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Technical Specification

**Digital cellular telecommunications system (Phase 2+);
Universal Mobile Telecommunications System (UMTS);
LTE;
Mandatory speech codec;
Adaptive Multi-Rate (AMR) speech codec;
Interface to Iu, Uu and Nb
(3GPP TS 26.102 version 9.0.0 Release 9)**

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1 Scope

The present document specifies the mapping of the AMR generic frame format (3GPP TS 26.101) to the Iu Interface (3GPP TS 25.415 [7]), the Uu Interface and the Nb Interface (3GPP TS 29.415). It further specifies the mapping of Enhanced Full Rate (GSM_EFR) coded speech and of PCM 64 kBit/s (ITU-T G.711 [9]) coded speech to the Nb Interface in a BICC-based circuit switched core network.

The present document also specifies the mapping of Full Rate (GSM_FR) coded speech and of Half Rate (GSM_HR) coded speech to the Nb Interface in a BICC-based circuit switched core network.

The present document also specifies the transport of the AMR Codec Types, the AMR-WB Codec Types, the GSM_EFR Codec, the GSM_FR Codec, the GSM_HR Codec and the ITU-T G.711 Codec over the A-Interface over IP (3GPP TS 48.002 [11]) and the Nb-Interface in a SIP-I -based circuit switched core network (3GPP TS 23.231 [12]).

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 25.415: "Iu Interface CN-UTRAN User plane Protocols".

[2] 3GPP TS 26.101: "AMR Speech Codec, Frame structure".

[3] 3GPP TS 23.107: "QoS Concept and Architecture".

[4] 3GPP TS 46.051: "Enhanced Full Rate (EFR) speech processing functions; General Description"

[5] 3GPP TS 28.062: "Inband Tandem Free Operation (TFO) of speech codecs; Service description; Stage 3".

[6] 3GPP TS 23.153: "Out of band transcoder control, Stage 2".

[7] 3GPP TS 29.415: "Core Network Nb Interface User Plane Protocols".

[8] ITU-T I.366.2: "AAL type 2 service specific convergence sublayer for trunking".

[9] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[10] 3GPP TS 29.414: "Core Network Nb data transport and transport signalling".

[11] 3GPP TS 48.002: "Base Station System - Mobile-services Switching Centre (BSS - MSC) interface; Interface principles".

[12] 3GPP TS 23.231: "SIP-I based circuit-switched core network; Stage 2".

[13] 3GPP TS 29.007: "General requirements on interworking between the Public Land Mobile Network (PLMN) and the Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN)".

[14] 3GPP TS 26.103: "Speech codec list for GSM and UMTS ".

[15] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)", J. Rosenberg and H. Schulzrinne.

[16] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications", H. Schulzrinne, S. Casner, R. Frederick and V. Jacobson.

[17] IETF RFC 3551 (2003): "RTP Profile for Audio and Video Conferences with Minimal Control", H. Schulzrinne and S. Casner.

[18] void

[19] IETF RFC 4566 (2006): "SDP: Session Description Protocol", M. Handley, V. Jacobson and C. Perkins.

[20] IETF RFC 4733 (2006): "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals", H. Schulzrinne and T. Taylor.

[21] IETF RFC 4867 (2007): "RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs", J. Sjoberg, M. Westerlund, A. Lakaniemi and Q. Xie.

[22] http://www.ietf.org/internet-drafts/draft-westerlund-avt-rtp-gsm-hr-00.txt "RTP Payload Format for GSM-HR".

[23] 3GPP TS 46.010: "Full rate speech; Transcoding".

[24] 3GPP TS 46.020: "Half rate speech; Half rate speech transcoding".

[25] 3GPP TS 46.041: "Half rate speech; Discontinuous Transmission (DTX) for half rate speech traffic channels".

[26] 3GPP TS 48.060: "In-band control of remote transcoders and rate adaptors for full rate traffic channels".

[27] 3GPP TS 48.061: "In band control of remote transcoders and rate adaptors for half rate traffic channels".

[28] 3GPP TS 46.012: "Full rate speech; Comfort noise aspect for full rate speech traffic channels".

[29] 3GPP TS 46.022: "Half rate speech; Comfort noise aspects for half rate speech traffic channels".

[30] 3GPP TS 46.062: "Comfort noise aspects for Enhanced Full Rate (EFR) speech traffic channels".

[31] 3GPP TS 26.093: "Adaptive Multi-Rate (AMR) speech codec; Source controlled rate operation".

[32] 3GPP TS 26.193: "Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Source controlled rate operation".

[33] 3GPP TS 48.008: "Mobile Switching Centre - Base Station System (MSC-BSS) interface".

[34] 3GPP TS 48.103: "Base Station System – Media GateWay (BSS-MGW) interface; User Plane transport mechanism".

[35] 3GPP TS 45.009: "Radio Access Network; Link adaptation"

[36] 3GPP TS 46.060: "EFR Speech Codec; Speech Transcoding Functions"

3 Definitions and abbreviations

3.1 Definitions

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

AMR Generic Frame Interface: this interface transports the AMR IF1 generic frame as defined in 3GPP TS 26.101

3.2 Abbreviations

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

AAL2	ATM Adaptation Layer 2
ACS	Active Codec Set
AMR	Adaptive Multi-Rate
AoIP	A-Interface user plane transport over RTP/UDP/IP
AS	Access Stratum
ATM	Asynchronous Transfer Mode
BFH	Bad Frame Handling
CDMA	Code Division Multiple Access
CMI	Codec Mode Indication
CMR/CMC	Codec Mode Request or Codec Mode Command
CN	Core Network
DRC	Downlink Rate Command
FDD	Frequency Duplex Division
FQC	Frame Quality Classification (Iu Interface)
FQI	Frame Quality Indication (AMR IF1)
GSM	Global System for Mobile communications
ITU-T	International Telecommunication Union – Telecommunication standardisation sector
MGW	Media GateWay
NboIP	Nb-Interface user plane transport over RTP/UDP/IP when SIP-I is used on Nc
PCM	Pulse Code Modulation, synonym for 64 kBit/s coded speech (see ITU-T G.711 [9])
PDC	Personal Digital Communication
PLMN	Public Land Mobile Network
QoS	Quality of Service
RAB	Radio Access Bearer
RAN	Radio Access Network
RF	Radio Frequency
RFC	RAB sub-flow Combination
RFCI	RFC Indicator
RFCS	RFC Set
RX	Receive
SCR	Source Controlled Rate
SDU	Source Data Unit
SID	Silence Insertion Descriptor
SMpSDU	Support Mode for Predefined SDU sizes
SPD	SPeech Decoder
SPE	SPeech Encoder
TC	Transcoder
TDD	Time Duplex Division
TDMA	Time Division Multiple Access
TFO	Tandem Free Operation
TrFO	Transcoder Free Operation
TX	Transmit
UE	User Equipment (terminal)
URC	Uplink Rate Command

4 General

The Iu-Interface is defined in two different variants for speech telephony,

a) for the ATM bearer with Iu-framing and
b) for the IP bearer with Iu-framing.

The Nb-Interface is defined in three different variants for speech telephony,

a) for the ATM bearer with Nb-framing in a BICC-based Core Network,
b) for the IP bearer with Nb-framing in a BICC-based Core Network and
c) for the IP bearer with RTP packetization in a SIP-I -based Core Network, also called NboIP.

The mapping of the AMR Speech Codec parameters to the Iu interface specifies the frame structure of the speech data exchanged between the RNC and the TC inside the MGW in case of normal operation. This mapping is independent from the radio interface in the sense that it has the same structure for both FDD and TDD modes of the UTRAN.

The mapping between the Speech Codec and the Radio Access Network within the UE is not an open interface and need not to be detailed.

The mapping on the Nb Interface in a BICC based Core Network is identical to the one on the Iu Interface in case of Transcoder Free Operation, with the MGW relaying the SDUs unaltered between Iu and Nb Interfaces.

PCM coded speech is mapped onto the Nb-Interface in packets of 40 octets (5ms packetization time) or 160 octets (20ms packetization time). With Nb-framing (i.e. in a BICC-based Circuit Switched Core Network, IP or ATM) the default packetization time for PCM-coded speech is 5ms; 20ms is an additional option. For NboIP (i.e. RTP packetization in a SIP-I -based Circuit Switched Core Network) the default packetization time for PCM-coded speech is 20ms; 5ms is an additional option.

The packetization time of PCM-coded speech for AoIP is 20ms without any other option.

For the 3GPP Codec Types (GSM_FR, GSM_HR, GSM_EFR, AMR and AMR-WB) the framing is always 20ms and also the packetization time is 20ms in all versions of the Nb-Interface and the A-Interface over IP. The mapping of GSM_FR, GSM_HR and GSM_EFR Speech Codec parameters is defined on the A Interface over IP and all versions of the Nb-Interface, but not on the Iu Interface.

5 RAB aspects

During the RAB Assignment procedure initiated by the CN to establish the RAB for AMR, the RAB parameters are defined. The AMR RAB is established with one or more RAB co-ordinated sub-flows with predefined sizes and QoS parameters. In this way, each RAB sub-flow Combination corresponds to one AMR frame type. For AMR, the first RAB sub-flow (sub-flow 1) corresponds with the Class A bits. In case there are three RAB sub-flows, the third RAB sub-flow (sub-flow 3) corresponds with the Class C bits. On the Iu interface, these RAB parameters define the corresponding parameters regarding the transport of AMR frames.

Some of the QoS parameters in the RAB assignment procedure are determined from the Bearer Capability Information Element used at call set up. These QoS parameters as defined in [3], can be set as follows:

Table 5-1: Example of mapping of BC IE into QoS parameters for UMTS AMR

RAB service attribute	RAB service attribute value			Comments
Traffic Class	Conversational			
RAB Asymmetry Indicator	Symmetric, bidirectional			Symmetric RABs are used for uplink and downlink
Maximum bit rate	12.2 / 10.2 / 7.95 / 7.4 / 6.7 / 5.9 / 5.15 / 4.75 kbit/s			This value depends on the highest mode rate in the RFCS
Guaranteed bit rate	12.2 / 10.2 / 7.95 / 7.4 / 6.7 / 5.9 / 5.15 / 4.75 kbit/s			One of the values is chosen, depending on the lowest rate controllable SDU format (note 2)
Delivery Order	Yes			(note 1)
Maximum SDU size	244 / 204 / 159 / 148 / 134 / 118 / 103 / 95 bits			Maximum size of payload field in Iu UP, according to the highest mode rate in the RFCS
Traffic Handling Priority	Not applicable			Parameter not applicable for the conversational traffic class. (note 1)
Source statistics descriptor	Speech			(note 1)
SDU Parameters	RAB sub-flow 1 (Class A bits)	RAB sub-flow 2 (Class B bits)	RAB sub-flow 3 (Class C bits)	The number of SDU, their number of RAB sub-flow and their relative sub-flow size is subject to operator tuning (note 3)
SDU error ratio	$7 * 10^{-3}$	-	-	(note 3)
Residual bit error ratio	10^{-6}	10^{-3}	$5 * 10^{-3}$	(note 3 – applicable for every sub-flow)
Delivery of erroneous SDUs	yes	-	-	Class A bits are delivered with error indication; Class B and C bits are delivered without any error indication.
SDU format information 1-9				(note 4)
Sub-flow SDU size 1-9	(note 5)	(note 5)	(note 5)	

NOTE 1: These parameters apply to all UMTS speech codec types.
NOTE 2: The guaranteed bit rate depends on the periodicity and the lowest rate controllable SDU size.
NOTE 3: These parameters are subject to operator tuning.
NOTE 4: SDU format information has to be specified for each AMR core frame type (i.e. with speech bits and comfort noise bits) included in the RFCS as defined in [2].
NOTE 5: The sub-flow SDU size corresponding to an AMR core frame type indicates the number of bits in the class A, class B and class C fields. The assigned SDU sizes shall be set so that the SCR operation is always possible.

The RAB parameters shall be set so that the SCR operation is always possible.
The conversational traffic class shall be used for the speech service, which is identified by the ITC parameter of the bearer capability information element in the SETUP message. This shall apply for all UMTS speech codec types.
The parameters traffic class, transfer delay, traffic handling priority and source statistics descriptor shall be the same for all speech codec types applicable for UMTS.

6 Iu Interface User Plane (RAN)

The data structure exchanged on the Iu interface are symmetrical, i.e. the structure of the uplink data frames is identical to that of the downlink data frames.

6.1 Frame structure on the Iu UP transport protocol

6.1.1 Initialisation

At the initialisation of the SMpSDU mode of operation, several parameters are set by the CN. The initialisation procedure is described in [1].

- RFCS:

  In the case of AMR, the RFCS corresponds to the Active Codec Set (ACS) plus potentially SCR authorised in the communication. Annex A of [1] gives an illustration of the usage of RFCI for AMR speech RAB. RFCS used in downlink may differ from that in uplink.

- Delivery of erroneous SDUs:

  This parameter shall be set to YES. Erroneous speech frames may be used to assist the error concealment procedures. Therefore, according to [1], PDU type 0 (containing a payload CRC) shall be used for transport of AMR data.

6.1.2 Time Alignment Procedure

The TC should adjust the timing of the speech data transmission in downlink direction according to the time alignment frames sent by the RNC.

Time alignment procedure shall be dismissed in case of TFO and TrFO.

6.2 Mapping of the bits

The mapping of the bits between the generic AMR frames and the PDU is the same for both uplink and downlink frames.

The following table gives the correspondence of the bit fields between the generic AMR frames at the TC interface and the PDU exchanged with the Iu transport layer.

Table 6-1: Mapping of generic AMR frames onto Iu PDUs

PDU field	Corresponding field within the generic AMR frame	Comment
PDU Type	N/A	Type 0
Frame Number	N/A	
FQC	Frame Quality Indicator	
RFCI	Frame Type	
Payload CRC	N/A	
Header CRC	N/A	
Payload Fields (N Sub-flows)	Class A or SID payload Class B Class C	
SDU #1	Most important speech bits come first	Mandatory
SDU #2	Next bits follow	Optional
…	…	Optional
SDU #N	Least important speech bits	Optional

The number of RAB sub-flows, their corresponding sizes, and their attributes such as "Delivery of erroneous SDUs" shall be defined at the RAB establishment and signalled in the RANAP RAB establishment request, as proposed in clause 5. The number of RAB sub-flows is corresponding to the desired bit protection classes. The total number of bits in all sub-flows for one RFC shall correspond to the total number given in 3GPP TS 26.101, generic AMR frame, format IF1, for the corresponding Codec Mode, respectively Frame Type.

Guidance for setting the number of bits in each RAB sub-flow according to their relative subjective importance is given in 3GPP TS 26.101.

The following two tables are examples of mapping of RAB sub-flows.

Table 6-2 gives three examples of sub-flow mapping.
The RFCI definition is given in order of increasing SDU sizes.

- Example 1 describes Codec Type UMTS_AMR, with all eight codec modes foreseen in the Active Codec Set (ACS) and provision for Source Controlled Rate operation (SCR). In this example, Blind Transport Format Detection is supported and the sub-flow mapping follows the 26.101 class division guidance.

- Example 2 describes Codec Type GSM_EFR, with one codec mode, including SCR.

- Example 3 describes Codec Type FR_AMR, including AMR SCR

Table 6-2: Example for AMR with SCR and three sub-flows, according to subjective class division indication of 3GPP TS 26.101

UMTS_AMR	GSM_EFR	FR_AMR	RAB sub-flows			Total size of bits/RAB sub-flows combination (Mandatory)	Source rate
RFCI Example 1	RFCI Example 2	RFCI Example 3	RAB sub-flow 1 (Optional)	RAB sub-flow 2 (Optional)	RAB sub-flow 3 (Optional)		
2		**2**	**42**	**53**	**0**	95	AMR 4,75 kbps
3			**49**	**54**	**0**	103	AMR 5,15 kbps
4		**3**	**55**	**63**	**0**	118	AMR 5,9 kbps
5		**4**	**58**	**76**	**0**	134	AMR 6,7 kbps
6			**61**	**87**	**0**	148	AMR 7,4 kbps
7			**75**	**84**	**0**	159	AMR 7,95 kbps
8		**5**	**65**	**99**	**40**	204	AMR 10,2 kbps
9	**2**		**81**	**103**	**60**	244	AMR 12,2 kbps
1		**1**	**39**	**0**	**0**	39	AMR SID
	1		**43**	**0**	**0**	43	GSM-EFR SID

Table 6-3 gives one example of sub-flow mapping that supports Equal Error Protection.
The RFCI definition is given in order of increasing SDU sizes.

- Example 4 describes Codec Type PDC_EFR and the corresponding Source Controlled Rate operation (SCR).

Table 6-3: Example of SDU sizes for PDC_EFR with SCR and Equal Error Protection

PDC_EFR	RAB sub-flow	Total size of bits/RAB sub-flows combination (Mandatory)	Source rate
RFCI Example 4	RAB sub-Flow 1 (Mandatory)		
	95	95	AMR 4,75kbps
	103	103	AMR 5,15kbps
	118	118	AMR 5,9kbps
2	134	134	AMR 6,7kbps
	148	148	AMR 7,4kbps
	159	159	AMR 7,95kbps
	204	204	AMR 10,2kbps
	244	244	AMR 12,2kbps
	39	39	AMR SID
	43	43	GSM-EFR SID
	38	38	TDMA-EFR SID
1	37	37	PDC-EFR SID

6.3 Frame handlers

Iu PDU Frame handling functions are described in 3GPP TS 25.415 [1]. This sections describe the mandatory frame handling functions at the AMR Generic frame interface.

6.3.1 Handling of frames from TC to Iu interface (downlink)

The frames from the TC in generic AMR frame format IF1 are mapped onto the Iu PDU as follows.

6.3.1.1 Frame Quality Indicator

The Frame Quality Indicator (FQI) from the TC is directly mapped to the Frame Quality Classification (FQC) of the Iu frame according to Table 6-4.

Table 6-4: FQI AMR to FQC Iu PDU mapping

FQI AMR	FQI value (1 bit)	FQC PDU	FQC value (2 bit)
GOOD	1	GOOD	00
BAD	0	BAD	01

6.3.1.2 Frame Type

The received Frame Type Index l is mapped onto the RFCI j thanks to the assigned RFCS table: the correspondence between Codec Mode, Frame Type Index l and RFCI j is defined at RAB assignment.

6.3.1.3 Codec Mode Indication

The Codec Mode Indication is not used.

6.3.1.4 Codec Mode Request

Codec Mode Request (CMR) in downlink direction is forwarded to the rate control procedure when it changes, or when it is commanded so by the TC in case of TFO, see 3G TS 28.062.

6.3.1.5 Optional internal 8 bits CRC

The internal AMR Codec CRC is not used on the Iu interface.

6.3.1.6 Mapping of Speech or Comfort Noise parameter bits

Let us define the N payload fields of the N sub-flows for RFCI j as follows:

$U_i(k)$ shall be the bits in sub-flow i, for k =1 to Mi

M_i shall be the size of sub-flow i, for i = 1 to N

d(k) shall be the bits of the speech or comfort noise parameters of the corresponding Frame Type l in decreasing subjective importance, as defined in the generic AMR frame format IF1, see TS 26.101 [2].

Then the following mapping in pseudo code applies:

$U_1(k)$ = d(k-1) with k = 1, … M_1

$U_2(k)$ = $d(k -1+ M_1)$ with k = 1, … M_2

$U_3(k)$ = $d(k-1 + M_2)$ with k = 1, … M_3

…

$U_N(k)$ = $d(k-1 + M_{N-1})$ with k = 1 … M_N

6.3.2 Handling of frames from Iu interface to TC (uplink)

The uplink Iu frames are mapped onto generic AMR frames, format IF1, as follows.

6.3.2.1 Frame Quality Indicator

At reception of Iu PDU the Iu frame handler function set the Frame Quality Classification according to the received FQC, Header-CRC check, and Payload-CRC check (see 25.415). AMR Frame Type and Frame Quality Indicator are determined according to the following table:

Table 6-5: FQC Iu PDU type 0 to AMR FQI and AMR Frame Type mapping

FQC	**FQC value** (2 bits)	**Resulting FQI**	**FQI value** (1 bit)	**resulting Frame Type**
GOOD	00	GOOD	1	from RFCI
BAD	01	BAD	0	NO_DATA
BAD Radio	10	BAD	0	from RFCI
Reserved	11	BAD	0	Reserved

6.3.2.2 Frame Type

The received RFCI j is mapped onto the Frame Type Index l thanks to the RFCS table.

6.3.2.3 Codec Mode Indication

The Codec Mode Indication is not used.

6.3.2.4 Codec Mode Request

The received Downlink Rate Control command (DRC) is mapped onto the Codec Mode Request (CMR) towards the AMR Codec. In case a new DRC is received it is mapped into the corresponding CMR of the generic AMR frame format. It is remembered by the TC until the next DRC is received. In each new frame that is sent to the AMR Codec, the stored CMR is resent, in order to control the Codec Mode for the downlink direction.

6.3.2.5 Optional internal 8 bits CRC

The internal AMR Codec CRC is not used on the Iu interface.

6.3.2.6 Speech and Comfort noise parameter bits

The speech and Comfort noise parameter bits are mapped from the sub-flows to the payload of the generic AMR frames with the reverse function of clause 6.3.1.6.

7 Uu Interface User Plane (UE)

The interface between the UE AMR speech codec (see 3GPP TS 26.101) and the Radio Access Network is an internal UE interface and is not detailed. The mapping is corresponding to the mapping described in clause 6 for the Iu interface.

Even though the details of Uu interface are not detailed, there are some functional requirements for the UE that need to be considered, when an AMR codec type (i.e. UMTS AMR2) is being used in a conversational speech call. These requirements are related to the mapping of AMR Generic frame format handling functions. The requirements are

1. The set of available codec modes (bitrates) that the UE may use are configured by UTRAN. The UE shall select, from the configured set of codec modes, a mode that is supported by the current TX power conditions as defined in 3GPP TS25.133. The highest available mode should be used for best speech quality.

2. The lowest configured codec mode is always to be considered supported.

3. When the codec mode is being adapted during a call, the used mode should be changed in a step-by-step fashion within the configured set of codec modes, i.e. by stepping one mode up or down within the configured set. This avoids disruptions on AMR decoding in GSM side, if TFO or TrFO operation is ongoing.

8 Nb Interface User Plane (CN) of a BICC-based Circuit Switched Core Network

The data structures exchanged on the Nb interface are symmetrical, i.e. the structures of the sent and received data frames are identical.

The Nb-Interface is defined in a BICC-based Core Network in two different variants, a) for the ATM bearer with Nb-framing, b) for the IP bearer with Nb-framing. The Nb-framing and the use of PDU Type 0 for the speech payload and PDU Type 14 [7] for AMR Rate Control is common for both versions of the Nb-Interface here. These two versions also share the principle of "Nb_Init", where the Nb-Interface is initialized on User Plane level and where the Initial Codec Mode for AMR and/or AMR-WB is signalled.

8.1 Frame structure on the Nb UP transport protocol

Delivery of erroneous SDUs for AMR- and AMR-WB-coded speech, as well as for GSM_FR-, GSM_HR-, GSM_EFR-coded speech and for PCM-coded speech on the Nb-Interface shall be set to: "YES" in a BICC-based Circuit Switched Core Network. Erroneous speech frames may be used to assist the error concealment procedures. Therefore, according to [1] and [7], PDU Type 0 (with payload CRC) shall be used for the transport of AMR, AMR-WB, GSM_FR, GSM_HR and GSM_EFR coded speech on the Nb interface.
PDU Type 0 (with payload CRC) shall be used for the transport of PCM coded speech on the Nb interface, too.

8.1.1 Initialisation

The initialisation procedure is used for support mode. At the initialisation several parameters are set by the CN. The initialisation procedure for the Nb Interface is described in [7].

8.1.2 Time Alignment Procedure

The handling of Time Alignment on the Nb Interface is described in [7].

The Time alignment procedure shall be dismissed in case of TFO and TrFO.

8.1.3 SID Frame Generation

All 3GPP Codec Types include a standardized Discontinuous Transmission (DTX) with Voice Activity Detection, Silence Description (by SID frames) and Comfort Noise Generation to fill the speech pauses. If speech inactivity is detected by the Encoder, then (speech) frames are not transmitted, but the transmission is suspended in order to save battery life time in the mobile station, reduce interference on the radio interface and reduce load on all links. The receiving Decoder fills these transmission pauses with Comfort Noise to minimize the contrast between pauses and active speech. Silence Descriptor (SID) frames need to be send during speech inactivity to keep the Comfort Noise decently well aligned with the background noise at sender side. This is especially important at the onset of the next talkspurt and therefore SID frames should not be too old, when speech starts again.

The generation of SID frames for the AMR and AMR-WB families of Codecs is determined by the Speech Encoder as specified in TS 26.093 [31], respectively TS 26.193 [32]. The radio subsystem does not influence this timing! SID frames come during speech pauses in uplink and downlink about every 160ms. Also an AMR Encoder in the Media Gateway generates and sends SID frames about every 160ms.

The generation of SID frames for GSM_FR, GSM_HR and GSM_EFR in the GSM radio network is determined by the GSM mobile station and the GSM radio subsystem, not primarily by the Speech Encoder! SID frames come during speech pauses in uplink from the mobile station about every 480ms. In downlink to the mobile station, when they are generated by the Speech Encoder of the GSM radio subsystem, SID frames are sent every 20ms to the GSM base station, which then picks only one every 480ms for downlink radio transmission. For other applications, like transport over Nb, it is more appropriate to send the SID frames less often than every 20ms, but 480ms may be too sparse. As a compromise it is recommended that an Encoder in the Media Gateway should generate and send SID frames every 160ms.

8.2 Mapping of the bits

8.2.1 Mapping for AMR frames

The mapping of the bits between the generic AMR frames and the PDU for the Nb Interface is identical to the mapping on the Iu Interface. In case of TrFO the MGW relays the AMR frames from the Iu Interface unaltered to the Nb Interface and vice versa, as described in [7].

8.2.2 Mapping for PCM Coded Speech

The mapping for the PCM coded speech in 5ms frames on the Nb Interface shall be as defined in Table 8-1.

Table 8-1: Mapping of PCM Coded Speech in 5 ms frames onto Nb PDU, Type 0

PDU field	Comment
PDU Type	Type 0 (with Payload CRC)
Frame Number	as defined in [7]
FQC	set to "good"
RFCI	initialise by MGW, see [7], one value required
Header CRC	as defined in [7]
Payload CRC	as defined in [7]
Payload Field	320 bits of PCM coded speech, in accordance with [8].

The mapping for the PCM coded speech in 20ms frames on the Nb Interface shall be as defined in Table 8-2.

Table 8-2: Mapping of PCM Coded Speech in 20ms frames onto Nb PDU, Type 0

PDU field	Comment
PDU Type	Type 0 (with Payload CRC)
Frame Number	as defined in [7]
FQC	set to "good"
RFCI	initialised by MGW, see [7], one value required
Header CRC	as defined in [7]
Payload CRC	as defined in [7]
Payload Field	4x320 bits of PCM coded speech, in accordance with [8].

5ms is the default packetization time to be supported for PCM encoded speech over Nb in a BICC based Core Network. 20ms is an additional optional packetization time for PCM encoded speech in a BICC based Core Network over IP Nb bearer that may be negotiated during bearer establishment as specified in [10].

NOTE: the use of 20ms packetization time will result in some call scenarios in higher delays over the speech path compared to the 5ms packetization time. This potentially higher delay should be taken into account in the overall end to end (ear to mouth) delay budget.

8.2.3 Mapping for GSM_EFR frames

The mapping of the bits between the generic GSM_EFR frames and the PDUs for the Nb Interface follows the same principles as the mapping of AMR frames. The PDU for the GSM_EFR speech frame is identical to the PDU for AMR Mode 12.2 kbps.

The PDU for the GSM_EFR SID frame is similar to the PDU for AMR SID, with 43 instead of 39 bits in the payload field. The contents of GSM_EFR SID is the Comfort Noise Parameter set (s(i)) as defined in [36]. The Comfort noise parameters are computed as described in [30] by the GSM_EFR speech encoder and are denoted as s(i) = {s(1),s(2),...,s(38), s(87),s(88),…,s(91)}. The notation $s(i)$ follows that of [36] (Table 6). The notation $d(j)$ = {d(1) … d(43)} of the SID frame is local to the present document and is formed as defined by the pseudo code below.

for j = 1 to 38
    $d(j) := s(j)$; /* LSP parameters in s(1) to s(38) */;

for j = 39 to 43
    $d(j) := s(j+48)$; /* fixed codebook gain parameter in s(87)-s(91) */

The payload within the PDU shall be the vector *d(j)* constructed above. The first bit in the vector *d(j)* shall be supplied first in the payload within the PDU.

NOTE: The Payload field for Nb frames for GSM_EFR in a BICC-based Circuit Switched Core Network is filled differently to the Payload field in RTP Packets according to RFC3551 [17], used in AoIP and NboIP.

8.2.4 Mapping for GSM_FR frames

The mapping of GSM_FR-coded speech in 20ms frames on the Nb Interface shall be as defined in Table 8.2.4.1.

Table 8.2.4.1: Mapping of GSM_FR-coded speech in 20ms frames onto Nb PDU, Type 0

PDU field	Comment
PDU Type	Type 0 (with Payload CRC)
Frame Number	as defined in [7]
FQC	see below
RFCI	initialise by MGW, see [7], two values required (Speech and SID)
Header CRC	as defined in [7]
Payload CRC	as defined in [7]
Payload Field	4 bits for "signature", 260 bits for Speech frames and 42 bits for SID frames, see below

Payload field:
The 260 bits of GSM_FR-coded speech (b1…b260) are defined in TS 46.010, chapter 1.7. They and a "signature" are copied into the 33 octets of the Payload field as follows. The four most significant bits (bit 8….5) of the first octet (octet 1) of the Nb Payload field are set to a "signature" of 0b1101 = 0xD. Then the four most significant bits (b6…b3) of the first GSM_FR parameter (LAR 1) are copied into the next bits (bit 4…1) of the first octet. The two least significant bits of the first GSM_FR parameter (LAR 1) are copied into the next octet (octet 2) into the 2 MSBs (bit 8…7), and so on. Each GSM_FR parameter is copied bit by bit with its most significant bit first. The least significant bit of the last GSM_FR parameter (b258 of RPE-pulse no.13) is placed in the LSB (bit 1) of octet 33.

The GSM_FR SID frames are defined in chapter 5.2 of [28] and in chapter 1.7 of [23] and are denoted as b(i) = {b(1),b(2),...,b(36), b(48),b(49),...,b(53)}. Each GSM_FR SID parameter is copied bit by bit with its most significant bit first. The notation $d(j)$ = {d(1) … d(42)} of the SID frame is local to the present document and is formed by the pseudo code below.

for j = 1 to 36
  $d(j) := b(j)$; /* averaged log area coefficients in $b(1)$ to $b(36)$ */;

for j = 37 to 42
  $d(j) := b(j+11)$; /* averaged block amplitude values in $b(48)$-$b(53)$ */

The payload within the PDU shall be the vector *d(j)* constructed above. The first bit in the vector *d(j)* shall be supplied first in the payload within the PDU.

NOTE: The Payload field for Nb frames for GSM_FR in a BICC-based Circuit Switched Core Network is filled differently to the Payload field in RTP Packets according to RFC3551 [17], used in AoIP and NboIP.

The F**QC bit** is set by the MGW depending on the call case:

1. FQC is set to "good", if the GSM_FR-compression and coding is performed within the MGW
2. FQC is set to "good", if GSM_FR-coded speech is received without frame quality indication
3. FQC is derived from the input frame, if FQC or a similar frame quality indication is specified there.
   In case of GSM_FR-coded speech received via TFO frames (see TS 28.062 [5]) the FQC bit is derived from the "Bad Frame Indication" (BFI) of these TFO frames. Speech frames and SID frames marked with BFI set to "good" shall be sent with FQC set to "good". Speech frames and SID frames marked with BFI set to "bad" shall not be sent in order to save bandwidth on Nb.

8.2.5 Mapping for GSM_HR frames

The mapping of GSM_HR-coded speech in 20ms frames on the Nb Interface shall be as defined in Table 8.2.5.1.

Table 8.2.5.1: Mapping of GSM_HR-coded speech in 20ms frames onto Nb PDU, Type 0

PDU field	Comment
PDU Type	Type 0 (with Payload CRC)
Frame Number	as defined in [7]
FQC	see below
RFCI	initialise by MGW, see [7], two values required (Speech and SID)
Header CRC	as defined in [7]
Payload CRC	as defined in [7]
Payload Field	112 bits for Speech frames and 33 bits for SID frames, see below

The 112 bits of GSM_HR-coded speech (b1…b112) are defined in TS 46.020, Annex B, in the order of occurrence. The first bit (b1) of the first parameter is placed in bit 8 (the MSB) of the first octet (octet 1) of the Nb Payload field; the second bit is placed in bit 7 of the first octet and so on. The last bit (b112) is placed in the LSB (bit 1) of octet 14.

The GSM_HR SID frames are defined in [24] and [29] and are denoted as b(i) = {b(1),b(2),...,b(33)}. The notation *d*(*j*) = {d(1) … d(33)} of the SID frame is local to the present document and is formed by the pseudo code as follows.

for *j* = 1 to 5
d(*j*) := *b*(*j*); /* R0 parameter in *b*(1) to *b*(5) */;

for *j* = 6 to 16
d(*j*) := *b*(*j*); /* LPC1 parameter in *b*(6)-*b*(16) */

for *j* = 17 to 25
d(*j*) := *b*(*j*); /* LPC2 parameter in *b*(17)-*b*(25) */

for *j* = 26 to 33
d(*j*) := *b*(*j*); /* LPC3 parameter in *b*(26)-*b*(33) */

The payload within the PDU shall be the vector *d(j)* constructed above. The first bit in the vector *d(j)* shall be supplied first in the payload within the PDU.

NOTE: The Payload field for Nb frames for GSM_HR in a BICC-based Circuit Switched Core Network is filled similar, but not identical to the Payload field in RTP Packets according to [22], used in AoIP and NboIP.

The FQC bit is set by the MGW depending on the call case:

1. FQC is set to "good", if the GSM_HR-compression and coding is performed within the MGW.
2. FQC is set to "good", if GSM_HR-coded speech is received without frame quality indication.
3. FQC is derived from the input frame, if FQC or a similar frame quality indication is specified there.
   In case of GSM_HR-coded speech received via TFO frames (see TS 28.062 [5]) the FQC bit is derived from the "Extended control bits" (XC1 to XC5) for 8kbps submultiplexing (specified in TS 48.061, chapter 5.2.4.1.1 and partly reprinted here for ease of reading) as defined in table 8.2.5.2.

Table 8.2.5.2: The FQC bit for GSM_HR-coded Nb frames derived from TFO frames

FQC	XC1	XC2	XC3	XC4	XC5	**Meaning** (in Abis frames with 8kbps submultiplexing)
good	0	0	0	0	0	Good speech frame with UFI = 0 (BFI=0, SID=0, TAF=1) (BFI=0, SID=0, TAF=0)
bad*	0	0	0	0	1	Unreliable speech frame (if speech decoder is in speech decoding mode) or unusable frame (if speech decoder is in comfort noise insertion mode) with UFI = 1 (BFI=0, SID=0, TAF=1) (BFI=0, SID=0, TAF=0)
good	0	0	0	1	0	Valid SID frame with UFI = 0 (BFI=0, SID=2, TAF=1) (BFI=0, SID=2, TAF=0)
bad	0	0	0	1	1	Invalid SID frame with UFI = 1 (BFI=0, SID=2, TAF=1) (BFI=0, SID=2, TAF=0)
bad	0	1	0	0	0	Invalid SID frame at TAF=0 with UFI = 0 (BFI=0, SID=1, TAF=0) (BFI=1, SID=1, TAF=0) (BFI=1, SID=2, TAF=0)
bad	0	1	0	0	1	Invalid SID frame at TAF=0 with UFI = 1 (BFI=0, SID=1, TAF=0) (BFI=1, SID=1, TAF=0) (BFI=1, SID=2, TAF=0)
bad	0	1	0	1	0	Invalid SID frame at TAF=1 with UFI = 0 (BFI=0, SID=1, TAF=1) (BFI=1, SID=1, TAF=1) (BFI=1, SID=2, TAF=1)
bad	0	1	0	1	1	Invalid SID frame at TAF=1 with UFI = 1 (BFI=0, SID=1, TAF=1) (BFI=1, SID=1, TAF=1) (BFI=1, SID=2, TAF=1)
bad*	0	1	1	0	0	Bad speech frame or unusable frame at TAF = 0 with UFI = 0 (BFI=1, SID=0, TAF=0)
bad*	0	1	1	0	1	Bad speech frame or unusable frame at TAF = 0 with UFI = 1 (BFI=1, SID=0, TAF=0)
bad*	0	1	1	1	0	Bad speech frame or unusable frame at TAF = 1 with UFI = 0 (BFI=1, SID=0, TAF=1)
bad*	0	1	1	1	1	Bad speech frame or unusable frame at TAF = 1 with UFI = 1 (BFI=1, SID=0, TAF=1)

Speech frames and SID frames marked in Table 8.2.5.2 with FQC set to "good" shall be sent.
Frames marked in Table 8.2.5.2 with FQC set to "bad*" or "bad" shall not be sent in order to save bandwidth on Nb.

NOTE**:** the abbreviations "UFI" (unreliable frame indication), "BFI" (bad frame indication), "SID" (Silence Descriptor) and "TAF" (Time Alignment Flag) are defined in 3GPP TS 46.041 [25].

8.3 Frame handlers

Nb PDU Frame handling functions are described in [7].

9 Nb Interface User Plane (CN) of a SIP-I -based Circuit Switched Core Network

9.1 Overview

The SIP-I -based Circuit Switched Core Network is specified in 3GPP TS 23.231 [12]. The User Plane in this Core Network is further specified in 3GPP TS 29.414 [10]. RTP is specified in IETF RFC 3550 [16].

RTP is used in a SIP-I -based Circuit Switched Core Network as framing protocol at the Nb-Interface (without Nb-framing protocol). The rules for the usage of RTP and RTCP in 3GPP TS 29.414 [10] are applicable in combination with further Codec-specific rules provided in the present specification.

Table 9.1.1 lists the applicable 3GPP Codec Types for a SIP-I -based Circuit Switched Core Network. Codecs for data transport are described in 3GPP TS 29.007 [13].

Table 9.1.1 Supported Codec Types in a SIP-I -based Circuit Switched Core Network

Payload Type Name	References	Remarks	Support
audio/AMR	IETF RFC 4867 [21]	Applicable for FR_AMR, HR_AMR, OHR_AMR, UMTS_AMR and UMTS_AMR2	Mandatory. Not all AMR Configurations are mandatory. Some Configurations are preferred, see below.
audio/AMR-WB	IETF RFC 4867 [21]	Applicable for FR_AMR-WB, OHR_AMR-WB, OFR_AMR-WB, UMTS_AMR-WB	Optional. AMR-WB is mandatory, if WB speech is supported. Not all WB Configurations are mandatory, see below
audio/GSM_EFR	IETF RFC 3551 [17]	Useful if an A-interface over IP is attached or TFO is used.	Optional
audio/GSM_FR	IETF RFC 3551 [17]	Useful if an A-interface over IP is attached or TFO is used.	Optional
audio/GSM_HR	[22]	Useful if an A-interface over IP is attached or TFO is used.	Optional
audio/PCMA	IETF RFC 3551 [17]	ITU-T G.711 Alaw	Mandatory
audio/PCMU	IETF RFC 3551 [17]	ITU-T G.711 ulaw	Mandatory
audio/telephone-event	IETF RFC 4733 [20]	Used to transport DTMF	Mandatory

The RTP "Payload Type" number for the Nb-Interface is either static (for PCMA, PCMU and GSM_FR) or determined by the MSC-S (dynamic Payload Type).

9.1.1 Time Alignment Procedure

Time Alignment (and AMR Phase Alignment) is not specified in a SIP-I -based Circuit Switched Core Network.

9.1.2 SID Frame Generation

All 3GPP Codec Types include standardized Discontinuous Transmission (DTX) with Voice Activity Detection, Silence Description (by SID frames) and Comfort Noise Generation to fill the speech pauses. If speech inactivity is detected by the Encoder, then (speech) frames are not transmitted, but the transmission is suspended in order to save battery life time in the mobile station, reduce interference on the radio interface and reduce load on all links. The receiving Decoder fills these transmission pauses with Comfort Noise to minimize the contrast between pauses and active speech. Silence Descriptor frames need to be send during speech inactivity to keep the Comfort Noise decently well aligned with the background noise at sender side. This is especially important at the onset of the next talkspurt and therefore SID frames should not be too old, when speech starts again.

The generation of SID frames for the AMR and AMR-WB families of Codecs is determined by the Speech Encoder as specified in TS 26.093 [31], respectively TS 26.193 [32]. The radio subsystem does not influence this timing! SID frames come during speech pauses in uplink and downlink about every 160ms. Also an AMR Encoder in the Media Gateway generates and sends SID frames about every 160ms.

The generation of SID frames for GSM_FR, GSM_HR and GSM_EFR in the GSM radio network is determined by the GSM mobile station and the GSM radio subsystem, not primarily by the Speech Encoder! SID frames come during speech pauses in uplink from the mobile station about every 480ms. In downlink to the mobile station, when they are generated by the Speech Encoder of the GSM radio subsystem, SID frames are sent every 20ms to the GSM base station, which then picks only one every 480ms for downlink radio transmission. For other applications, like transport over Nb, it is more appropriate to send the SID frames less often than every 20ms, but 480ms may be too sparse. As a compromise it is recommended that an Encoder in the Media Gateway should generate and send SID frames every 160ms.

9.1.3 Initial Codec Mode

NOTE: At the Nb-Interface in a SIP-I -based Core Network, direct RTP packetization without Nb-framing is applied. Therefore the use of PDU Type 0 for the speech payload and PDU Type 14 [7] for AMR Rate Control is here not applicable. Also the principle of "Nb_Init" is not applicable for a SIP-I -based Core Network.

The Initial Codec Mode for AMR and AMR-WB shall be derived by pre-defined rules from the AMR Configuration (Active Codec Set), see TS 45.009 [35], chapter 3.4.3 "Initial Codec Mode Selection at Call Setup and Handover".

Start of extract from TS 45.009 [35] for information and ease of reading:

"If the Initial Codec Mode is not signalled, then the default Initial Codec Mode is given by the following implicit rule. If the Active Codec Set contains:

1 mode, then this shall be the Initial Codec Mode;

2 or 3 modes, then the Initial Codec Mode shall be the most robust mode of the set (with lowest bit rate);

4 modes, then the Initial Codec Mode shall be the second most robust mode of the set (with second lowest bit rate."

End of extract from TS 45.009 [35].

In case of FR_AMR (Set 1), i.e. Config-NB-Code 1, see below, this is the AMR Mode with 5.90kbps.

9.2 AMR

AMR (FR_AMR, HR_AMR, OHR_AMR, UMTS_AMR and UMTS_AMR2) shall be packed according to IETF RFC 4867 [21].

The AMR Codec Types can be used in conversational speech telephony services in a number of different Codec Configurations. The set of preferred AMR Codec Configurations is defined in TS 28.062 [5], Table 7.11.3.1.3-2. One of these preferred Configurations, **Config-NB-Code 1**, is recommended for TFO-TrFO harmonisation between GSM and UMTS networks. This Configuration shall be supported in a SIP-I based circuit switched core network to ensure interoperability with an AoIP-based BSS. However, it is recommended that nodes in the core network support all AMR modes for maximum interoperability.

The bandwidth efficient mode of RFC 4867 shall be used. CRC and robust sorting shall not be applied.

To avoid delay, a single frame (Speech or SID_FIRST or SID_UPDATE or ONSET) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20ms shall be applied. No_Data frames should not be sent, except when needed for urgent Rate Control.

Nodes in the core network (e.g. MGWs) transcoding between AMR and some other Codec shall observe the following rules:

- An AMR Encoder (sender) in the core network shall obey an AMR Codec Mode change period of 40ms, i.e. Codec Mode changes by the AMR Encoder (sender) in this core network node are only permissible at every second frame. This ensures maximum interoperability with any AMR receiver.
- An AMR Decoder (receiver) shall, however, be able to accept Codec Mode changes at any time. Variations of the Codec Mode period in receive direction may happen due to handover or other events during a conversation. The UMTS_AMR Codec Type (only allowed in R99 UTRAN-only terminals) may change its Codec Mode any time. Other application of the AMR Codec Types (e.g. MTSI) may perform Codec Mode changes any time. This ensures maximum interoperability with any AMR sender.
- An AMR Encoder shall only change the Codec Mode to a neighbouring mode of the defined AMR Configuration (one step up or one step down), regardless which Rate Control command it receives. If necessary the AMR Encoder shall apply several Codec Mode changes in a row, if the received Rate Control command requests a change of more than one step. This ensures maximum interoperability with any AMR receiver, especially within GSM terminals.

- An AMR Decoder (receiver) shall, however, be able to accept Codec Mode changes in any step size. Variations of the Codec Mode in receive direction may happen due to handover or other events during a conversation. Other application of the AMR Codec Types (e.g. MTSI) may perform any Codec Mode changes. This ensures maximum interoperability with any AMR sender.

- DTX (SCR) shall be supported in send and receive direction.

AMR Rate Control shall use the CMR bits inside the RTP payload, both, in send and receive direction. RTCP shall not be used for AMR Rate Control in a CS core network.

Rate Control Commands coming from an Nb-Interface of a BICC-based Core Network, or an Iu-Interface, or an IMS-Interface, or an general VoIP-Interface, shall be converted to CMR and shall be send continuously inside RTP packets together with the next Speech or SID_FIRST or SID_UPDATE or ONSET frame.

NOTE: In a SIP-I -based Circuit Switched Core Network no Nb-framing is applied and so also no "PDU Type 14" [7] exists for Rate Control Commands.

It is allowed to send an artificially inserted No_Data frame to transport an urgent CMR in RTP. Please note that a GSM radio subsystem connected via AoIP can not send No_Data frames across the radio interface and will typically ignore such No_Data frames. The use of No_Data frames for CMR is especially helpful inside the Core Network at call setup to control the downlink mode for the Encoder inside the terminating MGW for the compression of the ringback tone or an announcement, when the originating MGW still blocks the speech path in forward direction to prevent fraud.

9.3 AMR-WB

AMR-WB (FR_AMR-WB, OHR_AMR-WB, OFR_AMR-WB, UMTS_AMR-WB) shall be packed according to IETF RFC 4867 [21].

The AMR-WB Codec Types can be used in conversational speech telephony services in a number of different Codec Configurations. The set of AMR-WB Codec Configurations is defined in TS 26.103 [14], Table 5.7-1. One of these Configurations, **Config-WB-Code** 0, shall be supported by all nodes supporting the AMR-WB codec in a SIP-I based circuit switched core network to ensure interoperability. However, it is recommended that nodes in the core network support all AMR-WB modes for maximum interoperability.

The bandwidth efficient mode of RFC 4867 [21] shall be used. CRC and robust sorting shall not be applied.

To avoid delay, a single frame (Speech or SID_FIRST or SID_UPDATE or ONSET) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20ms shall be applied. No_Data frames should not be sent, except when needed for urgent Rate Control.

Nodes in the core network (e.g. MGWs) transcoding between AMR-WB and some other Codec shall observe the following rules:

- An AMR-WB Encoder (sender) in the core network shall obey an AMR-WB Codec Mode change period of 40ms, i.e. Codec Mode changes by the AMR-WB Encoder (sender) in this core network node are only permissible at every second frame. This ensures maximum interoperability with any AMR-WB receiver.

- An AMR-WB Decoder (receiver) shall, however, be able to accept Codec Mode changes at any time. Variations of the Codec Mode period in receive direction may happen due to handover or other events during a conversation. Other application of the AMR-WB Codec Types (e.g. MTSI) may perform Codec Mode changes any time. This ensures maximum interoperability with any AMR-WB sender.

- An AMR-WB Encoder shall only change the Codec Mode to a neighbouring mode of the defined AMR-WB Configuration (one step up or one step down), regardless which Rate Control command it receives. If necessary the AMR-WB Encoder shall apply several Codec Mode changes in a row, if the received Rate Control command requests a change of more than one step. This ensures maximum interoperability with any AMR-WB receiver, especially within GSM terminals.

- An AMR-WB Decoder (receiver) shall, however, be able to accept Codec Mode changes in any step size. Variations of the Codec Mode in receive direction may happen due to handover or other events during a conversation. Other application of the AMR-WB Codec Types (e.g. MTSI) may perform any Codec Mode changes. This ensures maximum interoperability with any AMR-WB sender.

- DTX (SCR) shall be supported in send and receive direction.

AMR-WB Rate Control shall use the CMR bits inside the RTP payload, both, in send and receive direction. RTCP shall not be used for AMR-WB Rate Control in a CS core network.

Rate Control Commands coming from an Nb-Interface of a BICC-based Core Network, or an Iu-Interface, or an IMS-Interface, or an general VoIP-Interface, shall be converted to CMR and shall be send continuously inside RTP packets together with the next Speech or SID_FIRST or SID_UPDATE or ONSET frame.

NOTE: In a SIP-I -based Circuit Switched Core Network no Nb-framing is applied and so also no "PDU Type 14" [7] exists for Rate Control Commands.

It is allowed to send an artificially inserted No_Data frame to transport an urgent CMR in RTP. Please note that a GSM radio subsystem connected via AoIP can not send No_Data frames across the radio interface and will typically ignore such No_Data frames. The use of No_Data frames for CMR is especially helpful on the AoIP-Interface in uplink and inside the Core Network at call setup to control the downlink mode for the Encoder inside the terminating MGW for the compression of the ringback tone or an announcement, when the originating MGW still blocks the speech path in forward direction to prevent fraud.

9.4 GSM_EFR

GSM_EFR shall be packed according to IETF RFC 3551 [17].

The coding of SID frames is based on the coding of Speech frames by setting the 95 bits of the so called "SID-Codeword" all to "1", see TS 46.062 [30].

To avoid delay, a single frame (Speech or SID) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20 ms shall be applied. No_Data frames shall not be sent.

DTX shall be supported in send and receive direction.

GSM_EFR frames received from some interface (e.g. a GSM radio interface via TFO) with a bad frame indication set to "bad" shall not be forwarded on the Nb-Interface in a SIP-I -based Circuit Switched Core Network, but silently discarded.

NOTE: RFC 3551 [17] does not support the concept of Bad Frame Indication.

9.5 GSM_FR

GSM_FR shall be packed according to IETF RFC 3551 [17].

The coding of SID frames is based on the coding of Speech frames by setting the 95 bits of the so called "SID-Codeword" all to "0", see TS 46.012 [28].

To avoid delay, a single frame (Speech or SID) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20ms shall be applied. No_Date frames shall not be sent.

DTX shall be supported in send and receive direction.

GSM_FR frames received from some interface (e.g. a GSM radio interface via TFO) with a bad frame indication set to "bad" shall not be forwarded on the Nb-Interface in a SIP-I based Circuit Switched Core Network, but silently discarded.

NOTE: RFC 3551 [17] does not support the concept of Bad Frame Indication.

9.6 GSM_HR

GSM_HR shall be packed according to [22].

The options specified in [22] are not applied inside the Circuit Switched Core Network, but set to pre-defined values as follows:

A single frame (Speech or SID) shall be included in one RTP packet, FEC and Interleaving (redundancy) shall not be used, Encryption shall not be used, a packetization time of 20ms shall be applied. No_Data frames shall not be sent.

DTX shall be supported in send and receive direction.

GSM_HR frames received from some interface (e.g. a GSM radio interface via TFO) with a bad frame indication set to "bad" shall not be forwarded on the Nb-Interface in a SIP-I -based Circuit Switched Core Network, but silently discarded.

NOTE: [22] does not support the concept of Bad Frame Indication.

9.7 PCM

PCMU and PCMA shall be packed according to IETF RFC 3551 [17].

The PCM packetization time for a SIP-I -based Core Network is negotiated via SDP. The mandatory, default value is 20ms; 5ms is one other, optional value; no other packetization time shall be used. To avoid delay, a single frame of length equal to the packetization time shall be included in one RTP packet, Interleaving (redundancy) shall not be used.

The usage of DTX for PCM-coded speech is not recommended for NboIP.

9.8 Telephone-Event

Telephony-Event (DTMF) shall be encoded according to IETF RFC 4733 [20].

The audio/telephone-event payload type in IETF RFC 4733 [20] with default events and default rate shall be used to encode DTMF, if compressed speech is used in a SIP-I -based Core Network. Only in case of PCM-coded speech on NboIP the Telephone-Event is optional, i.e. also inband DTMF tones may be used (see TS 23.231 [12]).

10 A-Interface User Plane over IP

10.1 Overview

The A interface User Plane over IP (AoIP) is standardised in the 3GPP TS 48-series (mainly in TS 48.008) [33] for the Control Plane and in TS 48.103 [34] for the User Plane).

For AoIP the same Codecs as described in Clause 9 are applicable, except telephone-event, see table 10.1.1. Those Codecs shall also be applied in the same manner as described in Clause 9, unless otherwise specified in the present Clause 10.

Table 10.1.1 Supported Codec Types for the A interface User Plane over IP

Payload Type Name	References	Remarks	Support
audio/AMR	IETF RFC 4867 [21]	Applicable for FR_AMR, HR_AMR, OHR_AMR	Optional. Not all AMR Configurations are mandatory. Some Configurations are preferred, see chapter 9.
audio/AMR-WB	IETF RFC 4867 [21]	Applicable for FR_AMR-WB, OHR_AMR-WB, OFR_AMR-WB	Optional. AMR-WB is mandatory, if WB speech is supported. Not all AMR-WB Configurations are mandatory, see chapter 9
audio/GSM_EFR	IETF RFC 3551 [17]		Optional
audio/GSM_FR	IETF RFC 3551 [17]		Mandatory
audio/GSM_HR	[22]		Optional
audio/PCMA	IETF RFC 3551 [17]	ITU-T G.711 Alaw	Optional
audio/PCMU	IETF RFC 3551 [17]	ITU-T G.711 ulaw	Optional

The RTP "Payload Type" for AoIP is pre-determined by 3GPP TS 48.103 [34] for all Codec Types (static payload type).

10.1.1 Time Alignment Procedure

Time Alignment (and AMR Phase Alignment) is not specified for AoIP.

10.1.2 SID Frame Generation

All 3GPP Codec Types include standardized Discontinuous Transmission (DTX) with Voice Activity Detection, Silence Description (by SID frames) and Comfort Noise Generation to fill the speech pauses. If speech inactivity is detected by the Encoder, then (speech) frames are not transmitted, but the transmission is suspended in order to save battery life time in the mobile station, reduce interference on the radio interface and reduce load on all links. The receiving Decoder fills these transmission pauses with Comfort Noise to minimize the contrast between pauses and active speech. Silence Descriptor frames need to be send during speech inactivity to keep the Comfort Noise decently well aligned with the background noise at sender side. This is especially important at the onset of the next talk spurt and therefore SID frames should not be too old, when speech starts again.

The generation of SID frames for the AMR and AMR-WB families of Codecs is determined by the Speech Encoder as specified in TS 26.093 [31], respectively TS 26.193 [32]. The radio subsystem does not influence this timing! SID frames come during speech pauses in uplink and downlink about every 160ms. Also an AMR Encoder in the Media Gateway generates and sends SID frames about every 160ms.

The generation of SID frames for GSM_FR, GSM_HR and GSM_EFR in the GSM radio network is determined by the GSM mobile station and the GSM radio subsystem, not primarily by the Speech Encoder! SID frames come during speech pauses in uplink from the mobile station about every 480ms. In downlink to the mobile station, when they are generated by the Speech Encoder of the GSM radio subsystem, SID frames are sent every 20ms to the GSM base station, which then picks only one every 480ms for downlink radio transmission. For other applications, like transport over the A-Interface, it is more appropriate to send the SID frames less often than every 20ms, but 480ms may be too sparse. As a compromise it is recommended that an Encoder in the Media Gateway should generate and send SID frames every 160ms.

10.1.3 Initial Codec Mode

The Initial Codec Mode for AMR and AMR-WB shall be derived by pre-defined rules from the AMR Configuration (Active Codec Set), see TS 45.009 [35], chapter 3.4.3 "Initial Codec Mode Selection at Call Setup and Handover".

Start of extract from TS 45.009 [35] for information and ease of reading:

> "If the Initial Codec Mode is not signalled, then the default Initial Codec Mode is given by the following implicit rule. If the Active Codec Set contains:

1 mode,	then this shall be the Initial Codec Mode;
2 or 3 modes,	then the Initial Codec Mode shall be the most robust mode of the set (with lowest bit rate);
4 modes,	then the Initial Codec Mode shall be the second most robust mode of the set (with second lowest bit rate."

End of extract from TS 45.009 [35].

In case of FR_AMR (Set 1), i.e. Config-NB-Code 1, see below, this is the AMR Mode with 5.90 kbps.

10.2 AMR

AMR (FR_AMR, HR_AMR, OHR_AMR) shall be packed according to IETF RFC 4867 [21].

The AMR Codec Types can be used in conversational speech telephony services in a number of different Codec Configurations. The set of preferred AMR Codec Configurations is defined in TS 28.062 [5], Table 7.11.3.1.3-2. One of these preferred Configurations, **Config-NB-Code 1**, is recommended for TFO-TrFO harmonisation between GSM and UMTS networks. This Configuration shall be supported in an AoIP supporting BSS and an AoIP supporting Circuit Switched Core Network to ensure interoperability. However, it is recommended that a BSS and Circuit Switched Core Network supports all AMR modes for maximum interoperability.

The bandwidth efficient mode of RFC 4867 [21] shall be used. CRC and robust sorting shall not be applied.

To avoid delay, a single frame (Speech or SID_FIRST or SID_UPDATE or ONSET) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20ms shall be applied. No_Data frames should not be sent downlink across AoIP, except to transport an urgent CMR in RTP. The use of No_Data frames for CMR is especially helpful on the AoIP-Interface in uplink and inside the Core Network at call setup to control the downlink mode for the Encoder inside the terminating MGW for the compression of the ringback tone or an announcement, when the originating MGW still blocks the speech path in forward direction to prevent fraud.

Please note that a GSM radio subsystem can not send No_Data frames across the radio interface and will typically ignore such No_Data frames in downlink direction.

DTX (SCR) shall be supported in send and receive direction.

AMR Rate Control shall use the CMR bits inside the RTP payload, both, in send and receive direction. RTCP shall not be used for AMR Rate Control in a CS network.

10.3 AMR-WB

AMR-WB (FR_AMR-WB, OHR_AMR-WB, OFR_AMR-WB) shall be packed according to IETF RFC 4867 [21].

The AMR-WB Codec Types can be used in conversational speech telephony services in a number of different Codec Configurations. The set of AMR-WB Codec Configurations is defined in TS 26.103 [14], Table 5.7-1. One of these Configurations, **Config-WB-Code 0,** shall be supported by all nodes supporting the AMR-WB codec in an AoIP-supporting BSS and an AoIP-supporting Circuit Switched Core Network to ensure interoperability. However, it is recommended that nodes in the Core Network support all AMR-WB modes for maximum interoperability.

The bandwidth efficient mode of RFC 4867 [21] shall be used. CRC and robust sorting shall not be applied.

To avoid delay, a single frame (Speech or SID_FIRST or SID_UPDATE or ONSET) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20ms shall be applied. No_Data frames should not be sent downlink across AoIP, except to transport an urgent CMR in RTP. The use of No_Data frames for CMR is especially helpful on the AoIP-Interface in uplink and inside the Core Network at call setup to control the downlink mode for the Encoder inside the terminating MGW for the compression of the ringback tone or an announcement, when the originating MGW still blocks the speech path in forward direction to prevent fraud.

Please note that a GSM radio subsystem can not send No_Data frames across the radio interface and will typically ignore such No_Data frames in downlink direction.

DTX (SCR) shall be supported in send and receive direction.

AMR-WB Rate Control shall use the CMR bits inside the RTP payload, both, in send and receive direction. RTCP shall not be used for AMR-WB Rate Control in a Circuit Switched Core Network.

10.4 GSM_EFR

GSM_EFR shall be packed according to IETF RFC 3551 [17].

The coding of SID frames is based on the coding of Speech frames by setting the 95 bits of the so called "SID-Codeword" all to "1", see TS 46.062 [30].

To avoid delay, a single frame (Speech or SID) shall be included in one RTP packet, Interleaving (redundancy) shall not be used, and a packetization time of 20 ms shall be applied. No_Data frames shall not be sent.

DTX shall be supported in send and receive direction.

NOTE: RFC 3551 [17] does not support the concept of Bad Frame Indication. Therefore missing GSM_EFR frames in the AoIP downlink direction (e.g. discarded by a network node due to the missing bad frame indication) need to be properly treated within the BSS before sending downlink on the radio interface. Details are not specified.

10.5 GSM_FR

GSM_FR shall be packed according to IETF RFC 3551 [17].

The coding of SID frames is based on the coding of Speech frames by setting the 95 bits of the so called "SID-Codeword" all to "0", see TS 46.012 [28].

To avoid delay, a single frame (Speech or SID) shall be included in one RTP packet, Interleaving (redundancy)shall not be used, and a packetization time of 20 ms shall be applied. No_Data frames shall not be sent.

DTX shall be supported in send and receive direction.

NOTE: RFC 3551 [17] does not support the concept of Bad Frame Indication. Therefore missing GSM_EFR frames in the AoIP downlink direction (e.g. discarded by a network node due to the missing bad frame indication) need to be properly treated within the BSS before sending downlink on the radio interface. Details are not specified.

10.6 GSM_HR

GSM_HR shall be packed according to [22].

The options specified in [22] are not applied on AoIP, but set to pre-defined values as follows:
A single frame (Speech or SID) shall be included in one RTP packet, FEC and Interleaving (redundancy) shall not be used, Encryption shall not be used, a packetization time of 20ms shall be applied. No_Data frames shall not be sent.

DTX shall be supported in send and receive direction.

NOTE: [22] does not support the concept of Bad Frame Indication. Therefore missing GSM_HR frames in the AoIP downlink direction (e.g. discarded by a network node due to the missing bad frame indication) need to be properly treated within the BSS before sending downlink on the radio interface. Details are not specified.

10.7 PCM

PCMU and PCMA shall be packed according to IETF RFC 3551 [17].

A packetization time of 20ms shall be applied for PCM on AoIP. The packetization time is not negotiated for AoIP. To avoid delay, a single frame of 20ms shall be included in one RTP packet, Interleaving (redundancy) shall not be used.

The usage of DTX for PCM-coded speech is not allowed on AoIP.

Annex A (informative): Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
1999-12	6	SP-99563			Approved at TSG-SA#6 Plenary		3.0.0
2000-03	7	SP-000025	001	3	Introduction of QoS parameters used at RAB assignment	3.0.0	3.1.0
2000-03	7	SP-000025	002		Introduction of different RFCS set on Iu User Plane	3.0.0	3.1.0
2000-03	7	SP-000025	003	2	Introduction of Time Alignment	3.0.0	3.1.0
2000-12	10	SP-000575	005	1	AMR interface to Iu	3.1.0	3.2.0
2001-03	11	SP-010103	006	2	Removal of TFO and TrFO from Release 99, and removal of Initial Time Alignment	3.2.0	3.3.0
2001-03	11	SP-010103	008	1	Introduction of TFO and TrFO	3.3.0	4.0.0
2002-06	16				Version for Release 5	4.0.0	5.0.0
2002-12	18	SP-020689	012	2	Correction of RAB parameter assignment for AMR	5.0.0	5.1.0
2003-03	19	SP-030087	015	2	AMR Rate Adaptation of Rel-5	5.1.0	5.2.0
2004-04	25	SP-040645	016	1	Mapping of GSM_EFR SID on Nb Interface	5.2.0	6.0.0
2005-12	30	SP-050791	0017		20 ms packetisation time for PCM coded speech over IP Nb	6.0.0	7.0.0
2006-06	32	SP-060358	0018	1	Supplement of 20 ms packetisation time for PCM coded speech over IP Nb	7.0.0	7.1.0
2008-06	41	SP-080475	0019	2	Addition of CS over IP User Plane	7.1.0	8.0.0
2008-06	41	SP-080475	0020	1	Nb-framing for GSM_FR and GSM_HR	7.1.0	8.0.0
2008-12	42	SP-080678	0021	3	Corrections to CS over IP User Plane	8.0.0	8.1.0
2009-09	45	SP-090568	0023	1	Clarification of RAB sub-flow numbering for AMR	8.1.0	8.2.0
2009-12	46	SP-090703	0024	1	Correction of payload field size for mapping of GSM FR on Nb interface	8.2.0	8.3.0
2009-12	46				Version for Release 9	8.3.0	9.0.0

History

Document history		
V9.0.0	January 2010	Publication