmission timer. Actually, the process of multiplexing LPDUs to the slot and the dynamic MODCOD selection, as defined in 8.1.1, is also applicable to retransmitted fragments. In this case, the process of mapping LPDUs of the *active_block* to the PPDU payload continues until either all LPDUs of the *active_block* are added to the multiplex or no more room is left at the PPDU payload. In either case, the retransmission timeout is set and associated with the (fully or partially) transmitted active block.

In the event that the retransmission timer is triggered again, the retransmission handler resumes the retransmission of the active block in the previously non-retransmitted fragment, in the case of partial retransmission; or in the first fragment of the block, in the case of a previous full retransmission.

8.1.1.2.3.2 Receiver side

At reception, the UT parses the received multiplex of LPDUs and processes those addressed to it by examining both Dst and SrcL2 addresses. Those LPDU requiring ARQ support are forwarded to the ARQ handler which resolves to which NPDU or context they belong to by inspecting the L2 addresses, flow identifier and packet counter. Then the process indicated by the corresponding SDL and verbal walkthrough in [AD-02] is executed.

In order to keep the number of generated (transmitted) ARQ ACKs as low as possible, it is recommended that the receiver exploit the cumulative approach of the protocol as much as possible:

- The receiver waits to process all the LPDUs conveyed by the received burst before generating (*queuing*) the ARQ ACKs.

When there is an opportunity to transmit an ARQ ACK, the transmitter checks for queued ARQ ACKs and transmit them. In the case that more than one ARQ ACK belonging to the same NPDU are present in the queue, it is recommended that the scheduler only transmit the ARQ- ACK with the largest fragment counter value; the others are silently discarded.

8.1.1.3 Performances

Using the reference scheduler and an estimation of the ECAC traffic during the 6 hour peak, the following reference performances can be obtained:

The reference number of carriers that are required to satisfy the CoS of the reference traffic profile under the aeronautical propagation channel are shown in Table 8-2. In this table the latency delay introduced is also shown:

Frame Size	Beam1 Carriers	Beam2 Carriers	Beam3 Carriers	Beam4 Carriers	Beam5 Carriers	Voice Latency	
						Min	Max
1	1	1	8	4	1	372	562
2	1	1	8	4	1	372	548
3	1	1	10	6	1	372	638
4	1	1	11	5	2	372	723
5	1	1	12	6	2	372	808

Table 8-2: Reference number of carriers required in the FL

8.2 RTN Link Radio Resource Management

8.2.1 CDMA Codes Management

A RACH burst to be transmitted by a UT is defined by the following parameters related to code management (all of them are distributed by the GS to UTs through signalling tables):

- Complex scrambling code(s).
- Complex preamble sequence(s).
- A channelization code for the data channel (DCH).
- A channelization code for the auxiliary channel (ACH).

The complex scrambling and the complex preamble sequences are defined in pairs. A few pairs can be allocated to minimize collisions due to time alignment among bursts. A pair can be reused in different frequency bands.

The probability of time (code) collision (defined as the probability that at least two bursts are received with a time separation lower than a chip period) is manifested as follows,

$$P_{Tcoll} = 1 - e^{-\lambda / R_c}$$

where λ and R_c are the bursts arrival rate on the band and the chip rate, respectively. As an example, the probability of time collision is $6.24 \cdot 10^{-4}$ for a burst arrival rate of 100 bursts per second.

Note that, in practice, bursts collision is computed as the probability that at least two bursts collide in time, frequency and amplitude. Bursts colliding in time can be still typically discriminated thanks to the power, frequency and data content diversity. So, the time collision probability is not fully representative of the possible packet collision impact in terms of packet errors.

A single DCH channelization code and a single ACH channelisation code are defined per RACH Configuration ID. DCH and ACH channelisation codes must be different. Moreover, if different RACH Configuration IDs share a given frequency band, ACH channelisation codes must also be different.

The preamble spreading sequence is used to identify the SF of the received bursts (in the event that several RACH Configuration IDs with different SF are used in a given frequency band).

The ACH channelization code is used to fully discriminate the RACH Configuration ID of the received burst. Thus, the DW length of the received burst is identified by the ACH channelisation code used.

In the event of a system using full frequency reuse, RACH with different SFs are transmitted at different frequency bands. If no full frequency is selected, it is possible that RACH with different SFs share the same band depending on the traffic to be served.

8.2.2 Mixing spreading factors in the same Return Link Carrier (RLC)

In addition to the frequency reuse concept, it is possible to exploit the concept of mixing spreading factors length in the same RLC. This is an optional mode that could be used whenever the traffic rate for both spreading factors is by far lower than the channel throughput. In particular, the overall performance is mainly driven by the lower SF (namely 4) since it is not possible to get full advantage of the SIC.

Moreover, the receiver complexity increases in a system using the mixing mode as the SIC is obliged to seek for two different preambles, one for each spreading factor. Notice that using a single preamble for all the burst modes is not feasible in a real system as it diminishes the system efficiency by increasing the physical overhead near 50%.

In conclusion, it is recommended that burst modes with different spreading factors be segregated into different return frequency carriers (RLC).

8.2.3 Return Link reception mode

This section discusses the different GS reception modes that can be implemented for the System Reference model. The proposed reception modes assumes no spreading factor mixing in the same return link carrier (refer to8.2.2), while imposing assumptions on the frequency plan or frequency reuse factor.

– *Basic reception mode*: In this mode, there is no interaction between GS receivers, as they are defined in section 5.2.1 and section 5.2.4. When a receiver decodes a burst belonging to another beam, it strictly uses the just decoded data to feed its SIC. Other than this, the decoded data is discarded.

– *Enhanced reception mode*: In this mode, there is interaction between co-localized GS receivers, as they are defined in section 5.2.1 and section 5.2.4. When a receiver decodes a burst belonging to another beam, it uses the decoded data to feed its associated SIC but also sends it to other receivers. Therefore, within this mode receivers in a GS work in cooperation enhancing the system performance in the sense that the packet loss rate and latency are reduced as there are more receivers that can receive a burst. Notice that by reducing the PLR, the ARQ performance is also improved as the number of retransmissions is reduced.

The enhanced reception mode with full frequency reuse and no spreading factor mixing in the same return link carrier provides the best performances and is therefore the recommended configuration.

The following system assumptions and circumstances must come together for a system to exploit the enhanced reception mode:

– The underlying system provides user link frequency reuse, e.g., full frequency reuse.
– The user link carrier allocation in the different beams is provided so as some, or all, return link carriers are reused in the beam overlapping areas.
– A terminal located in a beam overlapping area transmits a RACH burst in a user link carrier that is reused in the overlapped beams.

8.2.3.1 Detection of duplicated bursts in enhanced mode

When the GS implements the enhanced reception mode, a burst transmitted by a UT can be received by multiples of the bank of co-localized receivers at the GS. Hence, the L2 process of the GS could receive duplicated versions of the LPDUs conveyed in the multiple received burst.

To prevent forwarding duplicated NPDUs to upper layers it is recommended that the GS implements one of the following mechanisms, which are listed in order of preference:

– Burst timestamping: The SIC time stamps the decoded packets using the time at which the burst enters the SIC. This information is used by the L2 process along to the L2 header of the LPDUs to identify duplicity as indicated below:
    - In the event that the L2 header does not match any of the LPDUs stored in the duplicity detection buffer, the LPDU is processed and stored in the duplicity buffer.
    - Otherwise, the time stamp of the matching LPDUs is checked. If the time stamp difference is smaller than half the duration of the smaller burst (75 ms for $RACH\_CR_{160}\_SF_4\_DB_{976}$), the packet is silently discarded.
    - Otherwise, the LPDU is processed and stored in the duplicity buffer.
- LPDU hashing: This is a process similar to the one described above, but instead of using time stamps generated by the SIC it is based on comparing the LPDUs, for example using a hash value computed over the whole LPDU.

LPDUs that are stored in the duplicity buffer should remain in it at least during the SIC processing window; then they can be removed.

8.2.4 RTN Link multiple access

This section addresses the steps and recommendations that UTs observe to transmit over the return link carriers, which are shared among terminals.

8.2.4.1 Multiple access parameters

The parameters that the terminal needs to support the RL multiple access procedure are signalled by the GSE through system signalling messages, e.g., Logon Table.

8.2.4.1.1 Congestion Control Parameters and Re-Transmission Timeout

The recommended values for the Congestion Control and Re-Transmission Timeout are shown in Table 8-3 for RACH with SF=16 and in Table 8-4 for RACH with SF=4 for the case of *CC_category_id* equal to '*Normal*'.

traffic_status	*tx_backoff*	*persistence*	*retransmission_timeout*
Low	50 ms	1.00	1.0 s
Medium	100 ms	0.75	1.1 s
High	150 ms	0.35	1.2 s
Congested	200 ms	0.05	1.3 s

Table 8-3: Recommended CC and ReTx Timeout parameters for SF=16

traffic_status	*tx_backoff*	*persistence*	*retransmission_timeout*
Low	165ms	0.75	1.0 s
Medium	210ms	0.65	1.1 s
High	240ms	0.50	1.2 s
Congested	300ms	0.20	1.3 s

Table 8-4: Recommended CC and ReTx Timeout parameters for SF=4

In the case of 'High' priority traffic, the following parameters are recommended:

- '*tx_backoff*' is set to 0 ms for all loads to indicate bypass of congestion control as defined in D018-COM-FUN-1610.
- '*persistence*' is set to 1 for all loads to indicate bypass of congestion control as defined in D018-COM-FUN-1610.
- '*retransmission_timeout*' as per Normal category.

8.2.4.1.2 Power Randomization parameters

The NCC controls the range of the explicit power randomization mechanism through the parameter "*lower_bound_power_dB",* which is a negative value as terminals transmit at full power plus this value.

The NCC sets this parameter to 0 to disable the explicit power randomization and rely on the implicit power randomization derived from aeronautical propagation channel and antenna pattern.

The actual value of *lower_bound_power_dB* depends on system constraints, such as effective link margin. Hence, different values depending on the SF could be required.

It is recommended that the explicit power randomization mechanism be exploited as it improves the power unbalance at the input of the SIC, improving its performances.

A recommended value for *lower_bound_power_dB* is -2 dB for both spreading factors.

8.2.4.1.3 Available channels

The NCC distributes through the *RTN_channel_configuration* signalling structure information about the return link carriers that the UT can use for the transmission.

- The type of traffic that can be transmitted in the return link channel, which is indicated through the field *RTN_channel_service_mask*.

It is recommended that the system should reserve a separated channel for the voice service to improve the performance of the voice service. For example, distributing a *RTN_channel_configuration* with the following configuration:

- *RTN_channel_service_mask* = b0001 (Voice)
- *N_supported_RACH_confs* = 2
    - RACH_config_id [0] = b0100 (RACH_CR$_{160}$_SF$_4$_DB$_{976)}$
    - RACH_config_id [1] = b0010 (RACH_CR$_{160}$_SF$_4$_DB$_{2048)}$

It is recommended that mixing different spreading factors in the same return link channel should be avoided, that is, not assigning RACH_config_ids of RACH burst configurations with different spreading factor in the same RTN_channel_configuration structure.

8.2.4.1.4 Traffic Status

For each of the available return link channel, the GS distributes its load condition through the *traffic_status* parameter. The UT uses this information to select, for each channel, the appropriate set of congestion control parameters.

It is recommended that the GS broadcast this information with a periodicity of 1 second.

8.2.4.1.4.1 Traffic Status measurement

The GS computes the load condition, or *traffic_status*, from noise rise measurement, as indicated in 5.2.5, and derives the *traffic_status* using the thresholds shown in Table 8-5.

traffic_status	Noise rise range (SF=16)	Noise rise range (SF=4)
Low	<150	<30
Medium	[150, 200)	[30, 35)
High	[200, 250)	[35, 40)
Congested	≥ 250	≥ 40

Table 8-5: Noise rise to load condition thresholds

8.2.4.2 UT transmitter procedure

This section provides the steps and recommendations on how the UT schedules and transmits information. The process is summarized next.

The UT extracts information concerning the NPDU or system signalling message that needs to be transmitted. (Throughout the procedure the term NPDU, unless explicitly stated, also refers to system signalling messages.)

This information is tagged to the NPDU, so it can be used at the different transmission stages:

- Identify the RACH burst types that can be used to transmit the NPDU, specified in D018-COM-FUN-2470 and described below in section 8.2.4.2.1.1.
- The congestion control procedure specified in D018-COM-FUN-1610 and described below in section 8.2.4.2.1.2.
- Scheduling policies as specified in D018-COM-FUN-1615, D018-COM-FUN-1625 and D018-COM-FUN-1640 and described below in section 8.2.4.2.1.3.
- Encapsulation and ARQ provision as indicated in sections 8.7.1.2 and 8.7.1.3 of [AD-02], and described in section 8.2.4.2.2.
- Return link carrier selection and power randomization, described in section 8.2.4.2.3 below.

8.2.4.2.1 Congestion Control and Scheduling

When an NPDU or signalling message arrives to the UT transmitter, its QoS parameters (TD95, ET and application specific information, as indicated in 4.6.5) are resolved by the network adaptation layer, as indicated in section 4.6.5. This information is used to derive the information required to drive the mapping to burst, congestion control and scheduling policies indicated below. Otherwise, the scheduler fallbacks to a best effort discipline as indicated in 8.1.1.1.1.

The QoS parameters of the NPDU must be identified and extracted, the latest at ALSAP, using any of the possible identification means.

As indicated in section 4.6.5, if the QoS parameters required to drive the scheduling policies cannot be determined, a best effort service policy is applied.

The same scheduling policies proposed for the FWD link in section 8.1.1.1 regarding the case of application messages spanning in multiple NPDUs are also applicable for the RTN link.

Finally, it is necessary to mention that the UT keeps a register of the 95th percentile of NPDUs latencies and the number of NPDUs successfully transmitted vs. arrived (continuity estimator), per CoS category, in order to measure its compliance with the required application QoS. This can be done on an SNMP MIB or log text files following any format. This is not standardized.

To support the description of the transmission procedure, some attributes are tagged to the received NPDU:

- The selected RACH burst type to transmit the NPDU is stored in attribute *rl_npdu_rach_id* using the values specified by the field *RACH_config_id* of the *RACH_burst_configurations* signalling structure (see section 11.10 of [AD-02]).

- The category assigned to the NPDU is stored in the attribute *rl_npdu_cc_category_id* using the values specified by the field *CC_category_id* of the *CC_config* structure signalling structure (see section 11.10 of [AD-02]).
  - Voice and system signalling indicated in D018-COM-FUN-1605 are assigned to category '*High'*.
  - Data traffic which is not voice is assigned to category '*Normal'.*
- The type of service is stored in the support variable *rl_npdu_service_id* according to the field *RTN_channel_service_mask* of the *RACH_channel_configurations* signalling structure (see section 11.10 of [AD-02]).
- The '*expiration_timeout*' is computed as indicated in D018-COM-FUN-1625 for the NPDUs that are constrained by an ET QoS parameter.

8.2.4.2.1.1 Mapping to burst type

In the first step, the NPDU is mapped to the corresponding RACH burst as summarized below:

- Voice packets are mapped to $RACH\_CR_{160}\_SF_{4}\_DB_{976}$ or $RACH\_CR_{160}\_SF_{4}\_DB_{2048}$ can be used. Specific recommendations on voice scheduling are provided in 8.2.4.2.1.3.2.
- Applications whose size is larger than, or equal to, 500 bytes are mapped to $RACH\_CR_{160}\_SF_{4}\_DB_{2048}$:
  - D_ALERT, COTRACT (Wilco), COTRACT (Interactive), ADS_C, ENGINE and FLIPINT application messages in the profile shown in section 4.4.
- Small messages, including system signalling, that fit into the $RACH\_CR_{160}\_SF_{16}\_DB_{288}$ without requiring fragmentation are mapped to this burst:
  - For example ARQ ACKs messages, compressed TL_ACKs or ACM signalling.
- Other information is sent using $RACH\_CR_{160}\_SF_{16}\_DB_{512}$.

The attribute *rl_npdu_rach_id* is set the selected RACH burst type.

8.2.4.2.1.2 Congestion Control

Before sending the NPDU to the scheduler, the UT computes and applies a random delay to the entities indicated in D018-COM-FUN-1605 following the procedure indicated in D018-COM-FUN-1610.

NB: System signalling messages not covered by D018-COM-FUN-1605 provide their own congestion control mechanism so they bypass this congestion control procedure and go directly to the scheduler.

The procedure specified in D018-COM-FUN-1610 to compute and apply the '*backoff_delay*' is applied. The following implementation is recommended.

- The transmitter identifies the sub-set of return link carriers in which the NPDU, or applicable system signalling message, can be transmitted. The attributes '*rl_npdu_rach_id'* and '*rl_npdu_service_id'* are used. For example, a FLIPINT message can be transmitted in the subset of carriers in which data services can be transmitted using the $RACH\_CR_{160}\_SF_{4}\_DB_{2048}$ RACH burst.

- From the subset of return link carriers identified in the previous step, the UT identifies the most congested carrier. The UT resolves this information by inspecting the *traffic_status* parameter of the subset of return link carriers.

  Note: Those carriers that are disabled through *traffic_status* equal to '*Channel Disabled*' are not considered.

- The '*tx_backoff*' and '*persistence'* parameters associated with the *traffic_status* of the most congested carrier and the category indicated by the attribute *rl_npdu_cc_category_id* are selected and applied as indicated in D018-COM-FUN-1610.

This congestion control procedure is also applied to the set of LPDUs selected for retransmission, as indicated in 8.2.4.2.2.3.

8.2.4.2.1.3 Priority Scheduler

The purpose of the priority scheduler is to try to satisfy the QoS requirements. When the QoS parameters are available to the scheduler, it provides EDF scheduling policies for the services constrained by TD95 and ET (application messages) and strict priority scheduling for signalling messages and voice. If the QoS requirements are not available, the scheduler adopts a best effort approach as indicated in 4.6.5.

For the NPDUs based on EDF scheduling, the scheduler computes a target time which is used to drive the EDF scheduler. The scheduler takes into account the estimation of the latency introduced by the propagation delay and GS processing time which is distributed by the GS through the parameter '*rl_path_and_processing_time'*.

The entities forwarded by the congestion control process (NPDUs, retransmitted ARQ blocks and system signalling messages) are scheduled as indicated below:

- Voice traffic, identified as application type equal to *voice* in section 4.6.5, has the highest priority and is transmitted using a strict priority policy. Voice service demands real time constraints, as much as possible a low end to end delay and jitter
- System signalling messages are also scheduled using a strict priority policy with less priority than voice but more priority than other data.
- Data traffic is transmitted using an EDF scheduling policy driven by the TD95 and ET QoS parameters.

Data that is scheduled using an EDF policy are selected for transmission based on target time which is driven by:

- The TD95 QoS parameter when the NPDU can be transmitted within the interval specified by TD95 - *rl_path_and_processing_time'* computed with reference to the arrival of the NPDU.
- The ET QoS parameter when the UT fails to satisfy the TD95 constraint.

The UT de-queues and drops NPDUs, including all their associated LPDUs, whenever it detects that the ET constraint cannot be satisfied.

8.2.4.2.1.3.1 High rate data scheduling considerations

It is recommended that the scheduling priority of the data marked as data high rate should be enhanced (see 4.6.5). For example, FLIPINT application messages in the profile shown in section 4.4.

This could be implemented by driving the EDF scheduler with a special target time (i.e., a time already in the past) or by applying a strict priority scheduling policy.

In the event of adopting the strict priority approach:

- The priority of the high rate data is kept below the signalling priority and voice priority.
- The UT also drops the NPDUs in the case that it fails to transmit it within the interval indicated by the ET QoS parameter.

8.2.4.2.1.3.2 Voice scheduling considerations

The UT buffers at least 4 VoIP datagrams and maps them typically to $RACH\_CR_{160}\_SF_{4}\_DB_{976}$. The first mapped voice LPDU contains the required L2 addresses (i.e., GES ID and UT ID) while the remaining voice LPDUs could use the address reuse configuration (AF='00') to reduce the size of the LPDU header.

Please refer to 8.1.1.1.2.1 concerning latency considerations.

8.2.4.2.2 Encapsulation and ARQ

As in the case of the FWD link encapsulation process, see section 8.1.1.2.1, the RTN link layer encapsulator (L2 module) first has to determine whether the incoming NPDU must be fragmented or not, in order to first set the S and E flags (Start and End bits) accordingly, as described in section 8.7.1.3 of [AD-02].

The AF field allows reusing the addresses specified on the previous LPDU in the same PPDU, e.g., when sending data and signalling LPDUs on the same PPDU to the same terminal.

The ARQ indicates whether the ARQ process is in use or not, as signalled by the network adaptation layer, which deduces whether ARQ is necessary or not by the continuity requirements (QoS parameter) of the incoming NPDU.

The C flag indicates the presence of CRC at LPDUs. As there is already a CRC at the physical layer level, this option is not needed, except if the application requires an integrity higher than 1e-12. The reassembly of fragmented LPDUs can be based on the alternative Seq Number field included when CRC is not used, as described in section 8.2.4.2.2.2.2. Note that the L2 CRC is computed using the same algorithm as the physical layer CRC-32.

In the RTN link, the L2 encapsulator can receive from the network adaptation layer the NPDU to encapsulate with the next QoS parameters attached (out of band), determined by the network adaptation layer for the NPDU: TD95, ET, ARQ need, L2 CRC need. The QoS parameters can be deduced by the network adaptation layer from incoming NPDU DSCP field, transport layer protocol or ports, or by other mechanisms, agreed upon at system level and configured in the

GS element. If the QoS parameters are not determined, the NPDU is treated following a best effort policy.

The UT selects and reuses FID values from the pool of available FID values. Notice that in the case of fragmentation just 8 FID values are available (3 bits used), while in the case of not having fragmentation a 4 bit FID field is used by the ARQ process.

Applications with the same QoS parameters can be assigned the same FID value (belong to the same flow), as the objective of the FID field is to allow the pre-emption of the transmission of packets with different QoS requirements. As there ARQ flag in the RTN link encapsulation, there is no need to get from system tables which RTN link FIDs are to be used with traffic requiring ARQ and which are for traffic not requiring ARQ, not like in the FWD link.

8.2.4.2.2.1 Fragmentation

The ARQ process is executed at LPDU level, i.e., after the fragmentation and before the reassembly of packet upon reception. It is assumed that before reassembly in the receiver, ARQ reorders the received fragments correctly. Re-assembled NPDUs will be delivered to the receiver network adaptation layer in the order of reception of the last fragment of the NPDUs.

The FID field, the Total Length together with the CRC-32 or Seq Number fields provides the possibility for parallel fragmentation and reassembly of several NPDUs.

8.2.4.2.2.1.1 General Process

To perform fragmentation and reassembly, which is required when the NPDU does not fit in the available resource offered in a single physical layer payload, the following rules apply:

1) All fragments carrying data of the same NPDU have the same Flow ID.
2) The first fragment of an NPDU with a given Flow ID has S equal 1 and E bit equal 0.
3) The fragments which are neither the first nor the last NPDU fragment, have S bit equal 0 and E bit equal 0.
4) The last fragment of an NPDU with a given Flow ID has S bit equal 0 and E bit equal 1.
5) All fragments of an NPDU with the same Flow ID are transmitted in order.

Whenever fragmentation is to be applied, assuming that the physical layer offers n_1 bytes of resource in its payload (n_1 being lower than the NPDU total length), the encapsulation process takes the first (n_1-L2_header_length) bytes of the NPDU and performs the following:

- If CRC-32 is used for this NPDU, it computes CRC value using a systematic 32-bit CRC encoder on all NPDU data.
- Forms a fragment with S bit set to value "1", E bit set to value "0".
- Appends the Address Format field with the corresponding value depending on addressing requirement of the message (cf. AF field values described in previous section and in section 8.7.1.3 of [AD-02]).
- Sets ARQ flag to “1” if ARQ is used for this NPDU or “0” if it’s not used, as signalled by the network adaptation layer.

- Sets the Length field to the value (n1-L2_header_length). L2_header_length takes into account the 4 mandatory bytes of the first fragment header + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (consistent with the Address Format field).
- Sets Flow ID to a value associated with the class of service of the corresponding NPDU.
- Appends the appropriate Protocol Type field (see PT field values in section 8.7.1.3 of [AD-02]).
- Sets C flag to 1 if CRC-32 is used for this NPDU.
- Sets the Total Length field to the number of bytes of the NPDU of the un-fragmented NPDU, adding the 4 bytes of CRC-32 field (if present).
- Appends the 2 bytes of the ARQ protocol field if used, not including the 4 bit FID field, as the 3 bit FID field is reused.
- Appends the 2, 3 or 4 bytes of the Source/Dest address field if used. The presence and content of this field depend on the AF field value (see AF field description in previous section and in section 8.7.1.3 of [AD-02]).
- Inserts the first (n1-L2_header_length) bytes from the NPDU data in the L2 Payload.
- Places this fragment into the current L1 frame as the first fragment or directly after any other fragment already present in the L1 frame.

If the resource offered next by the physical layer (n_x) does not permit transmitting the remaining bytes of the NPDU, the encapsulation process performs the following steps:

- Forms a fragment with S bit set to value "0", E bit set to value "0".
- Appends the Address Format field with the corresponding value depending on addressing requirement of the message (see AF field values described in previous section and in section 8.7.1.3 of [AD-02]).
- Sets ARQ flag to “1” if ARQ is used for this NPDU or “0” if it’s not used, as signalled by the network adaptation layer.
- Sets the Length field to the value (n_x-L2_header_length). L2_header_length takes into account the 2 mandatory bytes of the intermediate fragment header + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (consistent with the Address Format field).
- Sets Flow ID to a value associated with the class of service of the corresponding NPDU.
- Appends the 2 bytes of the ARQ protocol field if used, not including the 4 bit FID field, as the 3 bit FID field is reused.
- Appends the 2, 3 or 4 bytes of the Source/Dest address field if used. The presence and content of this field depend on the AF field value (see AF field description in previous section and in section 8.7.1.3 of [AD-02]).
- Inserts the following (n_x-L2_header_length) bytes from the NPDU data in the L2 Payload.
- Places this fragment into the current L1 frame as first fragment or directly after any other fragment already present in the L1 frame.

- If the remaining resource in the L1 payload is shorter than (L2_header_length+1) bytes then padding bits are inserted in the remaining bytes of the L1 payload. L2_header_length takes into account the 4 mandatory header bytes if a first fragment needs to be inserted or the 2 mandatory header bytes if the intermediate fragment needs to be inserted + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (in consistence with the Address Format to use).

The previous steps are repeated by the encapsulation process for all fragments that are not the last fragment (and not the first), i.e., until the remaining quantity of bytes of the NPDU to be transmitted is larger than resource offered by the physical layer.

If the resource offered by the physical layer is equal or larger than the remaining quantity of bytes of the NPDU, the encapsulation process performs the following steps:

- Forms a fragment with S bit set to value "0", E bit set to value "1".
- Appends the Address Format field with the corresponding value depending on addressing requirement of the message (see AF field values described in previous section).
- Sets ARQ flag to “1” if ARQ is used for this NPDU or “0” if it’s not used, as signalled by the network adaptation layer.
- Sets the Length field to the value (n_{x+}L2_trailer_length-L2_header_length). L2_header_length takes into account the 2 mandatory bytes of the intermediate fragment header + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (consistent with the Address Format field). L2_trailer_length is 1 byte if the Sequence Number field is used and 4 bytes if CRC-32 field is used.
- Sets Flow ID to a value associated with the class of service of the corresponding NPDU.
- Appends the 2 bytes of the ARQ protocol field if used, not including the 4 bit FID field, as the 3 bit FID field is reused.
- Appends the 2, 3 or 4 bytes of the Source/Dest address field if used. The presence and content of this field depend on the AF field value (see AF field description in previous section and in section 8.7.1.3 of [AD-02]).
- Inserts the last (n_x-L2_header_length) bytes from the NPDU data in the L2 Payload.
- Append CRC-32 computed in first step if C flag is set to 1 or the corresponding value to Seq Number field if C flag is set to 0. If Seq Number field is used, its value is computed as follows: The sender has a variable counter of sequence number for every Flow ID. The current value of the counter for the corresponding Flow ID is appended to the Seq Number value. Then the counter is incremented by 1 unit. When the counter reaches 255, the next increment cycles back to 0.
- Places this fragment into the current L1 frame as first fragment or directly after any other fragment already present in the L1 frame.
- If there is no other signalling/data to be sent by the UT or if the remaining resource in the L1 payload is shorter than (L2_header_length+1) bytes then padding bits are inserted in the remaining bytes of the L1 payload. L2_header_length takes into account the 4 mandatory header bytes if a first fragment needs to be inserted or the 2 mandatory

header bytes if an intermediate fragment needs to be inserted or the 2 mandatory header + 4 optional CRC trailer if a last fragment need to be inserted + 2 optional bytes of ARQ field (if used) + 2, 3 or 4 optional bytes for the addressing field (consistent with the Address Format to use).

8.2.4.2.2.2 Reassembly

8.2.4.2.2.2.1 Using CRC-32 field

When the last fragment of a NPDU is received (i.e., when S bit is 0 and E bit is 1), the following steps are executed:

- If there is no first fragment having the same Flow ID, all fragments having this specific Flow ID are thrown away.
- If there is a first fragment having the same Flow ID, a first check consists of verifying that the Total Length indicated in the first fragment corresponds to the sum of the Lengths of all the fragments since the last first fragment having the same Flow ID. If this is not the case, all fragments since the last start fragment are thrown away.
- If the previous step shows that the Total Length corresponds to the sum of the Lengths of all the fragments since the last first fragment having the same Flow ID, the CRC-32 field is used to check the integrity of the re-assembled NPDU. If the CRC detects corruption, all fragments since the last start fragment are thrown away; otherwise it is passed to the network adaptation layer.

8.2.4.2.2.2.2 Using the seq number field

The sender and receiver perform the following steps when using the sequence number method:

- The sender and the receiver need to initialize their respective sequence number counter for every Flow ID to the same value (e.g., to the value 0). This has to be done at logon and during the handover process to perform a seamless handover from the encapsulation point of view.
- As explained in the description of the fragmentation, the sender has a variable counter of sequence number for every Flow ID. When emitting an end fragment, the encapsulation process appends the value of this counter to the fragment tail and increments the counter. When the counter reaches 255, the next increment cycles back to 0.
- The receiver has a sequence number counter as well for every Flow ID. This counter contains the sequence number expected in the next packet (in Seq number field). When the receiver gets the last fragment of a packet it does the following:
    - If there is no first fragment having the same Flow ID, all fragments having this specific Flow ID are thrown away.
    - If there is a first fragment having the same Flow ID, the reassembly process checks that the Total Length indicated in the first fragment corresponds to the sum of the Lengths of all the fragments since the last first fragment having the same Flow ID. If this is not the case, all fragments since the last start fragment are thrown away.

- It compares the Seq Number field of the fragment to the counter for the given Flow ID. If they differ, all the fragments since the last start fragment are thrown away. If they are the same, the packet is passed to the upper layer.
- It sets the counter for the corresponding Flow ID to the value of sequence number received +1 with a cycle every 255.

8.2.4.2.2.3 ARQ considerations

It is strongly recommended that critical services such as FLIPINT use an ARQ block size of 1 LPDU to speed up as much as possible the retransmission process. Although other services can increase the size of the ARQ block (i.e., to 2 or 3), it is recommended that they should be kept to 1.

As indicated in D018-COM-FUN-1690, the transmitter selects all the LPDUs out of the *active_block* for retransmission when the *ReTx_Timer* of the *active_block* triggers and forwards it to the congestion control module, see 8.2.4.2.1.2.

When there is an opportunity to transmit an ARQ ACK, the transmitter checks for queued ARQ ACKs and transmit them. In the case that more than one ARQ ACK belonging to the same NPDU is queued, it is recommended that the scheduler only transmits the ARQ- ACK with the largest fragment counter value. The others are silently discarded.

In order to speed up the transmission of the ARQ-ACKs, the transmitter can concatenate (*piggyback*) ARQ-ACKs signalling messages to other LPDUs that are mapped to a PPDU payload in the event that there is room enough in the RACH burst's payload.

8.2.4.2.3 Carrier and Power randomization

Before the PPDU payload is forwarded to the physical transmitter, the following procedures are performed:

- The UT selects one carrier among the subset of carriers in which the selected RACH burst can be transmitted. This is done by randomly selecting one of the carriers in the subset following a uniform distribution.

  Note: Those carriers that are disabled through *traffic_status* equal to '*Channel Disabled*' are not considered.

Once the carrier is selected, the UT computes:

- The *lower_bound_power_dB* parameter associated with the selected RACH burst type (*rl_npdu_rach_id)* is identified. A random power factor '*k_dB*' value is then computed as indicated in D018-COM-FUN-3295 and tagged to the PPDU payload when forwarding it to the physical transmitter.
- The UT randomly selects the pair of '*preamble_sequence_generator*' and '*scrambling_sequence_generator*' to use to transmit the burst:
  - For the selected carrier (RTN_channel_id) and RACH burst (RACH_config_id), the UT resolves the set of '*preamble_scrambling_struct*' that is available. This is done by inspecting the signalling structures *RTN_channel_configuration* and *RACH_burst_configuration* distributed by the GS.

- Out of the available '*preamble_scrambling_struct*', the UT randomly selects one pair of '*preamble_sequence_generator*' and '*scrambling_sequence_generator*' values following a uniform distribution. These values are then tagged to the PPDU payload when it is forwarded to the physical transmitter.

8.2.5 Performances

Using the reference scheduler and an estimation of the ECAC traffic during the 6 hour peak, the following reference performances can be obtained:

The reference number of carriers required to satisfy the CoS of the reference traffic profile under the aeronautical propagation channel are shown in Table 8-6. In this table the latency delay introduced is also shown:

Reception Mode	SIC Efficiency	SF=16 channels	SF=4 channels	Special channels
Enhanced	95%	3	4	1
Enhanced	80%	3	4	1
Enhanced	70%	3	5	1
Basic	80%	4	5	1

Table 8-6: Reference number of carriers required in the RL

The voice latency obtained is in the range of 400 to 570 ms with a 99.9%.

The 95th percentile of the FLIPINT message is below 2.9 s.

9. NETWORK LAYER AND INTERFACE WITH UPPER LAYERS

9.1 ATN/OSI components and interfaces

For the ATN/OSI protocol stack, the satellite communication system acts effectively as a pass-through for 8208 encapsulated OSI data. ATN/OSI Network Registration is performed using IDRP and ISH protocol exchanges between the Airborne Router (AR) and the Air-Ground OSI Router (AGR). Data transfer involves the use of the Connection-Oriented 8208 Protocol (i.e., the establishment of virtual circuits). All data, including IDRP and ISH data, is transferred using 8208. The 8208 protocol provides a reliable transfer over the Air-Ground link, which is already provided by the link layer, but which is supported mainly for legacy reasons.

This section provides guidelines regarding how ATN/OSI support has to be implemented, both at aircraft and GS level, describing a set of reference components. It also provides a brief overview of the interworking between the network layer related protocol functions within the different system elements.

9.1.1 Protocol overview

The SATCOM system informs the UT about the 8208 address of the AGR during the logon procedure. This information is conveyed to the AR by the UT using a JOIN Event.

This triggers a Route Initiation procedure in the AR for this new link. The AR Mobile SNDCF issues a CALL REQUEST to establish a virtual circuit (VC) to the 8208 address of the AGR. The CALL REQUEST includes a logical channel identifier (LCN). The UT-OSI-Adapter (see Figure 9-1) uses this LCN to identify the specific virtual circuit (VC) and the associated link. Subsequent 8208 packets for the same VC are identified by the LCN.

The CALL REQUEST includes an Intermediate System Hello (ISH) data unit using the 8208 Fast Select facility. The ISH contains the L3 address (NET) of the AR.

The UT forwards the ISO/IEC 8208 packet to the GES, over the SATCOM system. Within the GES, the packet is forwarded to the AGR using the 8208 Called-Address as identifier. The same AGR will have separate identifiers to identify each Service Provider.

The peer 8208 layer in the AGR receives the CALL REQUEST and generates a CALL ACCEPT packet which is forwarded back by the GES to the UT, over the same link where the CALL REQUEST was received. The CALL ACCEPT packet contains an ISH with the L3 address (NET) of the AGR. L3 routing information is finally exchanged over 8208 using the IDRP routing protocol.

After this process, data exchange between the AR and the ATN/OSI based terrestrial network is possible.

During a handover, several 8208 virtual circuits are maintained in parallel, until the old link is released. The AR may continue to send 8208 Data Packets over the old VC until a CALL ACCEPT has been received for the new VC and routing data has been exchanged.

9.1.2 ATN/OSI reference components

Refer to Figure 9-1, which identifies reference components of an ATN/OSI implementation:

- AR-OSI-Adapter

- UT-OSI-Adapter
- UT-Core
- GES-Core
- GES-OSI-Adapter
- AGR-OSI-Adapter

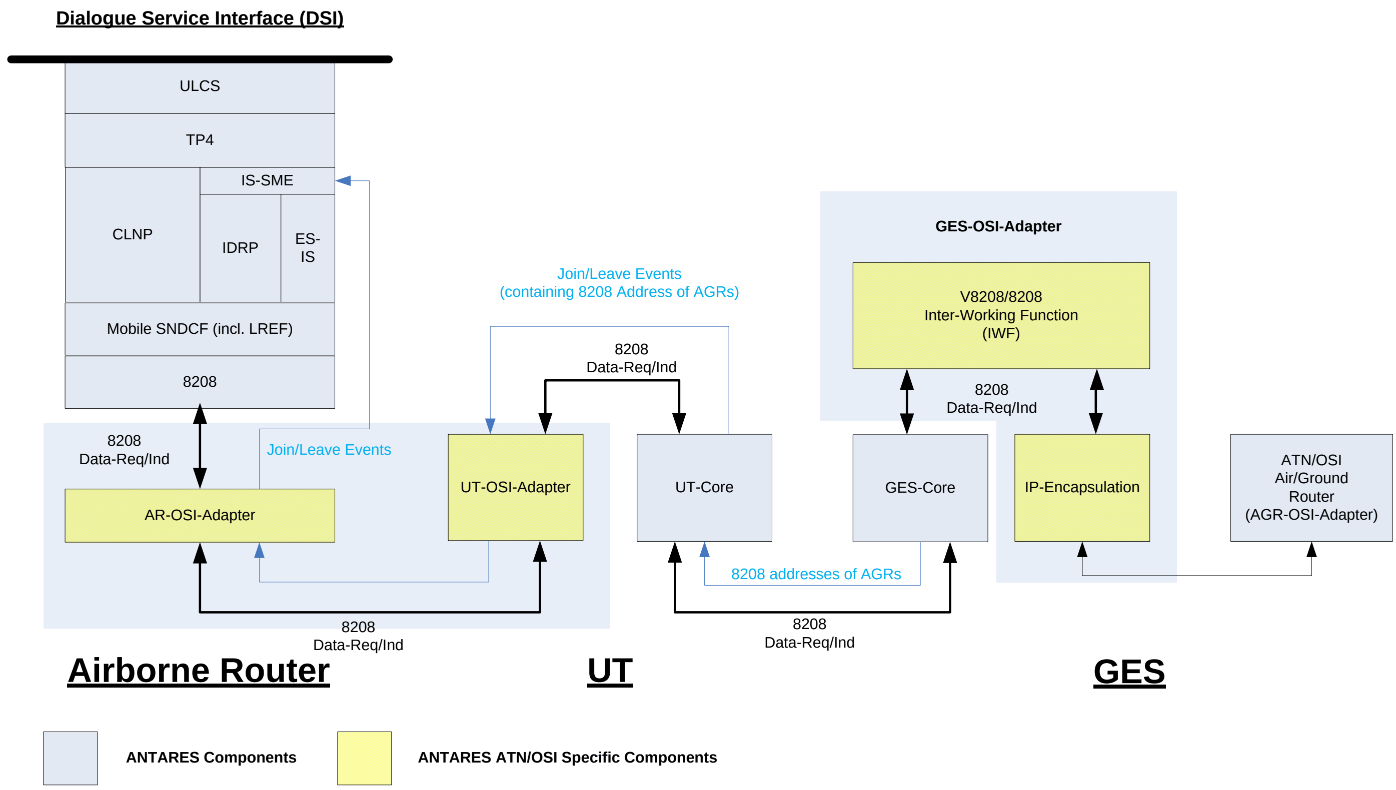


Figure 9-1: Network layer components (with AR-OSI-Adapter located in the AR)

Support for the ATN/OSI protocol stack requires an AGR compatible with existing VDL Mode-2 implementations, as VDLM2 Mobile SNDCF is used.

9.1.2.1 AR-OSI Adapter

The AR-OSI-Adapter is software located in one the following:

- AR
- Avionics unit which sits between the AR and UT
- UT

It is used to provide sub-network support for an AR and supports the following functions:

- Join Event handling

- Leave Event handling
- Data Transfer of 8208 packets

The AR-OSI-Adapter converts Logon/Logoff indications from the UT-OSI-Adapter into Join/Leave events directed to the IS-SME function within the AR.

It should readily adapt to existing VDLM2 AR, but this depends on the specific airborne implementations.

The AR-OSI-Adapter interfaces to the v8208 protocol layer within the AR. There are 2 approaches:

- Incorporate the AR-OSI-Adapter into the AR.
- Interface to the AR using the existing VDR/A429W3 Interface.

The second approach involves terminating the VDLM2 AVLC/LME/VME layer (i.e., the VDLM2 link layer) in the AR-OSI-Adapter and incorporating this element in the UT (or in a device between the AR and UT). It would be required to spoof the AVLC/LME/VME of a ground station including the generation of signalling (GSIFs messages) and handling identification (XID) exchanges. This will permit ANTARES to work with existing ARs without any changes to the AR.

In any case, for the sake of clarity in the following sections it will be assumed that the AR-OSI-Adapter is located in the AR.

9.1.2.2 UT-OSI-Adapter

The UT-OSI-Adapter provides support for inter-connecting the AR to the UT-Core. It supports:

- Generation of Join Events
- Generation of Leave Events
- Data transfer of 8208 packets
- Mapping between the CS Link Layer (L2) and OSI Layer 3

The Join and Leave Events are generated when a UT logon/logoff or handover occurs.

All downlink data transfers received from the AR-OSI-Adapter over A429W/AFDX consist of 8208 packets. The 8208 packets are then passed to the UT-Core A429W/AFDX for framing and downlinking.

All uplink legacy data transfers received from the UT-Core consist of 8208 packets. The 8208 packets are passed to the AR-OSI-Adapter over A429W/AFDX.

It also maps 8208 packets to the generic ANTARES link layer channels.

9.1.2.3 UT-Core and GES-Core

The UT-Core and GES-Core must support:

- OSI network registration
- OSI data transfer

The network registration support for OSI means that the UT-Core and GES-Core exchange OSI addressing and other information during logon/logoff and handovers.

The addressing information required is:

- AR 8208 address (provided by the AR)
- AGR 8208 address(es)
- Optionally, Preferred Service Provider Identifier (provided by the AR)

The Preferred Service Provider Identifier is used to route traffic from an aircraft to the appropriate ground-ground router, using Source Address Routing. The Preferred Service Provider can be selected by the AR as is currently the case.

OSI Network Registration Support will be restricted to the GES-Core advertising available 8208 AGR addresses, as is the case with VDL Mode-2 Networks.

9.1.2.4 GES-OSI-Adapter

The GES-OSI-Adapter handles 8208 data. Data transfer can be handled in the same way 8208 data is handled in a VDLM2 Ground Station: An inter-working function can inter-connect the airborne 8208 connections with connections to the AGR. However, this inter-working functionality is not strictly needed by ANTARES and would be supported only for legacy reasons.

Currently there are different types of VDL ground station to AGR interface supported by the three providers of ground stations, namely ARINC, SELEX-Communications and SITA. These interfaces are proprietary and are not standardised. The specification of this interface is also outside of the communication standard.

The protocol between the GES and AGR must be capable of separating the data streams for the different Air/Ground virtual circuits. It must ensure that uplink messages are sent to the correct aircraft on the correct channel.

One method is to use a single TCP/IP connection between GES and AGR. A header identifying the source/destination (link identifier) is added to each 8208 packet. Mapping tables in the GES and AGR are used to route to the correct aircraft and AGR 8208 address.

9.1.3 Redundancy support for ATN/OSI

The OSI RESET signalling message may be used to support GS redundancy. When a fail-over/switch-over occurs, the GS should increment the *Reset_counter* and broadcast the OSI RESET message on all forward link carriers assigned to the failing GES several times (to assure that all UT receive this message correctly). All retransmissions will carry the same *Reset_counter* value.

The CS specifies that upon reception of this packet, the UT locally triggers the reset of the 8208 virtual connections associated with the link where this OSI RESET was received. This may be done using the following method:

- When the UT receives a valid OSI RESET packet, the AR-OSI-ADAPTER could generate an 8208 RESET INDICATION (8208 Packet Type PTI = 0x1B with cause =0x00 and diagnostic code = 0x00) locally for each active 8208 Virtual Circuit and send it to the AR.

The UT should intercept the RESET CONFIRMATION (8208 Packet Type PTI = 0x1F) response from the AR and not transmit it over the satellite link.

9.1.4 QoS support for ATN/OSI

Currently, ATN/OSI applications use a single QoS.

Supporting multiple QoS requires multiple Virtual Circuits, one for each QoS type. It is not possible to have multiple QoS over a single VC because each QoS requires a separate data stream and separate 8208 packet acknowledgements.

The AR software would make a separate CALL-REQUEST for each QoS supported.

Two methods could be considered for identifying the QoS:

a) Include a QoS Parameter in the CALL-REQUEST. The QoS Parameter would be encoded as a priority facility code in the CALL-REQUEST packet to indicate Class of Service as follows:

- Facility Code 1 1 0 1 0 0 1 0 (0xD2)
- Length 0 0 0 0 0 0 0 1 (0x01)
- Class x xxxxx xx x

b) Identify the QoS by the Calling-Address. So some method should be agreed upon regarding how to encode the QoS in the Calling-Address.

On the airborne side, the AR-OSI-ADAPTER would maintain a mapping between VCs and QoS class. The VC is identified by the Logical Channel Number (LCN) in the 8208 header. The adapter would need to examine each CALL-REQUEST to obtain the LCN/QoS mapping and insert an entry in the mapping table. For 8208 packets other than the CALL-REQUEST packet, the QoS class would be determined by the LCN. The adapter would indicate the appropriate QoS class identifier with each 8208 packet sent to the UT. This QoS class identifier should be translated into those QoS parameters relevant for the CS (TD95, ET, etc.).

On the ground side, the AGR would extract the QoS from the CALL-REQUEST and it would maintain a mapping table.

At the CLNP routing level, the ATN security parameter in the CLNP packet would be used to determine the QoS and map it to the correct 8208 Virtual Circuit. The QoS for each CLNP packet is identified solely by the ATN-Security label, as defined in the ICAO SARPS.

9.1.5 Interactions between 8208 and CS link-layer ARQ

In the CS, both the CS link layer and the 8208 packet layer perform ARQ. The v8208 protocol assumes that the underlying layers will lose some NPDUs due to resets at frame level. If this happens, then the v8208 layer receiving side will issue a REJECT message and this will cause the sending side to re-send the missing packet.

At the link level, the CS uses a more efficient method for requesting retransmissions and thus the retransmission mechanism at the upper layer is usually not needed.

Interference between both retransmission mechanisms is unlikely assuming that retry timers at the v8208 level are far higher than retransmission time-outs defined at the CS Link Level. Typical v8208 timers are as follows:

Timer	Granularity	Min	Max
T20 (Restart Request Response Timer)	Seconds	60	360
T21 (Call Request Response Timer)	Seconds	60	360
T22 (Reset Request Response Timer)	Seconds	60	360
T23 (Clear Request Response Timer)	Seconds	60	360
T27 (Reject Response Timer)	Seconds	60	360

Table 9-1: v8208 Timer Values

9.2 ATN/IPS components and interfaces

9.2.1 Reference ATN/IPS end to end architecture

Figure 9-2 shows a possible ATN/IPS end to end architecture, which has been taken as reference for the CS specification:

- The reference model considers a multi mobile node scenario, i.e., that the SATCOM system provides, through the UT, more than one interface at IPv6 level, each of them with a different IP address.
- Mobile nodes (MN) are connected to the UT through an ARINC 664 (AFDX) link. It should be noted that other implementation options would be equally feasible (different physical interfaces per attached mobile node, support for PPP over Ethernet between the MN and the UT, etc.).
- It is assumed that aircraft network adaptation layer functions are fully implemented within the UT. Again, other implementation options would be equally feasible (especially if the MN is actually a router).
- UT normally communicates with a single GES, except during a handover situation, with a temporary connectivity through two GES.
- The A/G router, which is part of the GES, interfaces the G/G routers, which are part of the ground IPS network.
- The Home Agent (HA) is hosted by a mobility service provider, which can be accessed by the MN through the ground IPS network.
- The correspondent node (CN) is normally located in the ATC/AOC centres.

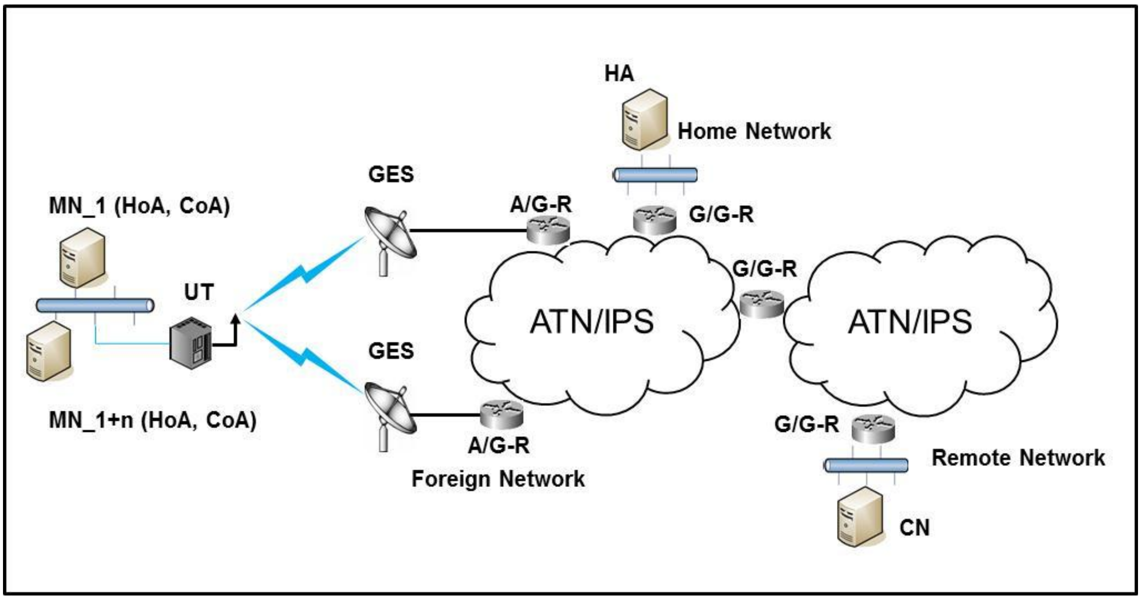


Figure 9-2: ATN/IPS end to end architecture

9.2.2 Recommendations regarding ROHC configuration

A number of guidelines are given below regarding the configuration of the ATN/IPS header compression scheme supported by the CS, the Robust Header Compression protocol (ROHC). It should be noted that, while these recommendations do not affect interoperability, they avoid a negative impact of ROHC on the system performance in terms of continuity and integrity degradation due to compression context error propagation and to signalling overheads.

Refer to RFC3095 for a more detailed description of the mentioned modes and configuration parameters.

It is recommended to:

- Exploit the enhanced performance of the associated ROHC instances.
- Use piggybacked or interspersed feedback information.
- Use U-O-mode for unicast IP packets. No recommendation for R-mode is made, because of higher associated feedback overheads and because U-O-mode is simpler and already achieves a good performance in the event of context damage.
- Use U-mode only for broadcast and multicast IP packets. However, header compression should generally not be utilized for ICMPv6, multicast and neighbour discovery protocol link local signalling, because of low compression efficiency compared to achievable compression efficiency for user traffic.
- Use a constant confidence value L or alternatively a variable confidence value L as a function of PLR (the more severe the PLR, the higher the L value). With L=3 at PLR=10^{-5}, a context damage probability lower than 10^{-15} can be achieved.
- Set the k and n values to “1” for a strong error detection scheme.

These recommendations are summarized in the table below:

Parameter	Description	Recommendation	Value
Operation mode	ROHC operation mode	a) Utilize the U-mode for broadcast and multicast IP packets b) Utilize the U-O-mode for unicast IP packets	a) U-mode b) U-O-mode
L-value	Confidence Value. No. of uncompressed ROHC packets sent consecutively by compressor, before establishment of full context state at decompressor can be assumed	a) Set L-value according to worst case PLR, if ROHC library requires a constant value. b) Implement L-value as a function of system PLR.	a) L=3, if PLR $<10^{-5}$ b) L=3, 10^{-6}<PLR<10^{-5} L=4, 10^{-5}<PLR<10^{-4} L=5, 10^{-4}<PLR<10^{-3} L=7, 10^{-3}<PLR<10^{-2}
IR-T	Periodic refreshment of the context in U-mode	Implement IR-T as a max no. of consecutive compressed packets	IR-T=100 packets
k, n values	k_1, n_1 and k_2, n_2 rules for context damage recovery mechanism Refer to section 5.3.2.2.3 of RFC3095.	Set the k, n values to “1”.	k_1 = n_1 = k_2 = n_2 = 1
Feedback	Feedback information in event of unsuccessful verification of compressed ROHC packets or negative validation of decompressed packets	Send feedback information as piggybacked or interspersed in both directions	Piggybacked or interspersed

Table 9-2: Recommendations for ROHC configuration

9.2.3 Acceleration of the IPv6 configuration process

The GS can accelerate the aircraft IPv6 configuration process by transmitting a Router Advertisement (RA) message (with contents as defined in RFC4861) over the FWD link just after the UT has established the link at L2. This RA message includes all configuration information relevant for L3 (network prefix, MTU, address configuration method, etc.).

The RA could be sent just after a successful logon completion or, in the case of an HO implying a change in network attachment point, when the new link has been established at L2 (refer to the *L2 UP* event included in the HO message sequence charts in [AD-02]).

Support of this feature allows reducing the periodicity of the (unsolicited) Router Advertisement messages that have to be sent regularly by a GES.

9.2.4 Address resolution guidelines and clarifications

Address resolution is required to provide a mapping between the destination L3 addresses included in the NPDU and the link-layer addresses used in the satellite system. These link-layer addresses are the UT ID or GSE ID used in the encapsulation headers and the satellite MAC addresses.

9.2.4.1 Resolution of the GS link-layer addresses

The GS satellite MAC address is provided at L3 by including a *Source Link-Layer Address* option in the Router Advertisement messages sent by the GS over the FWD link (periodically, if unsolicited, or as a response to a Router Solicitation message, refer to RFC4861). The aircraft will normally handle a single destination GS satellite MAC address (the one of the assigned GES), except during handovers.

The mapping between the GS satellite MAC address and the GSE ID is provided through logon/handover signalling.

9.2.4.2 Resolution of the UT link-layer addresses

The GS can obtain the UT ID associated with a certain destination CoA or link-local IP address directly, without the need for extra protocol exchanges and overheads, following the following method:

1. A CoA or link-local IP address is mapped to the UT ID once the first NPDU containing this IP address as source address is received over the return link. This is possible as the CS link layer includes the UT ID in the encapsulation header.
2. When an NPDU has to be forwarded to a UT, this mapping table is checked, looking for the UT ID associated with the destination CoA or link-local IP address. The NPDU is then forwarded to the correct UT.

Step 1 is always possible before step 2, because of the way addresses are configured in IPv6 (i.e., the first NPDU containing a new IPv6 address as source address is always transmitted first by the aircraft).

In any case, the CS does not impose this mechanism for address resolution. So the GS could also resolve the UT address by multicasting a Neighbour Solicitation message, as described in RFC4861. In this case, the UT destination satellite MAC address is provided through a Neighbour Advertisement message (also defined in RFC4861) transmitted by the aircraft.

The mapping between the UT satellite MAC address(es) and the UT ID is established through the logon/handover signalling.

9.2.4.3 Support for multiple satellite MAC addresses per UT

The CS provides support for scenarios where the aircraft IPv6 network layer uses multiple satellite link-layer interfaces although there is a single UT. Up to four different link-layer interfaces can be supported.

Each interface has its own UT satellite MAC address and can handle its own L3 processes (i.e., can have a different IPv6 address and use its own IPv6 configuration). However, the UT handles a single L2 and L1 process, common to all link-layer interfaces, and the GS considers the UT as a single entity.

The GS will be informed of these multiple UT satellite MAC addresses through the logon/handover signalling and therefore will be able to interpret correctly that an NPDU directed to one of these MAC addresses is destined to the associated UT. These MAC addresses will be the ones included as *Link-layer Address* in the NS/NA signalling messages defined in RFC4861.

An NPDU received by the aircraft over the satellite link does not carry any identifier of its associated UT satellite MAC address (the GS just identifies the destination UT by using the UT ID). So, on the FWD link, selection of the correct link-layer interface is done locally in the aircraft, following the following method:

1. A link-local or CoA IP address is mapped to a UT satellite MAC address once the first NPDU containing this address as source address is transmitted over the air interface.
2. Upon receiving an NPDU over the forward link, this mapping table is checked, looking for the satellite MAC address associated with the destination link-local or CoA IP address. The NPDU is then forwarded to the correct link-layer interface.

Step 1 is always possible before step 2, because of the way addresses are configured in IPv6 (i.e., the first NPDU containing a new IPv6 address as source address is always transmitted first by the aircraft).

This process is performed before header compression (step 1) and after header decompression (step 2) and is only necessary if there are multiple link-layer interfaces.

10. CONTROL PLANE

This section provides implementation guidelines for the handover and log-on procedures. Guidelines for other control procedures or mechanisms, e.g., ACM, RRM and synchronization, are provided in other sections. In particular, refer to section 6 for synchronization, section 7 for ACM and section 8 for RRM.

10.1 Handover Control Procedure

10.1.1 Handover detection

A handover can be triggered by:

1. The UT when it detects that it is losing coverage with its current beam/satellite and a new beam/satellite is available.
2. The system when it needs to perform an HO because of resource allocation issues, management/administrative/political/business and manual reasons (e.g., flight crew decisions, administrative decisions, maintenance tasks, etc.), redundancy considerations, etc.

This section provides recommendations for the first case (i.e., when the HO is triggered by the UT due to degradation of the current beam or satellite). As the system cannot rely on external navigation or positioning systems to perform the HO detection, it is necessary to devise a reliable mechanism adapted to mobile environments for accurately estimating the link quality of the signal received from the current beam and the ones from the adjacent beams/satellites.

Likewise in ACM, the main hurdle in implementing an HO detection mechanism based on Link Quality in an aeronautical environment is that there is not an univocal relation between the Link Quality (PER) and the estimated SNIR. The proposed solution to this problem is to solve it by using a mechanism which relies on the slow ACM mechanism (estimation of most spectral efficient MODCOD supported by a UT based on PER measurements at $N_{reduced}$LDPC decoder iterations).

This mechanism requires that the UT is able to receive at least 2 independent FLCs simultaneously; i.e., it must have at least 2 receivers (from now on the term “receiver” is used to refer to the capability of receiving one FLC):

- the first receiver is devoted to receiving the FLC assigned to the UT in the serving beam,
- while the second receiver is fully devoted to HO detection functions.

It is worth mentioning that both receivers have the capability of demodulating and decoding the FCH bursts in the FLC carriers (both receivers have the same capabilities).

The possible HO detection mechanism based on Link Quality measurements should be as follows:

- On one hand, the first receiver should track, demodulate and decode the FCH bursts transmitted through its assigned FLC and should adjust its preferred MODCOD according to the ACM mechanism proposed in section 7. Thus, at any instant, the UT knows the preferred MODCOD in its serving beam and the measured PER at $N_{reduced}$ LDPC decoder iterations.

- On the other hand, the UT, based on the information provided through the system tables, should tune its second receiver in order to detect FLC carriers from adjacent beams/satellites. For each of the detected FLC carriers, the UT should determine the most spectral efficient MODCOD is able to receive while assuring the target PER. This can be done by using the state machine reported in section 7.1.3 for the slow ACM mechanism. This means that:
  - The PER thresholds ($PER_{Thr\_Red\_\#N}$) and $N_{reduced\_\#N}$ LDPC Decoder iterations are determined as reported in section 7.1.1.
  - The Link Quality estimator (PER measurements) reported in section 7.1.2 is valid.

  Thus, for each adjacent beam/satellite, the UTs know the most efficient MODCOD it is able to receive keeping the target PER and the measured PER at $N_{reduced}$ LDPC decoder iterations.

  It is worth mentioning that, during this HO detection phase, the second receiver is not supposed to forward frames of adjacent beams/satellite carriers to upper layers.

- The HO detection mechanism should trigger an HO recommendation when the spectral efficiency of one of the adjacent beams/satellites is better than the current beam (i.e., when the preferred MODCOD of one of the adjacent beams is more efficient than the one in the current beam). The following information should be provided to the HO decision module for the current beam carrier and for all adjacent ones:
  - Preferred MODCOD
  - Number of correct packets after reduced number of LDPC decoder iterations for each MODCOD available on FLC (only those reliably estimated).

  These two information sets are proposed to be used as *Primary_link_quality* and *Additional_link_quality* fields in the HO recommendation message.

  It is worth noting that, in order to cover most of the beam/satellite HO scenarios, the second receiver should be able to monitor the Link Quality of 3 neighbour beams/satellites.

  In order to prevent loop instability (switch back and forth between 2 beams/satellites) a transitory state should be implemented in which it is forbidden to trigger another HO recommendation.

The HO detection mechanism reactivity time can be improved if, in addition to the preferred MODCOD of each FLC carrier, it also uses the measured PER after reduced number of LDPC decoder iterations to trigger an HO recommendation. It is mandatory to take into account the PER after reduced number of LDPC decoder iterations in order for seamless handover to take place in the situations where the destination beam uses only the most robust MODCOD or ACM is not supported.

Depending on the number of receivers of the UT (understanding this, as explained above, as the capability to fully receive several FLC), the HO detection process is deactivated during the HO execution process to devote the second receiver to the new link and allow parallel tx/rx over old and new links.

10.1.2 Handover Decision

The handover decision function is located in the GS, and should take the decision of the HO target destination, including target channel, beam and satellite, and GS target entities (or even deny HO operation) based on the following information when available:

- HO type and priority.
- Current resource availability (load level, congestion indications, etc.) and operational mode (active, degraded mode, etc.) of HO satellite channels and GS entities targets.
- Signal measurement results (HO candidate link quality).
- Aircraft position-based information (location within beam coverage area, aircraft speed and trajectory).
- GS and UT administrative/business/political preferences (i.e., preferred service provider for the aircraft or airline), possibly linked to aircraft position.
- Expected loss of satellite visibility, because of satellite movement (HEO/MEO satellites).
- UT elevation angle (HEO/MEO satellites), if aircraft position is available.
- Network layer preferences (based on network topology criteria, i.e., select GES with shortest path to destination).
- Management or maintenance instructions.
- Other (flight plan information).

This functionality requires real-time knowledge of system status and may imply communication with the different target candidates.

The HO decision may be taken also by the UT for the specific case of switching to a preferred SSP, by a manual trigger of the air crew, for example if flying over a given airspace. In this case, the UT executes the Direct LOGON procedure for the HO.

10.1.3 Handover Execution

The following cases regarding individual HO execution process have been defined:

HO Type	Description
1	Change of satellite service providers (SSP)
2	Beam/channel/satellite change within same GES and GES HO
3	Fast HO within same GES

Table 10-1: Individual handover types

Type 1 covers the case of handover between different Satellite Service Providers, which implies a change of NCC in which a logon process is required, and also all other cases in which the NCC changes even within the same SSP.

Type 2 covers all channel, beam, or satellite changes in which the NCC remains unchanged,with or without GES change.

Type 3 is a particularisation of the type 2 handover that may be applicable for cases in which GES does not change and only if L2 layer processes are implemented at GES as a unified process and not at modulator/demodulator level.

In addition to the 3 handover types defined previously, the UT may proceed by performing a direct logon into a new preferred SSP, triggered by manual air crew selection or by other APB reasons. In this case, after the successful logon in the new SSP, the UT proceeds to parallelise the tx/rx through old and new links until there is no pending traffic to the old link and the handover can be completed.

All the above handover procedures may be shortened depending on the architecture of the GSE involved in the handover. For the collocated NCC/GES (CNG) case or in the case of available ground to ground communications between GSE, some of the messages in the protocol become optional, the GS having the capability to signal to the UT if these messages are required or not.

The HO execution is based on a make-before-break approach, with the UT transmitting and receiving in parallel simultaneously through previous and new links. The procedures defined for HO execution rely on emptying L2 buffers towards the previous GSE before releasing the old link. A trigger to start the L3 update process when the new L2 link is available is defined, together with a timer to give enough time to complete the process before releasing the old link. Current L3 implementations for mobile IPv6 may not take advantage of this feature and start L3 update only when the old L2 link has been released. Nevertheless, this will allow improvement of the overall process for future enhanced IPv6 mobility detection techniques.

The HO execution phase is expected to be completed in less than 2 minutes for SSP change, GES and direct logon handovers for most of the cases, while fast handover is expected to be completed in less than 20 seconds in most of the cases.

In addition to the individual HO being defined, the bulk HO will be performed when all traffic in a serving satellite will be switched to a new satellite.

This process of bulk HO can be performed almost totally transparent to the UT involved, since only the Doppler pre-compensation and the ACM mechanism are affected (potentially, depending on the specific scenario). To cover this scenario, the GS can signal the UT that a reset in Doppler pre-compensation and ACM process is required, by means of two dedicated bits in the FWD_DD header.

10.2 Terminal Registration Control procedure

10.2.1 Logon reject

When the logon is stopped or rejected (i.e., UT not recognized), no message can be sent back to the UT. This permits the mitigation of the effects of malicious users' logon attempts. If there is a deficient transmission of data, a new attempt is made after a timer expiration. The attempt failures might be logged in the management system and an error might be raised after a certain number of attempts (this is a management matter).

10.2.2 Use of control code field

The CS defines an optional "Control Code" field in a number of logon signalling messages transmitted by the GS, namely: *Logon Initial Accept* and *Logon Validation Accept*. This field is not processed by the UT, which only replicates it in its own messages (*Logon Validation Request*, *Logon Accomplished ACK*) without modifications. This section provides guidelines regarding the purpose and use of this field.

The logon procedure supports scenarios with a decentralized GS architecture where GS elements are not connected through a WAN and thus cannot coordinate during a logon process. In this case, the UT itself is used to convey information between the different ground elements.

To provide a certain protection to the ground network against non-authorized users and mal-functioning UTs in this scenario, an optional system of key exchange can be included in the process, based on the "Control Code" field mentioned above.

The principle would be based on the following:

- The public keys of NCC and GES are shared and known by all the ground stations.
- In the *Logon Initial Accept* message, the NCC provides to the UT a control code encrypted with the GES public key.
- The UT provides this control code to the GES during the logon process (*Logon Validation Request* message). At this stage the GES can control if the UT has already performed its registration on the NCC.
- When this is checked, the GES provides to the UT a new control code encrypted with the public key of the NCC (*Logon Validation Accept* message).
- At the end of the logon process, the UT provides to the NCC this control code by using the *Logon Accomplished ACK* message (which is an optional message that should be enabled in this scenario).
- The NCC can decode this control code and can thus validate that this UT has correctly registered with the GES.

The identification of the FWD link resources assigned to the UT (FWD carrier identifier) would be included as part of this control code and would be decoded by the GES to obtain information about the FWD link resources assigned to the UT by the NCC.

An equivalent method is applied for the "Control Code" field included in log-off messages (*Logoff Request ACK*, *Logoff Confirm*).

10.2.3 Initial logon carrier detection

The logon procedure included in the CS specifies the behaviour of the UT once it has been able to detect a logon carrier (LOGON CARRIER STATE). In order to reach this state from the LOGOFF state, the UT can follow different approaches, depending on the configuration data entered in the UT:

- If the UT has been pre-configured with a set of logon parameters (as defined in requirement D018-COM-FUN-2500), it will try successively to lock each of the potential logon carriers until it succeeds or fails. A carrier is considered locked when the UT is able to receive data correctly (correct CRC).

- If no pre-configured data is available or the previous method fails, then it can follow the "cold start" process described below. It should be noted that, in any case, this process is optional.

10.2.3.1 Cold start process

The cold start mechanism assumes that the UT is attached to a module able to detect and analyze the carriers available in a given frequency band. In the rest of the section this module will be called the Signal Detector and Analyzer (SDA).

This module will receive the same RF signal as the UT (a 3dB divider could be used to ease the physical integration).

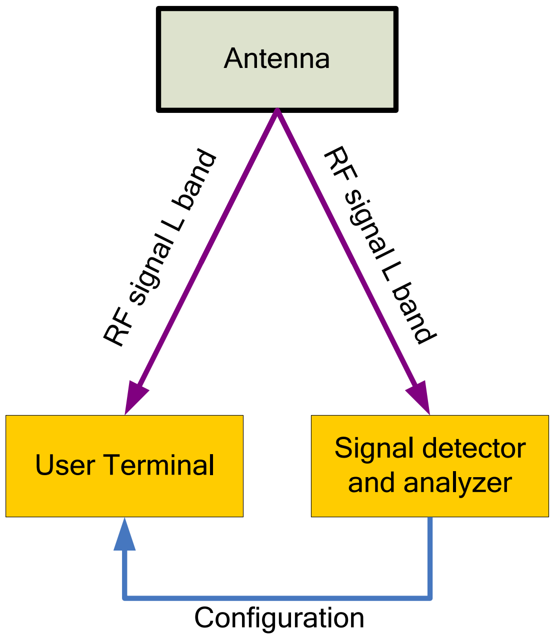


Figure 10-1 – SDA implementation

The cold start process would be the following:

1. The SDA performs an RF acquisition over a period of time (using maximum hold acquisition).
2. On this signal acquisition, the SDA will detect the carriers in term of centre frequency and bandwidth in the given frequency band.
3. The SDA will select the possible carriers that use the CS (with its specific bandwidth, carrier rates, modulation type, etc.), and will reject other carriers.
4. The SDA configures the UT with the FWD carrier frequency and rate and waits for signal lock and its identification as logon carrier.

5. If the previous step fails, the signal analyzer performs the same operation on the second carrier candidate and so on until the process is successful.

This process is illustrated in the following figure:

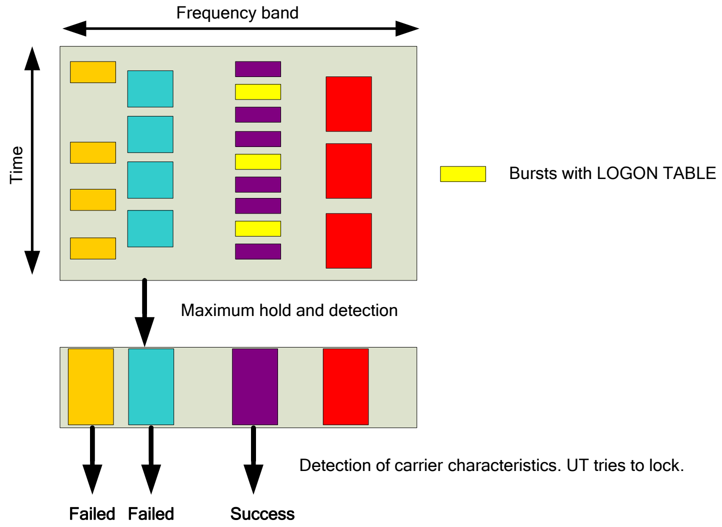


Figure 10-2 – Description of SDA function

10.3 Timeouts and retrials definition guidelines in control protocols

10.3.1 Generic Signalling Timeout

Most of signalling messages do not require a specific timeout value and neither specific number of retrials, as no significant different delay will be added for the process required to respond to any request message, neither in the UT or the GS, and the definition of a generic timeout value for all signalling messages responses and a generic number of retrials will suffice.

The default value for the timeout can be obtained taking into account:

- Delay both in FWD and RTN channels since a message is passed to L2 process until the message begins to be transmitted, taking into account schedule, encapsulation, congestion control and timeslot distribution to GS elements. It is assumed that signalling messages are transmitted with the highest priority, more than the highest CoS used for the services.
- Transmission time, taking into account burst type required to fit the complete message (including segmentation if required).
- Reception process delay.
- Round trip time for 2 messages (message + response).
- Process time in the GS/UT required to generate the response.

Roughly, a 4-5 second value will be suitable to cope with all the delays mentioned in most scenarios, but take into account that the final value will depend on system implementation. Such as, for example, FWD timeslot distribution to GS elements (i.e., previous value is suitable for a maximum delay caused by a timeslot distribution of approximately 1 second, but if the timeslot availability for GS is reduced, additional delay must be considered).

10.3.2 Specific HO Timeouts

L3 Update, Old Channel Empty and Connection Close Timeouts

As mentioned in the previous section, HO execution is based on a make-before-break approach, emptying L2 buffers towards the previous GSE before releasing the old link.

Two timers are involved in this process, *L3 Update* and *Old Channel Empty*. The purpose of the first one is to provide enough time to complete L3 update processes before taking into account buffer status (to avoid the case of the buffer being temporarily empty before the L3 update has been completely performed), while the second one is used to limit the maximum time in which both old and new links are established in parallel, releasing the old link in any case (even if old buffer is not empty) when the timer expires.

The actual values of both timers highly depend on L3 implementations for OSI and mobile IPv6:

- If the L3 update processes take advantage of the trigger generated when the new L2 link is established, a reasonable value for *L3 Update* timeout can be defined around 90 seconds, or even less, considering expected values for the L3 update processes. *Old Channel Empty* must include previous value and additional time to empty the buffer of pending traffic. Considering values of ET/TD95 for the defined services (maximum 57 s), a value of 150 seconds is recommended for *Old Channel Empty* timeout.
- If the L3 update processes do not take advantage of the L2 trigger and the L3 update process is not initiated until the old L2 link is released, then both *Old Channel Empty* timeouts can be minimized, in order to release the old L2 link immediately and perform the complete HO as fast as possible. In this case, the value of *L3 Update* timeout is not relevant.

The *Connection Close* timeout allows adding flexibility for the GS buffer emptying, such as for example in the case of very different values for ET/TD95 in GS and UT originated services. In most scenarios, *Connection Close* timeout can use the same value used for *Generic Signalling*.

10.3.3 Message Retransmission

The maximum number of retrials is defined taking into account the probability that a message in the FWD and in RTN link is lost or corrupted. Assuming an overall loss probability below 10E-3, a value of 3 retrials seems enough to decrease the probability of exceeding the number of retrials to a reasonable value (10E-9).

10.4 Guidelines regarding redundancy

Although the CS does not mandate how system redundancy has to be implemented, it provides some features that can be exploited in order to allow more seamless switchovers between active and redundant GS stations. These features are described below.

10.4.1 Support for redundancy at encapsulation level

The GSE ID L2 address field used in the encapsulation headers is structured as follows:

- The first 6 bits identify the GSE element from a UT point of view (refer to D018-COM-FUN-0469).
- The last two bits identify the specific station acting as the GSE element, in the event that there are multiple stations working in backup mode.

In the RTN link, the UT just uses the first six bits for the RTN link ARQ and encapsulation processes.

In the FWD link, however, the UT considers all bits of the GSE ID field and is able to support the FWD link ARQ and reassembly process at least for four different GSE ID values per link. This allows initiating FWD link transmissions from a GS station that was previously in back-up mode and which does not have the fast-changing L2 traffic context information (ARQ fragment and packet counters, etc.) of the failed GSE.

In fact, it allows even support for a seamless switch-over, as two different, redundant GS stations can transmit on the FWD link at the same time during a certain time period. In this way, the GS station going from active to back-up mode can empty its buffers and losses can be avoided.

10.4.2 Support for redundancy at network layer level (OSI RESET)

Refer to section 9.1.3.

10.5 Keep-alive mechanism

The keep-alive mechanism allows recovery from a non-nominal situation where the UT is not active anymore and the GS considers that it is still logged-on (e.g., due to a UT switch-off without previously executing a log-off procedure).

If there is no traffic coming from a UT during a certain time period, the GS can check that the UT is still active by:

- Using signalling mechanisms already defined in the CS which trigger a response by the UT (e.g., HO_info_request).
- Using a specific POLL REQUEST signalling message that has been defined in D018 for this purpose. The UT should answer with a POLL RESPONSE message.

If the UT does not answer to a certain number of these requests, the GS considers the UT as logged off.

11. APPENDIX A: AERONAUTICAL PROPAGATION CHANNEL

This section describes the propagation channel used for modelling the multipath effects on the mobile link as well as the aeronautical propagation channel scenarios.

11.1 Aeronautical Multipath Propagation Channel Model

The aeronautical multipath propagation channel is characterised by the combination of the three following components:

- A strong Line of Sight (LoS) component, present most of the time
- Local Scatters from the aircraft's fuselage (LS component)
- Delayed reflections from the ground (GR component)

These three components are shown in the following figure.

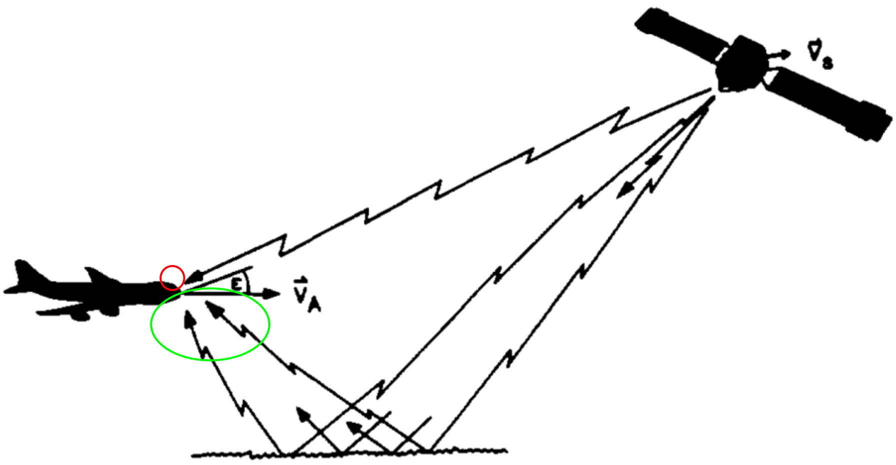


Figure 11-1: Illustration of the geometry of the aeronautical communication channel. Local scatters are illustrated with red, reflections from the ground with green.

The following figure shows the aeronautical multipath propagation channel model block diagram. The multipath channel is a two-tap model:

- The first tap (delay = 0) is divided into two components:
  - The LoS component
  - The Local Scatter component

  The behaviour of the first tap (LoS and LS components) corresponds to a Ricean fading process.
- The second tap, with delay = τ_{GR} with respect to the LoS component, is the ground reflection component.

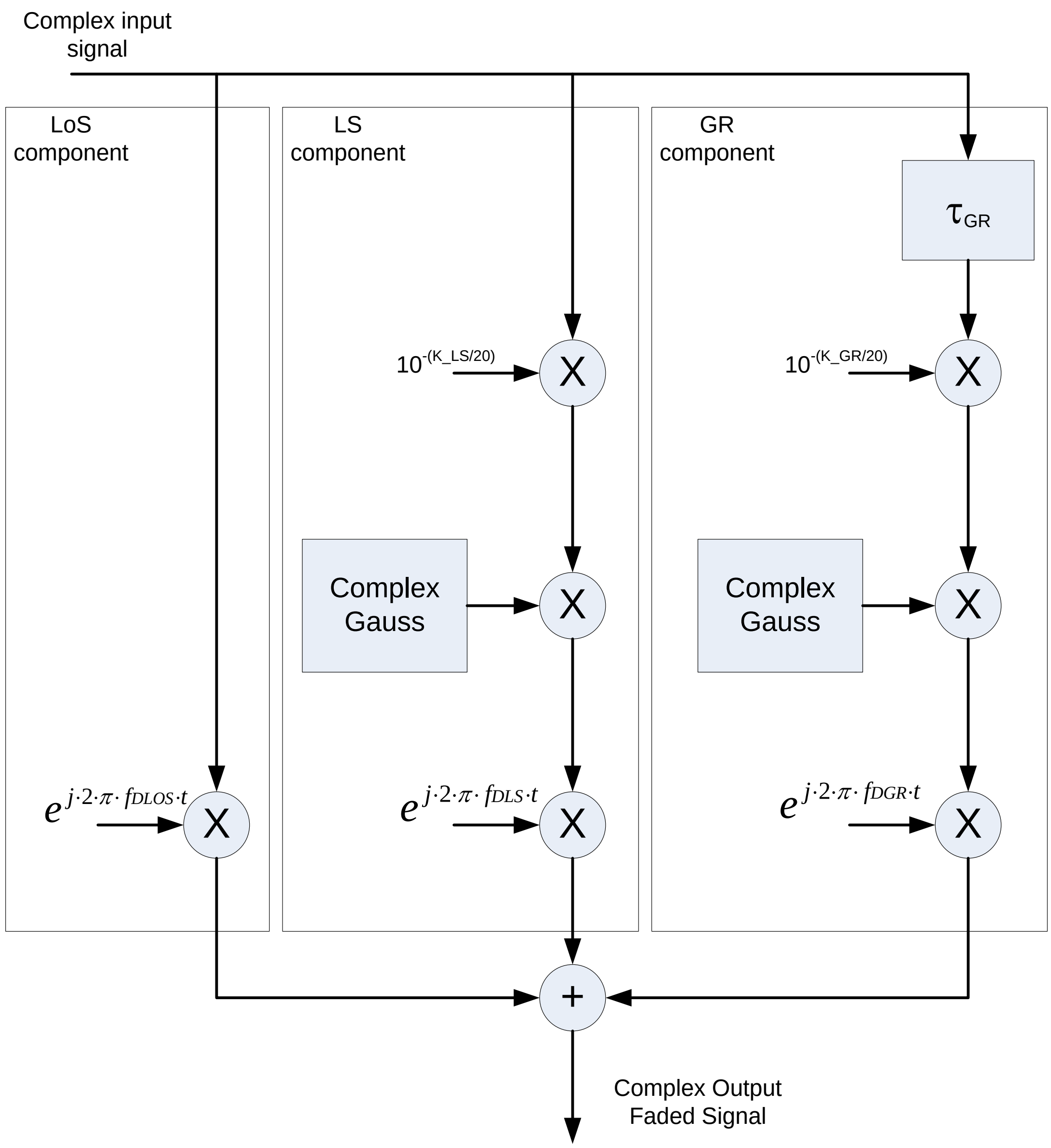


Figure 11-2: Block diagram for the aeronautical multipath propagation channel model

11.1.1 LoS component

The LOS component suffers a Doppler shift due to the relative movement of the aircraft with respect to the satellite (f_{DLOS}).

11.1.2 Local Scatter (LS) component

The Local Scatter component is modelled as a Rayleigh fading:

- The input signal to the LS component block is multiplied by a fixed gain ($10^{-(K\_LS/20)}$), where K_LS represents the ratio of the direct signal power (LoS component) to the total LS multipath power.

- Then, the resulting signal is multiplied by a complex Rayleigh fading process. The Rayleigh fading process is characterized by:
  - Doppler Spread (B_{LS}) of 1Hz
  - The LS Doppler spectrum is Gaussian and is given by

$$P_{LS}(f) = \frac{\sqrt{2}}{B_{LS}\sqrt{\pi}} e^{\left(-\frac{2f^2}{B_{LS}^2}\right)}$$

- Finally, a Doppler shift is added to the LS component (f_{DLS}).

11.1.3 Ground Reflection (GR) component

The Ground Reflection component is also modelled as a Rayleigh fading:

- The input signal is firstly delayed τ_{GR} with respect to the LoS component,
- Then the delayed signal is multiplied by a fixed gain ($10^{-(K\_GR/20)}$), where K_GR represents the ratio of the direct signal power (LoS component) to the total GR multipath power.
- The resulting signal is multiplied by a complex Rayleigh fading process. The Rayleigh fading process is characterized by:
  - Doppler Spread (B_{GR})
  - The GR Doppler spectrum is Gaussian and is given by

$$P_{GR}(f) = \frac{\sqrt{2}}{B_{GR}\sqrt{\pi}} e^{\left(-\frac{2f^2}{B_{GR}^2}\right)}$$

- Finally, a Doppler shift is added to the LS component (f_{DGR}).

11.2 Aeronautical Multipath Propagation Channel Scenarios

This section presents the configuration of the Aeronautical Multipath Propagation Channel Model for the different scenarios.

Aero Scenario Id	Aircraft type	LoS component parameters	LS component parameters			GR component parameters				Periodic fades (Rotary- wing only)		
		f_{DLOS} (Hz)	K_LS (dB)	B_{LS} (Hz)	f_{DLS} (Hz)	K_GR (dB)	B_{GR} (Hz)	τ_{GR}(μs)	f_{DGR} (Hz)	Fade Depth (dB)	Fade duration (ms)	Fading period (ms)
1	Fixed wing	0	14	1	0	20	32.8	11.5	4	N.A	N.A	N.A
2	Fixed wing	0	14	1	0	15	103.65	38.28	3.4	N.A	N.A	N.A
3	Fixed wing	0	14	1	0	10	32.8	11.5	4	N.A	N.A	N.A
4	Fixed wing	0	14	1	0	5	18.4	5.8	3.8	N.A	N.A	N.A
5	Rotary wing	0	14	1	0	20	11.24	6.37	4	7	10.8	54

Table 11-1: Aeronautical propagation scenarios

12. APPENDIX B: OVERHEADS ESTIMATION

This section describes an estimation of the CS L1 and L2 overheads for the FWD link considering the reference traffic profile for the ECAC area shown in section 4.4. Refer to section 4.4.4 for the considered overheads at L3.

Computed during multiple access simulations, the CS average L2 overhead at L2is 2.2% in the FWD link, which satisfies the requirement of an L2 overhead < 4%.

It has been seen that the L2 overhead depends a lot on the considered service message size, going from just 2% for FREETXT application messages (377 bytes in the FWD link) to 44% for transport layer ACKs (TL_ACKs of 16 bytes).

Measured also by the multiple access simulations, the proportion of bytes sent in the FWD link corresponding to ARQ ACKs (3 bytes per ARQ ACK) with respect to messages' bytes sent, as computed from traffic traces and section 4.4.4 assumptions, is on average 1.16%.

The next figures show a comparison of encapsulation efficiencies computed during the CS design phase for both FWD and RTN links.

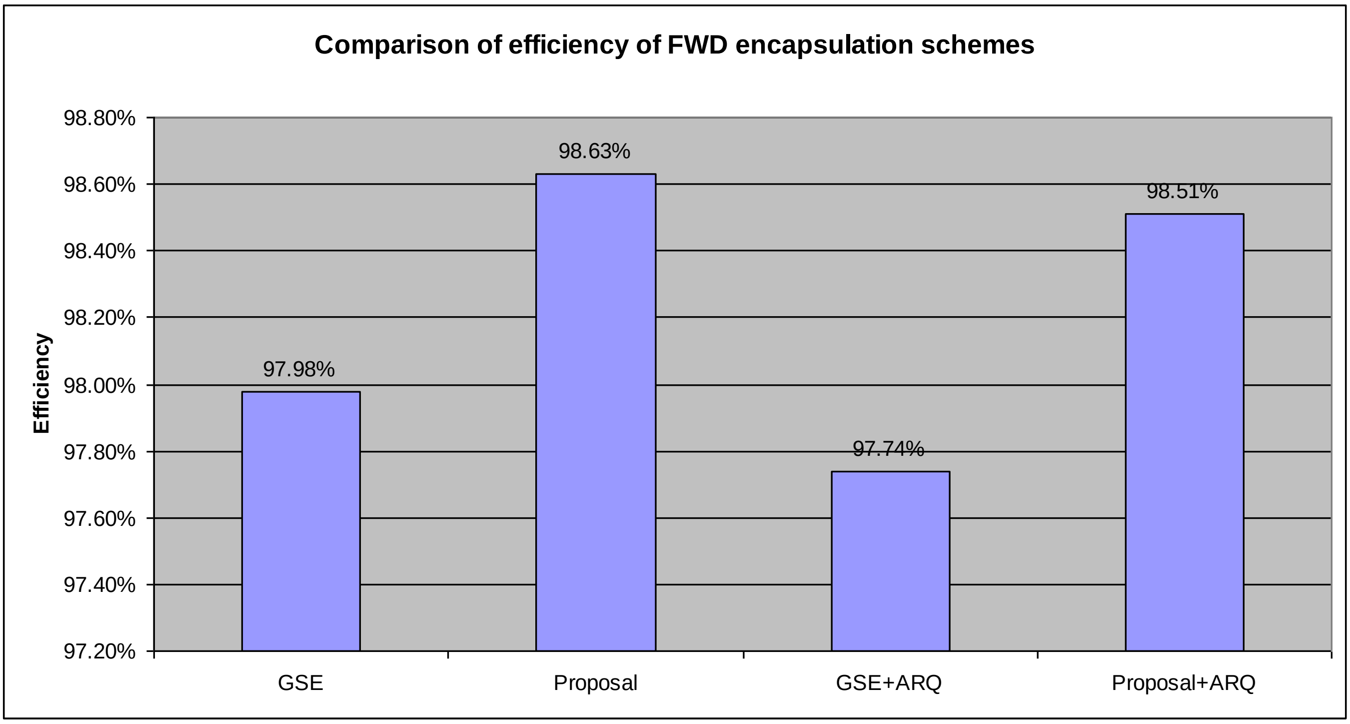


Figure 12-1: Efficiency of GSE and proposed FWD scheme with and without ARQ overhead for the 2025 traffic profile

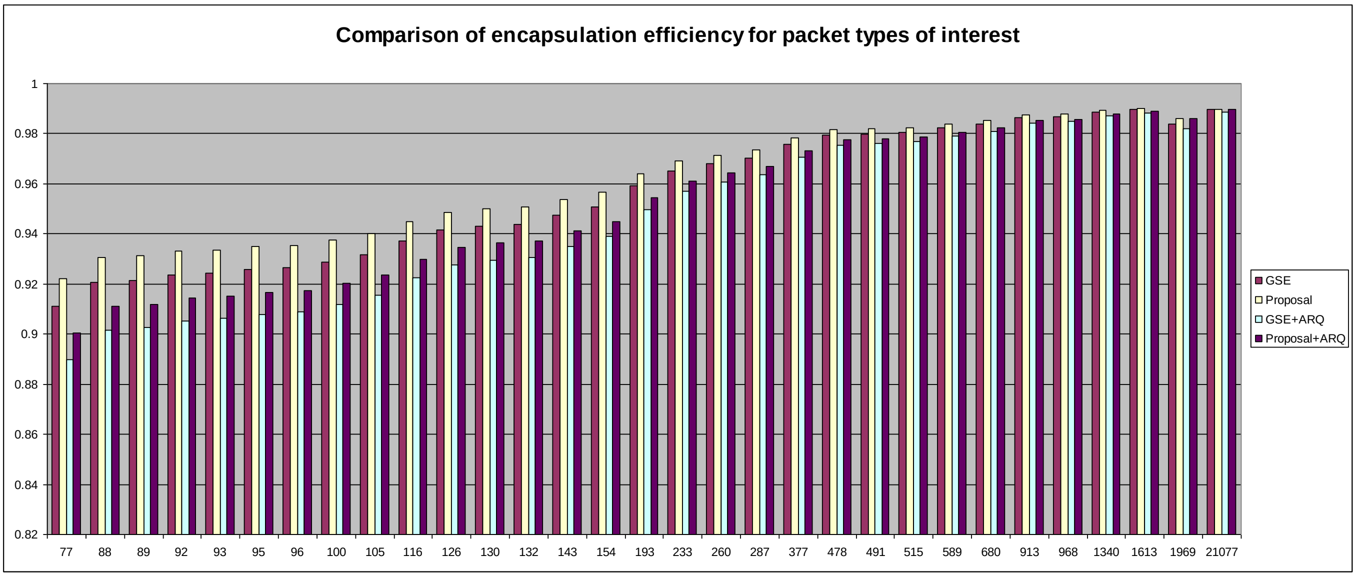


Figure 12-2: Efficiency of GSE and proposed FWD scheme with and without ARQ overhead for every packet size of the traffic profile

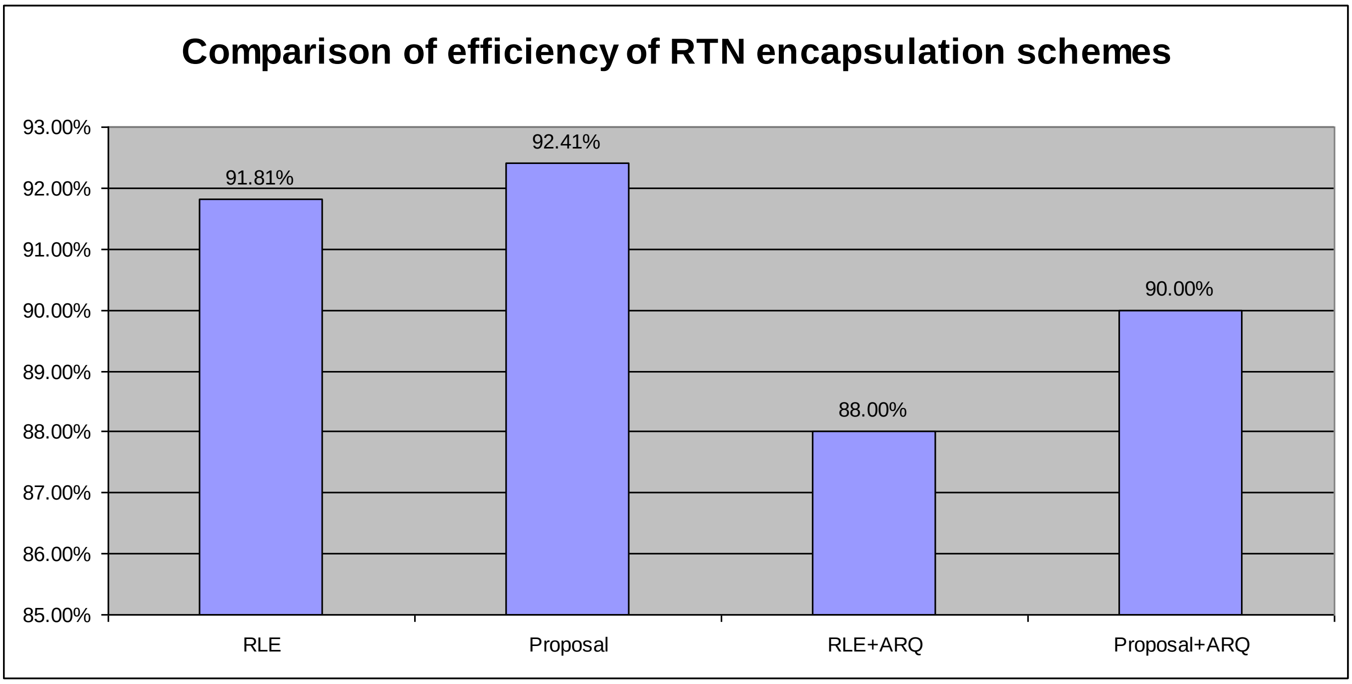


Figure 12-3: Efficiency of RLE and proposed RTN scheme with and without ARQ overhead for the 2025 traffic profile

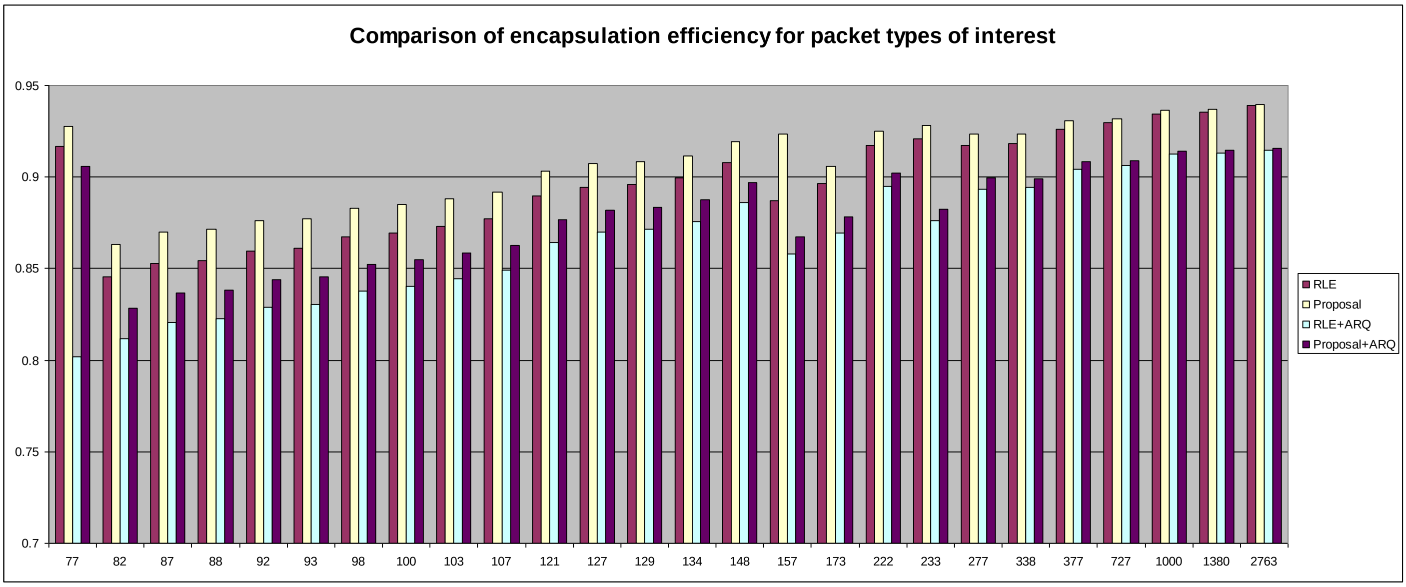


Figure 12-4: Efficiency of RLE and proposed RTN scheme with and without ARQ overhead for every packet size of the traffic profile

On the traffic profile for the Atlantic Ocean a unique message size is used to send ADS-C information over the RTN link. Every 1000-byte message sent over the RTN link is acknowledged at transport layer over the FWD link using a TL_ACK packet. These message sizes over RTN and FWD links are already used for other ATS services in the ECAC area, considered previously.

Over the RTN link this new traffic represents less than 1% of the overall traffic profile, which is negligible in terms of encapsulation efficiency. Moreover, the encapsulation efficiency for message size of 1000 bytes (around 94%) is higher than the mean efficiency over the RTN link (92.41%), meaning that this new traffic will slightly improve this mean efficiency.

Over the FWD link this new traffic represents less than 0.01% of the global traffic sent in the system, so it is completely negligible in terms of encapsulation efficiency. This means that it will not affect the performance previously presented concerning this efficiency.

The physical layer overheads are presented in **Table 12-1**.

FCH bursts		
Coverage region		**ECAC**
Forward link symbol rate		**160 kbaud**
Physical layer overheads	Burst duration	86.225 ms
	Guard time	20 symbols
	L1 header	64 symbols
	FWD_DD [extended]	24 bits [64 bits]
	CRC	32 bits
	Time slot	86.35 ms
	Guard time overhead	**0.14%**
	L1 header overhead	**0.46%**
	FWD_DD overhead (QPSK ¼)	**0.39% [1.04%]**

	CRC overhead (QPSK ¼)	**0.52%**
	Carrier bandwidth	192 kHz
	Guard band	600 Hz (GEO) 3600 Hz (HEO) 500 Hz (MEO)
	Channel bandwidth	192.6 kHz (GEO) 195.6 kHz (HEO) 192.5 kHz (MEO)
	Guard band overhead	**0.31% (GEO) 1.84% (HEO) 0.26% (MEO)**
	Preamble	100 symbols
	Preamble overhead	**0.72%**
	Postamble	24 symbols
	Postamble overhead	**0.17%**
	Pilot symbols field	24 symbols
	Pilot symbols update time	248 symbols
	Pilot symbols overhead	**9.68%**

Table 12-1: Physical layer overhead

Taking into account the physical layer signalling (cells in grey in **Table 12-1**), the resulting FLC signalling overhead is presented in **Table 12-2**. MODCOD QPSK 1/4 has been assumed for the calculations.

FLC physical layer signalling overhead	
FCH bursts	
FLC with no NCR distribution	**1.32%**
NCR insertion (extended FWD_DD inserted every N timeslots)	N = 23
FLC with NCR distribution (extended FWD_DD every 2 s)	**1.35%**

Table 12-2: FLC physical layer signalling overhead

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D023	Communication Standard Implementation Guidelines Document	**X**	**X**	**X**	**X**	**X**