ETSI TS 101 376-4-15 V3.4.1 (2017-03)

TECHNICAL SPECIFICATION

GEO-Mobile Radio Interface Specifications (Release 3);
Third Generation Satellite Packet Radio Service;
Part 4: Radio interface protocol specifications;
Sub-part 15: Packet Data Convergence Protocol (PDCP) specification;
GMR-1 3G 25.323

Reference

RTS/SES-00423

Keywords

3G, earth station, GMPRS, GMR, GPRS, GSM, GSO, interface, MES, mobile, MSS, MUX, radio, satellite, S-PCN, TDMA

ETSI

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

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

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

Contents

Intellectual Property Rights

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

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

Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Satellite Earth Stations and Systems (SES).

The contents of the present document are subject to continuing work within TC-SES and may change following formal TC-SES approval. Should TC-SES modify the contents of the present document it will then be republished by ETSI with an identifying change of release date and an increase in version number as follows:

Version 3.m.n

where:

- The third digit (n) is incremented when editorial only changes have been incorporated in the specification;
- The second digit (m) is incremented for all other types of changes, i.e., technical enhancements, corrections, updates, etc.

The present document is part 5, sub-part 1 of a multi-part deliverable covering the GEO-Mobile Radio Interface Specifications (Release 3); Third Generation Satellite Packet Radio Service, as identified below:

Part 1: "General specifications":

Part 2: "Service specifications";

Part 3: "Network specifications";

Part 4: "Radio interface protocol specifications":

Sub-part 1: "Mobile Earth Station-Gateway Station System (MES-GSS) Interface; GMR-1 04.001";

Sub-part 2: "GMR-1 Satellite Network Access Reference Configuration; GMR-1 04.002";

Sub-part 3: "Channel Structures and Access Capabilities; GMR-1 04.003";

Sub-part 4: "Layer 1 General Requirements; GMR-1 3G 44.004";

Sub-part 5: "Data Link Layer General Aspects; GMR-1 04.005";

Sub-part 6: "Mobile earth Station-Gateway Station Interface Data Link Layer Specifications; GMR-1 04.006";

Sub-part 7: "Mobile Radio Interface Signalling Layer 3 General Aspects; GMR-1 3G 04.007";

Sub-part 8: "Mobile Radio Interface Layer 3 Specifications; GMR-1 3G 44.008";

Sub-part 9: "Performance Requirements on the Mobile Radio Interface; GMR-1 04.013";

Sub-part 10: "Rate Adaptation on the Access Terminal-Gateway Station Subsystem (MES-GSS) Interface; GMR-1 04.021";

Sub-part 11: "Radio Link Protocol (RLP) for Data Services; GMR-1 04.022";

Sub-part 12: "Mobile Earth Station (MES) - Base Station System (BSS) interface; Radio Link Control/Medium Access Control (RLC/MAC) protocol; GMR-1 3G 44.060";

Sub-part 13: "Radio Resource Control (RRC) protocol; Iu Mode; GMR-1 3G 44.118";

Sub-part 14: "Mobile Earth Station (MES) - Base Station System (BSS) interface; Radio Link Control/Medium Access Control (RLC/MAC) protocol; Iu Mode; GMR-1 3G 44.160";

Sub-part 15: "Packet Data Convergence Protocol (PDCP) specification; GMR-1 3G 25.323";

Part 5: "Radio interface physical layer specifications";

Part 6: "Speech coding specifications";

Part 7: "Terminal adaptor specifications".

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

Introduction

GMR stands for GEO (Geostationary Earth Orbit) Mobile Radio interface, which is used for Mobile Satellite Services (MSS) utilizing geostationary satellite(s). GMR is derived from the terrestrial digital cellular standard GSM and supports access to GSM core networks.

The present document is part of the GMR Release 3 specifications. Release 3 specifications are identified in the title and can also be identified by the version number:

- Release 1 specifications have a GMR 1 prefix in the title and a version number starting with "1" (V1.x.x).
- Release 2 specifications have a GMPRS 1 prefix in the title and a version number starting with "2" (V2.x.x).
- Release 3 specifications have a GMR-1 3G prefix in the title and a version number starting with "3" (V3.x.x).

The GMR release 1 specifications introduce the GEO-Mobile Radio interface specifications for circuit mode Mobile Satellite Services (MSS) utilizing geostationary satellite(s). GMR release 1 is derived from the terrestrial digital cellular standard GSM (phase 2) and it supports access to GSM core networks.

The GMR release 2 specifications add packet mode services to GMR release 1. The GMR release 2 specifications introduce the GEO-Mobile Packet Radio Service (GMPRS). GMPRS is derived from the terrestrial digital cellular standard GPRS (included in GSM Phase 2+) and it supports access to GSM/GPRS core networks.

The GMR release 3 specifications evolve packet mode services of GMR release 2 to 3rd generation UMTS compatible services. The GMR release 3 specifications introduce the GEO-Mobile Radio Third Generation (GMR-1 3G) service. Where applicable, GMR-1 3G is derived from the terrestrial digital cellular standard 3GPP and it supports access to 3GPP core networks.

Due to the differences between terrestrial and satellite channels, some modifications to the GSM or 3GPP standard are necessary. Some GSM and 3GPP specifications are directly applicable, whereas others are applicable with modifications. Similarly, some GSM and 3GPP specifications do not apply, while some GMR specifications have no corresponding GSM or 3GPP specification.

Since GMR is derived from GSM and 3GPP, the organization of the GMR specifications closely follows that of GSM or 3GPP as appropriate. The GMR numbers have been designed to correspond to the GSM and 3GPP numbering system. All GMR specifications are allocated a unique GMR number. This GMR number has a different prefix for Release 2 and Release 3 specifications as follows:

- Release 1: GMR-n xx.zyy
- Release 2: GMPRS-n xx.zyy
- Release 3: GMR-1 3G xx.zyy

where:

- xx.0yy (z = 0) is used for GMR specifications that have a corresponding GSM or 3GPP specification. In this case, the numbers xx and yy correspond to the GSM or 3GPP numbering scheme.
- xx.2yy (z = 2) is used for GMR specifications that do not correspond to a GSM or 3GPP specification. In this case, only the number xx corresponds to the GSM or 3GPP numbering scheme and the number yy is allocated by GMR.
- n denotes the first (n = 1) or second (n = 2) family of GMR specifications.

A GMR system is defined by the combination of a family of GMR specifications and GSM and 3GPP specifications as follows:

- If a GMR specification exists it takes precedence over the corresponding GSM or 3GPP specification (if any). This precedence rule applies to any references in the corresponding GSM or 3GPP specifications.

NOTE: Any references to GSM or 3GPP specifications within the GMR specifications are not subject to this precedence rule. For example, a GMR specification may contain specific references to the corresponding GSM or 3GPP specification.

- If a GMR specification does not exist, the corresponding GSM or 3GPP specification may or may not apply. The applicability of the GSM and 3GPP specifications are defined in ETSI TS 101 376-1-2 [7].

1 Scope

The present document provides the description of the GMR-1 3G Packet Data Convergence Protocol (PDCP).

2 References

2.1 Normative references

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

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

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

The following referenced documents are necessary for the application of the present document.

In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest release and the latest version of that document up to and including Release 7.

In the case of a reference to a GMR-1 3G document, a non-specific reference implicitly refers to the latest version of that document in the same Release as the present document.

[1] ETSI TS 123 060: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); General Packet Radio Service (GPRS); Service description; Stage 2 (3GPP TS 23.060 Release 7)".

[2] ETSI TS 125 331 (Release 7): "Universal Mobile Telecommunications System (UMTS); Radio Resource Control (RRC); Protocol specification (3GPP TS 25.331 Release 7)".

[3] ETSI TS 125 322 (Release 7): "Universal Mobile Telecommunications System (UMTS); Radio Link Control (RLC) protocol specification (3GPP TS 25.322 Release 7)".

[4] IETF RFC 2507: "IP Header Compression".

[5] IETF RFC 3095: "RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed".

[6] ETSI TS 101 376-1-1: "GEO-Mobile Radio Interface Specifications (Release 2) General Packet Radio Service; Part 1: General specifications; Sub-part 1: Abbreviations and acronyms; GMPRS-1 01.004".

NOTE: This is a reference to a GMR-1 Release 2 Specification. See the introduction for more details.

[7] ETSI TS 101 376-1-2: "GEO-Mobile Radio Interface Specifications (Release 3); Third Generation Satellite Packet Radio Service; Part 1: General specifications; Sub-part 2: Introduction to the GMR-1 family; GMR-1 3G 41.201".

[8] Recommendation ITU-T V.44 (11-2000): "Data compression procedures".

[9] IETF RFC 768 (August 1980): "User Datagram Protocol".

[10] IETF RFC 791 (September 1981): "Internet Protocol".

[11] IETF RFC 3261 (June 2002): "SIP: Session Initiation Protocol".

2.2 Informative references

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

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

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

The following referenced documents are necessary for the application of the present document.

[i.1] ETSI TS 125 401: "Universal Mobile Telecommunications System (UMTS); UTRAN overall description (3GPP TS 25.401 Release 7)".

[i.2] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[i.3] ETSI TR 121 905: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Vocabulary for 3GPP Specifications (3GPP TR 21.905)".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in ETSI TR 121 905 [i.3], ETSI TS 101 376-1-2 [7] and the following apply:

N-context Refers collectively to both *N-context-C* and *N-context-D*
*N-context** Refers collectively to both *N-context-C** and *N-context-D**
N-context-C The compression context for downlink in RNC at any given point of time
N-context-C* The frozen snapshot of the compression context for downlink taken by RNC
N-context-C-static* The frozen snapshot of the static part of the compression context for downlink taken by RNC
N-context-D The decompression context for uplink in RNC at any given point of time
N-context-D* The frozen snapshot of the decompression context for uplink taken by RNC
N-context-D-static* The frozen snapshot of the static part of the decompression context for uplink taken by RNC
M-context Refers collectively to both *M-context-C* and *M-context-D*
*M-context** Refers collectively to both *M-context-C** and *M-context-D**
M-context-C The compression context for uplink in MES at any given point of time
M-context-C* The frozen snapshot of the compression context for uplink taken by MES
M-context-C-static* The frozen snapshot of the static part of the compression context for uplink taken by MES
M-context-D The decompression context for downlink in MES at any given point of time
M-context-D* The frozen snapshot of the decompression context for downlink taken by MES
M-context-D-static* The frozen snapshot of the static part of the decompression context for downlink taken by MES
M-HC Entity located in the mobile terminal that performs header compression for uplink (i.e. MES PDCP)
M-HCD Refers collectively to both *M-HC* and *M-HD*
M-HD Entity located in the mobile terminal that performs header decompression for downlink (i.e. MES PDCP)
N-HC Entity located in the network that performs header compression for downlink (i.e. RNC PDCP)
N-HCD Refers collectively to N-HC and N-HD
N-HD Entity located in the network that performs header decompression for uplink (i.e. RNC PDCP)

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in ETSI TS 101 376-1-1 [6] and the following apply:

AS	Access Stratum
CID	Context Identifier
C-SAP	Control Service Access Point
GMR-1 3G LLA	GMR1-3G Lower Layer Assisted
HC	Header Compression
HCL	Header Compression Lower layer
HCU	Header compression Upper layer
IETF	Internet Engineering Task Force
IP	Internet Protocol
L2	Layer 2 (data link layer)
MBMS	Multimedia Broadcast Multicast Service
M-HC	Mobile Header Compressor
M-HCD	Mobile Header Compressor/Decompressor
M-HD	Mobile Header Decompressor
NAS	Non Access Stratum
N-HC	Network Header Compressor
N-HCD	Network Header Compressor/Decompressor
N-HD	Network Header Decompressor
PDCP	Packet Data Convergence Protocol
PDU	Protocol Data Unit
PID	Packet Identifier
PPP	Point-to-Point Protocol
RAB	Radio Access Bearer
RB	Radio Bearer
RFC	Request for Comments
RLC	Radio Link Control
RNC	Radio Network Controller
RoHC LLA	Robust Header Compression Link Layer Assisted
RoHC	Robust Header Compression
RTP	Real Time Protocol
SDU	Service Data Unit
SE-VoIP	Spectrally Efficient Voice over IP
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network

4 General

4.1 Objective

The present document describes the functionality of the GMR-1 3G Packet Data Convergence Protocol (PDCP).

4.2 Overview on Sublayer Architecture

Figure 1 shows the model of the PDCP within the radio interface protocol architecture. The PDCP sublayer is defined for the PS domain only.

Every PS domain Radio Access Bearer (RAB) is associated with one Radio Bearer (RB), which in turn, is associated with one PDCP entity. Each PDCP entity is associated with one or two (one for each direction) RLC entities depending on the RB characteristic (i.e. unidirectional or bidirectional) and RLC mode. The PDCP entities are located in the PDCP sublayer.

Every PDCP entity uses zero, one or several different header compression and data compression protocols. Each individual PDCP entity uses at most one instance of each header compression protocol. Several PDCP entities may be defined for a MES with each using the same or a different set of header compression protocols. In this version of the specification, the following header compression protocols are supported:

- IETF RFC 2507 [4]
- IETF RFC 3095 [5]
- GMR-1 3G Lower Layer Assisted Header Suppression, which is defined in the present document
- Recommendation ITU-T V.44 Data Compression [8]

The PDCP sublayer is configured by the upper layer through the PDCP-C-SAP.

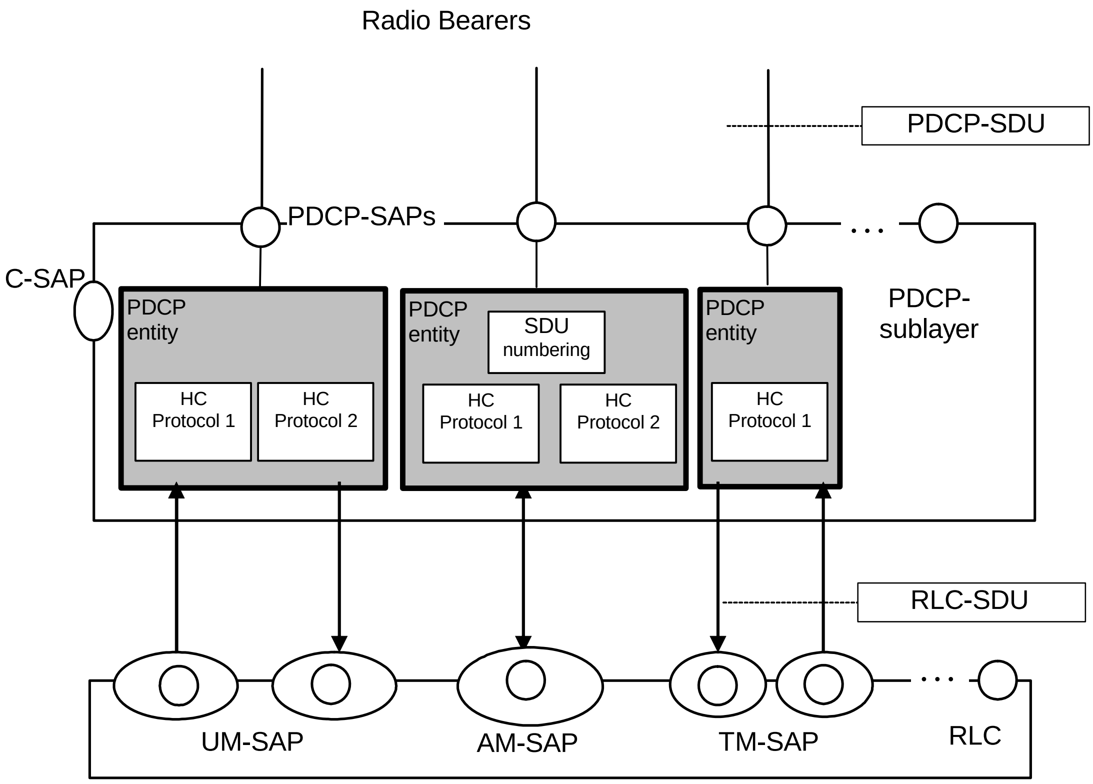

Figure 1: PDCP Structure

Figure 1 represents one possible structure for the PDCP sublayer and should not restrict implementation. A PDCP entity is mapped either to one AM RLC entity, or one or two UM or TM RLC entities. When a PDCP entity is mapped to two UM or TM RLC entities, each RLC entity is used for a different direction.

5 Functions

5.0 General

PDCP provides its services to the NAS at the MES or the relay at the Radio Network Controller (RNC).

The Packet Data Convergence Protocol shall perform the following functions:

- Header compression and decompression of IP data streams (e.g. TCP/IP and RTP/UDP/IP headers for IPv4 and IPv6) at the transmitting and receiving entity, respectively.

- Header suppression (of RTP/UDP/IP headers) of GMR-1 3G spectrally efficient VoIP (SE-VoIP) streams.
- Transfer of user data. This function is used for conveyance of data between users of PDCP services.
- Data compression.
- SRNS Relocation.

PDCP uses the services provided by the Radio Link Control (RLC) sublayer.

PDCP supports two kinds of compression, namely header compression and data compression. The header and data compressions for an IP packet are applied sequentially.

On the encoder side, every IP packet will arrive at the data compression entity first. After compressing the transport layer payload, the data compression entity attaches the compressed payload to the original Transport/IP header and delivers it to the header compression entity. If the payload is not compressed, it will be delivered to the header compression entity without any changes.

On the decoder side, the header decompression will decompress the Transport/IP header only and then delivers it along with the compressed or uncompressed payload to the data decompression entity. If the payload is compressed, the data decompressor will decompress it and deliver it to the upper layer along with the decompressed header. If the payload is not compressed, the data decompressor just passes it to the upper layer.

5.1 Header Compression

5.1.0 General

The header compression protocol is specific to the particular network layer, transport layer or upper layer protocol combinations e.g. TCP/IP and RTP/UDP/IP. The network layer protocol type, e.g. IP or PPP, is indicated during PDCP context activation as defined in ETSI TS 123 060 [1]. The header compression protocols and their parameters are configured by upper layers for each PDCP entity. Compressor- and decompressor-initiated signalling between peer PDCP entities during operation is accomplished through in-band signalling.

5.1.1 Mapping of PID Values

Depending on the configuration by upper layers (i.e. PDCP PDU type to be used and header compressor protocol), the PDCP sublayer shall be able to:

- Identify different types of header compression protocols
- If IETF RFC 2507 [4]:
  - Distinguish different header compression protocol packet types within a header compression protocol
- If IETF RFC 3095 [5]:
  - Distinguish different contexts for a header compression protocol

GMR-1 3G LLA Header Suppression protocol for spectrally efficient VoIP service is not mixed with other header compression protocols within a PDCP entity.

The above requirements are realized by utilizing the PID field in the PDCP PDU.

The mapping of the PID values shall follow the general rules listed below:

- PID values shall be mapped to the different packet types independently at each PDCP entity.
- PID value "0" shall indicate "no compression." PID value "0" shall be used in a PDCP PDU containing in its Data field a PDCP SDU that is unchanged by the Sender and that shall not be decompressed by the Receiver.

- PID values are mapped in ascending order, starting from 1, for every configured header compression protocol, in the order of configuration by the upper layer. The first available PID value is assigned to the first packet type of the header compression protocol as defined in the specification for this header compression protocol. PID values are mapped for all the specified packet types defined for the header compression protocol and in the order defined in clauses 5.1.2.2 and 5.1.3.3 for the respective header compression protocol.
- PID values are re-mapped for the PDCP entity after any reconfiguration of the header compression protocols for that entity.

Table 1 illustrates an example of the PID value mapping to the packet types when five arbitrary header compression methods are configured for one PDCP entity: IETF RFC 2507 [4], Methods A and B, IETF RFC 3095 [5], and Method C. Method A, Method B, and Method C are imaginary header compression protocols introduced for the purpose of illustration.

Table 1: Example of the PID Value Mapping Table

PID Value	Optimization Method	Packet Type
0	No header compression	-
1	IETF RFC 2507 [4]	Full header
2	IETF RFC 2507 [4]	Compressed TCP
3	IETF RFC 2507 [4]	Compressed TCP nondelta
4	IETF RFC 2507 [4]	Compressed non TCP
5	IETF RFC 2507 [4]	Context state
6	Method A	Packet Type 1 of Method A
7	Method A	Packet Type 2 of Method A
8	Method B	Packet Type 1 of Method B
9	Method B	Packet Type 2 of Method B
10	IETF RFC 3095 [5]	IETF RFC 3095 [5] packet format
11	Method C	Packet Type 1 of Method C
12	Method C	Packet Type 2 of Method C
13...31	Unassigned value	-

5.1.2 IP Header Compression (IETF RFC 2507)

5.1.2.0 General

The detailed operation of the IETF RFC 2507 [4] header compression protocol shall be as specified in IETF RFC 2507 [4]. The mechanisms related to error recovery and packet reordering are also described in IETF RFC 2507 [4]. These mechanisms shall be included in the functionality of the header compression supported by PDCP. The implementation of the IETF RFC 2507 [4] header compression functionality is not covered in the present specification and is left to the implementation.

5.1.2.1 Context Identifiers

Context identifiers for IETF RFC 2507 [4] shall only be included in the IETF RFC 2507 [4] packet types format, as defined in IETF RFC 2507 [4].

5.1.2.2 Mapping of PID Values for IETF RFC 2507

PID values shall be mapped to the IETF RFC 2507 [4] header compression packet types in the order presented in Table 2, where "n" is the number of PID values already mapped to other protocol packet types.

Table 2: Mapping of PID Values for IETF RFC 2507 [4] Header Compression Protocol

PID value	Optimization Method	Packet type
n+1	IETF RFC 2507 [4]	Full header
n+2	IETF RFC 2507 [4]	Compressed TCP
n+3	IETF RFC 2507 [4]	Compressed TCP non-delta
n+4	IETF RFC 2507 [4]	Compressed non-TCP
n+5	IETF RFC 2507 [4]	Context state

5.1.2.3 Management of Full Header Transmission

Transmission of a full header packet may be controlled by the lower layer information.

For a TCP stream, if the PDCP receives from lower layer the information of failed transmission of a single packet, the PDCP may send the next packet as a full header.

For a non-TCP stream, if the PDCP receives from lower layer the information of successful transmission of a full header packet, the PDCP may stop sending a full header packet that contains the same full header as the previously transmitted one.

5.1.3 Robust Header Compression (IETF RFC 3095)

5.1.3.0 General

The detailed operation of the, "Robust Header Compression (ROHC)" protocol shall be as specified in IETF RFC 3095 [5].

5.1.3.1 Context Identifiers

The context of the IETF RFC 3095 protocol is defined in [5]. IETF RFC 3095 [5] can be configured to support one or several contexts. Each context is identified by a value known as the context identifier (CID).

5.1.3.2 Void

5.1.3.3 Mapping of PID Values for IETF RFC 3095

The following PID value shall be mapped to the IETF RFC 3095 [5] header compression protocol as presented in Table 3, where n is the number of PID values already assigned to other protocol packet types.

Table 3: Mapping of PID Values for IETF RFC 3095 [5] Header Compression Protocol

PID value	Optimization Method	Packet type
n+1	IETF RFC 3095 [5]	IETF RFC 3095 [5] packet format

5.1.3.4 Void

5.1.3.5 Protocol Parameters

IETF RFC 3095 [5] has two types of parameters:

- Configuration parameters: these are mandatory and shall be configured between compressor and decompressor peers.
- Implementation parameters: these are optional and, when used, stipulate how IETF RFC 3095 [5] operates.

These parameters are categorized into four different groups, as defined below:

- M: Mandatory and configured by upper layers.

- MO: Parameters that shall be supported and when used can only be configured or triggered by upper layers.
- O: Optional IETF RFC 3095 [5] parameters that are not configured by upper layers. They may be used locally (i.e. Network and/or MES) for IETF RFC 3095 [5].
- N/A: Parameters that are not used in IETF RFC 3095 [5].

The usage and definition of the parameters shall be as specified below:

- MAX_CID (M): This is the maximum CID value that can be used. One CID value shall always be reserved for uncompressed flows.
- LARGE_CIDS: This is not configured by upper layers but inferred from the configured value of MAX_CID according to the following rule:
    - If MAX_CID > 15, then LARGE_CIDS = TRUE else LARGE_CIDS = FALSE.
- PROFILES (M): Profiles are used to define which profiles are allowed to be used by the MES in uplink. In downlink, all the profiles defined in [5] shall be supported.
- FEEDBACK_FOR (N/A).
- MRRU (M): Segmentation is not used by default.
- NO_OF_PACKET_SIZES_ALLOWED (N/A).
- PACKET_SIZES_ALLOWED (N/A).
- PAYLOAD_SIZES (O).
- NO_OF_PACKET_SIZES_USED (O).
- PACKET_SIZES_USED (O).
- CONTEXT_REINITIALIZATION (MO).
- MODE (O).
- CLOCK_RESOLUTION (O).
- REVERSE_DECOMPRESSION_DEPTH (M): Default value is that reverse decompression is not used.

5.1.3.6 Configuration by RRC

If the variable "PDCP_ROHC_TARGET_MODE" (see ETSI TS 125 331 [2]) is stored in the MES, and if applicable for the ROHC profile applied, the de-compressor shall only perform the operational state transitions defined in IETF RFC 3095 [5] to the stored mode.

If the variable "PDCP_ROHC_TARGET_MODE" (see ETSI TS 125 331 [2]) is not stored in the MES, the de-compressor shall not restrict the operational state transitions defined in IETF RFC 3095 [5].

5.1.4 GMR-1 3G LLA Header Suppression for Spectrally Efficient VoIP

GMR-1 3G LLA Header suppression for spectrally efficient VoIP is fully specified in Annex A.

5.1.5 Data Compression

Data compression is used to compress the payload of a non-encrypted IP datagram. Recommendation ITU-T V.44 [8] data compression procedure shall be used as the data compression algorithm. V.44 has two states of operations: stateful and stateless. The data compression is applied to the payload of the UDP/IP packet or TCP/IP packet. There are two data compression modes:

- Stateful mode

- Stateless mode

The stateful mode is typically used to compress a packet that is transported on a reliable link, while stateless mode is typically used to compress a packet that is transported on unreliable link.

For stateful compression mode, the decompression of the current packet will affect the decompression of the subsequent packets. Transmission error that may corrupt the dictionary of the current decompression will affect the decompression of the subsequent packets. Most TCP traffic is transported on L2 Acknowledged mode. Hence, since TCP packet is mostly free of transmission error, stateful compression mode is typically suitable for TCP packet.

For stateless compression mode, transmission error will not affect the decompression of the subsequent packets since each packet is decompressed independently from the previous packets. Typically stateless compression mode is applied to UDP packet since most UDP traffic is transported in L2 Unacknowledged mode.

RTP traffic is typically not compressed since usually RTP traffic, for example voice traffic, is already compressed.

V.44 Data Compression is specified in Annex D.

5.2 PDCP Setup

5.2.1 PDCP Protocol Information from RNC to MES

While establishing the radio bearer, the Radio Network Controller (RNC) sends RADIO BEARER SETUP message to the MES (see Figure 2). This message includes the PDCP Info IE that contains PDCP protocol info.

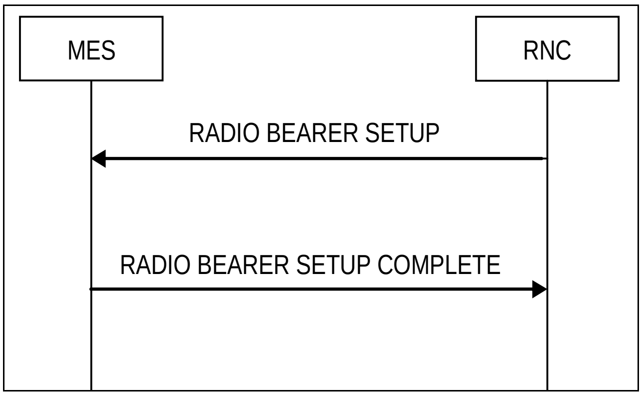

Figure 2: PDCP Info Exchange

1) The RNC sends the RADIO BEARER SETUP message to the MES. The message contains the IE and PDCP Info, which includes information on the PDCP protocol.

2) The MES replies with RADIO BEARER SETUP COMPLETE message.

5.2.2 PDCP Capability Information from MES to RNC

This indicates the algorithms and the value range of parameters supported by the MES PDCP. This information is sent to the RNC, either while setting up the RRC connection or while updating the MES capability information.

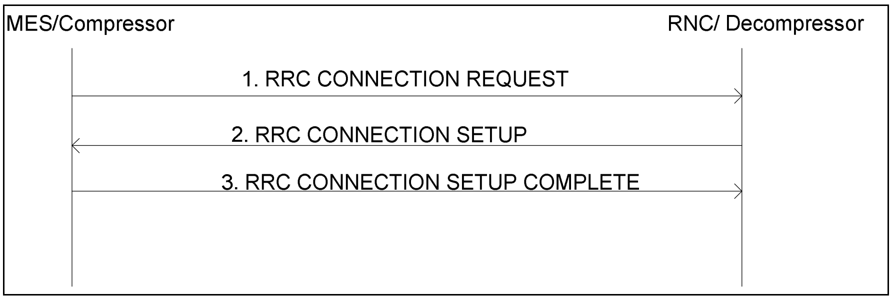

Figure 3: MES Capability Information

RRC Connection Setup

1) RNC initiates connection setup by sending RRC CONNECTION REQUEST message.

2) While setting up RRC Connection, the RNC sends the RRC CONNECTION SETUP message to the MES. This message includes the Capability Update Requirement information element.

3) If the Capability Requirement is set in the request message, the MES in its reply RRC CONNECTION SETUP COMPLETE message includes the MES RNC Iu mode Radio Access Capability IE. This IE is structured and coded according to the specification used for the corresponding system type. The IE includes the PDCP capability IE that includes the header compression parameters.

5.2.3 PDCP Entity Creation/Deletion

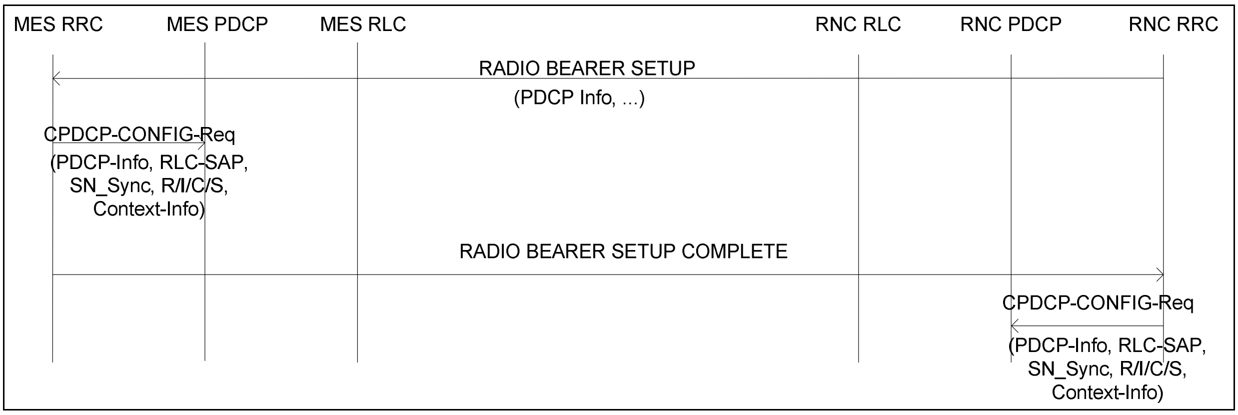

Figure 4: PDCP Entity Creation

PDCP Entity Creation

1) On the MES side, when the MES RRC receives a RADIO BEARER SETUP message, it shall use the primitive CPDCP-CONFIG-Req to configure the PDCP entity and the header compression protocols to be used.

2) On the RNC side, when the RNC RRC receives the RADIO BEARER SETUP COMPLETE message, it shall use the primitive CPDCP-CONFIG-Req to configure the PDCP entity.

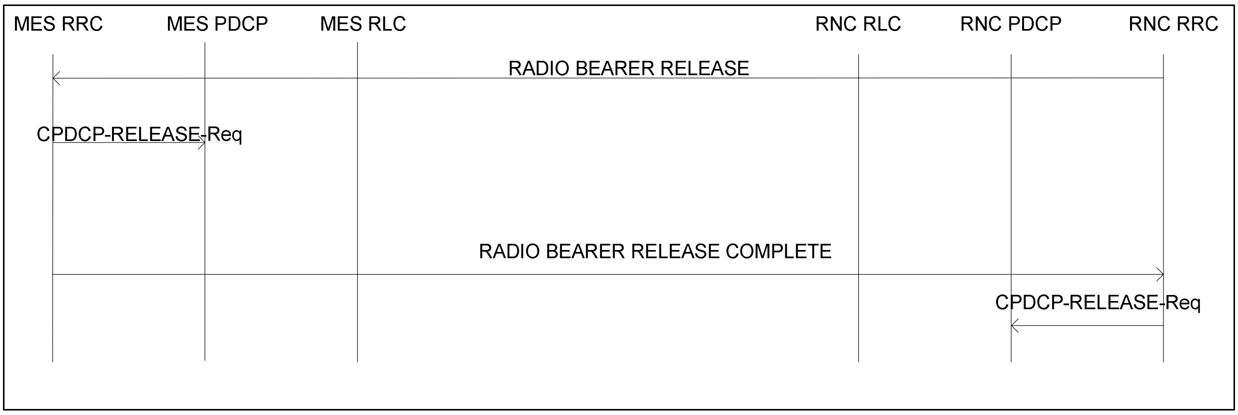

Figure 5: PDCP Entity Deletion

PDCP Entity Deletion

1) On the MES side, when the MES RRC receives a RADIO BEARER RELEASE message, it shall release all the resources associated with that radio bearer. The MES RRC sends CPDCP-RELEASE-Req message to the MES PDCP to release the PDCP entity.

2) On the RNC side, when the RNC RRC receives the RADIO BEARER RELEASE COMPLETE message, it shall release all the resources associated with that radio bearer. The RNC RRC sends CPDCP-RELEASE-Req message to the RNC PDCP to release the PDCP entity.

5.3 Data Transfer

5.3.0 General

If header compression is configured, the PDCP entity in the Sender shall:

- Perform header compression upon reception of a PDCP SDU from upper layer.
- Submit the PDCP PDU to lower layer in the sequence received from the upper layer.

When the PDCP entity at the Receiver receives the PDCP PDU from lower layers, it shall:

- Perform header decompression (if header compression is configured) of the PDCP PDU to obtain the PDCP SDU.
- Deliver the PDCP SDU to the upper layer in the order received from the lower layer.
- If the received PDCP PDU is of type PDCP SeqNum:
  - Follow the procedure in clause 5.6.2.

5.3.1 Data transfer over Acknowledged Mode RLC

Figure 6 shows the PDCP data transfer over acknowledged mode RLC.

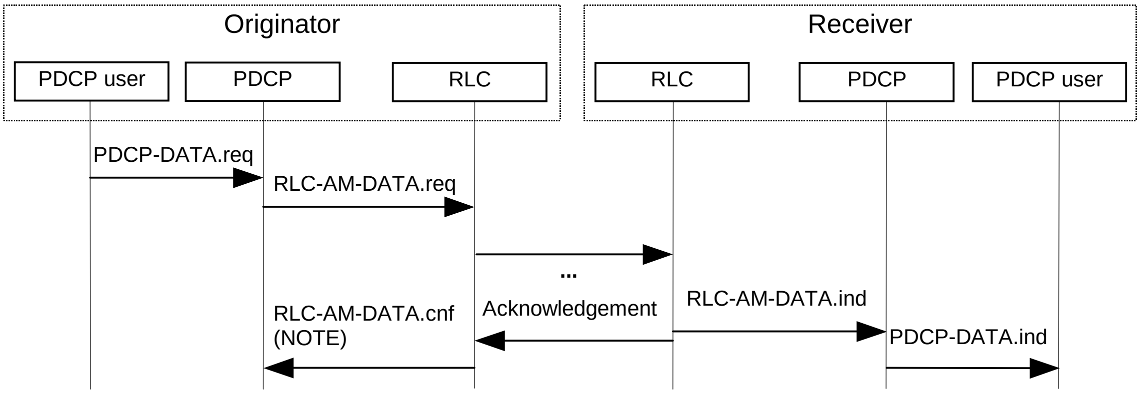

Figure 6: PDCP Data Transfer Over Acknowledged Mode RLC

NOTE: If the primitive RLC-AM-DATA.req is used with parameter CNF, the primitive RLC-AM-DATA.cnf is delivered. Otherwise, this primitive is not delivered.

5.3.2 Data transfer over Unacknowledged and Transparent Mode RLC

Figure 7 shows the PDCP data transfer over unacknowledged or transparent mode RLC.

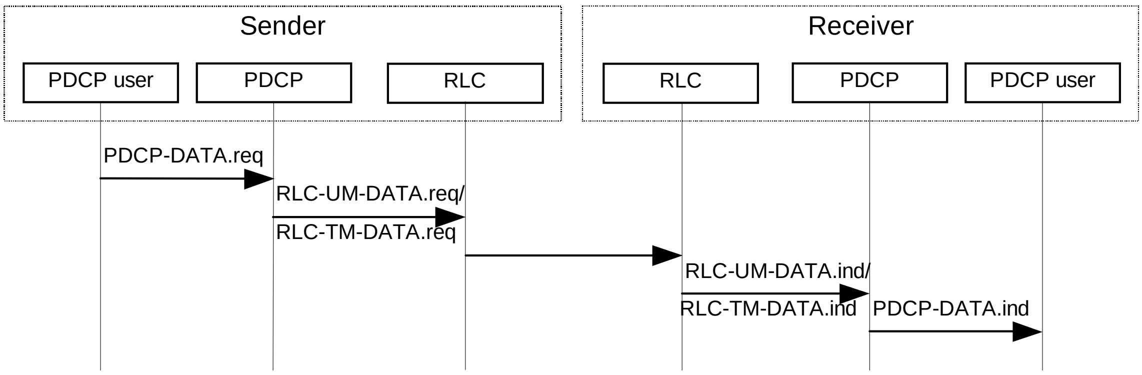

Figure 7: PDCP Data Transfer Over Unacknowledged or Transparent Mode RLC

5.3.3 RoHC Data Flow

Figure 8: RoHC Data Flow

An example message flow is shown in Figure 8. This example shows the following steps:

1) The PDCP entities on the compressor/decompressor side are created. They are passed appropriate parameters from the upper layer specifying the compression protocol to be used, lower layer protocol to be used, etc.

2) Initially, the compressor sends IR packets containing full context information to the decompressor.

3) The compressor moves to First Order (FO) state and starts sending IR-DYN packets containing the dynamic context information.

4) After it receives feedback acknowledging the packets (R-mode and optionally in O-mode), the compressor transitions to the Second Order (SO) state.

5) Once the decompressor receives the complete context information, it transitions to the Full Context (FC) state and start sending the compressed header packets Type 0/1.

6) The compressor then keeps sending the compressed header packets Type 0/1. The decompressor sends no feedback in case of correct Type 0/1 packets.

7) However, if it receives a packet that fails the CRC check.

8) It sends back a Negative Acknowledgement (NACK).

9) Upon a configurable timer expiry or receiving an explicit request from the decompressor, the compressor sends IR/IR-DYN packets to refresh the decompressor context.

10) The decompressor sends positive acknowledgement for such packets.

5.3.4 Compression/Decompression Flow in GMR-1 3G LLA Profile

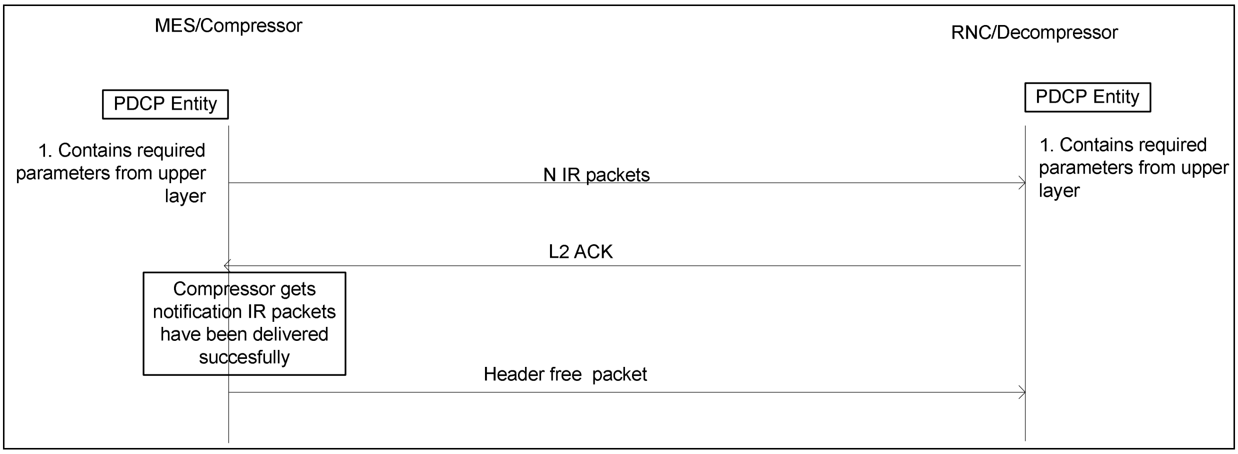

Figure 9: GMR-1 3G LLA Header SuppressionProfile

The two PDCP entities initially are configured by the upper layers with the header compression protocol to be used and the parameters for those protocols. The compressor sends N IR packets to the decompressor in RLC acknowledged mode to establish static context. This ensures reliable delivery of IR packets to the decompressor. Upon confirmation from L2 that all frames belonging to IR packets have been successfully delivered to the decompressor, the compressor informs the NAS regarding the successful creation of the PDCP contexts at the decompressor. For subsequent procedures, the compressor always sends an IR packet in RLC acknowledged mode to update or to create a new context at the decompressor. After this, the compressor proceeds with sending header free packets to the decompressor side.

5.4 SRNS Relocation

5.4.0 General

In case of Satellite Radio Network Subsystem (SRNS) Relocation, the upper layer indicates to PDCP to perform either the re-initialization or the context relocation of the compression protocols of an RB. In the present specification, context relocation is only applicable to GMR-1 3G LLA Header Suppression.

5.4.1 Lossless SRNS Relocation

Not supported in GMR-1 3G.

5.4.2 Context Relocation for GMR-1 3G LLA Header Suppressed Session

For the GMR-1 3G LLA Header Suppression, the snapshot of forward and return directions of the header compressor context and the frame number shall be transferred to the target RNC. The target RNC shall recreate header compressor context based on this information. The header compressor context is shown in Table 4.

Table 4: Header Compressor Context Relocation for GMR-1 3G LLA Header Suppression

Context Type	Field	Bits	Comment
Static IPv4	IP Version	4	
	ToS	8	Only if IP version = 4
	Source addr	32	Only if IP version = 4
	Dest addr	32	Only if IP version = 4
Static IPv6	IP Version	4	
	Traffic class	8	Only if IP version = 6
	Flow label	20	Only if IP version = 6
	Source addr	128	Only if IP version = 6
	Dest addr	128	Only if IP version = 6
Static UDP	Source port	16	
	Dest port	16	
Static RTP	RTP version	2	
	Payload type	7	
	SSRC	32	
Dynamic	IPv4 ID	16	Only if IP version = 4
	RTP sequence	16	
	RTP timestamp	32	
	Last frame number	32	Last frame number received in UL

After handover has been completed, the PDCP at the new RNC shall generate the dynamic parts of the context according to the following procedures:

- New IP Id = Old IP ID + (new FN - old FN)
- New RTP Seq = Old RTP Sequence Number + 1
- New RTP timestamp = Old RTP timestamp + (new FN - old FN) x 320
- The above procedure shall account for FN rollover

5.4.3 Context Relocation for All other Session Type

Not supported in GMR-1 3G.

5.5 Lossless DL RLC PDU Size Change

Not supported in GMR-1 3G.

5.6 General Procedures

5.6.1 PDCP Sequence Numbering

The value of the PDCP sequence number ranges from 0 to 65 535. The PDCP SN window size indicates the maximum number of PDCP SDUs, not confirmed to have been successfully transmitted to the peer entity by lower layer, that can be numbered at any given time. The PDCP SN window size is configured by upper layers. PDCP sequence numbers are set to "0" when the PDCP entity is set-up for the first time.

In the following, the "submission/reception of a PDCP SDU to/from lower layer" is used as a synonym for the submission/reception of a PDCP Data PDU or a PDCP SeqNum PDU to/from lower layer that carries in its data field a compressed or uncompressed PDCP SDU.

PDCP sequence numbers shall not be decremented in a PDCP entity.

5.6.2 PDCP Sequence Number Synchronization

Not supported in GMR-1 3G.

5.6.3 Sequence Number and Data Forwarding

Not supported in GMR-1 3G.

5.7 Header Compression and Decompression for MBMS

5.7.0 General

Not supported in GMR-1 3G.

5.7.1 Cell Change Inside the Same Cell Group

Not supported in GMR-1 3G.

5.7.2 Cell change between cell groups

Not supported in GMR-1 3G.

6 Services

6.1 Services Provided to Upper Layers

The following services are provided by PDCP to upper layers:

- Transfer of user data.
- Maintenance of PDCP SDU sequence numbers.

6.2 Services Expected from RLC Layer

For a detailed description of the following functions, see ETSI TS 125 322 [3]:

- Transparent data transfer Service.
- Unacknowledged data transfer Service.
- Acknowledged data transfer Service.

7 Elements for Layer-to-Layer Communication

7.0 General

The interaction between the PDCP layer and other layers are described in terms of primitives where the primitives represent the logical exchange of information and control between the PDCP layer and other layers. The primitives shall not specify or constrain implementations.

7.1 Control Plane

7.1.1 Primitives between PDCP and Upper Layers

The primitives between PDCP and upper layers are shown in Table 5.

Table 5: Primitives Between PDCP and Upper Layers

Generic Name	Parameter			
	Req.	Ind.	Resp.	Conf.
CPDCP-CONFIG	PDCP-Info, RLC-SAP SN_Sync, R/I/C/RS, Context-Info	Not Defined	Not Defined	Not Defined
CPDCP-CONTEXT	None	Not Defined	Not Defined	Context-Info
CPDCP-RELEASE	RLC-SAP	Not Defined	Not Defined	Not Defined
CPDCP-SN	PDCP SN	Not Defined	Not Defined	Not Defined

Each Primitive is defined as follows:

1) CPDCP-CONFIG-Req:

- CPDCP-CONFIG-Req is used to configure and - in the case of already existing PDCP entity - to reconfigure a PDCP entity and to assign it to the radio bearer associated with that entity.

2) CPDCP-RELEASE-Req:

- CPDCP-RELEASE-Req is used by the upper layers to release a PDCP entity.

3) CPDCP-SN-Req:

- This primitive is used at the network side. CPDCP-SN-Req is used to transfer the PDCP SN to PDCP.

4) CPDCP-CONTEXT-Req./Conf:

- CPDCP-CONTEXT-Req initiates specific actions in the source RNC in order to perform context relocation as a part of the SRNS relocation. The primitive is applicable only in the source RNC.

- CPDCP-CONTEXT-Conf is used to transfer the header compression context information from PDCP to upper layer in order to perform context relocation as a part of the SRNS relocation. The primitive is applicable only in the source RNC.

The following parameters are used in the primitives:

1) PDCP-Info:

- Contains the parameters for each of the header compression protocols configured to be used by one PDCP entity.

2) RLC-SAP:

- The RLC-SAP (TM/UM/AM) is used by the PDCP entity when communicating with RLC sublayer.

3) SN_Sync:

- Indicates that PDCP should start the PDCP SN synchronization procedure.

4) Next_Send_SN:

- The Send PDCP SN of the next PDCP SDU to be sent. There is one in the uplink (UL_Send PDCP SN) and one in the downlink (DL_Send PDCP SN). Refer to clause 5.4.1.

5) Next_Receive_SN:

- The Receive PDCP SN of the next PDCP SDU expected to be received. There is one in the uplink (UL_Receive PDCP SN) and one in the downlink (DL_Receive PDCP SN). Refer to clause 5.4.1.

6) PDCP SN:

- This includes a PDCP sequence number.

7) R/I/C/RS:

- Indicates that PDCP should Re-initialize (R)/Initialize (I) the header compression protocols. Alternatively (Context-relocation, C), it indicates that the MES PDCP shall perform specific actions related to context relocation during SRNS relocation. (RS) indicates to Re-initialize while keeping the static part of the header compression (only for IETF RFC 3095 [5]). The R/I/C/RS indication is given separately for each of the configured header compression protocols, if several exist for a given radio bearer.

8) Context-Info:

- Contains the header compression context information of each of the header compression protocols that are subject to the context relocation during SRNS relocation.

7.2 User Plane

7.2.1 Primitives Between PDPC and Upper Layer

PDCP interacts with upper layer in the user plane at PDCP-SAP.

Table 6: Primitives Between PDCP and Upper Layers in User Plane

Generic Name	Parameter			
	Req.	Ind.	Resp.	Conf.
PDCP-DATA	Data	Data	Not Defined	Not Defined

PDCP-DATA-Req./Ind.

PDCP-DATA-Req is used by the upper user-plane protocol layers to request a transmission of an upper layer PDU. PDCP-DATA-Ind is used to deliver a PDCP SDU that has been received to upper user plane protocol layers.

7.2.2 Primitives Between PDPC and RLC

Table 7: Primitives Between RLC and Upper Layers

Generic Name	Parameters			
	Req.	Ind.	Resp.	Conf.
RLC-AM-DATA	Data, CNF, DiscardReq, MUI, MES-ID type indicator	Data, DiscardInfo	Not Defined	Status, MUI
RLC-UM-DATA	Data, MES-ID type indicator, DiscardReq, MUI	Data	Not Defined	MUI
RLC-TM-DATA	Data, MES-ID type indicator, DiscardReq, MUI	Data, Error_Indicator	Not Defined	MUI

Each Primitive is defined as follows:

1) RLC-AM-DATA-Req/Ind/Conf:

- RLC-AM-DATA-Req is used by the upper layers to request transmission of an RLC SDU in acknowledged mode.

- RLC-AM-DATA-Ind is used by the AM RLC entity to deliver to the upper layers an RLC SDU that has been transmitted in acknowledged mode and to indicate to the upper layers the discarded RLC SDU in the peer RLC AM entity.

- RLC-AM-DATA-Conf is used by the AM RLC entity to confirm to the upper layers the reception of an RLC SDU by the peer-RLC AM entity or to inform the upper layers of a discarded SDU.

2) RLC-UM-DATA-Req/Ind/Conf:

- RLC-UM-DATA-Req is used by the upper layers to request transmission of an RLC SDU in unacknowledged mode.
- RLC-UM-DATA-Ind is used by the UM RLC entity to deliver to the upper layers an RLC SDU that has been transmitted in unacknowledged mode.
- RLC-UM-DATA-Conf is used by the UM RLC entity to inform the upper layers of a discarded SDU.

3) RLC-TM-DATA-Req/Ind/Conf:

- RLC-TM-DATA-Req is used by the upper layers to request transmission of an RLC SDU in transparent mode.
- RLC-TM-DATA-Ind is used by the TM RLC entity to deliver to the upper layers an RLC SDU that has been transmitted in transparent mode.
- RLC-TM-DATA-Conf is used by the TM RLC entity to inform the upper layers of a discarded SDU.

Primitive Parameters

The following parameters are used in the primitives:

1) The parameter data is the RLC SDU that is mapped onto the data field in RLC PDUs. When AM or UM RLC entities are used, the length of the data parameter is a multiple of 8 bits; otherwise when the TM RLC entity is used the length of the data parameter is a bit-string whose length need not be a multiple of 8 bits.

2) The parameter Confirmation Request (CNF) indicates whether the transmitting side of the AM RLC entity needs to confirm the reception of the RLC SDU by the peer-RLC AM entity. If required, once all AMD PDUs that make up the RLC SDU are positively acknowledged by the receiving AM RLC entity, the transmitting AM RLC entity notifies the upper layers.

3) The parameter Message Unit Identifier (MUI) is an identity of the RLC SDU, which is used to indicate which RLC SDU that is confirmed with the RLC-AM-DATA-Conf. primitive, or discarded with the RLC-AM/UM/TM-DATA-Conf. primitive.

4) The parameter DiscardInfo indicates to the upper layer the discarded RLC SDU in the peer-RLC AM entity. It is applicable only when in-sequence delivery is configured and it is to be used when the upper layers require the reliable data transfer.

5) The Error_Indicator parameter indicates that the RLC SDU is erroneous. Erroneous SDU will be discarded (see Table 7 and its explanation).

6) The parameter MES-ID type indicator indicates the RNTI type (U-RNTI or C-RNTI) to be used for the associated RLC SDU. This parameter is not required at the MES.

7) The parameter DiscardReq indicates whether the transmitting RLC entity needs to inform the upper layers of the discarded RLC SDU. If required, the transmitting RLC entity notifies the upper layers when the SDU is discarded.

8) The parameter Status is only applicable for AM operation. This parameter indicates whether a RLC SDU is successfully transmitted or discarded.

8 Elements for Peer-to-Peer Communication

8.1 Protocol Data Units

Different PDU formats are defined for the PDCP protocol: one not introducing any overhead to the (compressed) PDCP SDU, others introducing such overhead.

8.2 Formats

8.2.0 General

A PDCP PDU shall be a multiple of 8 bits if the RLC entity is configured for unacknowledged or acknowledged mode. Otherwise, if the RLC entity is configured for transparent mode, it is bit-aligned. In Tables 8, 9 and 10, bit strings are represented as follows: the first bit is the leftmost one on the first line of the table, the last bit is the rightmost on the last line of the table, and more generally, the bit string is to be read from left to right and then in the reading order of the lines.

SDUs are bit strings with any non-null length. If not compressed within PDCP an SDU is included from the first bit onward.

8.2.1 PDCP-No-Header PDU

The PDCP-No-Header PDU does not introduce any overhead to the PDCP SDU. The use of the PDCP-No-Header PDU is configured by the upper layer.

The format of the PDCP-No-Header PDU is shown in Table 8.

Table 8: PDCP-No-Header PDU

Data

8.2.2 PDCP Data PDU

The PDCP Data PDU is used to convey:

- Data containing an uncompressed PDCP SDU; or
- Header compression related control signalling; or
- Data that has been obtained from PDCP SDU after header compression.

The format of the PDCP Data PDU is shown in Table 9.

Table 9: PDCP Data PDU Format

<table>
<tr><td>PDU type</td><td>PID</td></tr>
<tr><td colspan="2">Data</td></tr>
</table>

8.2.3 PDCP SeqNum PDU

The PDCP SeqNum PDU is used to convey a PDCP SDU sequence number and:

- Data containing an uncompressed PDCP SDU; or
- Data that has been obtained from PDCP SDU after header compression.

The format of the PDCP SeqNum PDU is shown in Table 10.

Table 10: PDCP SeqNum PDU Format

PDU type	PID
Sequence number	
Data	

8.3 Parameters

8.3.0 General

If not otherwise mentioned in the definition of each field, then the bits in the parameters shall be interpreted as follows: the left most bit in the first row is the most significant bit; and the right most bit is the last row is the least significant bit.

Unless otherwise mentioned, integers are encoded in standard binary encoding for unsigned integers. In all cases, the bits appear ordered from MSB to LSB when read in the PDU.

8.3.1 PDU Type

Length: 3 bits.

The PDU Type field indicates the type of PDCP data PDU. The PID field identifies the exact header compression packet type and the setting of the PID values is described in clause 5.1.1.

Table 11: PDU Type Field Bit 0

Bit 0	PDU Type
0	Payload Data is uncompressed
1	Payload Data is compressed

Table 12: PDU Type Field Bit 1 and 2

Bit 1 and 2	PDU Type
00	PDCP Data PDU
01	PDCP SeqNum PDU
10 to 11	Reserved (PDUs with this encoding are invalid for this version of the protocol)

If the number of ROHC CIDs does not exceed 32, PDU type 10 (ROHC CID packet, Table 12) is used to identify the PDCP packet type. In this case the ROHC CID is carried in PID field defined in Table 1.

If the number of CIDs is greater than the number that can be accommodated in the PID field (Table 1), which is 32, of the PDCP PDU header, then the CID of 1 or 2 octets in the ROHC packet is used. In this case the PDU type bit 1 and 0 is 00 (PDCP Data PDU, Table 12).

Bit 2 of the PDU Type indicates the data compression mode. If this bit is 1, the data decompressor should reset the dictionary.

8.3.2 PID

Length: 5 bits.

The PID field indicates the used header compression and packet type or a context identifier.

Table 13: PID Field

Bit	Description
00000	No header compression
00001 to 11111	Dynamically negotiated header compression identifier, as described in clause 5.1.1

The PID field value indicates the used header compression protocol type and packet type, or CID. A specific header compression protocol may utilize a certain range of consecutive values from the PID field value space for different packet types. The Receiving PDCP entity should perform the necessary operation (e.g. header decompression) according to the PID field value.

8.3.3 Data

The Data field may include any one of the following:

- Uncompressed PDCP SDU
- Header compressed PDCP SDU
- Header compression protocol feedback information

8.3.4 Sequence number

Length: 16 bits

PDCP SDU sequence number.

9 Handling of Unknown, Unforeseen, and Erroneous Protocol Data

9.1 Invalid PDU Type

If a PDCP entity receives a PDCP PDU with a PDU Type set to Reserved (see clause 8.3.1), it shall:

- Discard the PDCP PDU

If a PDCP entity is not configured for lossless SRNS Relocation or lossless DL RLC PDU size change and it receives a PDCP SeqNum PDU, it shall:

- Discard the PDCP SeqNum PDU

9.2 Invalid PID value

If a PDCP entity receives a PDCP PDU with a PID value that is not mapped with a valid packet type (see clause 5.1.1), it shall:

- Discard the PDCP PDU

Annex A (normative): GMR-1 3G LLA Header Suppression

A.1 Introduction

A.1.0 General

A GMR-1 3G-enabled RNC shall only be deployed in All-IP UMTS networks.

GMR-1 3G introduces VoIP speech services by interfacing with a packet domain UMTS core network using the 3GPP standard Iu-PS interface (see ETSI TS 125 401 [i.1]). Real time protocol (RTP) (see IETF RFC 3550 [i.2]) is used to carry encoded speech data for VoIP services.

In order to ensure that the smallest hand-held terminals can close the link margin for the voice service, the service will be provided on PNB(1,6) or PNB(1,3) uplink channels. The downlink channels can be shared. GMR-1 3G spectrally-efficient VoIP service uses either a 2,45 or 4 kbps codec.

The satellite link is a resource-limited system in which any additional header information can significantly affect the total capacity. The IETF RFC 3095 [5] RoHC scheme that reduces RTP/UDP/IP header size to 1 to 4 octets is not sufficient and hence a header compression scheme is defined in the present document where no protocol header is transmitted over the GMR-1 3G satellite radio interface by completely suppressing the IP/UDP/RTP headers. This header suppression scheme requires GMR-1 3G lower layer assistance. The GMR-1 3G LLA Header Suppression scheme is supported in both uplink and downlink directions.

A.1.1 GMR-1 3G LLA Header SuppressionService Options

The GMR-1 3G LLA Header Suppression service transports IP/UDP/RTP for SE-VoIP service that is optimized for the GMR-1 3G radio interface.

The GMR-1 3G LLA Header Suppression that is supported on the satellite radio interface is an optimized version of the ROHC LLA profile. It is adapted for use on the GMR-1 3G Satellite Radio Interface as follows:

- Not all the packet types mentioned in ROHC LLA are used in the GMR-1 3G LLA.
- In the ROHC LLA, both static and dynamic contexts are established by exchanging packets between compressor and decompressor over the air. In the GMR-1 3G LLA, the radio interface packet exchange is required only to create the static context. The creation of dynamic context information is localized at the receiving side as described in the subsequent clauses of this annex.

A.2 Silence Suppression

The GMR-1 3G LLA Header Suppression is supported for voice codecs is operating either at 2,45 kbps or 4 kbps. The following paragraphs describe characteristics of the codec, including its silence packets and comfort noise generation concepts.

During a normal conversation, the participants typically alternate their speech, such that on average each direction of transmission is occupied about 50 % of the time. Discontinuous Transmission (DTX) is a mode of operation where the transmitters are switched on only for those frames that contain useful information. This is done for the following three purposes:

- In the MES, battery life will be prolonged, or a smaller battery could be used for a given operational duration.
- The average transmit power level over the radio interface is reduced, leading to better Radio Frequency (RF) spectrum efficiency.
- Satellite power utilization on the forward link is optimized.

However, to support the working of the GMR-1 3G LLA header suppression scheme, the DTX feature related to silence packets and comfort noise has been moved from the codec to the GMR-1 3G link layer.

The voice codec that is supported by GMR-1 3G LLA header suppression generates frames every 20 ms, each of which could be a silence pattern or a voice frame:

- On the network side, the Media gateway sends 20 ms voice packets to the RNC.
- On the MES side, during any active voice period, the codec generates voice frames. The packetization module creates an RTP packet for each 20 ms voice frame.
- If the upper layer sends the VoIP packet in 20 ms frame size, the lower layer at the RNC and at the MES will combine two 20 ms packets before transmitting it over the air.
- If the upper layer sends the VoIP packet in 40 ms frame size, the lower layer at the RNC and at the MES will not need to combine the packets, it just transmits it over the air.
- During an inactive voice period (silence period), a silence frame is generated every 20 ms. The packetization module creates an RTP packet for each 20 ms silence frame. The payload of a silence frame packet will always have a unique pattern: this is the silence descriptor (SID) that distinguishes silence frame packets from real voice packets. These silence frame packets have the same payload length as real voice packets.
- The silence frame and voice packets belong to the same RTP stream, and thus, maintain continuity of RTP sequence numbers.
- If the decoder does not receive any packets during its sampling period, it either generates comfort noise or error concealment packets. Error concealment packets are generated if the last received packet is a voice packet, while comfort noise is generated if the last received packet is a silence packet, The information on the last silence packet is used to generate subsequent silence packets in 20 ms interval when no packets are received. If decoder does not receive frames or receives frames with errors during its consecutive N sampling periods, the decoder controlling application mutes the decoder.

A.3 GMR-1 3G LLA Header Suppression Overview

A.3.0 General

In order to completely eliminate the compressed header in GMR-1 3G, all the functionality normally provided by the 1-octet header in the IETF RFC 3095 [5] scheme has to be provided by other means, typically by utilizing functionality provided by the lower layers. The entity that implements the interface between GMR-1 3G LLA profile and the lower layer is called the assisting layer (AL). The GMR-1 3G LLA profile that performs header compression is termed the HCU application (HC, Upper Layer). The assisting layer is realized by the HCL application (HC, Lower Layer), which adapts between the GMR-1 3G LLA implementation and the physical layer. This assistant layer is assumed to have knowledge of the physical layer synchronization.

A.3.1 Context and Functional Blocks

Figure A.1 shows the functional blocks involved in GMR-1 3G LLA Header Suppression at both the transmit and receive sides. For the purpose of the present document, the terms "transmitting side" and "receiving side" are used in relation to the type of interaction with the link layer.

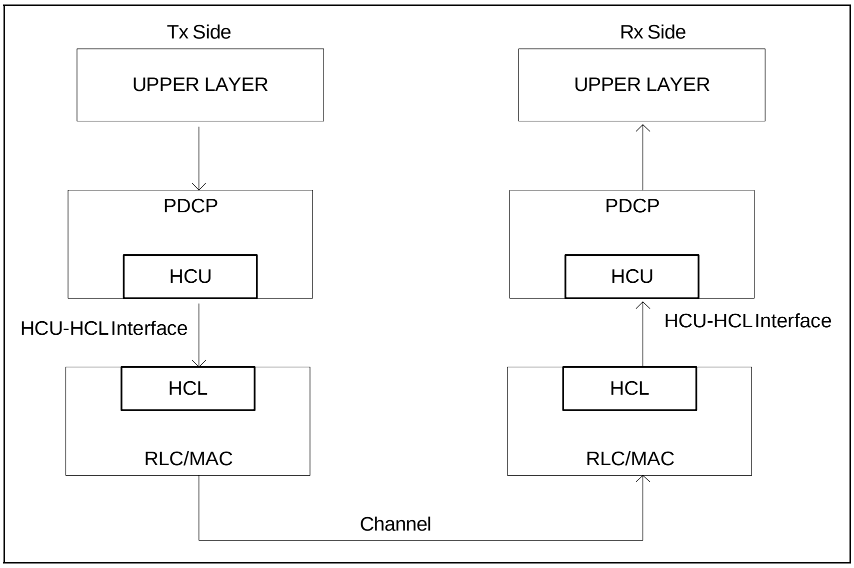

Figure A.1: GMR-1 3G LLA Functional Blocks

In GMR-1 3G, the HCU application resides in the PDCP layer and performs header compression and decompression. The HCL application resides in the RLC/MAC layer and provides link layer assistant functions.

Specifically, the transmitting side refers to the HCU-HCL pair for which the HCL application is supplying data to the link layer. Similarly, the receiving side refers to the HCU-HCL pair for which the HCL application is receiving data from the link layer.

GMR-1 3G LLA Header Suppression will be used only for the SE-VoIP flow. The PHY layer transmission bursts for a spectrally-efficient VoIP flow are allocated the same number of slots on each TDMA frame.

Packets will not have any header, nor any context identifier, and can only be sent from the compressor to decompressor.

On the uplink, due to the use of a dedicated channel, it is possible to achieve RLC transparent mode data transmission, i.e. it is not required to send any RLC/MAC headers. On the downlink, shared channels are used, so a RLC/MAC header should be present. However, on a 1x downlink channel, a dedicated channel is provided where a RLC/MAC header will not be present.

On a spectrally-efficient voice bearer channel, the burst may carry the following types of payloads:

- Real voice
- Silence frame

The real voice and silence frame packets are distinguished from each other by the silence identifier field. The silence frame payload always has a unique pattern known as SID (or the silence identifier).

The RRC entity for the spectrally-efficient VoIP service RAB will create one PDCP entity and two radio bearer IDs for the RLC/MAC, that is, one in transparent mode and the other in ACK mode. The context initialization packet (described later) will be sent via the ACK mode RLC instance, and header stripped packets will be sent via the RLC transparent mode.

A.3.2 HCU Application Overview

The HCU application is the process above the HCL application that generates and consumes data blocks. The HCU application refers to the header compression algorithm.

To achieve GMR-1 3G LLA Header Suppression, the header decompressor residing in the HCU application at the receiving side is required to regenerate some of the information needed to decompress the IP/UDP/RTP headers of header-free packets as follows:

- The HCU only obtains the Payload from the HCL.

- The HCU then regenerates the following information:
  - Packet type (voice or silence frame).
  - Vocoder Rate (2,45 kbps or 4 kbps).
  - RTP timestamp.

The HCU uses the following important functions supported by the HCL application:

- In-order delivery of data frames over the radio interface every 40 ms (hence no extra header is used for in-order delivery).
- Detection of packet losses over the radio interface.
- TDMA frame number.
- Recognition of voice or silence frame.

A.3.3 HCL Application Overview

To achieve GMR-1 3G LLA Header Suppression, the HCL application on the transmitting side:

- Sends voice packets or silence frame packets (with no headers).
- For silence frame packets, the first one is sent and then the HCL drops consecutive silence frame packets for the next one second. If there are many consecutive silence frame packets, one packet is sent over the air every second, thereby providing DTX operation.

The HCL application on the receiving side:

- Provides a loss indication to the associated connected service when a packet is damaged (e.g. when the physical frame is received with insufficient physical layer frame quality or is lost). The physical layer has a mechanism to detect a packet loss on a dedicated channel.

A.4 GMR-1 3G LLA Header Suppression End-to-End Flow

A.4.1 GMR-1 3G LLA Header Suppression Flow for Downlink

A.4.1.0 General

This clause describes the end-to-end flow for the GMR-1 3G LLA Header Suppression between the RNC and MES. On the downlink, either a DCH or a PDCH channel will be used for the SE-VoIP service. A DCH channel is only used if the channel is 1x.

A.4.1.1 Transmitting Side

The PDCP receives 40 ms or 20 ms RTP packets from the Core Network. These are handled as described below:

- The transmitter side in the PDCP creates a compression context for the downlink based on the packet initialization sent by the PDCP at the MES side. The compression context creation procedures are explained in clause A.4.2.
- The HCU on the transmitting side does not need to initialize the PDCP context at the peer HCU on the receiving side (at the MES). RTP packets received from the core network contain a 1 byte extension header between the normal RTP header and the payload. This extension header comprises 4 bits for frame type, 4 bits reserved, see Figures A.2 and A.3, and Table A.1.

- The PDCP shall compute the IP header checksum for IPv4 packets and UDP checksum for IPv6 packets. If there is a checksum verification error, the PDCP shall drop the packet. The PDCP should maintain statistics about how many RTP packets have been dropped.
- The PDCP shall classify RTP packets as voice, DTMF/Tone, or SID (Silence Insertion Descriptor) based on the well-known bit patterns of the payload.
- The transmitter side HCU application in the PDCP shall strip the RTP/UDP/IP header and the 1 byte extra header, and shall send the 40 ms or 20 ms payload to the RLC/MAC on the transparent mode interface. Along with the payload, the PDCP shall also send the timestamp from the RTP header, the classification (voice or SID), and the frame type information to the RLC/MAC.
- The HCL in the RLC/MAC shall send the packet to the PHY every 40 ms including an indication of the type of the packet (i.e. real voice, SID, DTMF/Tone, or frame erase). Note that if the packets come from the network every 20 ms, the RLC/MAC shall combine two 20 ms packets before sending it to the physical layer.
- The transmitter side HCL in the RLC/MAC provides the DTX operation, which means that it shall suppress silence frame packets in such a way that one such packet goes out to the physical layer once a second if there is a long silence period.
- If there is no packet required, the RLC/MAC schedules no header voice packet to be sent on the appropriate TBF and informs the physical layer.
- On dedicated channels, no RLC/MAC header is added. On a shared channel, although the same slot position, is used on every frame for a specific MES; a few bits of RLC/MAC header is added.
- When a packet is dropped by the PDCP or the HCL suppresses a silence frame packet, the RLC/MAC scheduler should schedule a packet from a best effort service (if available) on the slot where a voice or silence frame packet for this MES should have been sent. If there is no packet to fill the slot, the PHY layer should send a power control packet.
- The PHY classifies spectrally-efficient VoIP bursts from other bursts by using different unique words (UW). Note that IR packets and normal VoIP packets are sent on the same MAC slots.

IPv4/IPv6 Hdr (20/40 Bytes)	UDP Hdr (8 Bytes)	RTP Hdr (12 Bytes)	Ext Hdr (1 Byte)	Payload

Figure A.2: Position of Extension Header

Frame Type (4 bits)	Reserved (4 bits)

Figure A.3: Extension Header Contents

Table A.1: Extension Header Coding

Frame Type	Vocoder Rate
0	4,0 kbps
1	2,45 kbps
2-15	Reserved

A.4.1.2 Receiving Side

The receiving side for the downlink flow receives a headerless packet every 40 ms TDMA frame. These are handled as described below:

- PHY receives bursts on the downlink on timeslots allocated for the SE-VoIP service. On the downlink, the channel could be PDCH or DCH.
- PHY interprets whether a burst contains a suppressed SE-VoIP packet or other packet (including the IR packet and KAB3) by the value of the UW (Unique Word).
- PHY sends good packets to the RLC/MAC along with the TDMA frame number on which the packet was received, the burst classification information and the vocoder rate.
- Based on the burst classification information, the RLC/MAC decides whether it should send the SDU to the PDCP entity either through the ACK mode RLC instance or Transparent mode RLC instance. Any IR packets shall be sent to the ACK mode RLC instance. The RLC/MAC shall ignore any KAB3 bursts received from the PHY layer.
- The receiving side HCL in the RLC/MAC shall not split one RTP packet it receives every 40 ms into two data blocks. It passes the whole packet to the HCU layer.
- The HCU layer in the PDCP, upon receiving headerless packet from HCL, shall regenerate the IP/UDP/RTP header for the packet from the context information and send it to the upper layer.
- The HCU layer only has the static context. The dynamic context is handled as described below. The dynamic context contains the following information:
    - RTP Sequence number
    - IP Identification (in IPv4 packet headers)
    - RTP Timestamp
    - CRC
    - Traffic Class (in IPv6 packet headers)
    - Time-to-live (in IPv4 and Ipv6packet headers)
    - DF, RND, and NBO flags (in IPv4 packet headers)
    - Generic extension header list (in IPv4 and Ipv6 packet headers)
- The receiving side shall assume random number X as the start value for the RTP sequence number and random number Z as the start value for the RTP timestamp when the context is established. Thereafter, the HCU layer shall use the frame number to generate the subsequent RTP sequence numbers and timestamp every 40 ms:
    - The RTP timestamp shall be increased according to the size of the voice packetization. For example for packetization of 40 ms voice, sampled at 8 000 Hz, the RTP timestamp shall increase by 320 for each elapsed TDMA frame.
- For IP Identification, another random number Y is assumed as the IP ID when the context is established. Every time the sequence number is incremented, the IP ID is also incremented by the same amount. The traffic class, time-to-live, and the flags are regenerated locally at the receiving side. The extension headers are not present in VoIP packets so they do not need to be handled.
- The HCU in the PDCP sends the resulting 40 ms RTP packets (complete with regenerated headers) to the upper layer.
- If the vocoder does not receive the next packet after playing a voice packet, it shall play error concealment packets. If the vocoder does not receive the next packet after playing a SID packet, it shall play the same SID packet until receives the next packet.

A.4.2 GMR-1 3G LLA Header Suppression Flow for Uplink

A.4.2.1 Transmitting Side

The PDCP receives RTP packets from the upper layer (from vocoder) every 20 ms along with the vocoder rate information. These are handled as described below:

- The RTP payload types for the 4,0 kbps and the 2,45 kbps vocoder shall be unique.
- RTP packets received from the upper layer shall contain a 1 byte extension header between the normal RTP header and the payload comprising, 4 bits for frame type, 4 bits reserved, (see Figures A.2 and A.3, and Table A.1).
- The HCU in the PDCP classifies packets as either voice or silence (SID), based on the unique pattern of the payload. DTMF/Tone packets shall be treated as voice packets.
- The HCU shall suppress the headers of all packets, including the 1 byte extension header, and send each 20 ms packet to the HCL along with the following information:
    - RTP timestamp
    - Vocoder rate
    - Payload type: voice, DTMF/tone, SID
- The HCL shall combine two 20 ms packets into one 40 ms packet and shall send each 40 ms packet along with the classification and vocoder rate information to the PHY. If the two 20 ms packets combined into one 40 ms packet are of different types (i.e. voice and SID), the HCL shall indicate to the PHY the type as that of the first 20 ms packet.
- When HCL encounters the first SID packet after a sequence or one or more voice packets, it shall transmit the complete SID packet. When there are many consecutive SID packets, the HCL sends to the PHY only one 40 ms SID packet every one second. Thus the HCL provides the DTX functions.

NOTE: The PHY will send a power control frame if there is no other packet to send.

A.4.2.2 Receiving Side

The receiving side for the uplink, receives a headerless packet every TDMA frame.

- Every 40 ms TDMA frame the PHY detects the burst received on the DCH channels or PDCH channels. Two cases are possible:
    - PHY detects a good packet
    - PHY detects CRC error
- The PHY sends the good packets to the RLC/MAC. The PHY also provides vocoder rate information to the RLC/MAC.
- The RLC/MAC shall then send the packet to the appropriate PDCP entity along vocoder rate.
- The PDCP regenerates the IP/UDP/RTP header and adds this to the packet received from the RLC/MAC.
- The receiving side shall assume random number X as the start value for the RTP sequence number and random number Z as the start value for the RTP timestamp when the context is established. Thereafter, the HCU layer shall use the frame number to generate the subsequent RTP sequence numbers and timestamp every 40 ms:
    - The RTP timestamp shall be increased according to the size of the voice packetization. For example for packetization of 40 ms voice, sampled at 8 000 Hz, the RTP timestamp shall increase by 320 for each elapsed TDMA frame.
- The HCU in the PDCP sends 40 ms RTP packets towards the core network.

- If the vocoder does not receive packets after playing a voice packet, it shall play error concealment packets. If there is no packet in 60 ms or more, the voice decoder shall mute. If the vocoder does not receive packets after playing a SID packet, it shall play the same SID packet until it receives the next packet.

A.4.2.3 PDCP Context Creation

When the PDCP at RNC receives an uplink initialization packet to create the decompression context for the uplink, the PDCP at the RNC shall also use this initialization packet to create the compressor context for the downlink. The PDCP shall use the uplink destination IP address packet as its source IP address and the uplink source IP address packet as its destination IP address, see Figure A.4.

The RTP payload type shall uniquely identify either a 4,0 kbps or 2,45 kbps vocoder.

Similarly for the UDP port, the PDCP shall use the uplink destination UDP port packet as its source UDP port and the uplink source UDP port packet as its destination port. All other static headers in the RTP/UDP/IP headers shall be created by the PDCP. If the PDCP at the RNC side receives a RTP packet from the Core Network that cannot be associated with the existing compression context, the PDCP shall drop this RTP packet.

At the MES side, after a SIP PRACK message is received, the SIP application layer shall convey information from the SDP content to the PDCP to be used to create the PDCP context. The PDCP context shall be sent also to the RNC immediately. Hence the probability of conflict between sending PDCP initialization to the RNC and sending SIP 180 Ringing can be minimized (see IETF RFC 3261 [11]).

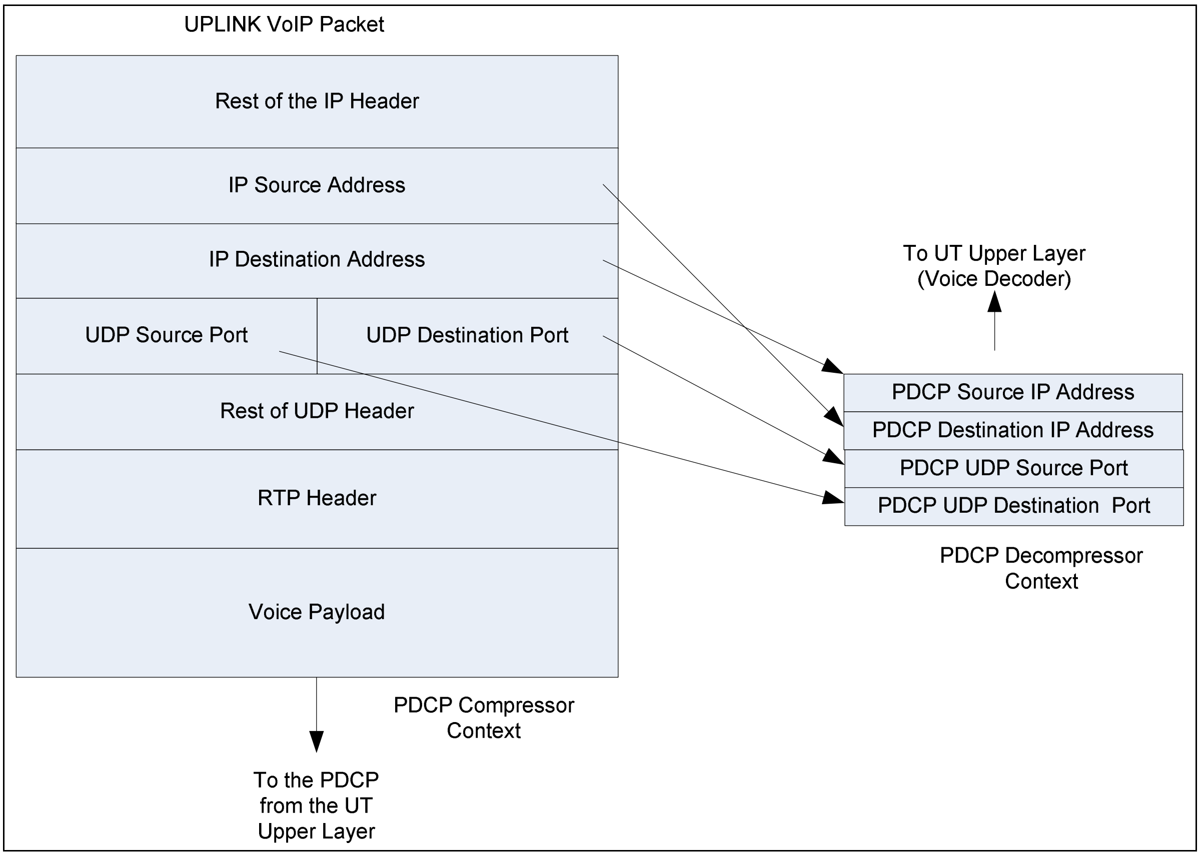

Figure A.4: PDCP Context Creation

The PDCP layer shall also send this context as an IR packet to the RNC to allow the RNC to create the decompressor context for the uplink traffic. The RNC shall use this information to create the static parts of the header decompression context. The RNC shall also use the IR packet to create context for the downlink direction by switching the IP/port source with IP/port destination as done in the MES.

Annex B (normative): ROHC Performance Testing

B.1 Introduction

B.1.0 General

This annex defines performance test cases for ROHC.

The ROHC profile within scope is profile 0x0001 for compression of RTP/UDP/IP headers only. This annex is not meant to bring incoherent limitations to implementations, and is not meant to create a sub-specification of IETF RFC 3095 [5] either, as a consequence of the requirements on performance that it defines. Note that this test is based on 20 ms frame size with 8 kHz sampling rate, hence the timestamp increase by multiple of 160. For 40 ms frame size, the timestamp calculations shown in this clause should be multiplied by a factor of two, i.e. the timestamp shall increase by multiple of 320.

B.1.1 Purpose of the Performance Testing

The purpose of the test cases is to ensure that ROHC implementations meets minimal requirements that can fairly be expected when subjected to an input sequence that includes frequently occurring and commonly observed changes in the values of header fields. The metrics used correspond to:

- The average compressed header size for an entire test sequence to assess an implementation's efficiency in terms of its overall compression ratio.
- The average compressed header size for different sub-sequences within each test case to assess an implementation's ability to minimize the variance in compressed header size with respect to the selection of the packet format used for individual packets.

The latter is to avoid a less desirable behaviour where a compressor would consistently and exclusively use two types of compressed header formats; one format that offers no compression but that completely updates and/or repairs the context, e.g. when the patterns of the header fields to be compressed do not allow optimal compression, and another format that provides the most efficient compression ratio otherwise. While it is noted that the use of larger headers is perfectly acceptable protocol-wise IETF RFC 3095 [5], the tests herein are meant to encourage compressor implementations to actively and efficiently implement compression.

These metrics are defined so that they are not impacted by an implementation's specific robustness algorithm(s) as well as to allow a wide range of compression strategies.

Compressor implementations are expected to implement robustness algorithms according to the optimistic approach for the U/O-modes of operation. The optimistic approach is the part of the selection of the packet format where a format that contains the necessary information to update a field is used a number N times, starting from the packet for which a new value has to be established in the decompressor context. While N is an implementation parameter, the metrics for each sequence in U/O-mode is expressed in terms of this parameter. Implementation should use the value N as an input parameter for the testing, to adapt to the expected robustness level required for the testing. The value of the parameter N is defined in each test case definition separately (the informative value is given in clause C.2.3). Similarly, R-mode operation requires that an update be conveyed to the decompressor until it gets acknowledged; however for R-mode, the relevant test cases provide explicit feedback messages when necessary.

The performance tests for ROHC as described in this clause are carried out by providing a sequence of uncompressed IP/UDP/RTP packets to the ROHC RTP compressor, together with artificially generated feedback messages that are synchronized with the packet sequence. All packets in these sequences are built on the same base structure, with most field values being constant. The performance test cases define different change patterns for three specific fields: the IPv4 IP ID, the RTP SN, and the RTP TS.

B.1.2 Input Sequence for Uncompressed Headers

The structure of the IPv4/UDP/RTP header and IPv6/UDP/RTP header is outlined in clause B.3, along with tables of the values to be used for each field. Fields with values marked ANY can have any value; these are the addressing fields that uniquely identify the flow of packets being compressed and their respective value does not otherwise affect the expected compression ratio as they are either sent in uncompressed form or completely omitted in compressed packets. The checksum values are dependent on the entire content of the packet and are calculated according to their respective protocol specifications, IETF RFC 768 [9], (B.3 Table UDP header fields) and IETF RFC 791 [10], (B.3 Table IPv4 header fields), which are referred to in the tables. For the input sequence, the UDP checksum with IPv4 shall always be enabled and thus have a non-zero value, i.e. the two octets of the UDP checksum are always included as part of the calculation of the compressed header size for both IPv4 and IPv6. Each test sequence defines specific values to create varying change patterns for the IPv4 IP ID, the RTP SN, and the RTP TS. A dummy payload of an arbitrary non-zero value shall be appended at the end of the header data, following the RTP header.

The outline of each test case follows the same format with respect to the input sequences and the requirements. Test 1a and 1b are base tests using a well-behaving flow of packets as one of the inputs. All subsequent tests are based on test 1a or test 1b, each with specific test events added to the packet flow of the base tests.

B.1.3 Feedback Format for the Test Cases

The feedback messages used in the test cases, when applicable, are artificially generated and synchronized with the input sequence of uncompressed packets. Feedback messages are generated according to the following format:

```
  0   1   2   3   4   5   6   7
+---+---+---+---+---+---+---+---+
| 1   1   1   1   0 |   Code    |  feedback type octet
+---+---+---+---+---+---+---+---+
|Acktype| Mode  |      SN       |
+---+---+---+---+---+---+---+---+
|              SN               |
+---+---+---+---+---+---+---+---+
| 0   0   0   1 | 0   0   0   1 |
+---+---+---+---+---+---+---+---+
|              CRC              |
+---+---+---+---+---+---+---+---+
```

Where:

- Code is set to 0x4 (indicates that feedback data above the type octet is 4 octets)
- Acktype is set to 0x0 (means ACK)
- Mode is set as defined by the test case
- SN is set as defined by the test case
- CRC is the 8-bit CRC computed over the entire feedback payload including any CID fields but excluding the packet type, the 'Size' field and the 'Code' octet, using the polynomial defined in IETF RFC 3095 [5].

NOTE: If the compressor uses the CID field in the compressed packet, the CID field should be included in the feedback packet and the Code and CID should be set as defined in IETF RFC 3095 [5].

B.1.4 Feedback Generation for Test Cases (R-Mode Only)

This clause defines a mechanism by which the test equipment shall dynamically generate feedback messages for each test case, once a transition to R-mode has been initiated and for the entire R-mode operation thereafter. Test cases may define additional feedback messages as input to the compressor.

The test equipment shall generate a feedback message when the ROHC packet type octet of the received compressed header matches any of the values as described in Table B.1. The compressed header type can be identified by inspecting the packet type octet of the compressed header, i.e. the first octet of the ROHC base header as defined in IETF RFC 3095 [5].

Table B.1: Bitmasks for Feedback Generation

Compressed Header Type (binary mask) (Note)	Outcome
01xxxxxx	Send feedback
110xxxxx	Send feedback
1111110x	Send feedback
Other values	No feedback
NOTE: The symbol 'x' means 'any value'.	

The feedback message shall be of the format as described in clause B.1.3 using:

- Mode is set to 0x3 (means R-mode)
- SN is set to the RTP SN corresponding to the received compressed header

The test equipment shall index the input sequence of uncompressed headers using the RTP Sequence Number, and it shall associate the correct RTP SN to each compressed header that it receives back from the compressor. The test equipment can derive the RTP SN by counting the number of received compressed headers.

NOTE: The purpose of this mechanism is only to provide feedback to the compressor when operating in R-mode; it is not meant to make further verifications of any specific ROHC functionality and applies only to the test cases defined in this annex.

B.1.5 Calculation of Compressed Header Size

The following fields shall be excluded from the calculation of the size of the compressed header in evaluation of compression performance:

- ROHC CID/add-CID octet(s)
- ROHC padding octets
- ROHC segmentation octets
- ROHC feedback octets, either piggybacked on the behalf of an associated decompressor or as feedback packets interspersed within the flow of compressed packets

B.2 Test Outline - RoHC RTP Profile 0x0001

B.2.1 Test 1a - Base Test of ROHC RTP O-Mode Compressor

B.2.1.1 Test Purpose

The purpose of the base test case is to verify that the compressor properly implements compression for a well-behaved IP/UDP/RTP packet flow, i.e. that it makes use of efficient compressed packet formats available to ROHC RTP [5] when operating in O-mode.

B.2.1.2 Sequence Details

A sequence consisting of 70 packets in total is used where all header fields are set according to the basic test packet structure, as described in clause B.3, with addition of the following:

1) The RTP Sequence Number is a linearly increasing counter with a packet-to-packet delta of 1, set to 0x0000 for the first packet and thus ending with 0x0045 (69) in the last packet of the sequence.

2) The RTP Time Stamp is a linearly increasing counter with a packet-to-packet delta of 160, set to 0x00000000 for the first packet and thus ending with 0x00002B20 (11040) in the last packet of the sequence.

3) The IP Identification is set to the same value as the RTP Sequence Number; this means that for IPv4 the IP-ID behaviour is not random, thus value(RND)=0 defined in IETF RFC 3095 [5] for both IPv4 and IPv6.

Between the 6th and 7th packets (SN=5 and SN=6) of the sequence, a ROHC feedback packet of feedback type 2 is to be given to the ROHC compressor to trigger an immediate transition to O-mode operation. The format of that packet is as follows:

```
  0   1   2   3   4   5   6   7
+---+---+---+---+---+---+---+---+
| 1   1   1   1   0 |   Code    |  feedback type octet
+---+---+---+---+---+---+---+---+
|Acktype| Mode  |      SN       |
+---+---+---+---+---+---+---+---+
|              SN               |
+---+---+---+---+---+---+---+---+
| 0   0   0   1 | 0   0   0   1 |
+---+---+---+---+---+---+---+---+
|              CRC              |
+---+---+---+---+---+---+---+---+
```

Where:

- Code is set to 0x4 (indicates that feedback data above the type octet is 4 octets).
- Acktype is set to 0x0 (means ACK).
- Mode is set to 0x2 (means O-mode).
- SN is set to 0x000.
- CRC is the 8-bit CRC computed over the entire feedback payload including any CID fields but excluding the packet type, the 'Size' field and the 'Code' octet, using the polynomial defined in IETF RFC 3095 [5].

NOTE: If compressor uses CID field in compressed packet, the CID field should be included in the feedback packet and the Code and CID should be set as defined in IETF RFC 3095 [5].

B.2.1.3 Test Requirement

Table B.2: Test Requirement for Test 1a

	Average Compressed Header Size, IPv4	Average Compressed Header Size, IPv6
SN < N + 1	44 octets	68 octets
SN > N	5 octets	5 octets

N is smaller than 8.

The sequence of expected compressed headers can be illustrated as follows (informative):

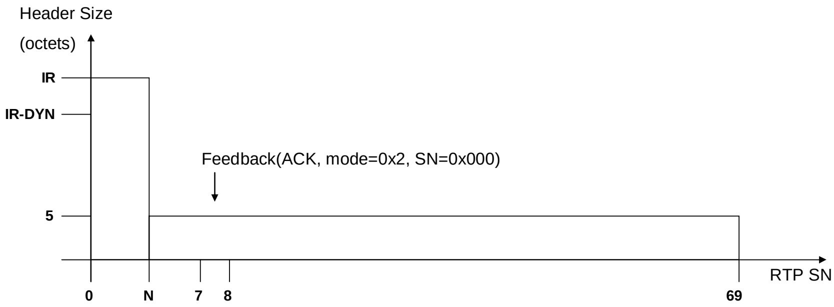

Figure B.1: Expected Outcome for Test 1a

B.2.2 Test 1b - Base Test of ROHC RTP R-Mode Compressor

B.2.2.1 Test Purpose

The purpose of the base test case is to verify that the compressor properly implements compression for a well-behaved IP/UDP/RTP packet flow, i.e. that it makes use of efficient compressed packet formats available to ROHC RTP [5] when operating in R-mode.

B.2.2.2 Sequence Details

A sequence consisting of 70 packets in total is used where all header fields are set according to the basic test packet structure, as described in clause B.3, with the addition of the following:

1) The RTP Sequence Number is a linearly increasing counter with a packet-to-packet delta of 1, set to 0x0000 for the first packet and thus ending with 0x0045 (69) in the last packet of the sequence.

2) The RTP Time Stamp is a linearly increasing counter with a packet-to-packet delta of 160, set to 0x00000000 for the first packet and thus ending with 0x00002B20 (11040) in the last packet of the sequence.

3) The IP Identification is set to the same value as the RTP Sequence Number; this means that for IPv4 the IP-ID behaviour is not random, thus value(RND)=0 defined in IETF RFC 3095 [5] for both IPv4 and IPv6.

Between the 6th and 7th (SN=5 and SN=6) packet of the sequence, a ROHC feedback packet of feedback type 2 is to be given to the ROHC compressor to initiate transition to R-mode operation. The format of that packet is as follows:

```
  0   1   2   3   4   5   6   7
+---+---+---+---+---+---+---+---+
| 1   1   1   1   0 |   Code    |  feedback type octet
+---+---+---+---+---+---+---+---+
|Acktype| Mode  |      SN       |
+---+---+---+---+---+---+---+---+
|              SN               |
+---+---+---+---+---+---+---+---+
| 0   0   0   1 | 0   0   0   1 |
+---+---+---+---+---+---+---+---+
|              CRC              |
+---+---+---+---+---+---+---+---+
```

Where:

- Code is set to 0x4 (indicates that feedback data above the type octet is 4 octets).
- Acktype is set to 0x0 (means ACK).

- Mode is set to 0x3 (means R-mode).
- SN is set to 0x000.
- CRC is the 8-bit CRC computed over the entire feedback payload including any CID fields but excluding the packet type, the 'Size' field and the 'Code' octet, using the polynomial defined in IETF RFC 3095 [5].

B.2.2.3 Test Requirement

Table B.3: Test Requirement for Test 1b

	Average Compressed Header Size, IPv4	Average Compressed Header Size, IPv6
SN < N + 1	44 octets	68 octets
N < SN < 8	5 octets	5 octets
7 < SN < x+1	9 octets	9 octets
SN > x+1	5 octets	5 octets

N shall be smaller than 8. The value of x is the RTP SN for which the test equipment generates the first feedback message that corresponds to a packet with SN > 7, as described in clause B.1.4.

The sequence of expected compressed headers can be illustrated as follows (informative):

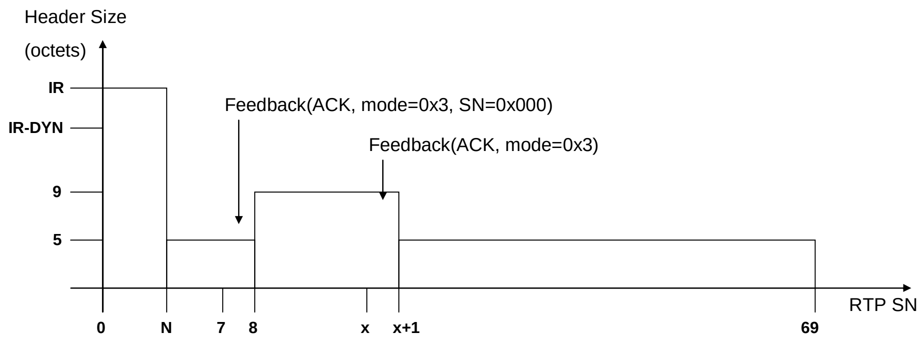

Figure B.2: Expected Outcome for Test 1b

B.2.3 Void

B.2.4 Void

B.2.5 Test 3a - Re-Establishment of TS Function after DTX in O-Mode

B.2.5.1 Test Purpose

The purpose of the TS re-establishment test case is to verify that the compressor re-establishes the proper TS value after a DTX period, i.e. that it uses efficient header formats available to ROHC RTP [5] when operating in O-mode.

B.2.5.2 Sequence Details

The test sequence is the same as in clause B.2.1, with the following exceptions:

1) The RTP Time Stamp is a linearly increasing counter with a packet-to-packet delta of 160, set to 0x00000000 for the first packet.

2) For packet with SN = 20, TS is increased to represent a 32 (0,64 seconds) packet skip (i.e. TS is increased by 32 x 160) and is thus set to (20 + 32) x 160 = 8 320 (0x00002080). Then TS continues to grow as stated in 1, above.

3) For packet with SN = 30, TS is increased to represent a 128 (2,56 seconds) packet skip (i.e. TS is increased by 128 x 160) and is thus set to (30 + 32 + 128) x 160 = 30 400 (0x000076C0). Then TS continues to grow as stated in 1, above.

4) For packet with SN = 40, TS is increased to represent a 2 048 (40,96 seconds) packet skip (i.e. TS is increased by 2 048 x 160) and is thus set to (40 + 32 + 128 + 2 048) x 160 = 359 680 (0x00057D00). Then TS continues to grow as stated in 1, above.

5) TS thus ends at 364 320 (0x00058F20) in the last packet of the sequence with RTP sequence number 69.

B.2.5.3 Test Requirement

Table B.4: Test Requirement for Test 3a

	Average Compressed Header Size, IPv4	Average Compressed Header Size, IPv6
SN < 20	See Test 1a	
19 < SN < 20 + N **29 < SN < 30 + N**	10 octets	10 octets
39 < SN < 40 + N	10 octets	10 octets
Other SN values	5 octets	5 octets

The sequence of expected compressed headers can be illustrated as follows (informative):

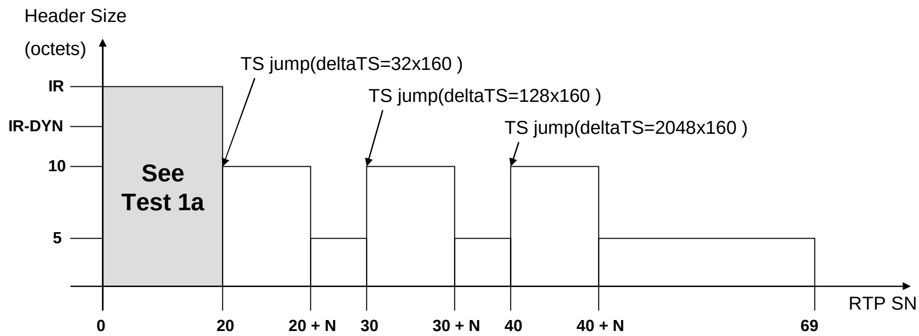

Figure B.3: Expected Outcome for Test 3a

B.2.6 Test 3b - Re-Establishment of TS Function after DTX in R-Mode

B.2.6.1 Test Purpose

The purpose of the TS re-establish test case is to verify that the compressor re-establishes the proper TS value after a DTX period, i.e. that it uses the efficient header formats available to ROHC RTP [5] when operating in R-mode.

B.2.6.2 Sequence Details

The test sequence is the same as in clause B.2.2, with the following exceptions:

1) The RTP Time Stamp is a linearly increasing counter with a packet-to-packet delta of 160, set to 0x00000000 for the first packet.

2) For packet with SN = 20, TS is increased to represent a 32 (0,64 seconds) packet skip (i.e. TS is increased by 32 x 160) and is thus set to (20 + 32) x 160 = 8 320 (0x00002080). Then TS continues to grow as stated in 1, above.

3) For packet with SN = 30, TS is increased to represent a 128 (2,56 seconds) packet skip (i.e. TS is increased by 128 x 160) and is thus set to (30 + 32 + 128) x 160 = 30 400 (0x000076C0). Then TS continues to grow as stated in 1, above.

4) For packet with SN = 40, TS is increased to represent a 2 048 (40,96 seconds) packet skip (i.e. TS is increased by 2 048 x 160) and is thus set to (40 + 32 + 128 + 2 048) x 160 = 359 680 (0x00057D00). Then TS continues to grow as stated in 1, above.

5) TS thus ends at 393 120 (0x0005FFA0) in the last packet of the sequence with RTP sequence number 249.

B.2.6.3 Test Requirement

Table B.5: Test Requirement for Test 3b

	Average Compressed Header Size, IPv4	Average Compressed Header Size, IPv6
SN < 20	See Test 1b	
19 < SN < 20 + x **29 < SN < 30 + x**	10 octets	10 octets
39 < SN < 40 + x	10 octets	10 octets
Other SN values	5 octets	5 octets

The sequence of expected compressed headers can be illustrated as follows (informative):

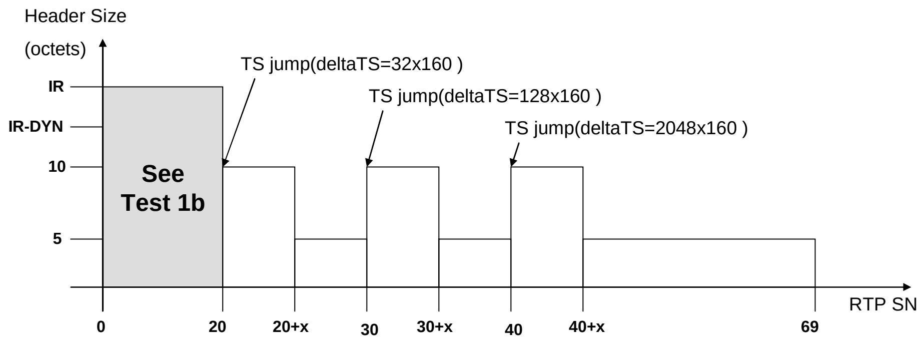

Figure B.4: Expected Outcome for Test 3b

B.2.7 Test 4a - Compressor Response to Single Lost Packets in O-Mode

B.2.7.1 Test Purpose

The purpose of this test is to verify that the compressor does not panic just because there is a single missing packet, i.e. the compressed packet size should not increase due to such events.

B.2.7.2 Sequence Details

The test sequence is the same as in clause B.2.1, with the following exception:

- Packets with SN 20, 30, and 40 are removed from the sequence.

B.2.7.3 Test Requirement

Maximal compressed header overhead for the test are the same as in clause B.2.1.

B.2.8 Test 4b - Compressor Response to Single Lost Packets in R-Mode

B.2.8.1 Test Purpose

The purpose of this test is to verify the compressor properly handles a single missing packet, i.e. the compressed packet size should not increase due to such events.

B.2.8.2 Sequence Details

The test sequence is the same as in clause B.2.2, with the following exception:

- Packets with SN 20, 30, and 40 are removed from the sequence.

B.2.8.3 Test Requirement

Maximal compressed header overhead for the test are the same as in clause B.2.2.

B.2.9 Void

B.2.10 Void

B.2.11 Test 6a - TS Function During DTX with Varying Delta in O-Mode

B.2.11.1 Test Purpose

The purpose of this test case is to verify that the compressor properly handles variations in the function between the TS value and the SN during and after a DTX period, during which SID packets are sent periodically, i.e. that it uses efficient header formats available to ROHC RTP [5] when operating in O-mode.

B.2.11.2 Sequence Details

The test sequence is the same as in clause B.2.1, with the following exceptions:

1) The RTP Time Stamp is a linearly increasing counter with a packet-to-packet delta of 160, set to 0x00000000 for the first packet.

2) For packets SN = 20, 21 and 22, TS is increased to represent a 7 (0,14 seconds) packet skip (i.e. TS is increased by 7 x 160) and is thus set to (20 + 7) x 160 = 4 320 (0x000010E0), (21 + 7 + 7) x 160 = 5 600 (0x000015E0) and (22 + 7 + 7 + 7) x 160 = 6 880 (0x00001AE0), respectively.

3) For packets SN = 30, 31, 32, 33 and 34, TS is increased to represent a 7 (0,14 seconds) packet skip (i.e. TS is increased by 7 x 160) and is thus set to 9 280 (0x00002440), 10 560 (0x00002940), 11 840 (0x00002E40), 13 120 (0x00003340) and 14 400 (0x00003840), respectively.

4) For packets SN = 40, 41, 42, 43, 44, 45, and 46, TS is increased to represent a 7 (0,14 seconds) packet skip (i.e. TS is increased by 7x160) and is thus set to 16 480 (0x00004060), 17 760 (0x00004560), 19 040 (0x00004A60), 20 320 (0x00004F60), 21 600 (0x00005460), 22 880 (0x00005960) and 24 160 (0x00005E60), respectively.

5) TS thus ends at 27 840 (0x00006CC0) in the last packet of the sequence with RTP sequence number 69.

B.2.11.3 Test Requirement

Table B.6: Test Requirement for Test 6a

	Average Compressed Header Size, IPv4	**Average Compressed Header Size, IPv6**
SN < 20	See Test 1a	
19 < SN < 23 + N **29 < SN < 35 + N** **39 < SN < 47 + N**	15 octets	15 octets
Other SN values	5 octets	5 octets

The sequence of expected compressed headers can be illustrated as follows (informative):

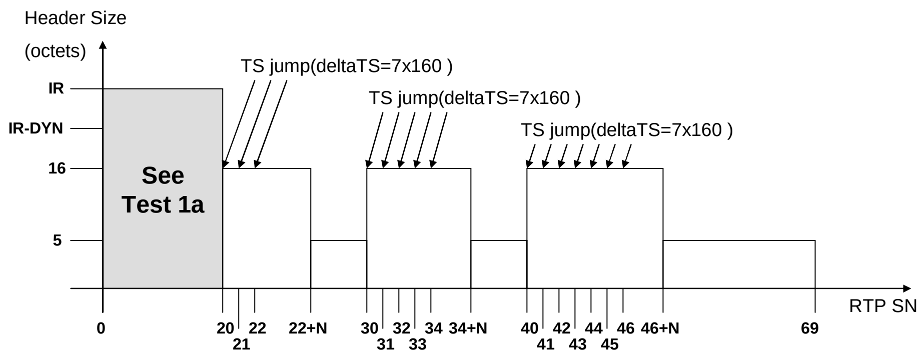

Figure B.5: Expected Outcome for Test 6a

B.2.12 Test 6b - TS Function During DTX with Varying Delta in R-Mode

B.2.12.1 Test Purpose

The purpose of this test case is to verify how efficiently the compressor handles variations in the function between the TS value and the SN during and after a DTX period, during which SID packets are sent periodically; i.e. that it uses the efficient header formats available to ROHC RTP [5] when operating in R-mode.

B.2.12.2 Sequence Details

The test sequence is the same as in clause B.2.2, with the following exceptions:

1) The RTP Time Stamp is a linearly increasing counter with a packet-to-packet delta of 160, set to 0x00000000 for the first packet.

2) For packets SN = 20, 21 and 22, TS is increased to represent a 7 (0,14 seconds) packet skip (i.e. TS is increased by 7 x 160) and is thus set to (20 + 7) x 160 = 4 320 (0x000010E0), (21 + 7 + 7) x 160 = 5 600 (0x000015E0) and (22 + 7 + 7 + 7) x 160 = 6 880 (0x00001AE0), respectively.

3) For packets SN = 30, 31, 32, 33 and 34, TS is increased to represent a 7 (0,14 seconds) packet skip (i.e. TS is increased by 7x160) and is thus set to 9 280 (0x00002440), 10 560 (0x00002940), 11 840 (0x00002E40), 13 120 (0x00003340) and 14 400 (0x00003840), respectively.

4) For packets SN = 40, 41, 42, 43, 44, 45, and 46, TS is increased to represent a 7 (0,14 seconds) packet skip (i.e. TS is increased by 7 x 160) and is thus set to 16 480 (0x00004060), 17 760 (0x00004560), 19 040 (0x00004A60), 20 320 (0x00004F60), 21 600 (0x00005460), 22 880 (0x00005960) and 24 160 (0x00005E60), respectively.

5) TS thus ends at 27 840 (0x00006CC0) in the last packet of the sequence with RTP sequence number 69.

B.2.12.3 Test Requirement

Table B.7: Test Requirement for Test 6b

	Average Compressed Header Size, IPv4	**Average Compressed Header Size, IPv6**
SN < 20	See Test 1b	
19 < SN < 23 + x **29 < SN < 35 + x** **39 < SN < 47 + x**	15 octets	15 octets
Other SN values	5 octets	5 octets

The sequence of expected compressed headers can be illustrated as follows (informative):

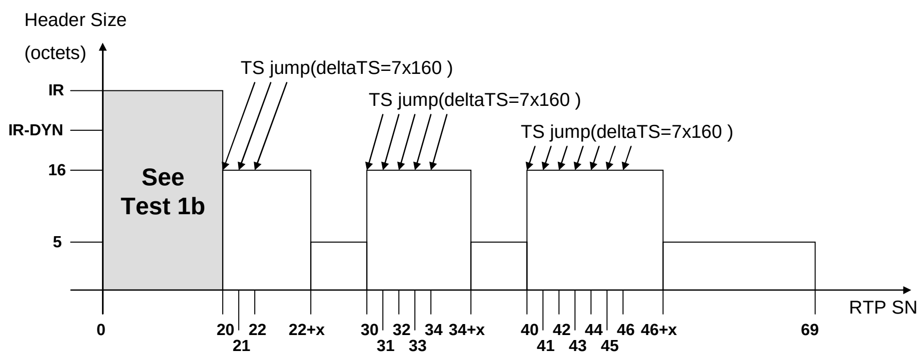

Figure B.6: Expected Outcome for Test 6b

B.2.13 Test 7a - SRNS Relocation in O-Mode

B.2.13.0 General

SRNS Relocation for RoHC is Not supported in this version of GMR-1 3G.

B.2.13.1 Test Purpose

Void.

B.2.13.2 Sequence Details

Void.

B.2.13.3 Test Requirement

Void.

B.3 Test Packet Structures

IPv6/UDP/RTP

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|Version| Traffic Class |           Flow Label                  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|         Payload Length        |  Next Header  |   Hop Limit   |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                                                               |
+                                                               +
|                                                               |
+                         Source Address                        +
|                                                               |
+                                                               +
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                                                               |
+                                                               +
|                                                               |
+                      Destination Address                      +
|                                                               |
+                                                               +
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          Source Port          |       Destination Port        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          Length               |       Checksum                |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

```
IPv4/UDP/RTP
                        1                   2                   3
    0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |Version|  IHL  |Type of Service|          Total Length         |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |         Identification        |R|D|F|      Fragment Offset    |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |  Time to Live |    Protocol   |         Header Checksum       |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |                       Source Address                          |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |                    Destination Address                        |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |                    Options                    |    Padding    |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |          Source Port          |       Destination Port        |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |          Length               |       Checksum                |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |V=2|P|X|  CC   |M|     PT      |       sequence number         |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |                           timestamp                           |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |           synchronization source (SSRC) identifier            |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+


IPv6 header fields
      +----------------------+-------------+----------------+
      | Field                | Size (bits) |     Value      |
      +----------------------+-------------+----------------+
      | Version              |      4      |      0x6       |
      | Traffic Class        |      8      |      0x00      |
      | Flow Label           |     20      |      0x00000   |
      | Payload Length       |     16      |      0x0034    |
      | Next Header          |      8      |      0x11      |
      | Hop Limit            |      8      | Test dependent |
      | Source Address       |    128      |      ANY       |
      | Destination Address  |    128      |      ANY       |
      +----------------------+-------------+----------------+

IPv4 header fields
      +----------------------+-------------+----------------+
      | Field                | Size (bits) |     Value      |
      +----------------------+-------------+----------------+
      | Version              |      4      |      0x4       |
      | Header Length (IHL)  |      4      |      0x5       |
      | Type Of Service      |      8      |      0x00      |
      | Packet Length        |     16      |      0x0048    |
      | Identification       |     16      | Test dependent |
      | Reserved flag (R)    |      1      |      0x0       |
      | Don't Fragment (D)   |      1      |      0x1       |
      | More Fragments (F)   |      1      |      0x0       |
      | Fragment Offset      |     13      |      0x0000    |
      | Time To Live         |      8      | Test dependent |
      | Protocol             |      8      |      0x11      |
      | Header Checksum      |     16      |  See RFC 791   |
      | Source Address       |     32      |      ANY       |
      | Destination Address  |     32      |      ANY       |
      +----------------------+-------------+----------------+
```

UDP header fields

Field	Size (bits)	Value
Source Port	16	ANY
Destination Port	16	ANY
Length	16	0x0034
Checksum	16	See RFC 768

RTP header fields

Field	Size (bits)	Value
Version (V)	2	0x2
Padding (P)	1	0x0
Extension (X)	1	0x0
CSRC Counter (CC)	4	0x0
Marker (M)	1	0x0
Payload Type (PT)	7	0x60
Sequence Number	16	Test dependent
Timestamp	32	Test dependent
SSRC	32	ANY

Annex C (informative): Reference Model for Generating ROHC Performance Requirements

C.1 Introduction

ROHC compressor and decompressor may use a set of parameters in order to operate (e.g. L confidence parameter, dynamic FOtimer and static IRtimer, K1 out of N1, K2 out of N2…) that may be optimized for a given application (e.g. Voice over IP, Videotelephony over IP, Interactive Gaming over IP).

NOTE: L confidence parameter allows setting the number of times an IR or IR-Dyn packet is transmitted and FOtimer and IRtimer are used in order to determine when a transition to a lower compressor state is necessary: The dynamic timer FOtimer triggers SO state to FO state transition and the static IRtimer triggers SO/FO state to IR state transition.

C.2 For Voice-over-IP (VoIP) Optimization

C.2.1 ROHC Parameters Optimizations for VoIP

For the support of VoIP in the network, ROHC compressor and decompressor parameter values (L confidence parameter, dynamic FOtimer and static IRtimer, K1 out of N1, K2 out of N2) may be used. Values used in ROHC reference model are given in clause C.2.2 for O-mode:

- The initialization phase duration
- The reaction delay to decompression failure
- The header compression ratio (compressed header size/uncompressed header size)
- The error rates (in UDP and PDCP layers)
- The amount of transferred data (including ROHC compressed packets and feedbacks)
- The resource usage (transport block occupancy in the RLC layer)

C.2.2 Parameter Setting for ROHC Reference Model for VoIP

The following parameters setting is applied in the reference model for ROHC performance tests of VoIP application:

Table C.1: ROHC Parameters Setting for VoIP

ROHC Parameter	O-Mode
L	2
FOtimer	0,12 seconds
IRtimer	0,12 seconds
K1 / N1	2 / 20
K2 / N2	1 / 1

C.2.3 Setting the Parameter Value N in Test Cases for VoIP

In test cases for VoIP to evaluate ROHC Compression performance for both O- and R-mode, the parameter N (defined in annex A) is set to 4.

Annex D (normative): Data Compression

D.1 Overview

V.44 data compression used in PDCP has two main parts: control function and data compression function.

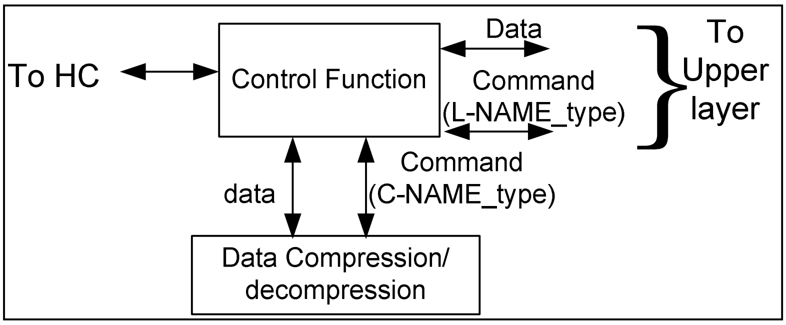

Figure D.1: V.44 Data Compression Functional Blocks

Control function in V.44 data compression communicates with the external entities, for example error control function, for command, and data transfer, as well. Data compression function encodes or decodes data.

D.2 Control Function

Refer to Recommendation ITU-T V.44 [8].

D.3 Data Compression Function

D.3.0 General

The data compression function consists of an encoder and a decoder. The encoder transfers the compressed data to decoder at the other end of the connection. Since most communication systems support bidirectional data transfer, each communicating party will typically have both an encoder and a decoder. Furthermore, in a case where one of the parties might have more resources, for example RNC can have more resources than MES, the compression might not be symmetric, i.e. the RNC to MES direction might have a better compression than the MES to RNC direction.

D.3.1 Encoding

Refer to Recommendation ITU-T V.44 [8] for encoding operation.

If the encoder cannot compress the data, it shall forward the data unchanged to the Header Compression entity. The encoder shall compare the size of the encoder output to the size of the input to see whether the compression was successful or not. If the output size of the encoder is larger than the input size, then the compression is considered to have failed and the data shall be forwarded unchanged.

D.3.2 Decoding

Refer to Recommendation ITU-T V.44 [8] for decoding operation.

For uncompressed binary codes, the decoder does not do any decoding; it simply forwards it to upper layer through the control function.

D.4 Packet Technique and Multi-Packet Technique

V.44 can operate in two different techniques: Packet Technique and Multi-Packet Technique. In packet mode, each packet is processed independently.

Packet Technique is also called stateless mode. Packet Technique and stateless mode will mean the same thing in the present document.

In Multi-Packet Technique, several packets or portion of packets are processed as a continuation. Multi-Packet Technique and stateful mode will mean the same thing in the present document.

The main differences between Packet Technique and Multi-Packet Technique are summarized in Table D.1.

Table D.1: Packet Mode and Multi-Packet Mode Differences

Packet Mode	Multi-Packet Mode
Does not require guaranteed delivery	Requires guaranteed delivery
Encoder and decoder dictionaries are initialized every packet	Encoder and decoder dictionaries are initialized at the beginning or if control function requests re-initialization
Encoder and decoder histories are not required	Encoder and decoder require histories

Multi-packet mode provides better compression with the expense of higher complexity compared to Packet mode.

Since Packet/Multi-Packet Technique does not support transparent mode if the encoder fails, it shall send the original (uncompressed) data with the first bit of the PDU type set to 0 to indicate that the data is not compressed and the dictionary needs to be reset, see clause 8.3.1.

For the GMR-1 3G LLA Header Suppression SE-VoIP case, the packet will not go through the data compression function. When packets arrive at PDCP layer, there shall be screening that checks the type of packet. For packets classified as SE-VoIP, there shall be no data compression.

D.5 Transfer

Refer to Recommendation ITU-T V.44 [8].

D.6 Communication between Control Function and Data Compression Function

Refer to Recommendation ITU-T V.44 [8].

D.7 Communication between Peer Data Compression Function

Refer to Recommendation ITU-T V.44 [8].

D.8 Configuration Parameters

The encoder and decoder shall agree on a set of parameters to be able to function properly. These parameters can be negotiated. However, in order to simplify the procedures, a set of default parameters should be configured on encoder and decoder.

Since the RNC has more resources than the MES, the configurations for the RNC-to-MES direction and the configuration for the MES-to-RNC direction are not the same. Tables D.2 and D.3 show the default configuration RNC-to-MES and the default configuration for MES-to-RNC respectively.

Table D.2: Data Compression Parameters: RNC-to-MES

Parameters	Default Values	Description
N_2 ($N_{2T} = N_{2R}$)	1 525	Total number of codewords (node-tree) or string of consecutive characters, greater than the maximum length packet expected, in bytes
N_7 ($N_{7T} = N_{7R}$)	255	Maximum string length
N_8 ($N_{8T} = N_{8R}$) (Note 1)	4 x N_2	Length of history, aggregation of N_8 and N_2 if possible to achieve better compression, in bytes
N_1	Derived from N_2	Maximum codeword size in bits
N_4	256	Number of characters in alphabet
N_5	4	Number of control code and first available codewords
C_1	N_5	Initial values
C_2	6	Initial current codewords size bits
C_3 (Note 2)	64	Initial threshold for changing codewords size in bits, 2^{C_2}
C_4	0	Initial "current history position"
C_5	7	Initial current ordinal size in bits
NOTE 1: For Packet Technique N_8 is not applicable. NOTE 2: Only for encoder.		

Table D.3: Data Compression Parameters: MES-to-RNC

Parameters	Default Values	Description
N_2 ($N_{2T} = N_{2R}$)	1 525	Total number of codewords (node-tree), greater than the maximum length packet expected, in bytes
N_7 ($N_{7T} = N_{7R}$)	255	Maximum string length
N_8 ($N_{8T} = N_{8R}$) (Note 1)	2 x N_2	Length of history, aggregation of N_8 and N_2 if possible to achieve better compression, in bytes
N_1	Derived from N_2	Maximum codeword size in bits
N_4	256	Number of characters in alphabet
N_5	4	Number of control code and first available codewords
C_1	N_5	Initial values
C_2	6	Initial current codewords size bits
C_3 (Note 1)	64	Initial threshold for changing codewords size in bits, 2^{C_2}
C_4	0	Initial "current history position"
C_5	7	Initial current ordinal size in bits
NOTE 1: For Packet Technique N8 is not applicable. NOTE 2: Only for encoder.		

For each pair of MES-RNC compression/decompression, there are only two types of context created for each RAB: one for TCP and one for UDP. This means that all TCP flows shall go to the TCP context, and all UDP flows shall go to the UDP context. For example, if a MES open Web browser session in which a RAB will be created and there are several TCP flows opened for this RAB, only one TCP context shall be created to cater for all the TCP flows for that RAB. Similarly, one UDP context shall be created for all the UDP flows.

Annex E (informative): Bibliography

ETSI TS 125 303: "Universal Mobile Telecommunications System (UMTS); Interlayer procedures in Connected Mode (3GPP TS 25.303 Release 7)".

IETF RFC 3096: "Requirements for robust IP/UDP/RTP header compression".

IETF RFC 4362: "RObust Header Compression (ROHC): A Link-Layer Assisted Profile for IP/UDP/RTP".

IETF RFC 3408: "Zero-byte Support for Bidirectional Reliable Mode (R-mode) in Extended Link-Layer Assisted RObust Header Compression (ROHC) Profile".

ETSI TS 101 376-4-13: "GEO-Mobile Radio Interface Specifications (Release 3); Third Generation Satellite Packet Radio Service; Part 4: Radio interface protocol specifications;
Sub-part 13: Radio Resource Control (RRC) protocol; Iu Mode; GMR-1 3G 44.118".

ETSI TS 101 376-4-14: "GEO-Mobile Radio Interface Specifications (Release 3); Third Generation Satellite Packet Radio Service; Part 4: Radio interface protocol specifications;
Sub-part 14: Mobile Earth Station (MES) - Base Station System (BSS) interface; Radio Link Control/Medium Access Control (RLC/MAC) protocol; Iu Mode; GMR-1 3G 44.160".

ETSI TS 101 376-4-12: "GEO-Mobile Radio Interface Specifications (Release 3); Third Generation Satellite Packet Radio Service; Part 4: Radio interface protocol specifications;
Sub-part 12: Mobile Earth Station (MES) - Base Station System (BSS) interface; Radio Link Control/Medium Access Control (RLC/MAC) protocol; GMR-1 3G 44.060".

ETSI TS 125 323: "Universal Mobile Telecommunications System (UMTS); Packet Data Convergence Protocol (PDCP) specification (3GPP TS 25.323 Release 7)".

History

Document history		
V3.3.1	December 2012	Publication
V3.4.1	March 2017	Publication