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TECHNICAL SPECIFICATION

Satellite Earth Stations and Systems (SES);
Family SL Satellite Radio Interface (Release 1);
Part 3: Control Plane and User Plane Specifications;
Sub-part 9: Initiation and Operation of User Plane

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Foreword

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

The present document is part 3, sub-part 9 of a multi-part deliverable. Full details of the entire series can be found in ETSI TS 102 744-1-1 [10].

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).

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Introduction

This multi-part deliverable (Release 1) defines a satellite radio interface that provides UMTS services to users of mobile terminals via geostationary (GEO) satellites in the frequency range 1 518,000 MHz to 1 559,000 MHz (downlink) and 1 626,500 MHz to 1 660,500 MHz and 1 668,000 MHz to 1 675,000 MHz (uplink).

1 Scope

The present document defines the necessary control plane behaviour for instantiation and initiation of the User Plane entities, and also describes the operation of the User Plane entities for the Family SL satellite radio interface between the Radio Network Controller (RNC) and the User Equipment (UE) used in the satellite network.

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 reference document (including any amendments) applies.

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

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

[1] ETSI TS 124 008: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); Mobile radio interface Layer 3 specification; Core network protocols; Stage 3 (3GPP TS 24.008 Release 4)".

[2] ETSI TS 126 103: "Universal Mobile Telecommunications System (UMTS); Speech Codec List for GSM and UMTS (3GPP TS 26.103 Release 4)".

[3] ETSI TS 123 014: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Support of Dual Tone Multi Frequency (DTMF) signalling (3GPP TS 23.014 Release 4)".

[4] ETSI TS 125 415: "Universal Mobile Telecommunications System (UMTS); UTRAN Iu Interface User Plane Protocols (3GPP TS 25.415 Release 4)".

[5] ETSI TS 124 007: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Mobile radio interface signalling layer 3; General Aspects (3GPP TS 24.007 Release 4)".

[6] ETSI TS 125 323: "Universal Mobile Telecommunications System (UMTS); Packet Data Convergence Protocol (PDCP) specification (3GPP TS 25.323 Release 4)".

[7] ETSI TS 127 007: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); AT command set for User Equipment (UE) (3GPP TS 27.007 Release 4)".

[8] ETSI TS 127 001: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS) (3GPP TS 27.001 Release 4)".

[9] ETSI TS 122 002: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN) (3GPP TS 22.002 Release 4)".

[10] ETSI TS 102 744-1-1: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 1: General Specifications; Sub-part 1: Services and Architectures".

[11] ETSI TS 102 744-1-4: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 1: General Specifications; Sub-part 4: Applicable External Specifications, Symbols and Abbreviations".

[12] ETSI TS 102 744-3-2: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 3: Control Plane and User Plane Specifications; Sub-part 2: Bearer Control Layer Operation".

[13] ETSI TS 102 744-3-4: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 3: Control Plane and User Plane Specifications; Sub-part 4: Bearer Connection Layer Operation".

[14] ETSI TS 102 744-3-5: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 3: Control Plane and User Plane Specifications; Sub-part 5: Adaptation Layer Interface".

[15] ETSI TS 102 744-3-6: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 3: Control Plane and User Plane Specifications; Sub-part 6: Adaptation Layer Operation".

[16] ETSI TS 102 744-3-8: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 3: Control Plane and User Plane Specifications; Sub-part 8: NAS Layer and User Plane Operation for MBMS Services".

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 reference document (including any amendments) applies.

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

[i.1] IETF RFC 2507: "IP Header Compression".

[i.2] IETF RFC 3095: "RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed".

[i.3] IETF RFC 2508: "Compressing IP/UDP/RTP Headers for Low-Speed Serial Links".

[i.4] ETSI TR 123 910: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); Circuit Switched Data Bearer Services (3GPP TR 23.910 Release 4)".

3 Symbols and abbreviations

3.1 Symbols

For the purposes of the present document, the symbols given in clause 3 of ETSI TS 102 744-1-4 [11] apply.

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in clause 3 of ETSI TS 102 744-1-4 [11] apply.

4 User Plane Operation

4.0 Overview

The present document specifies the behaviour of the User Plane in both the RNC and UE and covers the following:

- the provision of a circuit switched (CS) 4 kbit/s voice service using a satellite optimized low data rate voice codec;

- the provision of a circuit switched (CS) 64 kbit/s Unrestricted Digital Information (UDI)/Restricted Digital Information (RDI)/3,1 kHz audio service (BS 30); and
- the provision of a packet switched (PS) service employing the Packet Data Convergence Protocol (PDCP).

The present document also covers the elements of the Call Control signalling related to the provision of the above circuit switched services.

Figure 4.1 and Figure 4.2 illustrate the CS and PS User Plane stack architectures.

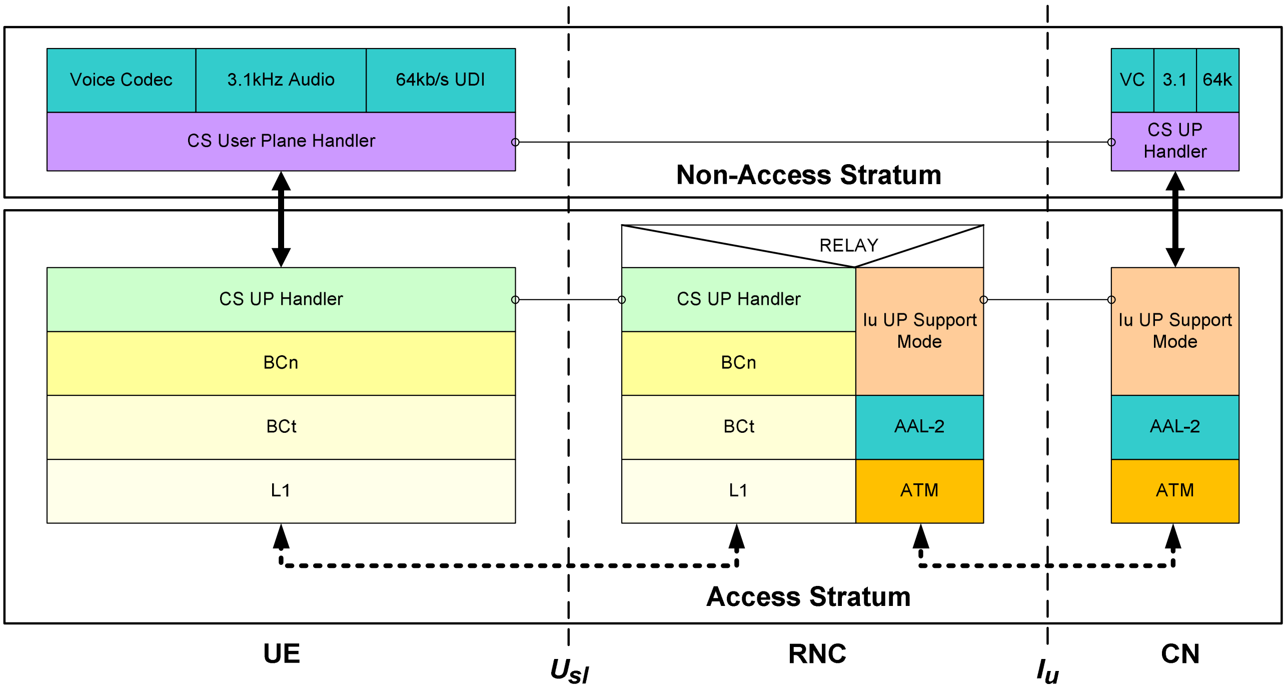

Figure 4.1: Circuit Switched User Plane Protocol Stack Layering showing NAS Layer Entities

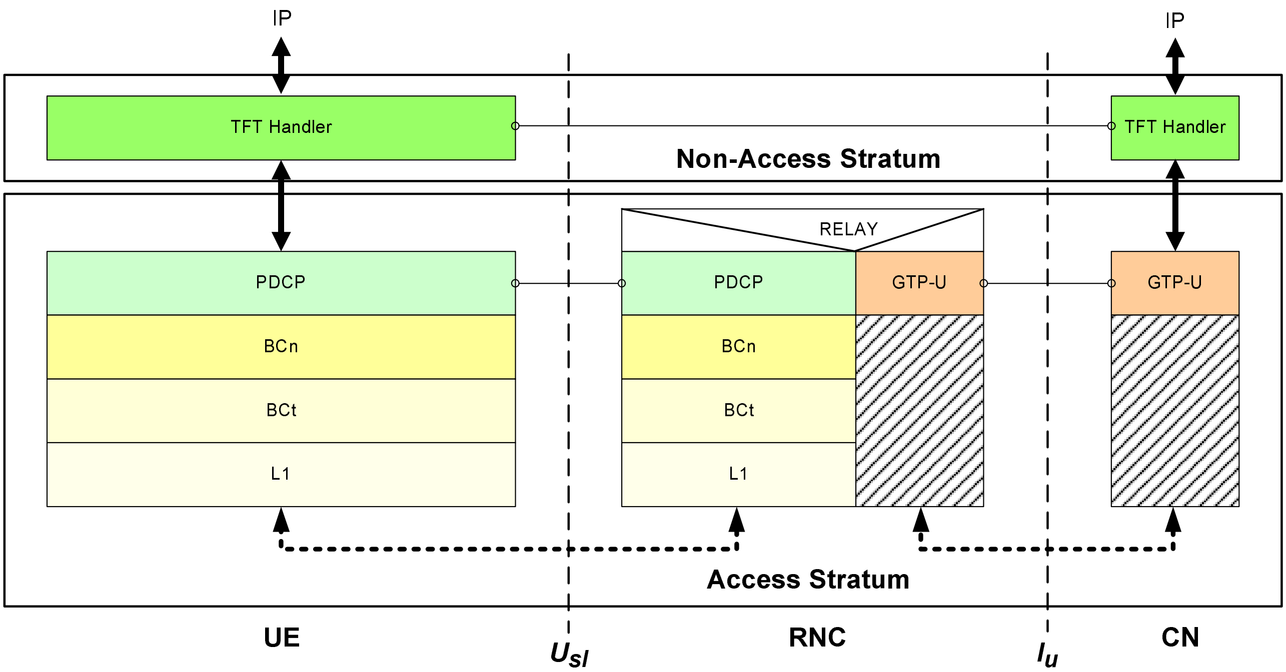

Figure 4.2: Packet Switched User Plane Protocol Stack Layering

4.1 Outline of Circuit Switched User Plane

4.1.0 General

The circuit switched user plane makes use of the CSH (Circuit Switched Handler) to provide services to the Non Access Stratum at the UE or the Relay function at the RNC. CSH is a satellite network specific sublayer and is specified in this and following clauses.

4.1.1 Architecture

Figure 4.3 shows the architecture of the CSH layer at the UE side. Figure 4.4 shows the architecture of the CSH at the RNC side. The CSH layer consists of a CSH Manager and one data handler per CS connection. Two types of data handler are defined.

The Adaptation Layer, a Radio Bearer Control (RBC) entity, configures the CSH sublayer through the CSH-RBC-SAP. The CSH Manager retrieves the information required to select the appropriate data handler and DTX (discontinuous transmission) type from the Call Control layer via the CSH-CC-SAP.

The data handlers supported by the CSH layer are:

- 4 kbit/s voice circuit switched handler (VCSH)
- 64 kbit/s circuit switched handler (BCSH)

The CSH Manager, based on the requests received from RBC and information provided from the Call Control Layer, shall create a data handler.

The 4 kbit/s voice circuit switched data handler is described in clause 5.1 of the present document. The 64 kbit/s circuit switched data handler is described in clause 6.1 of the present document.

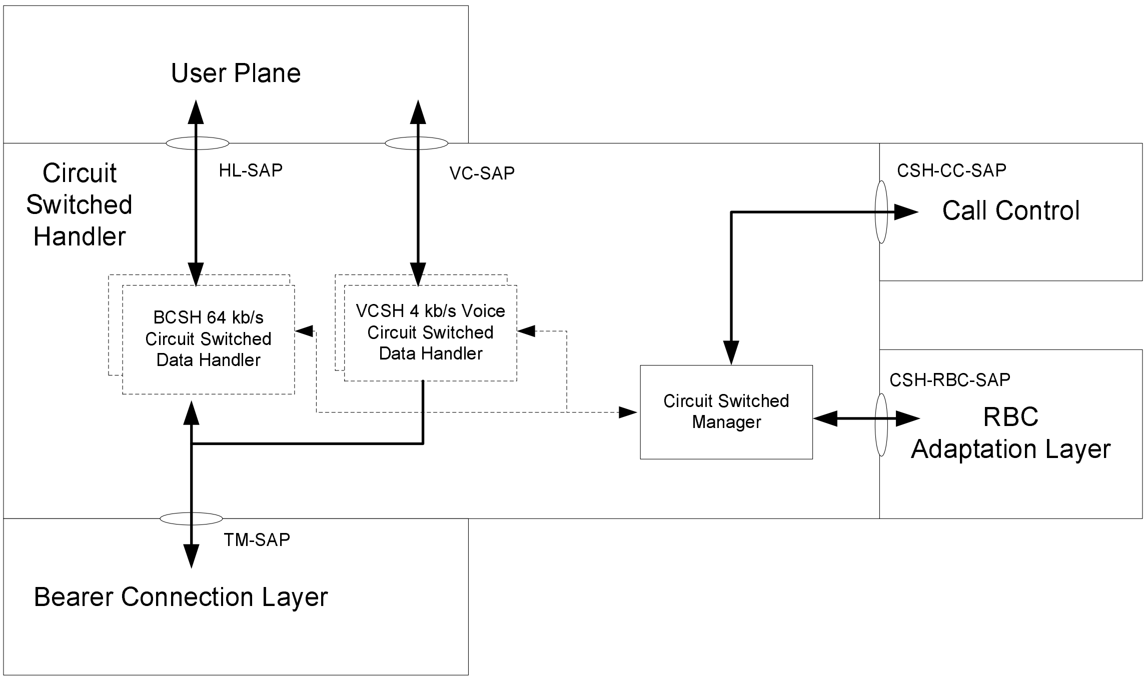

Figure 4.3: Circuit Switched Handler Layer Architecture (UE Side)

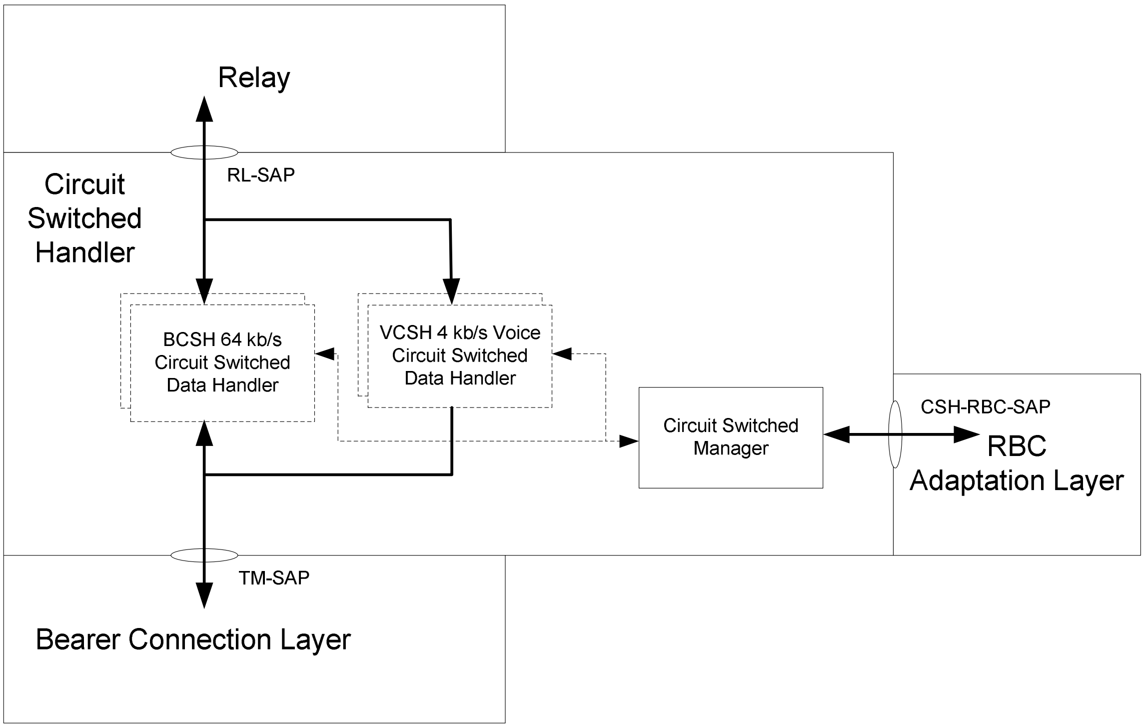

Figure 4.4: Circuit Switched Handler Layer Architecture (RNC Side)

4.1.2 Functions

The CSH Layer provides its services to the Non Access Stratum (NAS) at the UE or the relay at the Radio Network Controller (RNC).

The CSH Layer shall perform the following functions:

- Concatenation and separation of circuit switched data frames (satellite optimized low data rate voice codec frames or 640 bit higher layer frames for 64 kbit/s circuit switched services).
- Discontinuous transmission.
- Transfer of user data. This function is used for conveyance of data between users of CSH services.

CSH uses the services provided by the Bearer Connection sublayer.

4.1.3 Services

4.1.3.1 Services Provided to Upper Layers

The following services are provided by the CSH Layer towards the upper layers:

- Transfer of user data.

4.1.3.2 Services Expected from Bearer Connection Layer

The following services are expected from the Bearer Connection Layer:

- Transparent data transfer service.

4.1.4 Elements for layer-to-layer communication

4.1.4.0 General

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

4.1.4.1 Control Plane Primitives between the CSH layer and adaptation layer

The control primitives between the CSH layer and the adaptation layer are shown in Table 4.1 for the UE side and Table 4.2 for the RNC side.

Table 4.1: Control Plane Primitives between the CSH Layer (CSH Manager) and the adaptation layer RBC entity (UE side)

SAP	Generic Name	Parameter			
		_REQ	_IND	_RESP	_CNF
CSH-RBC-SAP	CSH_RBC_CONFIG	BCnID(Data-SAP), ForwardCSFramesPerPDU, ReturnCSFramesPerPDU, Ret_DTX	Not Defined	Not Defined	BCnID (Data-SAP), CircuitSwitched CallType
CSH-RBC-SAP	CSH_RBC_RELEASE	BCnID (Data-SAP)	Not Defined	Not Defined	Not Defined

Table 4.2: Control Plane Primitives between CSH Layer (CSH Manager) and the adaptation layer RBC entity (RNC side)

SAP	Generic Name	Parameter			
		_REQ	_IND	_RESP	_CNF
CSH-RBC-SAP	CSH_RBC_CONFIG	BCnID (Data-SAP), ForwardCSFramesPerPDU, ReturnCSFramesPerPDU, Fwd_DTX, CircuitSwitchedCallType	Not Defined	Not Defined	Not Defined
CSH-RBC-SAP	CSH_RBC_RELEASE	BCnID (Data-SAP)	Not Defined	Not Defined	Not Defined

Each Primitive is defined as follows:

1) CSH_RBC_CONFIG_REQ

CSH_RBC_CONFIG_REQ is used by RBC to configure a CSH entity and to assign it to the bearer connection associated with that entity.

2) CSH_RBC_CONFIG_CNF

CSH_RBC_CONFIG_CNF is used by the CSH on the UE side to forward to RBC the CircuitSwitchedCallType (retrieved from the CC layer) which RBC shall return with the RBC:EstablishAck to configure the RNC side CSH Data Handler (i.e. select the CSH data handler/DTX Algorithm on the RNC side appropriate for 4 kbit/s voice, UDI, RDI or 3,1 kHz audio services).

3) CSH_RBC_RELEASE_REQ

CSH_RBC_RELEASE_REQ is used by RBC to release a CSH entity.

The following parameters are used in the primitives:

1) BCnID (Data-SAP)

The BCn-SAP (Transparent Mode) used by CSH when communicating with the Bearer Connection (BCn) Layer.

2) ForwardCSFramesPerPDU

The number of satellite optimized low data rate codec frames (80 bits) or higher layer 64 kbit/s frames (640 bits) concatenated into a single CSH Protocol Data Unit (PDU) for transmission via the Bearer Connection Layer in the forward direction.

3) ReturnCSFramesPerPDU

The number of satellite optimized low data rate codec frames (80 bits) or higher layer 64 kbit/s frames (640 bits) concatenated into a single CSH PDU for transmission via the Bearer Connection Layer in the return direction.

NOTE 1: The values of *ForwardCSFramesPerPDU* and *ReturnCSFramesPerPDU* are passed to the UE in the Establish message during the Radio Access Bearer (RAB) setup. They are set independently of each other and apply for the lifetime of the RAB (i.e. the duration of the call) only. ForwardCSFramesPerPDU and ReturnCSFramesPerPDU have a value range of 1 to 8 frames per PDU. The values of these two parameters are operator configurable at the RNC for the purpose of making the most efficient use of the selected bearer type.

4) Ret_DTX

Boolean Indicating that DTX is or is not to be invoked for return direction transmission.

5) Fwd_DTX

Boolean Indicating that DTX is or is not to be invoked for forward direction transmission.

NOTE 2: Discontinuous Transmission is operator configured at the RNC. Discontinuous transmission in the forward and return directions may be enabled independently. The DTX flags are passed from the RNC to the UE in the RBC: Establish message during the RAB setup. The invocation of DTX applies for the lifetime of the RAB (i.e. the duration of the call) only.

6) CircuitSwitchedCallType

The following Circuit Switched Call Types are supported:

- 4 kbit/s Speech Call using the satellite optimized low data rate voice codec (CircuitSwitchedCallType = "type-4kbits-speech")
- 64 kbit/s Circuit for Pulse Code Modulation (PCM) coded analog modem, analog fax or high quality speech (CircuitSwitchedCallType = "type-3pt1khz-audio")
- 64 kbit/s Circuit for Integrated Service Digital Network (ISDN) Data (CircuitSwitchedCallType = "type-udi-isdn")
- 56 kbit/s Circuit for ISDN Data (CircuitSwitchedCallType = "type-rdi-isdn").

The mapping of Call Control Bearer Capability Information Element to CircuitSwitchedCallType is presented in annex A.

The mechanism for determining the required circuit switched service at the border of the satellite network (i.e. terminal equipments attached to the satellite network UE or external networks) is discussed in annex C.

The CircuitSwitchedCallType = "type-4kbits-speech" triggers the Circuit Switched Manager to invoke the data handler for the 4 kbit/s voice circuit (VCSH) as described in clause 5.

The CircuitSwitchedCallTypes = "type-3pt1khz-audio", "type-udi-isdn" or "type-rdi-isdn" invoke the data handler for the 64 kbit/s service (BCSH) as described in clause 6.

4.1.4.2 Control Plane Primitives between CSH and Call Control Layer

The control primitives between the CSH Manager and the Call Control layer at the UE are shown in Table 4.3.

NOTE: The Service Access Point interface to the Call Control layer, shown here, is only for the purpose of illustrating the requirement to pass Circuit Switched Call Type information between CC and CSH and does not specify or constrain implementation. The SAP is not specified in [1].

Table 4.3: Control Plane Primitives between CSH and the Call Control Layer (UE side)

SAP	Generic Name	Parameter			
		_REQ	_IND	_RESP	_CNF
CSH-CC-SAP	CSH_CC_CONFIG	-	Not Defined	Not Defined	CircuitSwitchedCallType

Each Primitive is defined as follows:

1) CSH_CC_CONFIG_REQ

   CSH_CC_CONFIG_REQ is used by the CSH Manager to request the type of circuit switched call from the Call Control layer.

2) CSH_CONFIG_CNF

   CSH_CC_CONFIG_CNF is used by the CC entity to respond to a CSH_CC_CONFIG_REQ message from the CSH Manager with the details of the type of circuit switched call being setup.

The CSH Manager uses the CircuitSwitchedCallType to determine what type of CSH Data Handler and DTX algorithm to configure appropriate to the call.

The following parameter is used in the primitives:

1) CircuitSwitchedCallType

   This parameter is as defined in clause 4.1.4.1.

4.1.4.3 User Plane Primitives between CSH and upper/lower layers

The user plane primitives between CSH and upper/lower layers are described in clause 5 for the 4 kbit/s voice circuit switched handler, and in clause 6 for the 64 kbit/s circuit switched handler, respectively.

4.1.4.4 Return Direction BCt User Plane Maintenance Bursts Prior to CC:Connect Message

Following the establishment of a Circuit Switched User Plane connection, to avoid BCt time-out through the UE_Chk Mode mechanism, it is permissible for the UE to transmit BCtPDUs with zero payload as maintenance bursts from the time when radio resources are established (i.e. when the UE sends an RBC:EstablishAck) until the CC:Connect is exchanged and the user connection is attached.

In other words: BCt may send BCtPDUs with zero payload as maintenance bursts when no data is received from CSH - until the user connection is attached, after this, CSH shall deliver data for the maintenance bursts as described in the present document. The BCtPDU with zero payload will satisfy UeChkMode in the RNC and since there is no payload, no user data is forwarded from the RNC which meets 3GPP requirements.

5 CS User Plane for the 4 kbit/s Voice Service

5.1 General Architecture

5.1.0 Overview

The protocol stack for the User Plane providing a circuit switched connection for the low bit rate voice service is shown in Figure 5.1.

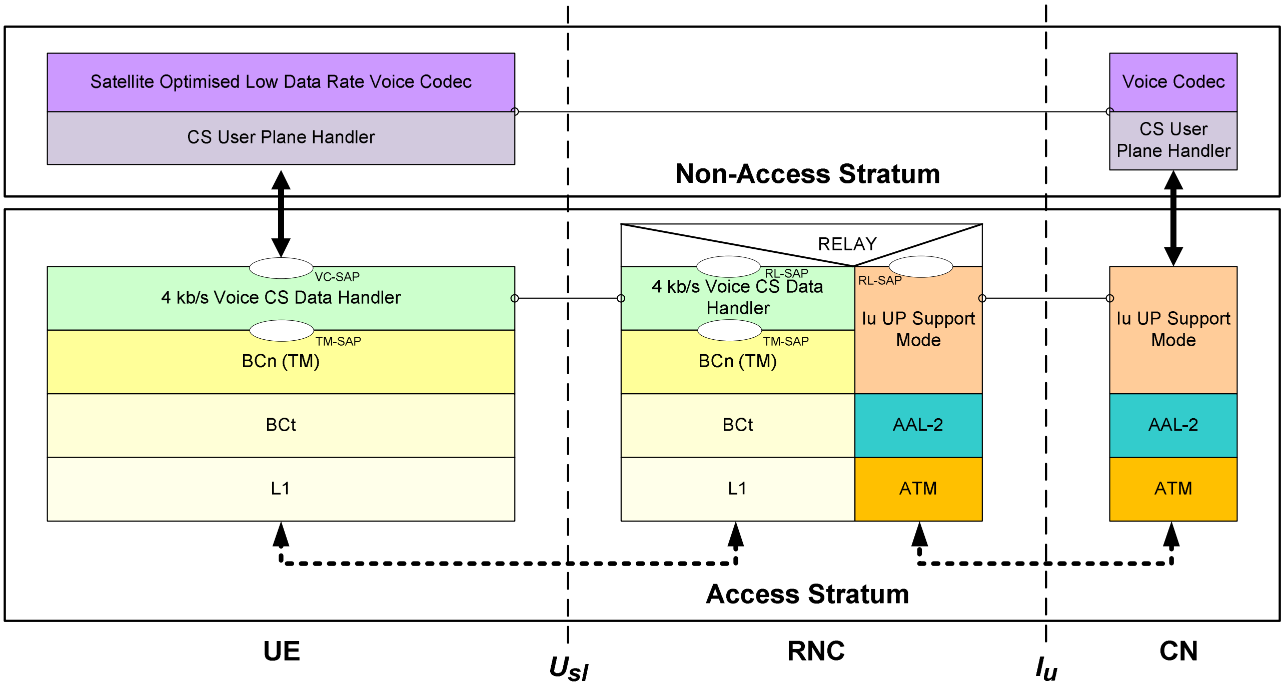

Figure 5.1: Circuit Switched User Plane Protocol Stack with Voice Codec

On the radio interface, the Bearer Connection Layer (BCn) is configured to operate in Transparent Mode (TM) while the Bearer Control Layer (BCt) allocates capacity so that the Bearer Connection Layer (BCn) data queue can be served at a constant bit rate. The 4 kbit/s Voice Circuit Switched Data Handler (VCSH) resides in the Circuit Switched Handler Layer (CSH) layer above the Bearer Connection Layer.

At the UE (User Equipment) side the VCSH is responsible for:

- Forwarding of Voice Codec Frames (VC Frames) from the Bearer Connection Layer to the Voice Codec and vice versa.
- Handling the transmission and reception of multiple voice frames per Circuit Switched Handler PDU (CSH PDU).
- Management of Discontinuous Transmission (DTX) from the UE (for the purpose of battery power conservation) during periods of inactivity.
- Handling of lost frames from the RNC (Radio Network Controller) due to blockages etc. as well as discontinuous transmissions by the RNC.

At the RNC side, the VCSH is responsible for:

- Forwarding of VC Frames from the Bearer Connection Layer via the Relay function to the Iu User Plane Protocol entity and vice versa.
- Handling the transmission and reception of multiple voice frames per CSH PDU.
- Management of Lost Frames from the CN (core network).
- Management of Discontinuous Transmission from the CN (for the purpose of using available bandwidth for other connections) during periods of inactivity.
- Handling of lost frames from the UE (due to blockages etc.) as well as discontinuous transmission by the UE.

The satellite optimized low data rate voice codec receives/transmits voice codec frames (VC frames) of 80 bits every 20 ms, resulting in an aggregate stream of 4 kbit/s.

As mentioned above the VCSH shall concatenate a configurable number of VC Frames into a single CSH PDU before passing this PDU to the Bearer Connection Layer. The parameters *ForwardCSFramesPerPDU* and *ReturnCSFramesPerPDU* specify the number of VC Frames per CSH PDU for the forward (to UE) and return (from UE) direction respectively.

5.1.1 Configuration

Configuration of the VCSH shall be as described in clause 4.1.4.

The values of *ForwardCSFramesPerPDU* and *ReturnCSFramesPerPDU* are passed to the UE in the RBC: Establish message during the RAB setup. They are set independently of each other and apply for the lifetime of the RAB (i.e. the duration of the call) only.

The RBC: Establish message indicates whether Discontinuous Transmission is enabled. Discontinuous transmission in the forward and return directions may be enabled independently by the network operator. The capability to perform DTX at RNC and UE is mandatory in both transmit and receive directions for the 4 kbit/s voice service when the feature enabled.

5.2 CS User Plane Handler (VCSH) for satellite optimized low data rate codec - UE Side

5.2.1 Services Provided to Upper Layers

5.2.1.0 General

The CS User Plane Handler (VCSH) in the UE provides services directly to the satellite optimized low data rate voice codec through the Voice Codec Service Access Point (VC-SAP), as shown in Figure 5.2.

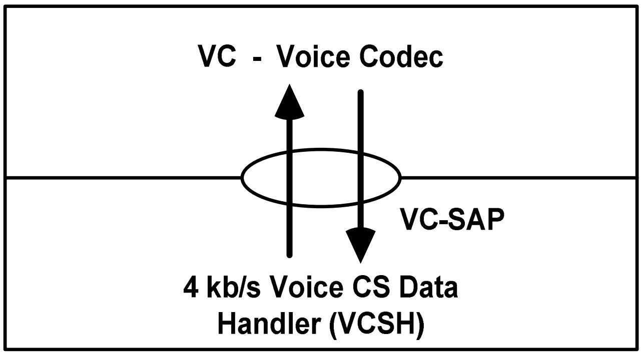

Figure 5.2: Service Access Points to Upper Layers

5.2.1.1 Service Primitives at VC-SAP

The service primitives at VC-SAP are shown in Table 5.1.

Table 5.1: CSH(VCSH)::VC Service Primitives at VC-SAP [UE]

Primitive	Direction	Parameters
VC_CSH_DATA_REQ *The Voice Codec requests CSH to send a VC Frame to its peer in the RNC*	To CSH from VC	VC Frame, DTX_FLAG
VC_CSH_DATA_IND *CSH passes a VC Frame to the VC or indicates to the VC that a VC Frame is absent*	From CSH to VC	CHOICE{VC Frame, LOST_FLAG}

5.2.2 Services Expected from Lower Layers

5.2.2.0 General

The VCSH in the UE uses Transparent Mode message transport services provided by the Bearer Connection Layer to communicate with its peer in the RNC through the TM-SAP, as shown in Figure 5.3.

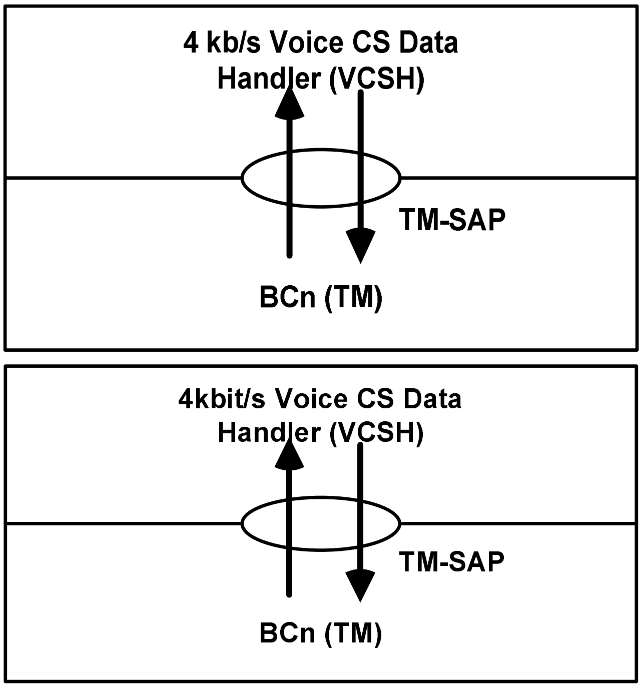

Figure 5.3: Service Access Points to Lower Layers

The Bearer Connection ID (BCnID) is the reference or handle to a particular SAP at the VCSH - Bearer Connection Layer boundary.

5.2.2.1 Service Primitives at TM-SAP

The service primitives at TM-SAP are shown in Table 5.2.

Table 5.2: CSH(VCSH)::BCN Service Primitives at TM-SAP [UE]

Primitive	Direction	Parameters
BCn_TM_DATA_REQ *CSH requests BCn to transmit the CSH-PDU (containing one or more VC Frames) in Transparent Mode*	To BCn from CSH	CSH-PDU, BCnID, DiscardReq, MUI
BCn_TM_DATA_IND *BCn passes a CSH-PDU (containing one or more VC Frames) to CSHin transparent mode*	From BCn to CSH	CHOICE{CSH-PDU, Err_ind}

NOTE: BCn_TM_DATA_CNF is not used by CSH (DiscardReq = FALSE) and therefore not shown in this table.

5.2.3 VCSH [UE] Architecture and Behaviour

5.2.3.0 General

The VCSH [UE] consists of a transmit path and a receive path, both being independent of each other. A first-in-first-out (FIFO) buffer is present in both the transmit path and receive path to handle the transmission and reception of multiple VC frames per CSH PDU. Each FIFO has an associated Control Unit, which handles the access to the FIFO.

In order to maintain minimum buffering delay and to avoid the problems associated with asynchronous clocks it is assumed that the Voice Codec Clock is synchronized to the forward frames received on the radio interface and that the VCSH [UE] behaviour is triggered at regular intervals by the Voice Codec (i.e. every 20 ms). Details of the clock synchronization are implementation specific and outside the scope of the present document.

Bit ordering of the satellite optimized low data rate codec payload frames shall be such that the 80 bits of each voice frame are transmitted with no overhead and in order with bit 0 transmitted first and bit 79 transmitted last.

The VCSH entity is created and initialized by the CSH Manager on command from the Adaptation Layer through a service access point. Service Access Point and Primitives for configuring of circuit switched user plane protocol entities by the Adaptation Layer are specified in clause 4.1, which includes a description of the configuration of the number of voice codec frames to be transmitted/received per CSH PDU.

5.2.3.1 [UE] Transmit Behaviour [Return]

The VCSH [UE] transmit path buffers ReturnCSFramesPerPDU VC frames (satellite optimized low data rate codec frames) and their associated DTX_FLAGs received via the VC-SAP from the VC at a rate of one VC frame per 20 ms interval (see Figure 5.4). If at least one of the DTX_FLAGs associated with a VC frame is low, all the frames are concatenated and passed to the BCn (in transparent mode, TM) via the TM-SAP. If no DTX_FLAG is low (and return direction discontinuous transmission has been enabled in the Establish message), the VC PDUs are discarded, thus effecting discontinuous transmission. In order to ensure that a Bearer Connection is not erroneously cleared by the RNC, the UE shall periodically transmit, at least once every 1,5 seconds, a CSH-PDU of user data containing ReturnCSFramesPerPDU VC frames in the return direction during extended periods of DTX.

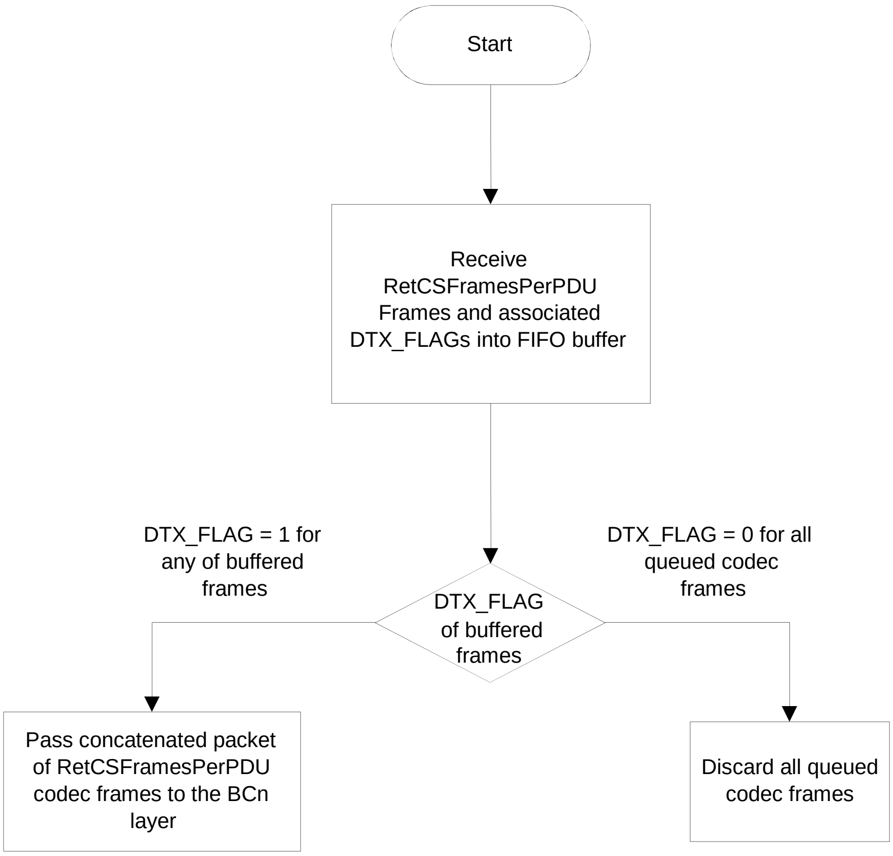

Figure 5.4: UE Return Direction (transmit) VCSH Behaviour for Satellite Optimized Low Data Rate Codec

5.2.3.2 [UE] Receive Behaviour [Forward]

When the VCSH in the UE receives a Data-PDU from BCn, via the TM-SAP, it shall separate the VC frames and place them in the receive FIFO buffer (see Figure 5.5). At successive 20 ms intervals the VCSH shall pass a single VC frame up to the VC via the VC-SAP. If the buffer is empty, the VCSH shall instead call the VC via the VC-SAP with condition LOST_FLAG. Note that the UE VCSH in the receive direction shall not distinguish between voice, Self Imposed Delay (SID), tone or dummy (type 61) codec frames, but shall instead rely on the satellite optimized low data rate codec above to handle the frames appropriately.

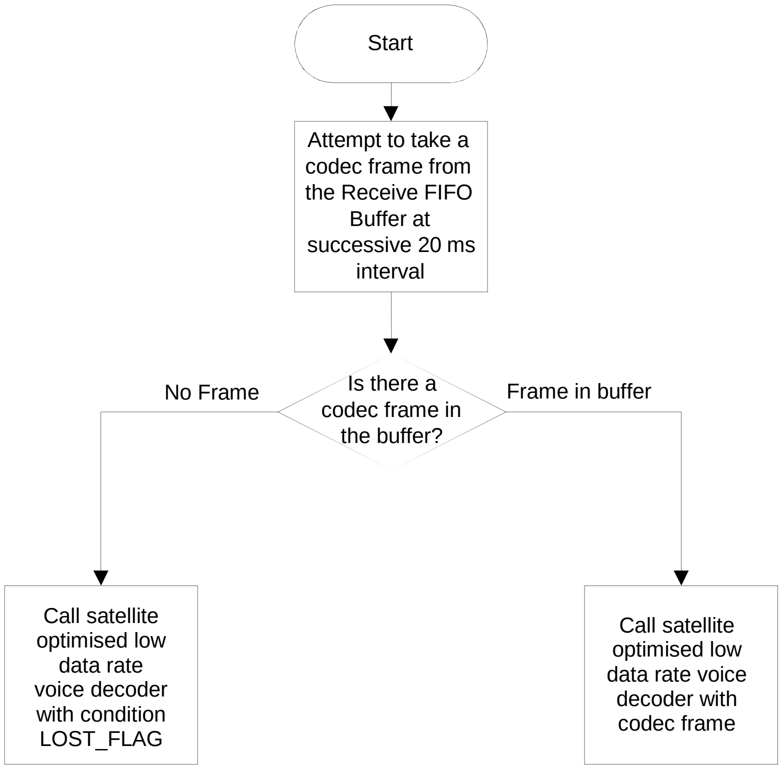

Figure 5.5: UE Forward (receive) Direction VCSH Behaviour for Satellite Optimized Low Data Rate Codec

5.3 Circuit Switched User Plane Handler for Satellite Optimized Low Data Rate Codec - RNC Side

5.3.1 Services Provided to Upper Layers

5.3.1.0 General

The VCSH in the RNC provides services to the satellite optimized low data rate voice codec indirectly via a relay to the Iu UP layer, through the RL-SAP, as shown in Figure 5.6.

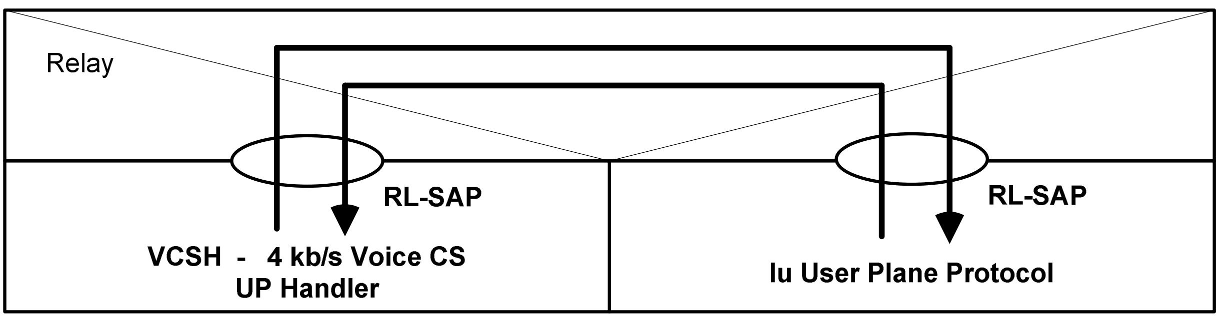

Figure 5.6: Service Access Points to Upper Layers [RNC]

5.3.1.1 Service Primitives at RL-SAP [RNC]

The service primitives at RL-SAP are shown in Table 5.3.

Table 5.3: CSH(VCSH)::IuUP Service Primitives at RL-SAP [RNC]

Primitive	Direction	Parameters
RL_CSH_DATA_REQ *The Iu UP layer requests CSH to send a VC Frame to its peer in the UE or the IuUP sends a NO_DATA frame to the CSH*	To CSH from IuUP	CHOICE{VC Frame, NO_DATA}
RL_CSH_DATA_IND *CSH passes a VC frame to Iu UP or passes a NO_DATA frame to IuUP*	From CSH to IuUP	CHOICE{VC Frame, NO_DATA}

5.3.2 Services Expected from Lower Layers

5.3.2.0 General

The VCSH in the RNC uses Transparent Mode message transport services provided by the Bearer Connection Layer to communicate with its peer in the UE through the TM-SAP, as shown in Figure 5.7.

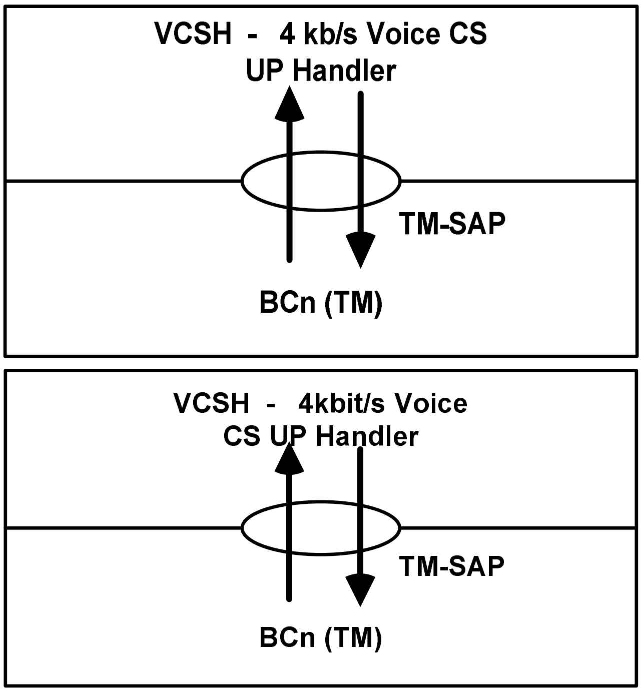

Figure 5.7: Service Access Points to Lower Layers [RNC]

5.3.2.1 Service Primitives at TM-SAP

The service primitives at TM-SAP are shown in Table 5.4.

Table 5.4: CSH(VCSH)::BCN Service Primitives at TM-SAP [RNC]

Primitive	Direction	Parameters
BCn_TM_DATA_REQ *CSH requests BCn to transmit the CSH-PDU in Transparent Mode*	To BCn from CSH	CSH-PDU, BCnID, DiscardReq, MUI
BCn_TM_DATA_IND *BCn passes a CSH PDU to CSH that has been received in transparent mode*	From BCn to CSH	CHOICE{CSH-PDU, Err_ind}

NOTE: BCn_TM_DATA_CNF is not used by CSH (DiscardReq = FALSE) and therefore not shown in this table.

5.3.3 VCSH [RNC] Architecture and Behaviour

5.3.3.1 Iu UP Configuration

For Voice operation, the Iu UP (Iu Interface User Plane Protocol) supports predefined PDU sizes.

5.3.3.2 [RNC] Transmit Behaviour [Forward]

When the VCSH in the RNC receives an Iu UP frame from the Iu UP layer (transmitted from the CN) via the RL-SAP then it shall determine if the frame is a NO_DATA Iu UP frame or contains a single VC frame (see Figure 5.8). If the received frame is a VC frame (SID, tone or voice), the VCSH shall place the received frame in the FIFO transmit buffer. If the received frame is a NO_DATA frame then the VCSH shall generate a dummy codec frame and store it in the FIFO transmit buffer.

If no Iu UP frame is received via the RL-SAP at the 20 ms interval, the VCSH shall assume that the Iu UP frame was lost in transit and shall generate a dummy codec frame and store it in the FIFO transmit buffer.

When ForwardCSFramesPerPDU are present in the transmit buffer, the VCSH shall check that at least one of the buffered frames is a valid VC frame. If so, it will then concatenate all the frames in the buffer and pass them in a single CSH PDU to the BCn layer via the TM-SAP. If none of the frames in the transmit FIFO buffer are valid frames, and forward discontinuous transmission has been enabled, then the VCSH will discard all the frames in the buffer. This mechanism thus effects discontinuous transmission.

Bit ordering of satellite optimized low data rate codec payload frames shall be such that the 80 bits of each voice frame is transmitted with no overhead/framing and in order with bit 0 transmitted first and bit 79 transmitted last.

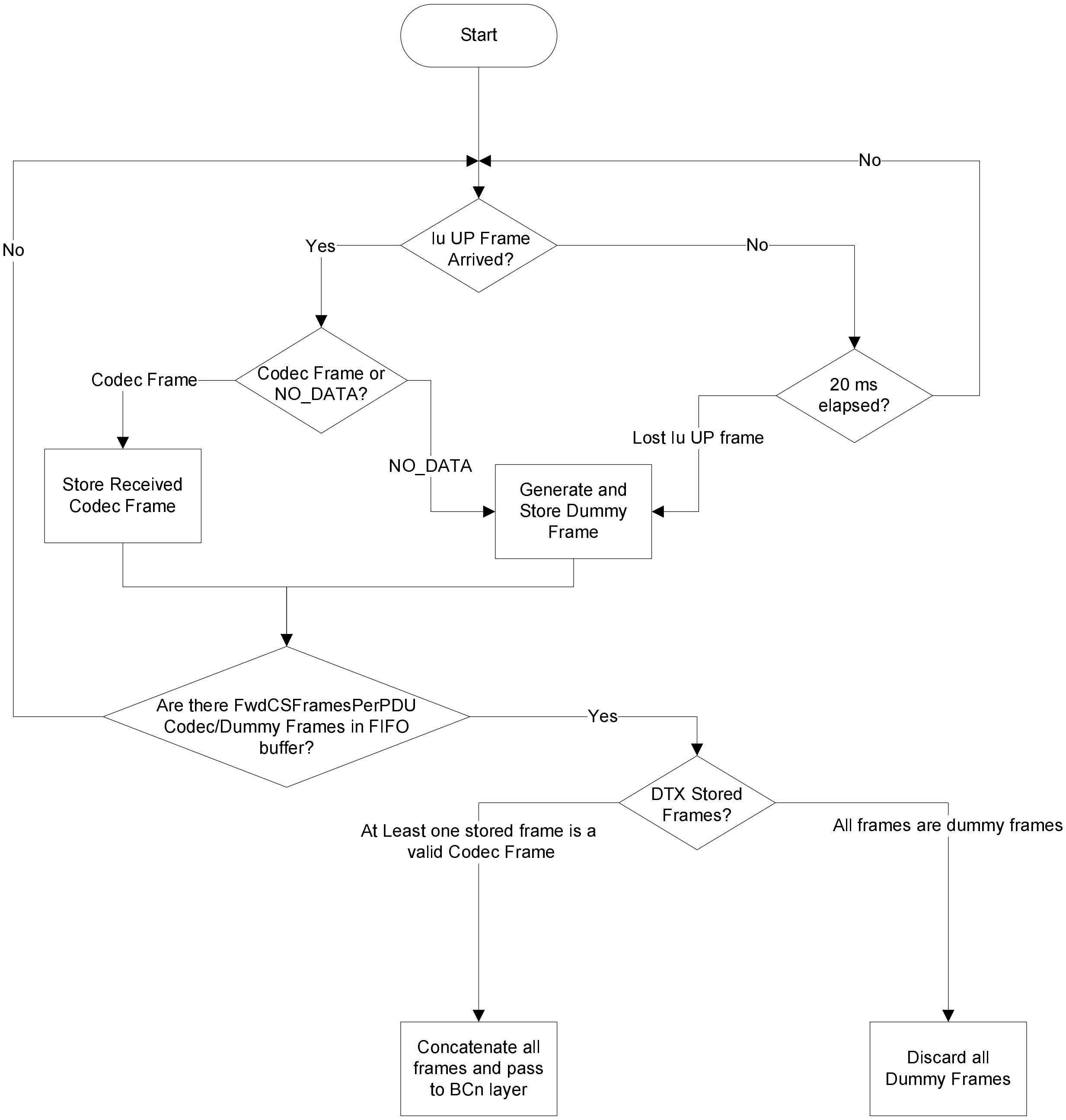

Figure 5.8: RNC Transmit behaviour (Forward Direction) for Satellite Optimized Low Data Rate Codec

5.3.3.3 [RNC] Receive Behaviour [Return]

The RNC VCSH awaits delivery of a CSH PDU via the TM-SAP from the BCn layer (see Figure 5.9). On arrival of a CSH PDU the VCSH separates the codec frames within the PDU and places them in the FIFO receive buffer. At successive 20 ms intervals, one VC frame is removed from the buffer and handed to the Iu UP layer via the relay function (RL-SAP). If the buffer is instead empty then the VCSH shall invoke a 'NO_DATA' frame at the Iu UP protocol for transmission to the CN.

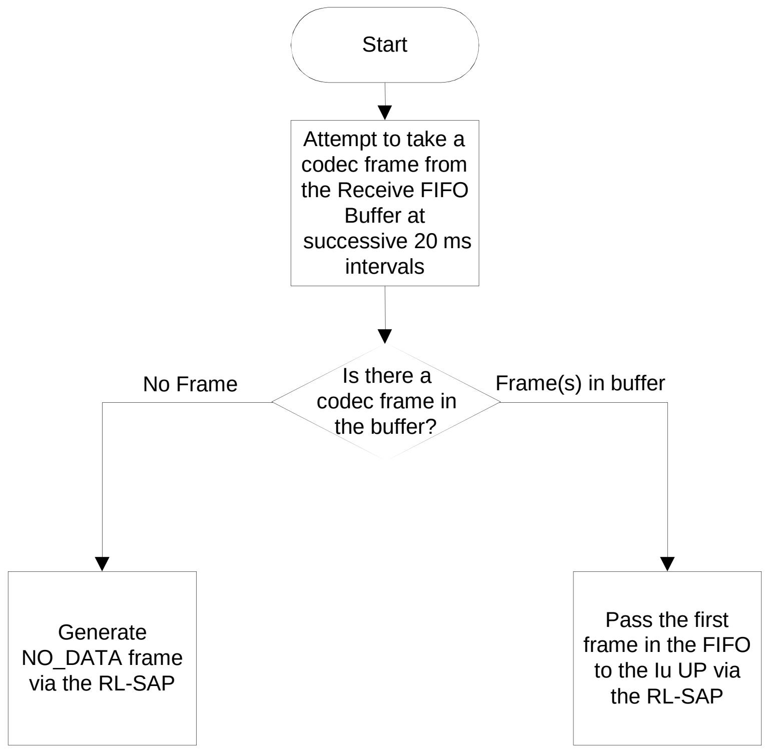

Figure 5.9: RNC Return Direction Behaviour for Satellite Optimized Low Data Rate Codec

5.4 Codec Type Signalling for Satellite Optimized Low Data Rate Voice Codec

5.4.1 Call Control Protocol Supported Codec Information Element

In order to inform the core network Mobile Switching Centre (MSC) of the capability of the UE to support the satellite optimized low data rate voice codec, both the Call Control SETUP (mobile originated case) and CALL CONFIRMED (mobile terminated case) messages generated by the UE shall include the Supported Codec List IE.

Table 5.5: Supported Codec List Information Element in SETUP or CALL CONFIRMED message (refer to [1], clause 9.3.23.2 and clause 9.3.2)

IEI	Information element	Type/Reference	Presence	Format	Length
40	Supported Codecs	Supported Codec List	O	TLV	5-n

Table 5.5 indicates the Information Element within the SETUP message that contains the Supported Codec List. For UEs that support the satellite optimized low data rate voice codec, the presence of this IE is mandatory. The generic format of the Supported Codec list IE is presented in Table 5.6.

Table 5.6: Generic Format of Supported Codec List Information Element (refer to [1], clause 10.5.4.32)

Supported Codec List IEI	octet 1
Length Of Supported Codec list	octet 2
System Identification 1 (SysID 1)	octet 3
Length Of Bitmap for SysID 1	octet 4
Codec Bitmap for SysID 1, bits 1 to 8	octet 5
Codec Bitmap for SysID 1, bits 9 to 16	octet 6
System Identification 2 (SysID 2)	octet j
Length Of Bitmap for (SysID 2)	octet j+1
Codec Bitmap for (SysID 2), bits 1 to 8	octet j+2
Codec Bitmap for (SysID 2), bits 9 to 16	octet j+3
System Identification x (SysID x)	octet m
Length Of Bitmap for (SysID x)	octet m+1
Codec Bitmap for (SysID x), bits 1 to 8	octet m+2
Codec Bitmap for (SysID x), bits 9 to 16	octet m+3

In the Supported Codec List IE the supported codecs are coded in the first and second octets of the Codec bitmap as shown in Table 5.7. A bit set to '1' indicates support for the named codec, a bit set to '0' indicates the absence of support.

Table 5.7: Generic Codec Bitmap in Supported Codec List (see [2])

bit 8	bit 7	bit 6	bit 5	bit 4	bit 3	bit 2	bit 1	
TDMA EFR	UMTS AMR 2	UMTS AMR	HR AMR	FR AMR	GSM EFR	GSM HR	GSM FR	Octet 1

bit 16	bit 15	bit 14	bit 13	bit 12	bit 11	bit 10	bit 9	
(reserved)	(reserved)	(reserved)	(reserved)	(reserved)	(reserved)	(reserved)	PDC EFR	Octet 2

Reserved bit 16 in Octet 2 of the Codec Bitmap is used to signal the capability to use the satellite optimized low data rate voice codec in the UE, as presented in Table 5.8. A UE that sets bit 16 to '1' makes it known to the network that it supports the satellite optimized low data rate voice codec.

Table 5.8: Variant of Octet 2 in Codec Bitmap to indicate presence of Satellite Optimized Low Data Rate Voice Codec Capability

bit 16	bit 15	bit 14	bit 13	bit 12	bit 11	bit 10	bit 9	
satellite optimized low data rate voice codec	(reserved)	(reserved)	(reserved)	(reserved)	(reserved)	(reserved)	PDC EFR	Octet 2

An example of the Supported Codec List IE for a UE supporting the satellite optimized low data rate voice codec is shown in Table 5.9.

Table 5.9: Example Supported Codec List IE Indicating Satellite Optimized Low Data Rate Voice Codec Capability in UE

Octet 1	0100 0000	Supported Codec List IEI
Octet 2	0000 0100	Length of the supported codec list: 4 octets
Octet 3	0000 0100	System Identification: UMTS
Octet 4	0000 0010	Length of the codec bitmap: 2 octets
Octet 5	xxxx xxxx	Codec bitmap, bits 1 to 8
Octet 6	1xxx xxxx	Codec bitmap, bits 9 to 16: satellite optimized low data rate voice codec, bit 16

5.4.2 NAS Synchronization Indicator - Satellite Optimized Low Data Rate Voice Codec

If the Supported Codec List IE is received, the CN will select one of the available codecs from the codec list and indicate this to the UE using the NAS Synchronization Indicator IE when setting up a Radio Access Bearer.

For speech calls where the UE has indicated the capability to support the satellite optimized low data rate voice codec, the network will select this codec by sending the NAS Synchronization Indicator IE set to the appropriate value.

In the RANAP (Radio Access Network Application Part) 'RAB ASSIGNMENT REQUEST' message from the CN to the RNC, the IE 'RAB Parameters' contains the 'NAS Synchronization Indicator'. This IE is defined in ASN.1 as follows:

```
NAS-Synchronisation-Indicator ::= BIT STRING (SIZE(4))
```

The values defined for different codec types are shown in Table 5.10.

Table 5.10: NAS Synchronization Indicator Values

Bitstring	Codec Type to be used
0000	GSM Full Rate (13,0 kbit/s)
0001	GSM Half Rate (5,6 kbit/s)
0010	GSM Enhanced Full Rate (12,2 kbit/s)
0011	Full Rate Adaptive Multi-Rate
0100	Half Rate Adaptive Multi-Rate
0101	UMTS Adaptive Multi-Rate
0110	UMTS Adaptive Multi-Rate 2
0111	TDMA Enhanced Full Rate (7,4 kbit/s)
1000	PDC Enhanced Full Rate (6,7 kbit/s)
1001 - 1110	reserved for future use
1111	Satellite Optimized Low Data Rate Voice Codec

Note that the value '1111' is assigned to indicate the selection of the satellite optimized low data rate voice codec and is valid within the satellite network only.

The RNC transparently forwards the NAS Synchronization Indicator to the UE when setting up the Bearer Connection to the UE for the user plane of the voice call, using the Adaptation Layer AVP 'RAB Info'. This mechanism is described in ETSI TS 102 744-3-5 [14].

Upon successful reception of NAS Synchronization Indicator at the UE, the UE will activate the indicated codec.

5.5 DTMF Transmission

Except during use of the satellite optimized low data rate voice codec and the 64 kbit/s '3,1 kHz audio' service, transmission of mobile originated DTMF tones is carried out using the 3GPP DTMF protocol as specified in [3].

For speech calls where the satellite optimized low data rate voice codec is activated, the 3GPP DTMF protocol shall be disabled. Instead the satellite optimized low data rate voice codec algorithm's own tone detection and transmission algorithm shall be employed.

During 64 kbit/s '3,1 kHz audio' calls, DTMF will be carried in-band and will not make use of the 3GPP DTMF protocol.

6 CS User Plane for the 64 kbit/s Synchronous Circuit Switched Service

6.1 General Architecture

The protocol stack for the User Plane providing a circuit switched connection for the UDI/RDI/3,1 kHz services is shown in Figure 6.1. This service offers a 64 kbit/s synchronous transparent bearer to the higher layer, equivalent to an ISDN B channel.

The BCSH operates using similar features to those previously described for the 4 kbit/s voice service (VCSH) in that a configurable number of user data frames may be concatenated into a single CSH PDU for transmission in a single BCtPDU. The BCSH also provides a discontinuous transmission mechanism for the purposes of battery power and bandwidth conservation. The BCSH will also substitute dummy user plane traffic in the event of lost frames. Note however that the details of the discontinuous transmission mechanism and TTI (Transmission Timing Interval) timing are different to those previously described for 4 kbit/s voice.

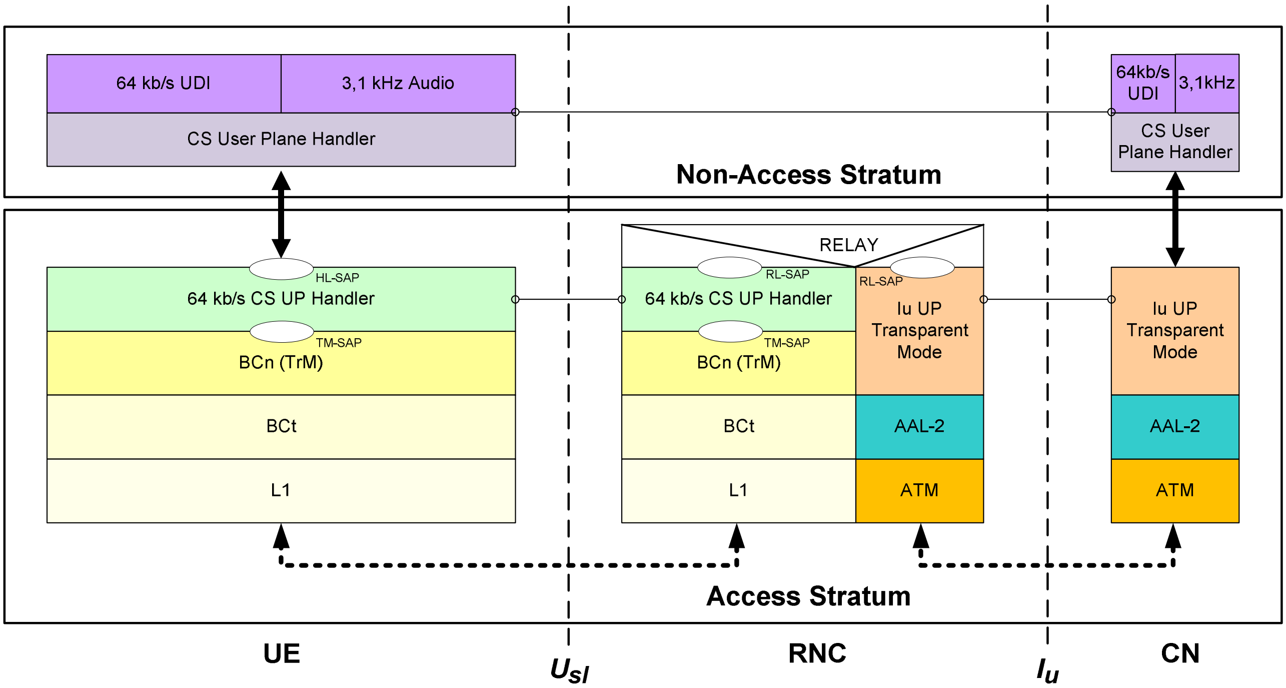

Figure 6.1: Circuit Switched User Plane Protocol Stack for 64 kbit/s Service

On the radio interface, the Bearer Connection Layer (BCn) is configured to operate in Transparent Mode (TM) while the Bearer Control Layer (BCt) allocates capacity so that the BCn data queue can be served at a constant bit rate. The BCSH resides above the Bearer Connection Layer.

At the UE side the BCSH is responsible for:

- Forwarding of Data Frames from the Bearer Connection Layer to the Higher Layer and vice versa.
- Handling the transmission and reception of multiple user data frames per Connection Layer PDU.
- Management of Discontinuous Transmission from the UE (for the purpose of battery power conservation) during periods of inactivity.
- Handling of lost frames from the RNC (due to blockages etc.) as well as discontinuous transmissions by the RNC.

At the RNC side, the BCSH is responsible for:

- Forwarding of user data frames from the Bearer Connection Layer via the Relay function to the Iu User Plane Protocol entity and vice versa.
- Handling the transmission and reception of multiple user data frames per Connection Layer PDU.
- Management of Lost Frames from the CN.
- Handling of lost frames from the UE (due to blockages etc.) as well as discontinuous transmissions by the UE.

On the UE side the layer above the BCSH provides a service which transmits/receives a synchronous data stream of 64 000 bits/s equivalent to one ISDN-B Channel. On the RNC side the data to/from the Core Network is passed in equally sized, equally spaced frames (of 640 bits transmitted every 10 ms). Since synchronization shall be maintained with this framing structure, the BCSH layer shall also concatenate multiples of 640 bit user data frames for transmission in both the forward and return direction.

As mentioned above the BCSH shall concatenate a configurable number of User Data Frames into a single CSH PDU before passing this PDU to the Bearer Connection Layer. The parameters *ForwardCSFramesPerPDU* and *ReturnCSFramesPerPDU* specify the number of User Data Frames per CSH PDU for the forward (to UE) and return (from UE) direction respectively.

The values of *ForwardCSFramesPerPDU* and *ReturnCSFramesPerPDU* are passed to the UE in the Establish message during the RAB setup. They are set independently of each other and apply for the lifetime of the RAB (i.e. the duration of the call) only. For further details refer to ETSI TS 102 744-3-6 [15].

6.2 64 kbit/s CS User Plane Handler (BCSH) - UE Side

6.2.1 64 kbit/s BCSH Services Provided to Upper Layers [UE]

6.2.1.0 General

The BCSH in the UE provides services directly to the higher layer through the Higher Layer Service Access Point (HL-SAP), as shown in Figure 6.2.

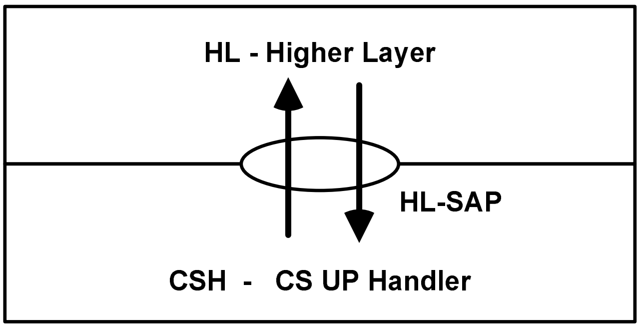

Figure 6.2: 64 kbit/s CS Service Access Points to Upper Layers [UE]

6.2.1.1 64 kbit/s CS Service Primitives at HL-SAP [UE]

The 64 kbit/s CS Service Primitives at HL-SAP are shown in Table 6.1.

Table 6.1: 64 kbit/s CSH(BCSH)::HL Service Primitives at HL-SAP [UE]

Primitive	Direction	Parameters
HL_CSH_DATA_REQ *The Higher Layer requests CSH to send a User Data Frame to its peer in the RNC*	To CSH from HL	User Data Frame
HL_CSH_DATA_IND *CSH indicates to the Higher Layer that it has received a PDU (User Data Frame) from the peer entity in the RNC*	From CSH to HL	User Data Frame

6.2.2 64 kbit/s CS Service, Services Expected from Lower Layers [UE]

6.2.2.0 General

The BCSH in the UE uses Transparent Mode message transport services provided by the Bearer Connection Layer to communicate with its peer in the RNC through the TM-SAP, as shown in Figure 6.3.

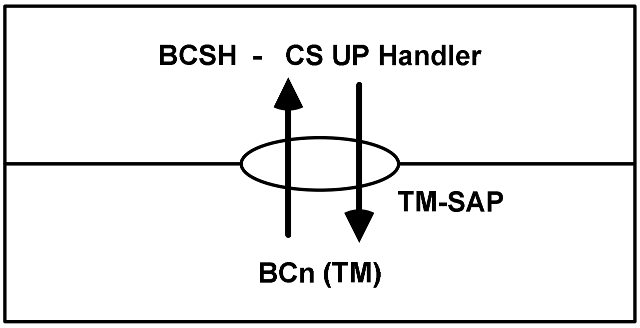

Figure 6.3: 64 kbit/s CS Service Access Points to Lower Layers

The Bearer Connection ID (BCnID) is the reference or handle to a particular SAP at the CSH- Bearer Connection Layer boundary.

6.2.2.1 64 kbit/s CS Service Primitives at TM-SAP [UE]

The 64 kbit/s CS Service Primitives at TM-SAP are shown in Table 6.2.

Table 6.2: 64 kbit/s CSH(BCSH)::BCN Service Primitives at TM-SAP [UE]

Primitive	Direction	Parameters
BCn_TM_DATA_REQ *CSH requests BCn to transmit the CSH PDU(containing one or more user data frames) in Transparent Mode*	To BCn from CSH	CSH-PDU, BCnID, DiscardReq, MUI
BCn_TM_DATA_IND *BCn indicates to CSH that a CSH PDU (containing one or more user data frames) has been received in transparent mode*	From BCn to CSH	CHOICE{CSH-PDU, Err_ind}

NOTE: BCn_TM_DATA_CNF is not used by CSH (DiscardReq = FALSE) and therefore not shown in this table.

6.2.3 64 kbit/s BCSH [UE] Architecture and Behaviour

6.2.3.0 General

The BCSH [UE] consists of a transmit and a receive path, both being independent of each other. A first-in-first-out (FIFO) buffer is present in both the transmit and receive path to handle the transmission and reception of single or multiple user data frames per Connection Layer PDU. Each FIFO has an associated Control Unit that handles the access to the FIFO.

In order to maintain minimum buffering delay and to avoid the problems associated with asynchronous clocks it is assumed that the Higher Layer Clock is synchronized to the forward frames received on the radio interface and that the BCSH [UE] behaviour is triggered at regular intervals by the Higher Layer (i.e. every 10 ms). Details of the clock synchronization are implementation specific and outside the scope of the present document.

The BCSH entity is created and initialized by the Adaptation Layer through a service access point. Service Access Point and Primitives for configuring of user plane protocol entities by the adaptation layer are in ETSI TS 102 744-3-6 [15].

6.2.3.1 64 kbit/s [UE] Transmit Behaviour [Return]

The BCSH transmit path in the UE receives ReturnCSFramesPerPDU user data frames (each of 640 bits) from the higher layer via the HL-SAP (where the frames are intended for onward transmission towards the CN). The reception of user data frames is synchronized to the radio interface transmission time at 10 ms intervals.

At an interval of (10 ms × ReturnCSFramesPerPDU) (where the interval is synchronized to the radio interface transmission time), the BCSH will remove ReturnCSFramesPerPDU from the FIFO receive buffer and concatenate these frames into a single CSH PDU. It will then pass the resulting CSH PDU to the BCn (TM) layer via the TM-SAP.

The BCSH will operate a discontinuous transmission algorithm. This algorithm is described further in clause 6.6.

6.2.3.2 64 kbit/s [UE] Receive Behaviour [Forward]

When the UE BCSH receives a CSH PDU passed up from BCn, via the TM-SAP, it shall separate it into its constituent User Data Frames and place them in the receive FIFO buffer (each CSH PDU will contain ForwardCSFramesPerPDU User Data Frames intended for the higher layer). At successive 10 ms intervals, the BCSH shall remove a single User Data Frame from the buffer and pass it up to the Higher Layer via the HL-SAP. The BCSH will operate a discontinuous transmission algorithm: If the buffer is empty the BCSH will locally generate idle-frame data until further user data arrives. The DTX algorithm is further described in clause 6.6.

6.3 64 kbit/s CS User Plane Handler (BCSH) - RNC Side

6.3.1 64 kbit/s Services Provided to Upper Layers [RNC]

6.3.1.0 General

The BCSH in the RNC provides services to the higher layer indirectly via a relay to the Iu UP layer, through the RL-SAP, as shown in Figure 6.4.

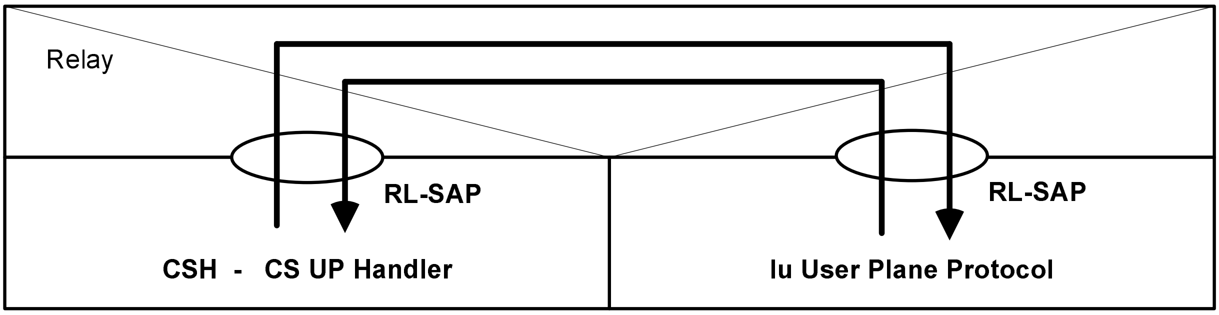

Figure 6.4: 64 kbit/s Service Access Points to Upper Layers [RNC]

6.3.1.1 64 kbit/s Service Primitives at RL-SAP [RNC]

The 64 kbit/s Service Primitives at RL-SAP are shown in Table 6.3.

Table 6.3: 64 kbit/s CSH(BCSH)::IuUP Service Primitives at RL-SAP [RNC]

Primitive	Direction	Parameters
RL_CSH_DATA_REQ *The Iu UP layer requests CSH to send a User Data Frame received from the CN to the UE.*	To CSH from IuUP	User Data Frame
RL_CSH_DATA_IND *CSH sends a User Data Frame (received from the UE), towards the Iu UP for onward transmission to the CN*	From CSH to IuUP	User Data Frame

6.3.2 64 kbit/s Services Expected from Lower Layers [RNC]

6.3.2.0 General

The BCSH in the RNC uses Transparent Mode message transport services provided by the Bearer Connection Layer to communicate with its peer in the UE through the TM-SAP, as shown in Figure 6.5.

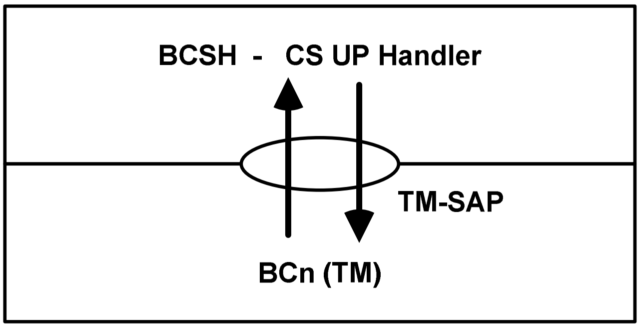

Figure 6.5: 64 kbit/s Service Access Points to Lower Layers [RNC]

6.3.2.1 64 kbit/s Service Primitives at TM-SAP [RNC]

The 64 kbit/s Service Primitives at TM-SAP are shown in Table 6.4.

Table 6.4: 64 kbit/s CSH(BCSH)::BCN Service Primitives at TM-SAP [RNC]

Primitive	Direction	Parameters
BCn_TM_DATA_REQ *CSH requests BCn to transmit the CSH PDU (containing one or more User Data Frames) in Transparent Mode*	To BCn from CSH	CSH-PDU, BCnID, DiscardReq, MUI
BCn_TM_DATA_IND *BCn indicates to CSH that a CSH PDU (containing one or more User Data Frames) has been received in transparent mode*	From BCn to CSH	CHOICE{CSH-PDU, Err_ind}

NOTE: BCn_TM_DATA_CNF is not used by CSH (DiscardReq = FALSE) and therefore not shown in this table.

6.3.3 64 kbit/s BCSH [RNC] Architecture and Behaviour

6.3.3.1 Iu UP Configuration

The Iu UP is operated in transparent mode (see [4]). The Core Network receives/transmits Iu UP Frames as specified in [i.4] of 640 bits with a transmission time interval (TTI) of 10 ms resulting in an aggregate stream of 64 000 bits/s.

6.3.3.2 64 kbit/s [RNC] Transmit Behaviour [Forward]

The BCSH in the RNC shall receive Transparent Mode Iu UP PDUs from the CN via the RL-SAP. The Transparent Mode Iu UP PDUs are 640 bit User Data Frames arriving at 10 ms intervals. The BCSH will place the User Data Frames in the FIFO receive buffer. Note that, unlike the case of the Voice Codec service, the 64 kbit/s service is a synchronous transparent bearer; therefore the Iu UP does not pass NO_DATA Iu UP frames.

At an interval of (10 ms × ForwardCSFramesPerPDU) (where the interval is synchronized to the radio interface transmission time), the BCSH will remove ForwardCSFramesPerPDU from the FIFO receive buffer and concatenate these frames into a single CSH PDU. It will then pass the resulting CSH PDU to the BCn (TM) layer via the TM-SAP for transmission in a single BCtPDU.

The BCSH will operate a Discontinuous Transmission algorithm. This algorithm is described further in clause 6.6.

6.3.3.3 64 kbit/s [RNC] Receive Behaviour [Return]

When the RNC BCSH receives a CSH PDU passed up from BCn, via the TM-SAP, it shall separate it into its constituent User Data Frames and place them in the receive FIFO buffer (each CSH PDU contains ReturnCSFramesPerPDU User Data Frames intended for the CN). At 10 ms intervals, the BCSH shall remove a single User Data Frame from the buffer and pass it up to the Relay/Iu UP via the RL-SAP. In the absence of buffered frames, the RNC BCSH will generate repeat idle frames locally for transmission towards the CN until the carrier is re-established. The DTX Algorithm is further described in clause 6.6.

6.4 Plesiochronous and Transmission Schedule Buffer Considerations

6.4.0 General

Plesiochronous and transmission schedule buffering will be required at the User Equipment to support the bit synchronous 64 kbit/s bearer for UDI/RDI and 3,1 kHz audio services.

Plesiochronous control shall be exercised to interface the slightly different clock rates between the received data clocks and local data clocks.

Transmission schedule buffering is required to ensure that a CSH PDU is always available for transmission at the scheduled Transmission Time Interval (TTI).

The satellite network Radio Resource Mechanism schedules transmissions for Circuit Switched services at intervals synchronous with the forward bearer frame timing. Transmit time intervals are scheduled at intervals of (10 ms × ForwardCSFramesPerPDU) and (10 ms × ReturnCSFramesPerPDU) in the forward and return directions, respectively.

Where DTX is enabled it shall be permitted to re-centre buffers during periods where transmission is inhibited in order to minimize the frequency of slip/stuff during wanted user traffic.

6.4.1 Plesiochronous/transmission schedule Buffer Requirements at the UE

NOTE: The requirements shown are the mandatory default settings. UE manufacturers may optionally allow the ability to reconfigure these values, to make a different delay/slip trade-off to better suit specific CS Call Types.

The requirements for the UE receive plesiochronous buffer are as follows:

1) The minimum time between clock slips shall be 30 minutes, there shall be no clock slips for the first 30 minutes of a call.

2) The maximum delay shall be 10 ms, this corresponds to a half full buffer delay of 5 ms.

3) An integral number of octets shall be used during a buffer slip or stuff in order to maintain byte integrity.

The requirement for the UE transmission schedule buffer is as follows:

1) The UE shall not miss a schedule transmit-time for a CSH PDU. Transmission of a partial CSH PDU shall be forbidden. If it proves necessary to slip or stuff in order to ensure the transmission of a full sized CSH PDU this shall be done so on a boundary of an integral number of octets.

Note that the NT1 clock at the UE may need to have a stability of at least $2,5 \times 10^{-6}$.

6.4.2 Plesiochronous/transmission schedule Buffer Requirements at the RNC

The RNC shall minimize jitter/wander in sending of consecutive Iu UP PDUs of 640 bits (80 octets) for the 64 kbit/s CS services towards the CN. Transmission timing on the Iu Interface shall be synchronized to the RNC clock.

No explicit requirement for the RNC transmission schedule buffer is stated here since the RNC and MSC clocks are locked together and the RNC generates the forward frame timing. The RNC shall therefore not miss transmit schedules. If it proves necessary to slip or stuff in order to ensure the transmission of a full sized CSH PDU this shall be done so on a boundary of 80 octets.

6.5 Discontinuous Transmission (DTX) for 64 kbit/s Service

6.5.1 General Introduction

In order to achieve power/bandwidth efficiencies in 64 kbit/s Circuit Switched calls, discontinuous transmission will be activated for 64 kbit/s CS bearer connections in the Circuit Switched Handler (BCSH) protocol.

When a repeated sequence of bytes is detected, the transmitter stops transmitting on a CSH PDU boundary. The receiver sees the absence of CSH PDUs and generates a local stream until user data is received or the call is cleared.

6.5.2 64 kbit/s DTX - Mandatory and Optional Requirements

The capability to operate with Discontinuous Transmission for the 64 kbit/s service is mandatory for the Radio Access Network (RAN) in both forward and return directions. The capability to operate with Discontinuous Transmission for the 64 kbit/s service in the UE is optional in the return direction and mandatory in the forward direction. Enabling of the mechanism to operate in either forward or return directions or both shall be configurable by the network operator.

No capability to inform the RNC of a UE's return direction DTX capability is assumed. A UE incapable of DTX operation in the return direction shall operate without return direction DTX when commanded to use it by the RNC.

6.5.3 Enabling/Disabling and Selecting DTX algorithm

As previously noted in clause 4.1, the enabling or disabling of forward and return DTX shall be communicated by the RNC to the UE in the RBC: Establish message. The parameter CircuitSwitchedCallType is retrieved from the Call Control layer at the UE. CircuitSwitchedCallType is returned by the UE to the RNC in the RBC: EstablishAck message. In this way the RNC and UE are informed of both the requirement to invoke DTX and the DTX algorithm to be used.

6.5.4 Terminology

The abbreviation DTX shall be equivalent to the term Discontinuous Transmission. The term Carrier Activation is not used in the present document because it implies transmission in a SCPC (Single Channel Per Carrier) channel, which is not applicable in the satellite network.

6.5.5 Choosing the appropriate DTX algorithm

The parameter CircuitSwitchedCallType indicates to BCSH which DTX algorithm to use. The mapping between CircuitSwitchedCallType and DTX algorithm is presented in Table 6.5.

Table 6.5: DTX Algorithm Selection for the BCSH

CircuitSwitchedCallType	DTX Algorithm	Clause
type-3pt1khz-audio	DTX for 3,1 kHz Audio Services - Voice Activated DTX algorithm	6.6.7
type-udi-isdn	DTX for 64 kbit/s UDI Service - Data Activated DTX Algorithm	6.6.6
type-rdi-isdn	None	None

NOTE: CircuitSwitchedCallType = "type-4kbits-speech" invokes the VCSH data handler with its dedicated DTX algorithm, described in clause 5. It is therefore not discussed further here.

The 64 kbit/s circuit switched bearer is used to carry both 3,1 kHz PCM coded audio traffic (e.g. high quality voice or fax) and UDI/RDI Data. The Call Control Bearer Capability identifies the type of data being carried.

Two Discontinuous Transmission algorithms are applicable for 64 kbit/s Circuit Switched services. The appropriate DTX algorithm is selected according to the Bearer Capability signalled in the CC:Setup message to initiate a circuit switched call.

DTX for the 56 kbit/s data service (RDI) is currently not defined. Note that this function would require a higher processing overhead because a bit-oriented protocol would be necessary to identify each 8-bit user data segment from a frame where each octet contains only 7 bits of user data.

6.5.6 DTX for 64 kbit/s UDI Service - Data Activated DTX Algorithm

This mechanism can be described as data activated switching. Data from the bearer is only transmitted during bursts of 64 kbit/s "active" (non idle-flag) user data (where user data not classed as "active" is defined by (i) in the algorithm shown in Figure 6.6) sent from transmitter to receiver. The discontinuous transmission will only occur on calls that use the HDLC (High Level Data Link Control) family of protocols, i.e. where signalled as ITC = UDI. The information transfer capability (ITC) is a parameter in the bearer capability description that is used in the signalling between the terminal and the network to set up a bearer.

Note that the CSH transmit entities are configured to concatenate (ReturnCSFramesPerPDU × 640) bits and (ForwardCSFramesPerPDU × 640) bits per CSH PDU signalled in the RBC Establish Message from the RNC, as appropriate to the direction of transmission.

At the receiver, the algorithm shown in Figure 6.7 will be utilized to analyze the received data and inject idle traffic in the absence of received frames.

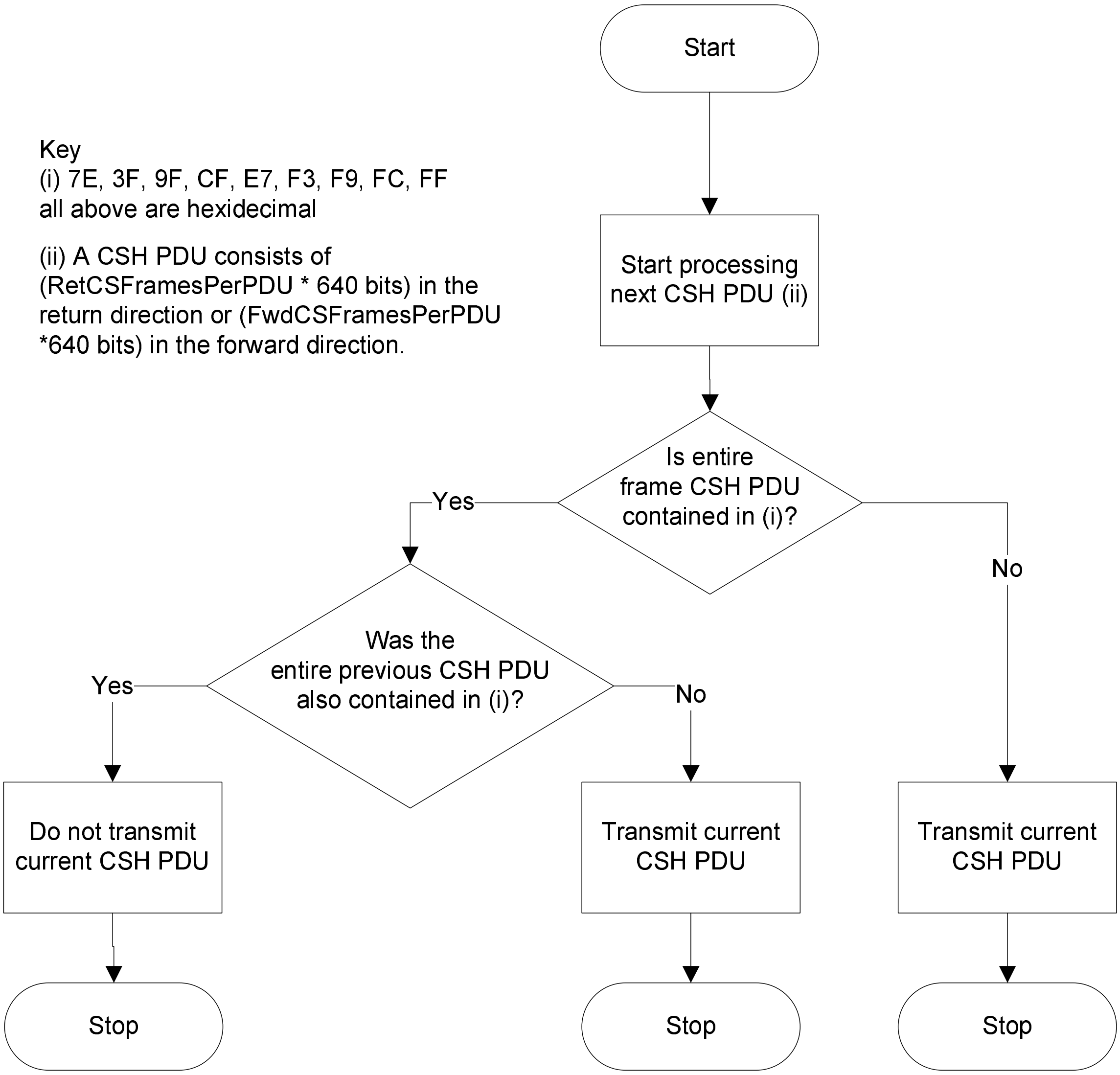

Figure 6.6: Discontinuous Transmission for 64 kbit/s UDI - Transmit Side

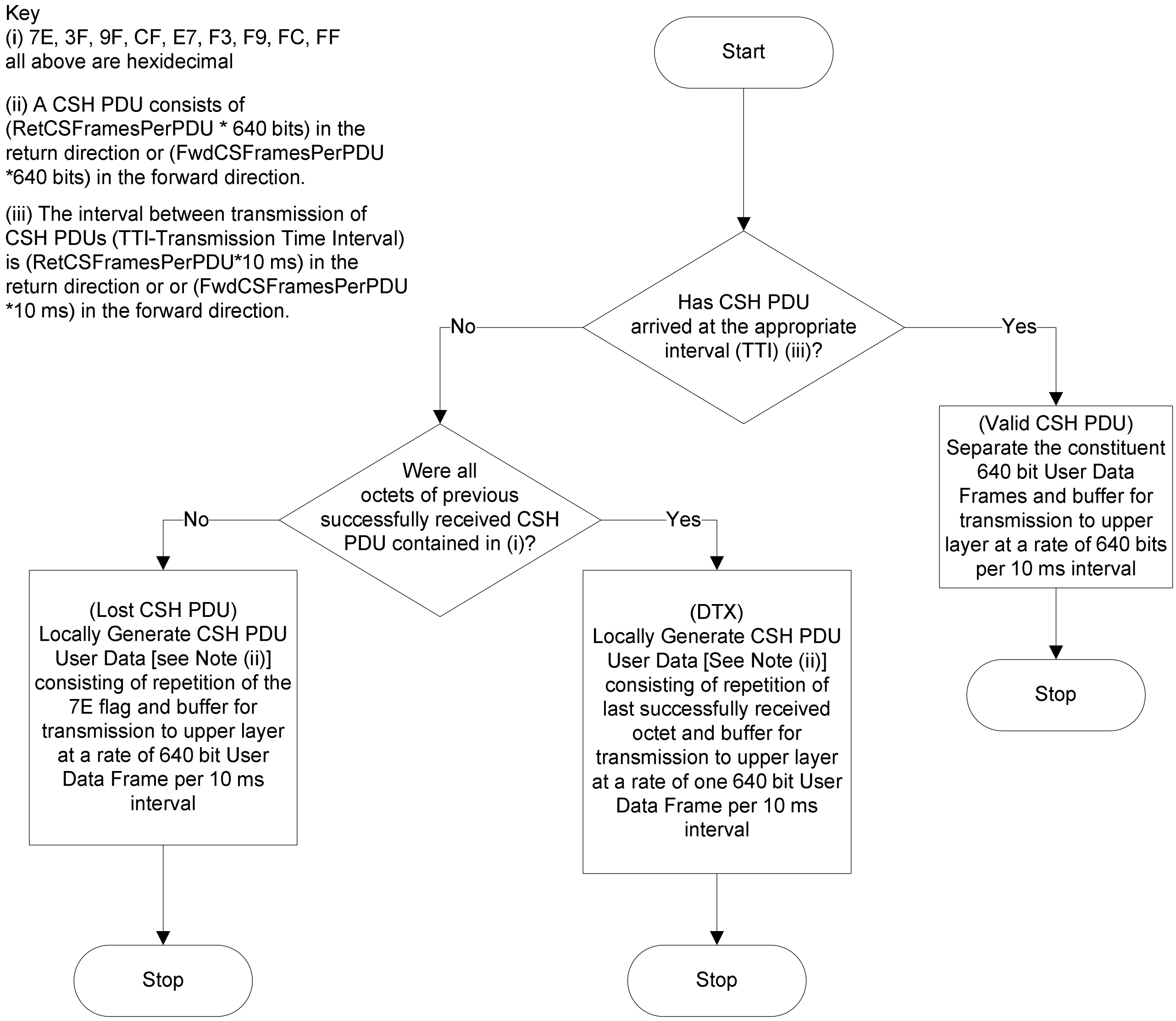

Figure 6.7: Discontinuous Transmission for 64 kbit/s UDI - Receive Side

6.5.7 DTX for 3,1 kHz Audio Services - Voice Activated DTX algorithm

DTX for 3,1 kHz audio services can be described as voice activated switching whereby the forward transmission of voice frames is turned off in the absence of analogue signals. The transmission of voice frames is restored whenever the equivalent analogue level of the PCM signal is above the threshold level X (where X shall be adjustable in the range -25 dBm0 to -45 dBm0 at the RAN for the forward direction and X shall be fixed at a level of -40 dBm at the UE in the return direction) for a period greater than (ReturnCSFramesPerPDU × 10 ms) in the return direction or (ForwardCSFramesPerPDU × 10 ms) in the forward direction (this shall be defined as "active" signal mode). The transmission of voice frames shall be turned off when the PCM signal is below the threshold level X for the same period (this shall be defined as "non-active" signal mode). DTX shall be implemented such that there will be no clipping at the start of signals.

At the receiver the absence of a received CSH-PDU containing 3,1 kHz audio traffic at the expected TTI (transmit time interval) shall result in the local generation of a low level -40 dBm0 (comfort noise) signal for a period of (ReturnCSFramesPerPDU × 10 ms) in the return direction or (ForwardCSFramesPerPDU × 10 ms) in the forward direction

6.5.8 Maintenance Burst for 64 kbit/s DTX (UDI and 3,1 kHz Audio)

In order to ensure that a Bearer Connection is not erroneously cleared by the RNC due to extended periods of DTX , the transmitter shall periodically transmit a CSH-PDU of user data containing (ReturnCSFramesPerPDU × 640 bits) in the return direction. This maintaining burst shall be transmitted at an interval of no greater than 1,5 seconds.

6.5.9 Behaviour for CSH and DTX Algorithm Selection for 64 kbit/s CS Bearer

The RBC:Establish message sent by the RNC to the UE to set up a Radio Bearer contains the CSHConfigurationParam AVP (see ETSI TS 102 744-3-5 [14], clause 3.4.8). This mechanism is common to both circuit switched 64 kbit/s and 4 kbit/s speech.

The RBC:EstablishAck message returned by the UE to the RNC acknowledges the Establish message. In the case of the 64 kbit/s bearer, the UE also uses this message to inform the RNC of the ITC (which is already known to the UE from Call Control) in an Adaptation Layer AVP.

The RNC and UE then configure the appropriate DTX algorithms locally and begin transmission.

7 Packet Switched User Plane

7.0 Overview

A simplified stack diagram showing the satellite network user plane layers is shown in Figure 7.1.

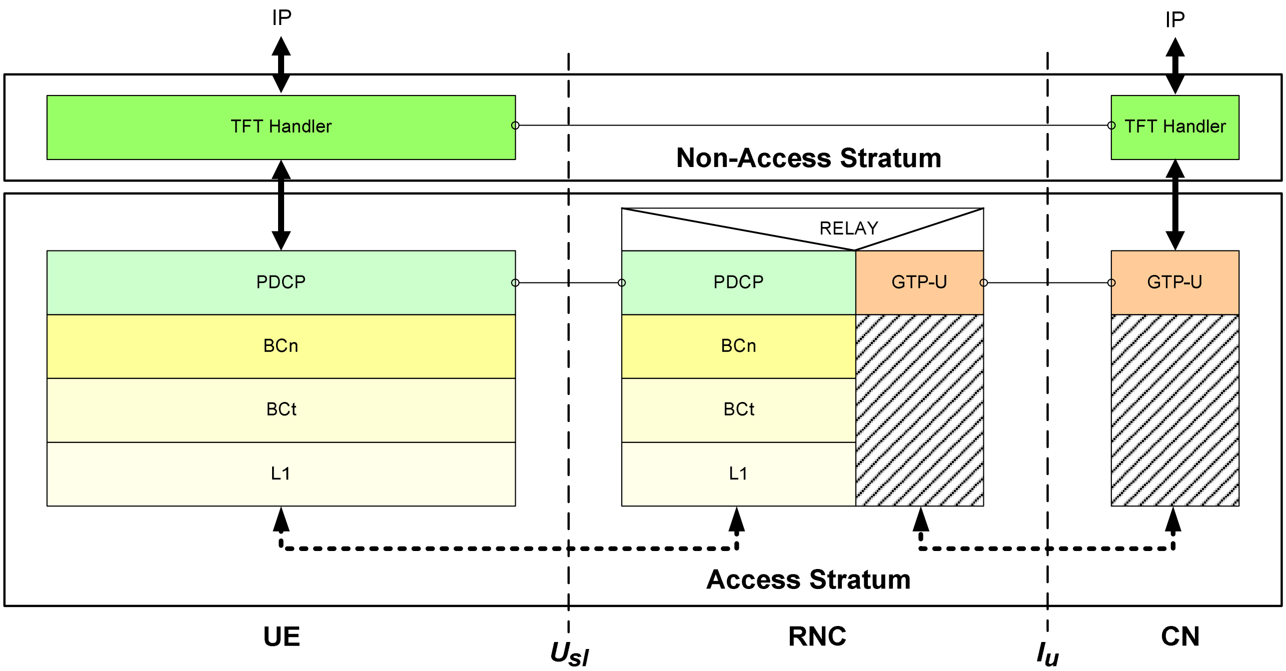

Figure 7.1: User Plane PS Domain

The User Plane behaviour of Bearer Connection and Bearer Control layers is specified in ETSI TS 102 744-3-4 [13] and ETSI TS 102 744-3-2 [12] respectively. The PDCP layer is configured by the Adaptation Layer as specified in ETSI TS 102 744-3-6 [15], and the TFT (Traffic Flow Template) layer is configured by NAS, as described in ETSI TS 124 007 [5] for PS domain and in ETSI TS 102 744-3-8 [16] for the Broadcast/Multicast (BM) domain. The User Plane operation of both the PDCP layer and the TFT layer are described in clauses 7.1 and 7.2 respectively.

7.1 PDCP Protocol

7.1.0 General

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;
- transfer of user data. This function is used for conveyance of data between users of PDCP services;

- maintenance of PDCP sequence numbers for radio bearers that are configured to support lossless SRNS (Serving Radio Network Subsystem) Relocation.

PDCP provides its services to the NAS at the UE or the Relay Function at the Radio Network Controller (RNC).

PDCP directly uses the RLC-like Acknowledged Mode (AM), Unacknowledged Mode (UM), or TM data transport services provided by the Bearer Connection (BCn) Layer. There is no intermediate data handling function and therefore the user plane implementation of PDCP is unchanged from the specifications in [6].

PDCP entities are created and initialized by the Adaptation Layer through a control service access point (CPDCP-SAP). Service Access Point and Primitives for configuring of user plane protocol entities by the Adaptation Layer are in ETSI TS 102 744-3-6 [15].

7.1.1 Disabling of IP Header Compression Per Connection

7.1.1.0 General

A backwards-compatible RNC feature is offered that permits IP header compression to be enabled/disabled per connection at the UE for the purposes of interoperability and troubleshooting. The feature permits support of an AT command option in ETSI TS 127 007 [7] (clause 10.1.1, AT CGDCONT) which cannot otherwise be utilized. The feature permits a UE to enable/disable IP header compression without needing to re-register with a new PDCP capability. The feature also allows IP header compression to be enabled/disabled per connection from the UE.

7.1.1.1 RNC Behaviour (Mandatory)

For a bi-directional AM or UM connection the RNC will not compress IP headers in the forward direction until it has received an IP header compressed PDCP PDU in the uplink direction. When the RNC has not yet started its compressor, it will send IP headers uncompressed with the PDCP header byte having the value 0. The feature is backwards compatible with standard PDCP implementations, since the current UE behaviour is to compress IP headers (if enabled by PDCPInfoParam AVP signalling).

7.1.1.2 UE Behaviour (Optional)

The current and default behaviour will be to compress IP headers in the return direction if enabled by PDCPInfoParam AVP signalling. Optionally, the UE may implement a configuration parameter which forces the UE to send PDCP PDUs with uncompressed IP headers in a specified Packet Data Protocol (PDP) context. As a consequence of the RNC behaviour described above it will not receive compressed IP headers from the RNC in that context.

7.1.2 PDCP Behaviour for operation with BM domain

The PDCP behaviour for operation with the BM domain is simplified in that IP Header compression is disabled in the PDCP layer, and takes place within the TFT layer.

The functionality of the PDCP entity for operation with the BM domain is to map the TFT Filter ID to the PDCP-Info field at the interface to the Bearer Connection Layer. Within the relay function of the RNC, the PDCPinfo parameter is mapped into the optional GTPu header parameter defined for this purpose (reference the Iu_{BM} specification).

7.2 TFT Handler

7.2.1 UE User Plane Protocol Stack Overview

A more detailed presentation of the BM domain-specific user-plane of the UE protocol stack, showing the references that are utilized at each layer is shown in Figure 7.2.

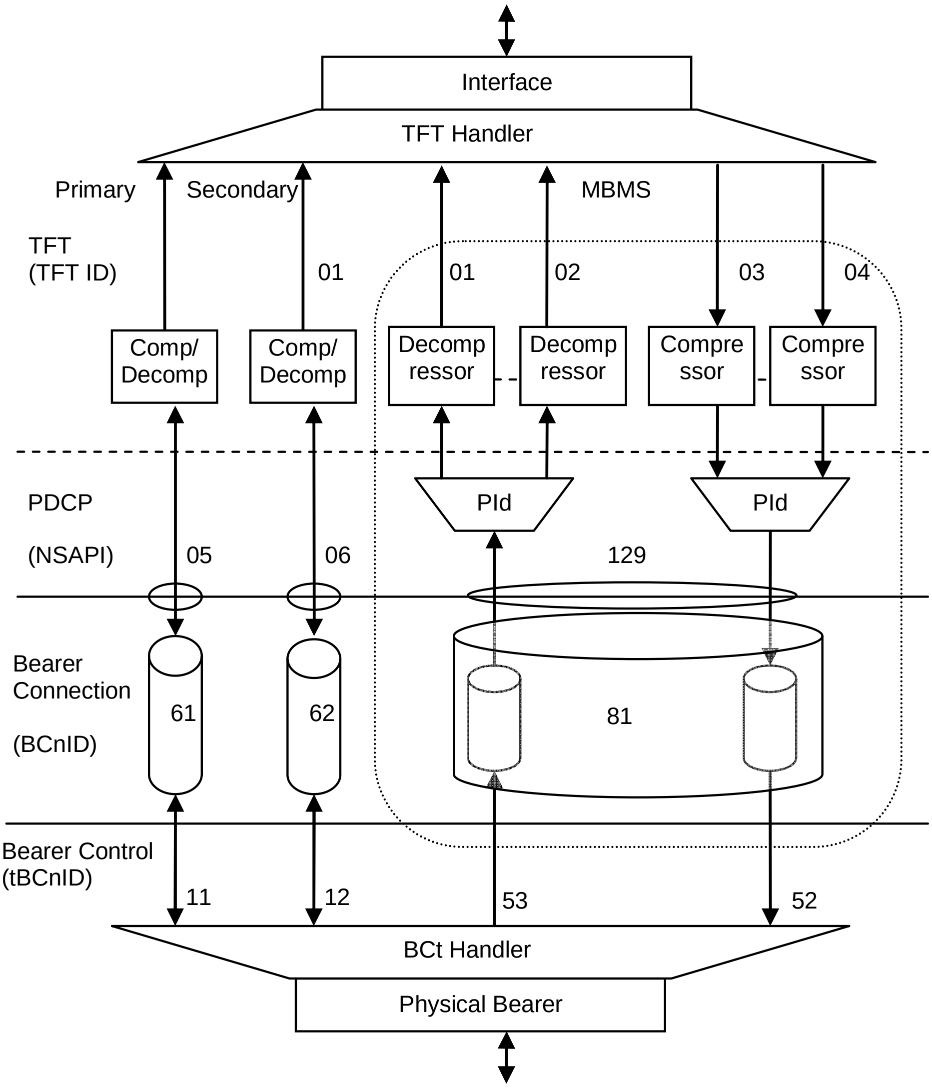

Figure 7.2: User Plane BM Domain - UE Protocol Stack

The TFT Handler, while having generic forwarding functionality in the PS domain, also includes IP header compression functionality. For the primary PDP contexts, the IP header compression is based upon either RFC 2507 [i.1] or RFC 3095 [i.2], based upon capabilities negotiated during PDP Context Activation. For secondary PDP contexts and Multimedia Broadcast Multicast Service (MBMS) bearer contexts, IP header compression and decompression in the TFT layer is initialized with information derived from the Traffic Filter Templates specified during PDP Context Activation. The TFT Filter ID is utilized to identify the particular compressor state information that shall be applied.

7.2.2 TFT Based Compression Approach

7.2.2.0 General

For Secondary PDP Contexts and MBMS contexts, the compressor state for each Traffic Filter Template shall be as described in this clause. It should be noted that in principle this approach could be combined with other compression algorithms such as RFC 2507 [i.1]/RFC 2508 [i.3] or RFC 3095 [i.2].

7.2.2.1 IPv4 operation

7.2.2.1.1 IP Version

The compressor shall delete the IP version. The decompressor shall reinstate the IP version.

7.2.2.1.2 IHL

If the Internet Header Length (IHL) has a value of 4, then it shall be deleted by the compressor, and reinstated by the decompressor. If the value is not equal to 4 then the packet shall be sent uncompressed.

7.2.2.1.3 ToS (/DSCP)

If the value is specified in the TFT then it shall be deleted by the compressor and reinstated by the decompressor using the specified value. Otherwise the ToS field shall be present.

7.2.2.1.4 Total Length

This field is deleted by the compressor and reinstated by the decompressor - it shall be derived from the link layer packet length.

7.2.2.1.5 Identification

This field should normally always be sent. This is defined as DELTA for Protocol = TCP, else RANDOM. TCP shall not be used for Multimedia Broadcast Multicast Services (MBMS).

For multicast services, deletion and reinstatement of Identification should take place for non-fragmented compressed packets.

7.2.2.1.6 Flags/Fragment Offset

If Flags = 0, then Flags and Fragment offset are deleted by the compressor and reinstated by the decompressor. If Flags not equal to zero, then packet is sent uncompressed.

7.2.2.1.7 TTL

TTL (Time To Live) is indicated as being 'NOCHANGE' in RFC 2507 [i.1]. This can be done with a full negotiated compressor state. However deletion and reinstatement of a fixed value (e.g. 255) while considered an option, it is potentially problematic and requires further analysis.

Typically the UE will be at one end of a communications network. It is possible that the TTL value could be set to a relatively low value (e.g. 15) in the downlink decompressor, and to a relatively high value (e.g. 240) in the uplink decompressor.

It is recommended that this asymmetric approach is adopted for UDP traffic, but not for any other type of traffic. Compression of this value should be negotiated. Failing that, the downlink value should be statically configurable in the UE. The uplink value should be statically configurable in the Core Network.

7.2.2.1.8 Protocol

If specified in the TFT, then this shall be deleted by the compressor and reinstated by the decompressor.

7.2.2.1.9 Header Checksum

This shall be deleted by the compressor and recalculated by the decompressor.

7.2.2.1.10 Source Address

In the uplink direction the compressor shall delete the Source Address, and the PDP address shall be reinstated by the Decompressor.

In the downlink direction, if the Remote IP Address Netmask specified in the TFT is 255.255.255.255, then the Source Address shall be deleted by the compressor, and the Remote IP Address as specified shall be reinserted by the decompressor.

For any other value of Remote IP Address Netmask, only the non-static elements of the Source IP address are sent (i.e. the value that is not masked) by the compressor. The decompressor adds the received bits to the base address of the Remote IP Address as specified in the TFT.

7.2.2.1.11 Destination Address

In the downlink direction the compressor shall delete the Source Address. For unicast PDP contexts the decompressor shall reinstate the PDP address. For MBMS contexts, the MBMS address shall be reinstated by the decompressor.

In the uplink direction, if the Remote IP Address Netmask is 255.255.255.255, then the destination IP address shall be deleted by the compressor, and the destination IP address as specified in the TFT shall be reinserted by the decompressor.

For any other value of Remote IP Address Netmask, only the non-static elements of the Destination IP address are sent (i.e. the value that is not masked) by the compressor. The decompressor adds the received bits to the destination IP base address as specified in the TFT.

7.2.2.1.12 Source Port

In the uplink direction, if the Local Port is specified within the TFT as a single value, then this shall be deleted by the compressor and reinstated by the decompressor. If a Local Port Range is specified, then the field width shall be determined by the range value, for instance if the Local Port Range is specified as 5 060 to 5 063, then three bits shall be sent. The compressor shall subtract the base value of the Local Port Range, and take the transmit the least significant bits as determined by the range value. The decompressor shall add the received value to the base value for the Local Port.

In the downlink direction the compressor and decompressor behaviour shall be the same, but shall use Remote Port and Remote Port Range.

7.2.2.1.13 Destination Port

In the uplink direction, if the Remote Port is specified within the TFT as a single value, then this shall be deleted by the compressor and reinstated by the decompressor. If a Remote Port Range is specified, then the field width shall be determined by the range value, for instance if the Remote Port Range is specified as 5 060 to 5 063, then three bits shall be sent. The compressor shall subtract the base value of the Remote Port Range, and take the transmit the least significant bits as determined by the range value. The decompressor shall add the received value to the base value for the Remote Port.

In the downlink direction the compressor and decompressor behaviour shall be the same, but shall use Local Port and Local Port Range.

7.2.2.1.14 UDP Length

If the Protocol Type is specified as UDP, and the packet is not fragmented then this field shall be deleted by the compressor and derived by the decompressor from the link layer length information. If the packet is fragmented it shall not be compressed, and therefore the UDP length shall remain intact.

7.2.2.1.15 UDP CRC

This value shall be deleted by the compressor and recalculated by the decompressor, noting that the packet will be discarded in the link layer if the Media Access Control (MAC) layer CRC or the link layer reassembly mechanism fails.

7.2.2.1.16 SPI

If specified within the TFT, the compressor shall delete the SPI (Security Parameter Index) and the decompressor shall reinstate the SPI.

7.2.2.1.17 Padding

The compressor shall pad the compressed header to an octet boundary using padding bits which shall be zeros. The decompressor shall ignore the padding bits. Note that padding occurs prior immediately before UDP payload.

7.2.2.2 IPv6 operation

7.2.2.2.1 IP Version

The compressor shall delete the IP version. The decompressor shall reinstate the IP version.

7.2.2.2.2 Traffic Class

If specified by the TFT, the compressor shall delete the Traffic Class field, and the decompressor shall reinstate the Traffic Class field.

7.2.2.2.3 Flow Label

If specified by the TFT, the compressor shall delete the Flow Label field, and the decompressor shall reinstate the Flow Label field.

7.2.2.2.4 Payload Length

This field is deleted by the compressor and reinstated by the decompressor - it shall be derived from the link layer packet length.

7.2.2.2.5 Next Header

If equal to the value specified in the TFT, then this field shall be deleted by the compressor and reinstated by the decompressor.

7.2.2.2.6 Hop Limit

This field shall always be sent.

7.2.2.2.7 Source Address

If the source IP address mask is FF::FF, then the source address is deleted by the compressor, and the source address is reinserted by the decompressor.

For any other value of source IP address mask, only the non-static elements of the address are sent (i.e. the value that is not masked) by the compressor. The decompressor adds the received bits to the Source IP base address specified in the TFT.

7.2.2.2.8 Destination Address

If the destination IP address mask is FF::FF, then the destination address is deleted by the compressor, and the destination address is reinserted by the decompressor.

For any other value of destination IP address mask, only the non-static elements of the address are sent (i.e. the value that is not masked) by the compressor. The decompressor adds the received bits to the destination IP base address specified in the TFT.

7.2.2.2.9 Source Port

If the Source Port is specified within the TFT as a single value, then this shall be deleted by the compressor and reinstated by the decompressor. If a source port range is specified, then the field width shall be determined by the range value, for instance if the source port range is specified as 5 060 to 5 063, then three bits shall be sent. The compressor shall subtract the base value of the source port range, and take the transmit the least significant bits as determined by the range value. The decompressor shall add the received value to the base value for the source port.

7.2.2.2.10 Destination Port

If the Destination Port is specified within the TFT as a single value, then this shall be deleted by the compressor and reinstated by the decompressor. If a destination port range is specified, then the field width shall be determined by the range value, for instance if the destination port range is specified as 5 060 to 5 063, then three bits shall be sent. The compressor shall subtract the base value of the destination port range, and take the transmit the least significant bits as determined by the range value. The decompressor shall add the received value to the base value for the destination port.

7.2.2.2.11 UDP Length

If the Protocol Type is specified as UDP, and the packet is not fragmented then this field shall be deleted by the compressor and derived by the decompressor from the link layer length information. If the packet is fragmented it shall not be compressed, and therefore the UDP length shall remain intact.

7.2.2.2.12 UDP CRC

This value shall be deleted by the compressor and recalculated by the decompressor, noting that the packet will be discarded in the link layer if the MAC layer CRC or the link layer reassembly mechanism fails.

7.2.2.2.13 SPI

If specified within the TFT, the compressor shall delete the SPI and the decompressor shall reinstate the SPI.

7.2.2.2.14 Padding

The compressor shall pad the compressed header to an octet boundary using padding bits which shall be zeros. The decompressor shall ignore the padding bits.

Annex A (normative): Circuit Switched Call type - Supported Bearer Capability Information Element

A.1 Introduction

The satellite network access stratum provides support for four general classes of Circuit Switched Call (which are distinguished in the Access Stratum by the parameter CircuitSwitchedCallType). This parameter allows the access stratum to select the appropriate Circuit Switched Data handler (VCSH or BCSH) and DTX algorithm. This annex tabulates the four Circuit Switched Call Classes and their distinguishing Bearer Capability Information Element parameters. Support in the access stratum for other Circuit Switched Call Types (such as Non-Transparent services) specified in UMTS is not incorporated in this release of the satellite network standard.

A.2 4 kbit/s Speech Telephony Service [CircuitSwitchedCallType = 'type-4kbits-speech']

The Bearer Capability Information Element for all speech telephony calls using the 4 kbit/s voice codec shall be coded as Teleservice 11 or 12 (basic telephony or emergency calls) as specified in [8] (clause B.1.8). The signalling to distinguish the 4 kbit/s voice codec shall follow the rules defined in clause 5.4 of the present document.

A.3 PCM Coded 64 kbit/s Service [CircuitSwitchedCallType = 'type-3pt1khz-audio']

The satellite network specific Bearer Capability Information Element for the PCM coded 64 kbit/s service is specified in annex B.

A.4 64 kbit/s UDI Service [CircuitSwitchedCallType = 'type-udi-isdn']

The Bearer Capability Information Element for the transparent 64 kbit/s service signalled with Information Transfer Capability ='UDI' is specified in [8] (clause B.1.3.1.5).

A.5 56 kbit/s RDI Service (also 56 kbit/s UDI with V.110 Rate Adaptation) [CircuitSwitchedCallType = 'type-rdi-isdn']

The Bearer Capability Information Element for the transparent 56 kbit/s service signalled with Information Transfer Capability ='RDI'), or signalled with ITC='UDI' with Rate Adaptation = V.110, Fixed Network User Rate = 56 kbit/s) is specified in [8] (clause B.1.3.1.3).

The following diagram (Figure A.1) illustrates the passage of the parameter CircuitSwitchedCallType for configuring the CSH at the UE and subsequently at the RNC.

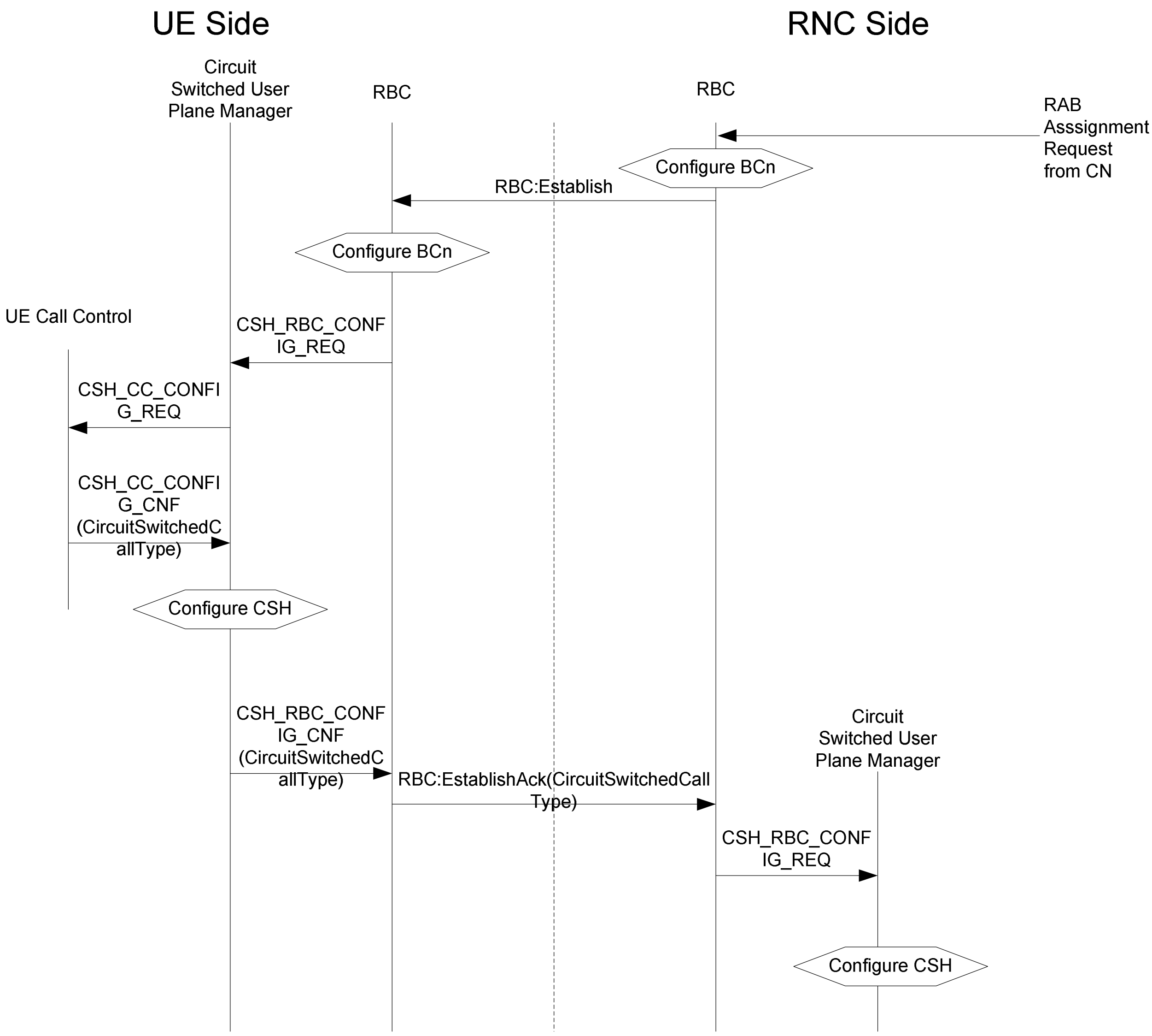

Figure A.1: Configuration of the CSH entity for Circuit Switched Calls with the CircuitSwitchedCallType parameter

Annex B (normative): Circuit Switched Bearer Capability for Satellite Network Specific PCM Coded 64 kbit/s Bearer Service

B.1 Introduction

This annex specifies the bearer capability information to be used in signalling exchanges between the terminal and the network for a satellite network specific 64 kbit/s bearer service providing transmission of PCM coded information such as speech or fax. The 3GPP specifications do not support such a service.

B.2 Support of PCM coded Fax, Data and Speech

Depending upon the user terminal capability a fax machine can be connected either directly to the terminal via a two wire interface or via an ISDN terminal adapter. In the former case the user terminal itself performs the PCM coding. At the fixed network side, the destination fax machine may be connected via the Public Switched Telephone Network (PSTN) or the ISDN and no interworking function is required in the satellite network.

B.3 64 kbit/s Bearer Service in UMTS

In ETSI TS 122 002 [9], a number of tables are provided which describe the set of bearer services supported by the 3GPP specifications. Table B.1 and Table B.2 reproduced here, describe the use of the 64 kbit/s bearer.

Table B.1: BS 30 transparent in regular mode for digital interworking

Fixed Network User Rate	Access Structure	Information Transfer Capability	Rate Adaptation	QoS Attribute	Notes
1,2 kbit/s	Synch	UDI	V.110	T	See note
2,4 kbit/s	Synch	UDI	V.110	T	See note
4,8 kbit/s	Synch	UDI	V.110	T	See note
9,6 kbit/s	Synch	UDI	V.110	T	See note
14,4 kbit/s	Synch	UDI	V.110	T	See note
19,2 kbit/s	Synch	UDI	V.110	T	See note
28,8 kbit/s	Synch	UDI	V.110	T	See note
38,4 kbit/s	Synch	UDI	V.110	T	See note
48 kbit/s	Synch	UDI	V.110	T	See note
56 kbit/s	Synch	UDI	V.110	T	
56 kbit/s	Synch	RDI	-	T	
64 kbit/s	Synch	UDI	-	T	
NOTE: Only applicable in GSM.					

Table B.2: BS 30 transparent for Multimedia

Fixed Network User Rate	Access Structure	Information Transfer Capability	Rate Adaptation	QoS Attribute	Notes
28,8 kbit/s	Synch	3,1 kHz Audio	H.223 & H.245	T	
32,0 kbit/s	Synch	UDI	H.223 & H.245	T	
33,6 kbit/s	Synch	3,1 kHz Audio	H.223 & H.245	T	See note
56 kbit/s	Synch	RDI	H.223 & H.245	T	
64 kbit/s	Synch	UDI	H.223 & H.245	T	
NOTE: 33,6 kbit/s FNURs is applicable only for UTRAN.					

These tables summarize the bearer capability parameter settings that are considered valid for UMTS and GSM. It is clear from these tables that the ITC value is to be set to UDI when the fixed network user rate (FNUR, this the data rate required to be set up in the fixed network) is 64 kbit/s. The UMTS specifications do not anticipate that the 64 kbit/s bearer will be used for any services that require an ITC value of 3,1 kHz.

B.4 Bearer Capability for 64 kbit/s Bearer Service using ITC = 3,1 kHz

In addition to the UMTS bearer services outlined in annex A of ETSI TS 102 744-1-1 [10], user terminals shall support a 64 kbit/s bearer with ITC = 3,1 kHz Audio. The bearer capability information to be used in conjunction with this service is defined in Figure B.1.

The Lower Layer Compatibility Information shall be set as defined in clause B.2.3.2 of ETSI TS 127 001 [8].

The terminal shall provide this bearer capability information to the network in the SETUP message for mobile originated calls. The network shall provide this bearer capability information to the terminal in the SETUP message for mobile terminated calls.

	Oct.											
ITC	3/5a	UDI	3,1 kHz	FAX3	Speech	RDI						
		Layer/protocol related										
SAP	5	NA	I.440									
RA	5/5a	NA	PIAFS	V.120	V.110	H.223 & H.245	X.31 Flag	No				
SA	6	A	S									
CE	6c	NA	NT	bothNT	bothT	T						
S	4	NA	SDU	unstructured								
U1L2P	7	ISO6429	COPnoFlct	NAV								
DC	4	NA	DC	NO								
NDB	6a	NA	7	8								
NPB	6b	NA	odd	even	0	1	none					
NSB	6a	NA	1	2								
		Radio channel related										
RCR	3	dualHR	dualFR	FR								
UR 1)	6a	NA	0,3	1,2	2,4	4,8	9,6					
IR 1)	6b	NA	8	16								
MT	6c	none	V.21	V.22	V.22bis	V.26ter	V.32	auto 1	NAV			
OMT 4)	6d	no other MT	V.34	NAV								
FNUR 1, 4)	6d	9,6	14,4	19,2	28,8	32	33,6	38,4	48	56	64	NAV
ACC 1, 2)	6e	none	4,8	9,6	14,4	NAV 3)						
MaxNumTCH 1)	6e	1	2	3	4	5	6	7	8	NAV 3)		
WAIUR	6f	NA	9,6	14,4	19,2	28,8	38,4	43,2	57,6	NAV		
UIMI	6f	NA	not. req.	upto 1	upto 2	upto 3	upto 4	NAV				
ACC ext. 1, 2)	6g	28,8	32,0	43,2	NAV 3)							
ASYM	6g	no. pref.	u1 biased	d1 biased	NAV							

Figure B.1: Definition of Bearer Capability for 64 kbit/s bearer service with ITC = 3,1 kHz

Annex C (normative): Circuit Switched Signalling of Information Transfer Capability to the External (PSTN/ISDN) Network and Terminal Equipment (TE) Attached to the User Terminal

C.1 Distinguishing between Speech, UDI/RDI and 3,1 kHz Audio

This clause discusses mechanisms to select and signal the different Circuit Switched telecommunication services from a mobile (Terminal Equipment) TE, attached to a UE, to the Core Network, and similarly from external terrestrial networks to the CN.

A TE attached to a UE making circuit switched calls may be connected by one of a variety of interfaces, for example via an ISDN S/T bus, a serial USB, or Bluetooth connection, etc.

A TE connected to the UE via a serial-type interface (USB, RS232, etc.) will typically use the 3GPP AT command set to select the appropriate Circuit Switched bearer type. Messages in [7] are defined to unambiguously select the UMTS bearer/teleservice required and their mappings to UMTS Call Control protocol bearer capability is specified in [8].

A TE attached to a UE via an ISDN S/T bus interface will exchange signalling with the UE using the DSS1 (Q.931) protocol. When requesting or answering a circuit switched call the TE will exchange Bearer Capability information with the UE regarding the call type using the DSS1 signalling. This DSS1 signalling will be mapped to UMTS Call Control signalling at the UE. A conversion between Call Control and DSS1 Bearer Capability and in particular the 'Information Transfer Capability' field is necessary to ensure that the appropriate behaviour is signalled between the UE and TE.

There are differences in how the ITC field is handled in the passage of signalling at call setup between TE, UE, Core Network and external PSTN/ISDN. It is important that this field is handled correctly to allocate the necessary bandwidth and interworking functions. Table C.1 illustrates this mapping for a mobile-originated call.

Table C.1: Mobile Originated Calls - Service/Bearer/Signalling Mapping

Service Required by the Mobile User	Information Transfer Capability signalled by a TE attached to the UE	UMTS Call Control ITC signalled from the UE to the satellite network MSC	Access Stratum bearer Provided	ITC signalled towards ISDN/PSTN
Speech	BC: ITC = Speech (For ISDN devices that only generate ITC = 3,1 kHz audio, *compressed speech could be signalled to the UE either by a dialling modifier or multiple subscriber number, depending on the UE vendor's implementation)*	4 kbit/s Speech Telephony Service [CircuitSwitchedCallType='type-4kbits-speech'] (CC BC: ITC = speech)	low data-rate speech with DTX algorithm	BC: ITC = Speech *(where CN transcodes to/from 64 kbit/s mu-law or A-law PCM towards the fixed network)*
PCM coded 64 kbit/s channel e.g. for analog modem or analog fax or high quality voice	BC: ITC= 3,1 kHz audio	PCM Coded 64 kbit/s Service [CircuitSwitchedCallType = 'type-3pt1khz-audio'] (CC BC: ITC=3,1 kHz audio)	64 kbit/s, no inter-working function with optional voice activated DTX	BC: ITC=3,1 kHz audio *(i.e. indicating PCM transcoding to audio tones is required at terminating end)*
64 kbit/s ISDN unrestricted, 8 kHz integrity	BC: ITC=UDI, UR=64 kbit/s	[CircuitSwitchedCallType='type-udi-isdn'] (CC BC: ITC=UDI, FNUR=64 kbit/s, RA=None) or for 3G-H.324/M: (CC BC: ITC=UDI, FNUR=64 kbit/s, = H.223&H.245)	64 kbit/s, no inter-working function with optional data activated DTX	BC: ITC=UDI
56 kbit/s effective data rate ISDN, restricted, 8 kHz integrity	(BC: ITC=RDI) or (BC=UDI, RA=V.110, UR=56 kbit/s)	[CircuitSwitchedCallType='type-rdi-isdn'] (CC BC: ITC=RDI, FNUR=56 kbit/s, RA=None) or equivalently (CC BC: ITC=UDI, FNUR=56 kbit/s, RA=V.110) or for 3G-H.324/M: (CC BC: ITC=RDI, FNUR=56 kbit/s, = H.223&H.245)	64 kbit/s, no-interworking function, last bit of each data octet set to 1 (ref [i.4]), effective data rate 56 kbit/s	BC: ITC=RDI

For Mobile Terminated calls originating in the ISDN, the calling party can indicate the service they require among the supported ones using the conventional DSS1/ISUP signalling of Bearer Capability (BC), mapped to the appropriate UMTS service at the gateway to the satellite network.

If the incoming call originates from the PSTN or an intervening PSTN network does not carry the ISUP signalling then an alternative mechanism shall be employed to signal the required service.

Two approaches are standardized in UMTS depending on whether the mobile subscriber's USIM subscription supports multi-numbering.

If the subscription does not support multiple MSISDNs for different service type then the subscriber can temporarily configure their UE to a specific CircuitSwitchedCallType for the next incoming call. The message announcing the call from the network to the UE does not specify the service but the UE indicates it in return.

Where the subscription does support multiple MSISDNs for different service types then the destination MSISDN is mapped by the CN to a specific Bearer Service/Teleservice and will initiate a call setup with the required signalling for the service. An example of this mapping is shown in Table C.2.

Table C.2: Mobile Terminating Calls - Service/Bearer/Signalling Mapping

Service Required by the Fixed User	Called Number	ISDN/PSTN	Network Service Provided	UMTS Call Control ITC signalled from the satellite network MSC to the UE	Information Transfer Capability signalled by the UE towards an attached TE
Speech	MSISDN for Voice	BC: ITC = Speech *(where the CN transcodes to/from 64 kbit/s mu-law or A-law PCM towards the fixed network)*	4 kbit/s satellite optimized low data rate voice codec speech with DTX algorithm	4 kbit/s Speech Telephony Service [CircuitSwitchedCallType= 'type-4kbits-speech'] (CC BC: ITC = speech)	BC: ITC = Speech where the UE transcodes to/from low data rate speech to a format appropriate to the attached TE
PCM coded 64 kbit/s channel e.g. for analog modem or analog fax or high quality voice	Alternative MSISDN for Fax/Modem Calls or high quality voice	BC: ITC=3,1 kHz audio *(i.e. indicating PCM transcoding is required at terminating end)*	64 kbit/s, no inter-working function for PCM data with optional voice activated DTX	PCM Coded 64 kbit/s Service [CircuitSwitchedCallType = 'type-3pt1khz-audio'] (CC BC: ITC=3,1 kHz audio)	BC: ITC= 3,1 kHz audio *(i.e. indicating PCM transcoding to audio tones is required at terminating end, either in the UE or TE)*
64 kbit/s ISDN unrestricted, 8 kHz integrity	Alternative MSISDN for 64 kbit/s unrestricted ISDN	(Ignored by MSC when BC inferred from dialled MSISDN) otherwise BC: (ITC=UDI, UR=64 kbit/s)	64 kbit/s, no inter-working function with optional data activated DTX	[CircuitSwitchedCallType= 'type-udi-isdn'] (CC BC: ITC=UDI, FNUR=64 kbit/s, RA=None) or for 3G-H.324/M: (CC BC: ITC=UDI, FNUR=64 kbit/s, = H.223&H.245)	BC: ITC=UDI, UR=64 kbit/s
56 kbit/s effective data rate ISDN, restricted, 8 kHz integrity	Alternative MSISDN for 56 kbit/s ISDN	(Ignored by MSC when BC inferred from dialled MSISDN) otherwise (BC: ITC=RDI) or (BC=UDI, RA=V.110, UR=56 kbit/s)	64 kbit/s, no-interworking function, last bit of each data octet set to 1 (ref [i.4]), effective data rate 56 kbit/s	[CircuitSwitchedCallType= 'type-rdi-isdn'] (CC BC: ITC=RDI, FNUR=56 kbit/s, RA=None) or for 3G-H.324/M: (CC BC: ITC=RDI, FNUR=56 kbit/s, = H.223&H.245)	(BC: ITC=RDI) or (BC=UDI, RA=V.110, UR=56 kbit/s)

Note that 'Speech' calls inter-working with external networks will by default make use of the 4 kbit/s voice codec service described above, where the core network interworking function will provide transcoding for A-law or mu-law PCM at 64 kbit/s facing the terrestrial network.

History

Document history		
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