

ETSI TS 126 103 V13.0.0 (2016-01)



**Digital cellular telecommunications system (Phase 2+);
Universal Mobile Telecommunications System (UMTS);
Speech codec list for GSM and UMTS
(3GPP TS 26.103 version 13.0.0 Release 13)**



Reference

RTS/TSGS-0426103vd00

Keywords

GSM,UMTS

ETSI

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

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

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

Important notice

The present document can be downloaded from:
<http://www.etsi.org/standards-search>

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at
<http://portal.etsi.org/tb/status/status.asp>

If you find errors in the present document, please send your comment to one of the following services:
<https://portal.etsi.org/People/CommitteeSupportStaff.aspx>

Copyright Notification

No part may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm except as authorized by written permission of ETSI.

The content of the PDF version shall not be modified without the written authorization of ETSI.
The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2016.
All rights reserved.

DECT™, **PLUGTESTS™**, **UMTS™** and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members.
3GPP™ and **LTE™** are Trade Marks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.
GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

Intellectual Property Rights

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

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

Foreword

This Technical Specification (TS) has been produced by ETSI 3rd Generation Partnership Project (3GPP).

The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under <http://webapp.etsi.org/key/queryform.asp>.

Modal verbs terminology

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

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

Contents

Intellectual Property Rights	2
Foreword.....	2
Modal verbs terminology.....	2
Foreword.....	4
1 Scope	5
2 Normative references	5
3 Definitions and Abbreviations.....	6
3.1 Definitions	6
3.2 Abbreviations	6
4 General	7
5 3GPP Codec List for OoBTC in a BICC-based Circuit Switched Core Network and for AoIP.....	9
5.1 GSM Full Rate Codec Type (GSM FR)	9
5.2 GSM Half Rate Codec Type (GSM HR).....	9
5.3 GSM Enhanced Full Rate Codec Type (GSM EFR).....	9
5.4 Five Adaptive Multi-Rate Codec Types (FR AMR, HR AMR, UMTS AMR, UMTS AMR2, OHR AMR)	10
5.5 TDMA Enhanced Full Rate Codec Type (TDMA EFR).....	13
5.6 PDC Enhanced Full Rate Codec Type (PDC_EFR).....	13
5.7 Four Adaptive Multi-Rate Wideband Codec Types (FR AMR-WB, UMTS AMR-WB, OFR AMR-WB, OHR AMR-WB)	13
5.8 MuMe Dummy Codec (3G.324M).....	16
5.9 MuMe2 Dummy Codec (3G.324M2).....	17
5.10 Codec Extension.....	17
5.11 CSData Dummy Codec (AoIP)	17
6 Codec List for the Call Control Protocol.....	17
6.1 System Identifiers for GSM and UMTS.....	17
6.2 Codec Bitmap	18
6.3 Selected Codec Type	19
7 3GPP Codecs for OoBTC in a SIP-I -based Circuit Switched Core Network	20
7.1 Overview	20
7.2 AMR.....	20
7.3 AMR-WB	21
7.4 GSM_EFR.....	21
7.5 GSM_FR	21
7.6 GSM_HR.....	21
7.7 PCM	22
7.8 Telephone-Event	22
Annex A (informative): Example Supported Codec List for UMTS	23
Annex B (informative): Change history	25
History	26

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

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

Version x.y.z

where:

- x the first digit:
 - 1 presented to TSG for information;
 - 2 presented to TSG for approval;
 - 3 or greater indicates TSG approved document under change control.
- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
- z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

The present Technical Specification outlines the Codec Lists in 3GPP including both systems, GSM and UMTS, to be used by the Out of Band Transcoder Control (OoBTC) protocol to set up a call or modify a call in **Transcoder Free Operation (TrFO)** and in "transcoder at the edge" scenarios.

The TS also specifies the SDP description of 3GPP Codecs to be used within a SIP-I -based circuit switched core network as specifies in 3GPP TS 23.231 [14].

The TS further specifies the coding of the Supported Codec List Information Elements for the UMTS radio access technology.

The TS further reserves the Code Point for the CSData (dummy) Codec Type for the negotiation of A-Interface Type and the RTP redundancy for CS Data and Fax services, see 3GPP TS 48.008 [23].

The Supported Codec List IE includes Codec_Types from the TDMA and PDC systems, to support TFO or TrFO between UMTS and TDMA, or UMTS and PDC.

2 Normative 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 26.090: "AMR Speech Codec; Speech Transcoding Functions".
- [2] 3GPP TS 26.093: "AMR Speech Codec; Source Controlled Rate Operation".
- [3] 3GPP TS 26.101: "Mandatory Speech Codec Speech Processing Functions; AMR Speech Codec Frame Structure".
- [4] 3GPP 46.0xx: "Enhanced Full Rate Codec Recommendations".
- [5] 3GPP 26.0xx: "Adaptive Multi-Rate Codec Recommendations".
- [6] "ITU Q.765.5: "Use of Application Transport Mechanism for Bearer Independent Call Control"
- [7] 3GPP TS 28.062: "In-band Tandem Free Operation (TFO) of Speech Codecs, Stage 3 - Service Description".
- [8] 3GPP TS 23.153: "Out of Band Transcoder Control - Stage 2".
- [9] 3GPP TS 24.008: "Mobile radio interface layer 3 specifications, Core Network Protocols"
- [10] 3GPP TS 26.190: "AMR Wideband Speech Codec; Speech Transcoding Functions".
- [11] 3GPP TS 26.193: "AMR Wideband Speech Codec; Source Controlled Rate Operation".
- [12] 3GPP TS 26.201: "Mandatory Speech Codec Speech Processing Functions; AMR Wideband Speech Codec Frame Structure".
- [13] 3GPP TS 23.172: "CS multimedia service UDI/RDI fallback and service modification; Stage 2".
- [14] 3GPP TS 23.231: "SIP-I based circuit-switched core network; Stage 2".

- [15] 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)".
- [16] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)", J. Rosenberg and H. Schulzrinne.
- [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. Sjöberg, M. Westerlund, A. Lakaniemi and Q. Xie.
- [22] IETF RFC 5993 (2010) "RTP Payload Format for Global System for Mobile Communications Half Rate (GSM-HR)".
- [23] 3GPP TS 48.008: "Mobile Switching Centre - Base Station System (MSC-BSS) interface".
- [24] 3GPP TS 26.102: "Adaptive Multi-Rate (AMR) speech codec; Interface to Iu, Uu and Nb".

3 Definitions and Abbreviations

3.1 Definitions

Codec Type: defines a specific type of a speech Coding algorithm, applied on a specific radio access technology (e.g. GSM FR, (GSM) FR AMR).

Codec Mode: defines a specific mode of a Codec Type (e.g. 12,2 kBit/s Mode of the (GSM) FR AMR).

Codec Configuration: defines a specific set of attributes to a certain Codec Type (e.g. the combination of ACS and DTX="on" for (GSM) FR AMR).

Organisation Identifier (OID): Identifies the standard organisation (e.g. 3GPP) producing a specification for a Codec List. ITU-T is responsible for maintaining the list of Organisation Identifiers.

System Identifier (SysID): Identifies the radio access technology (e.g. GSM or UMTS) for which the supported Codec List is defined.

Other definitions are given in TS 23.153 [8].

3.2 Abbreviations

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

ACS	Active Codec (mode) Set
AoIP	A-Interface User Plane over IP
BWM	BandWidth Multiplier
CoID	Codec IDentifier
CSDData	Circuit Switched Data and Fax dummy Codec

DTX	Discontinuous Transmission
GSM	Global System for Mobile communication
MuMe	Multi-Media
NboIP	Nb-Interface User Plane transport over IP in a SIP-I -based network
OID	Organisation IDentifier (e.g. ITU-T, 3GPP)
OoBTC	Out of Band Transcoder Control
PDC	Personal Digital Communication (synonym for ...)
RX	Receive
SCR	Source Controlled Rate operation (synonym to DTX)
SID	Silence Descriptor
SysID	System Identifier
TDMA	Time Division Multiple Access (synonym for ...)
TFO	Tandem Free Operation (also sometimes called "Transcoder-Through" or "Codec-Bypass")
TrFO	Transcoder Free Operation
TX	Transmit
UMTS	Universal Mobile Telecommunications System

4 General

The present Technical Specification (TS) outlines the 3GPP internal Codec Lists for both, GSM and UMTS, to be used by the Out of Band Transcoder Control (OoBTC) protocol a BICC-based Circuit Switched Core Network to set up a call or modify a call in Transcoder Free Operation (TrFO). The Codec List is also used in the Codec Negotiation for the A-Interface User Plane over IP (AoIP), see 3GPP TS 48.008 [23].

The TS specifies the SDP parameters for the 3GPP Codecs for OoBTC in a SIP-I -based Circuit Switched Core Network, see 3GPP TS 23.231 [14].

The TS further specifies the coding of the Supported Codec List Information Elements as defined in 3GPP TS 24.008 for the UMTS radio access technology.

Transcoder Free Operation allows the transport of speech signals in the coded domain from one user equipment (UE) to the other user equipment through the radio access network (RAN) and core network (CN), possibly through a transit network (TN). This enables high speech quality, low transmission costs and high flexibility.

The necessary Codec Type selection and resource allocation are negotiated out of band before and after call setup. Possible Codec (re-)configuration, Rate Control and DTX signalling may be performed after call setup by additional inband signalling or a combination of inband and out-of-band signalling.

Up to release '99 GSM does not support Transcoder Free Operation, but specifies the Tandem Free Operation (TFO). Tandem Free Operation enables similar advantages, but is based on pure inband signalling after call setup. The parameters defined in this Technical Specification allow interaction between TrFO and TFO. They further provide an evolutionary path for GSM towards Transcoder Free Operation.

The GERAN and UTRAN standards define fourteen different Speech Codec Types, see table 4.1.

In addition to these Speech Codec Types some "dummy" Codec Types are defined to support the negotiation for data, fax and multimedia applications.

Table 4.1: Support of Codec Types in Radio Access Technologies

	TDMA EFR	UMTS AMR 2	UMTS AMR	(GSM) HR AMR	(GSM) FR AMR	GSM EFR	GSM HR	GSM FR
CoID	0x07	0x06	0x05	0x04	0x03	0x02	0x01	0x00
GERAN GMSK	not defined	not possible	not possible	yes, 1..4 modi	yes, 1..4 modi	yes	yes	yes
GERAN 8PSK	not defined	not possible	not possible	not defined	not defined	not defined	not defined	not defined
UTRAN	not defined	yes, 1..8 modi 1..4 modi recomm.	R99, UTRAN- only UEs	not defined	not defined	not defined	not defined	not defined

	Codec Extension		OHR AMR-WB	OFR AMR-WB	OHR AMR	UMTS AMR-WB	FR AMR-WB	PDC EFR
CoID	0x0F	0x0E	0x0D	0x0C	0x0B	0x0A	0x09	0x08
GERAN GMSK	reserved	spare	not defined	not defined	not defined	not possible	yes ³ modi	not defined
GERAN 8PSK	reserved	spare	yes, 3 modi	yes, 3 modi	yes, 1..4 modi	not possible	not defined	not defined
UTRAN	reserved	spare	not defined	not defined	not defined	yes 3..4 modi	not defined	not defined

CoID is reprinted here in hexadecimal notation. It is defined in section 5.

Up to date the following Code Points are defined:

Table 4.2. Defined Code Points

Hexadecimal Notation	Binary Notation	Codec Name	Remark
0x00h	0x0000.0000	GSM_FR	
0x01h	0x0000.0001	GSM_HR	
0x02h	0x0000.0010	GSM_EFR	
0x03h	0x0000.0011	(GSM) FR_AMR	
0x04h	0x0000.0100	(GSM) HR_AMR	
0x05h	0x0000.0101	UMTS_AMR	
0x06h	0x0000.0110	UMTS_AMR2	
0x07h	0x0000.0111	TDMA_EFR	
0x08h	0x0000.1000	PDC_EFR	
0x09h	0x0000.1001	(GSM) FR_AMR-WB	
0x0Ah	0x0000.1010	UMTS_AMR-WB	
0x0Bh	0x0000.1011	OHR_AMR	
0x0Ch	0x0000.1100	OFR_AMR-WB	
0x0Dh	0x0000.1101	OHR_AMR-WB	
0x0Eh	0x0000.1110	Spare, for future use	
0x0Fh	0x0000.1111	Codec Extension	For AoIP and TFO
0x10h ... 0xFCh	0x0001.0000 ... 0x1111.1100	Spare, for future use	
0xFDh	0x1111.1101	CSDData	For AoIP only
0xFEh	0x1111.1110	MuMe2	For OoBTC only
0xFFh	0x1111.1111	MuMe	For OoBTC only

5 3GPP Codec List for OoBTC in a BICC-based Circuit Switched Core Network and for AoIP

The definition of the common Codec List for Out of Band Transcoder Control (3GPP TS 23.153, [8]) in 3GPP for GSM and UMTS follows the specifications given in ITU Q.765.5: The most preferred Codec Type is listed first, followed by the second preferred one, and so on. An informative example for a codec list for UMTS can be found in Annex A.

The Codec IDentification codes (CoIDs) are specified in two versions: the long form (8 bits) for the use in OoBTC and the short form (the 4 LSBs of the long form) for the use in TFO and AoIP.

5.1 GSM Full Rate Codec Type (GSM FR)

The Codec IDentification (CoID) code is defined to be: FR_CoID := 0x0000.0000.

The GSM Full Rate Codec Type has no additional parameters.

For information (for exact details see GSM Recommendations):

The GSM Full Rate Codec Type supports one fixed Codec Mode with 13.0 kBit/s.

DTX may be enabled in uplink and in downlink independently of each other. DTX on or off is defined by the network on a cell basis and can not be negotiated at call setup or during the call. The DTX scheme uses one SID frame to mark the end of a speech burst and to start Comfort Noise Generation. Identical SID frames for comfort noise updates are sent in speech pauses about every 480 ms, aligned with the cell's TDMA frame structure. The defined Tandem Free Operation allows the reception of GSM FR DTX information for the downlink direction in all cases. The TFO respectively TrFO partner is prepared to receive DTX information as well.

5.2 GSM Half Rate Codec Type (GSM HR)

The Codec IDentification (CoID) code is defined to be: HR_CoID := 0x0000.0001.

The GSM Half Rate Codec Type has no additional parameters.

For information (for exact details see GSM Recommendations):

The GSM Half Rate Codec Type supports one fixed Codec Mode with 5.60 kBit/s.

DTX may be enabled in uplink and in downlink independently of each other. DTX on or off is defined by the network on a cell basis and can not be negotiated at call setup or during the call. The DTX scheme uses one SID frame to mark the end of a speech burst and to start Comfort Noise Generation. Identical SID frames for comfort noise updates are sent in speech pauses about every 480 ms, aligned with the cell's TDMA frame structure. The defined Tandem Free Operation allows the reception of GSM HR DTX information for the downlink direction in all cases. The TFO respectively TrFO partner shall be prepared to receive DTX information as well.

5.3 GSM Enhanced Full Rate Codec Type (GSM EFR)

The Codec IDentification (CoID) code is defined to be: EFR_CoID := 0x0000.0010.

The GSM Enhanced Full Rate Codec Type has no additional parameters.

For information (for exact details see GSM Recommendations):

The GSM Enhanced Full Rate Codec Type supports one fixed Codec Mode with 12.2 kBit/s.

DTX may be enabled in uplink and in downlink independently of each other. DTX on or off is defined by the network on a cell basis and can not be negotiated at call setup or during the call. The DTX scheme uses one SID frame to mark the end of a speech burst and to start Comfort Noise Generation. It is important to note that the Comfort Noise parameters for this start of the comfort noise generation are calculated at transmitter side from the previous eight speech frames. A DTX hangover period needs to be applied therefore at transmitter side before sending the first SID frame.

SID frames with incremental information for comfort noise updates are sent in speech pauses about every 480 ms, aligned with the cell's TDMA frame structure. The defined Tandem Free Operation allows the reception of GSM EFR DTX information for the downlink direction in all cases. The TFO respectively TrFO partner shall be prepared to receive DTX information as well.

5.4 Five Adaptive Multi-Rate Codec Types (FR AMR, HR AMR, UMTS AMR, UMTS AMR2, OHR AMR)

The Adaptive Multi-Rate Codec algorithm is applied in GERAN-GMSK, GERAN-8PSK and UTRAN in five different Codec Types.

The Codec IDentification (CoID) codes are defined to be:

FR_AMR_CoID := 0x0000.0011.
 HR_AMR_CoID := 0x0000.0100.
 UMTS_AMR_CoID := 0x0000.0101.
 UMTS_AMR_2_CoID := 0x0000.0110.
 OHR_AMR_CoID := 0x0000.1011.

The AMR Codec Types can be used in conversational speech telephony services in a number of different configurations. The set of preferred configurations is defined in TS 28.062, 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, it is mandatory for an AoIP-supporting BSS, see 3GPP TS 48.008 [23], an AoIP-supporting BICC-based Circuit Switched Core Network and for any SIP-I -based Circuit Switched Core Network.

The Single Codec Information Element for AMR Codec Types may have several additional parameters. These parameters are optional in the Supported Codec List (BICC) and in the Available Codec List (BICC), but these parameters shall specify exactly one AMR Configuration for the Selected Codec (BICC), see [8].

Active Codec Set, ACS: eight bits.

Each bit corresponds to one AMR Mode. Setting the bit to "1" means the mode is included, setting the bit to "0" means the mode is not included in the ACS.

Note: Except for HR_AMR all eight AMR modes may be selected, for the HR_AMR only the six lower modes.

Supported Codec Set, SCS: eight bits.

Each bit corresponds to one AMR Mode, as in the ACS. Setting the bit to "1" means the mode is supported, setting the bit to "0" means the mode is not supported. The SCS shall at least contain all modes of the ACS.

Maximal number of codec modes in the ACS, MACS: three bits.

MACS shall be used in the Supported Codec List (BICC) and the Available Codec List (BICC), when it is necessary to restrict the maximum number of modes for the (future) Selected Codec (BICC).

For FR AMR, HR AMR and OHR AMR one up to four, for the UMTS AMR and UMTS AMR2 one up to eight Codec Modes are allowed.

Coding: "001": one, "010": two, ... "111": seven, "000": eight Codec Modes allowed.

Optimisation Mode for ACS, OM: one bit.

OM indicates, whether the sending side supports the modification (optimisation) of its offered ACS for the needs of the distant side.

Coding: "0": Optimisation of the ACS not supported, "1": Optimisation of the ACS supported.

If OM is specified as "Optimisation of the ACS not supported", then SCS and MACS have no meaning for this Single Codec Information Element; then the SCS shall at least contain all modes of the offered ACS; MACS shall be equal to or larger than the number of modes in the offered ACS.

Usage of this Single Codec Information Element in OoBIC.

In the Single Codec Information Element for the Selected Codec (BICC) the ACS shall be specified exactly.

For FR AMR, HR_AMR and OHR AMR at least one, but not more than four modes shall be included.

For UMTS AMR and UMTS AMR2 at least one, but not more than four modes should be included.

OM shall be set to "Optimisation of the ACS not supported".

In the Single Codec Information Element for the Supported Codec List (BICC) and the Available Codec List (BICC) one of the following codings shall be used

- either all parameters (ACS, SCS, MACS and OM) are omitted.
Then per default all possible AMR modes shall be treated as included in ACS and SCS, MACS shall be treated as set to its allowed maximum and OM shall be treated as set to "Optimisation of the ACS supported".
- or only the ACS is specified:
Then per default all possible AMR modes shall be treated as included in the SCS, MACS shall be treated as set to its allowed maximum and OM shall be treated as set to "Optimisation of the ACS supported".
- or ACS and SCS are specified.
Then per default MACS shall be treated as set to its allowed maximum and OM shall be treated as set to "Optimisation of the ACS supported".
- or all parameters (ACS, SCS, MACS and OM) are specified.

Procedures in OoBTC

The procedures for handling of these Single Codec Information Element in the originating, intermediate and terminating nodes are specified in TS 23.153 [8].

The "Single Codec" information element consists of 5 to 8 octets in case of the AMR Codec Types (table 5.4):

Table 5.4: Coding of "Single Codec" for the Adaptive Multi-Rate Codec Types

Octet	Parameter	MSB 8	7	6	5	4	3	2	1 LSB
1 m	Single Codec	Single Codec (see ITU-T Q.765.5)							
2 m	Length Indication	3, 4, 5, 6							
3 m	Compat. Info	Compatibility Information							
4 m	OID	ETSI OID (See ITU-T Q.765.5 [6])							
5 m	CoID	FR_AMR_CoID, HR_AMR_CoID, UMTS_AMR_CoID, UMTS_AMR_2_CoID, OHR_AMR_CoID							
6 o	ACS	12.2	10.2	7.95	7.40	6.70	5.90	5.15	4.75
7 o	SCS	12.2	10.2	7.95	7.40	6.70	5.90	5.15	4.75
8 o	OM, MACS	(spare)	(spare)	(spare)	(spare)	OM	MACS		

with "m" = mandatory and "o" = optional

For information on GSM procedures (for exact details see GSM Recommendations):

The GSM AMR Codec Types comprise eight (Full Rate), respectively six (Half Rate) different Codec Modes: 12,2 ... 4,75 kBit/s.

The active Codec Mode is selected from the Active Codec Set (ACS) by the network (Codec Mode Command) with assistance by the mobile station (Codec Mode Request). This Codec Mode Adaptation, also termed Rate Control, can be performed every 40 ms by going one Codec Mode up or down within the ACS. The Codec Modes in uplink and downlink at one radio leg may be different. In Tandem Free Operation both radio legs (A and B) are considered for the optimal selection of the active Codec Mode in each direction (uplink A and then downlink B, respectively vice versa) by the "Distributed Rate Decision" algorithm. The worst of both radio legs determines the highest allowed Codec Mode, respectively the maximally allowed rate ("Maximum Rate Control"). All rate control commands are transmitted inband: on the radio interface, the BTS-TRAU interface and the TRAU-TRAU interface.

The Active Codec Set is configured at call setup or reconfigured during the call. It consists of one up to maximally four Codec Modes (MACS) at a given time, selected from the Supported Codec Set. The maximal number of Codec Modes and the Supported Codec Set may be constrained by the network to consider resources and radio conditions.

The Active Codec Sets in uplink and downlink are identical.

First, at start up of Tandem Free Operation, Active Codec Sets, the Supported Codec Sets, the MACSs and the OMs are taken into account to determine the optimal common Active Codec Set. In a later phase the Codec Lists of both radio

legs may be taken into account to find the optimum configuration. For exact details see 3GPP TS 28.062. All configuration data and update protocols are transmitted inband.

The DTX scheme of the Adaptive Multi-Rate Codec Type marks with a specific SID_FIRST frame the end of a speech burst. SID_FIRST does not contain Comfort Noise parameters. This SID_FIRST starts the comfort noise generation with parameters that are calculated at receiver side (!) from the latest received seven speech frames. A DTX hangover period needs to be applied therefore at transmitter side before sending of this SID_FIRST.

Absolutely coded SID_UPDATE frames follow about every eighth frame (160 ms) in speech pauses. SID_UPDATE frames are sent independently of the cell's TDMA frame structure and are related only to the source signal.

An ONSET frame (typically) precedes in uplink direction the beginning of a new speech burst. DTX on or off is defined by the network on a cell basis. The defined Tandem Free Operation allows the reception of GSM-AMR DTX information for the downlink direction in all cases.

Note: The DTX scheme of the Enhanced Full Rate Codec Type is not compatible with the DTX scheme of the Adaptive Multi-Rate Codec Type in Codec Mode 12.2 kBit/s, although the speech modes of these two Codec Types are bit exact identical.

Informative for terminals of R99 that support only UTRAN access ("UTRAN-only" terminals):

UTRAN-only terminals of R99 may either use UMTS AMR or UMTS AMR2 as default speech version in UTRAN access.

Normative for terminals that support GSM and UTRAN radio access ("dual-mode" terminals):

Dual-mode terminals of R99 and onwards shall use the UMTS AMR2 as the default speech version in UTRAN access. They need not to support the UMTS AMR, because the UMTS AMR2 in terminals is a fully compatible replacement.

Normative for all UMTS terminals of REL-4 and onwards: The UMTS AMR2 shall be the default speech version in UTRAN access in all terminals, UTRAN-only and dual-mode (GSM and UTRAN) of REL-4 and onwards.

For information on UMTS procedures (for exact details see 3GPP TS 28.062 (TFO) and 3GPP TS 23.153 (TrFO)):

The active Codec Mode is selected from the Active Codec Set (ACS) by the network. This Codec Mode Adaptation, also termed Rate Control, can be performed for the UMTS AMR every 20 ms by going to another Codec Mode within the ACS. For the UMTS AMR 2 this Codec Mode Adaptation can be performed every 20ms for the downlink traffic channel, but only every 40ms for the uplink radio channel. The UE selects at call setup one of the two possible phases for Codec Mode Adaptation (odd or even frames). During the call changes of the Codec Mode in uplink direction are only allowed in this selected phase. Rate Control commands received in downlink direction are considered at the next possible phase.

By this definition the UMTS AMR 2 Codec Type is TFO and TrFO compatible to the FR AMR, HR AMR, OHR AMR and UMTS AMR 2 Codec Types. In any multi-mode configuration the UMTS_AMR shall be regarded as only compatible to itself, not to any other AMR codec Type, to avoid incompatibilities in TFO-TrFO-TFO interworking scenarios. In single mode configuration, UMTS AMR and UMTS AMR 2 are compatible, when both codec types use the same single rate ACS.

The Codec Modes in uplink and downlink at one radio leg may be different. In Tandem Free Operation or Transcoder Free Operation both radio legs (A and B) are considered for the optimal selection of the active Codec Mode in each direction (uplink A and then downlink B, respectively vice versa) by a "Distributed Rate Decision" algorithm. The worst of both radio legs determine the highest allowed Codec Mode, respectively the maximally allowed rate. All rate control commands are transmitted inband on the Iu and Nb interfaces and out of band on the radio interface.

The Active Codec Set is configured at call setup or reconfigured during the call. It consists of one up to maximally eight Codec Modes (MACS) at a given time, selected from the Supported Codec Set. The maximal number of Codec Modes and the Supported Codec Set may be constrained by the network to consider resources and radio conditions.

The Active Codec Sets in uplink and downlink are typically identical.

At call setup the Originating Side sends the AMR parameter set (included in the Codec List). The Terminating side then selects a suitable ACS from the given information and sends it back. In case the terminating side does not support TrFO a transcoder is allocated in the path at a suitable position, preferably as close as possible to the terminating side. This transcoder may by inband signalling install a Tandem Free Operation after call setup. Then, at start up of Tandem Free Operation, both Active Codec Sets, the Supported Codec Sets, the MACSs and the OMs are taken into account to determine the optimal common Active Codec Set. In a later phase the Codec Lists of both radio legs may be taken into account to find the optimum configuration. All configuration data and update protocols are transmitted inband on the TFO interface, but out of band within the UMTS network. For information on Tandem Free Operation see 3GPP TS 28.062 and on Transcoder Free Operation see 3GPP TS 23.153.

The SCR scheme of the Adaptive Multi-Rate Codec Types mark with a specific SID_FIRST frame the end of a speech burst. SID_FIRST does not contain Comfort Noise parameters. This SID_FIRST starts the comfort noise generation with parameters that are calculated at receiver side (!) from the latest received seven speech frames. A DTX hangover period needs to be applied therefore at transmitter side before sending of this SID_FIRST.

Absolutely coded SID_UPDATE frames follow about every eighth frame (160 ms) in speech pauses. SID_UPDATE frames are sent independently of the cell's timing structure and are related only to the source signal.

An ONSET frame does (typically) not exist in UMTS networks, but may be received in TFO from the distant partner. It marks the beginning of a speech burst. The uplink SCR operation is always activated for UMTS AMR and UMTS AMR2 codec types. The defined Tandem Free Operation and Transcoder Free Operation allows the reception of AMR SCR information for the downlink direction in all cases.

The SCR scheme of the UMTS AMR2 Codec Type is fully compatible to the SCR scheme of the UMTS AMR in UMTS and the DTX schemes of the FR AMR, HR AMR and OHR AMR Codec Types.

5.5 TDMA Enhanced Full Rate Codec Type (TDMA EFR)

The Codec Identification (CoID) code is defined to be: TDMA_EFR_CoID := 0x0000.0111.

The TDMA Enhanced Full Rate Codec Type has no additional parameters.

For information (for exact details see TDMA Recommendations):

The TDMA Enhanced Full Rate Codec Type supports one fixed Codec Mode with 7.4 kBit/s. This codec mode is bit exact identical with AMR codec mode at 7.4 kBit/s.

In a TDMA system DTX may be enabled in uplink, but not in downlink. The DTX scheme uses one SID frame to mark the end of a speech burst and to start or continue Comfort Noise Generation.

The defined Tandem Free Operation allows the reception of TDMA EFR DTX information for the downlink direction in all cases. In TDMA systems the transcoder has to generate comfort noise in speech like frames to be sent downlink. In UMTS the downlink DTX shall always be supported and the transcoder can therefore stay transparently in TFO.

5.6 PDC Enhanced Full Rate Codec Type (PDC_EFR)

The Codec Identification (CoID) code is defined to be: TDMA_EFR_CoID := 0x0000.1000.

The PDC Enhanced Full Rate Codec Type has no additional parameters.

For information (for exact details see PDC Recommendations):

The PDC Enhanced Full Rate Codec Type supports one fixed Codec Mode with 6.7 kBit/s. This codec mode is bit exact identical with AMR codec mode at 6.7 kBit/s.

In a PDC system DTX may be enabled in uplink, but not in downlink. The DTX scheme uses one SID frame to mark the end of a speech burst and to start or continue Comfort Noise Generation.

The Tandem Free Operation allows the reception of PDC EFR DTX information for the downlink direction in all cases. In PDC systems the transcoder has to generate comfort noise in speech like frames to be sent downlink. In UMTS the downlink DTX shall always be supported and the transcoder can therefore stay transparently in TFO.

5.7 Four Adaptive Multi-Rate Wideband Codec Types (FR AMR-WB, UMTS AMR-WB, OFR AMR-WB, OHR AMR-WB)

The Adaptive Multi-Rate - WideBand Codec algorithm is applied in GERAN-GMSK, GERAN-8PSK and UTRAN in four different Codec Types.

The Codec Identification (CoID) codes are defined to be:

FR_AMR-WB_CoID := 0x0000.1001.

UMTS_AMR-WB_CoID := 0x0000.1010.

OFR_AMR-WB_CoID := 0x0000.1100.
 OHR_AMR-WB_CoID := 0x0000.1101.

The AMR-WB Codec Types can be used in conversational speech telephony services in a number of different configurations. The set of allowed configurations is defined in Table 5.7-1.

Table 5.7-1: Allowed Configurations for the Adaptive Multi-Rate – Wideband Codec Types

Configuration → (Config-WB-Code)	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
↓ Codec Mode																
23,85					1	1										
15,85			1	1												
12,65	1	1	1	1	1	1										
8,85	1	1	1	1	1	1										
6,60	1	1	1	1	1	1										
OM	F	A	F	A	F	A										
FR_AMR-WB, OHR_AMR-WB	Y															
OFR_AMR-WB, UMTS_AMR-WB	Y	Y	Y	Y	Y	Y										

The "1" in the table indicates that the Codec Mode is included in the Active Codec Set of the Configuration.

The parameters "OM" (Optimisation Mode) define whether the indicated Configuration can be changed to any of the other Allowed ones (OM == A) or if the change is Forbidden (OM == F).

The "Y" in the table indicates, which Configuration is defined for which Codec Type.

Please note that Configurations 0 to 5 are immediately fully compatible with respect to TFO/TrFO due to the specification of Maximum Rate Control.

Table 5.7-2 defines the Coding of the "Single Codec" information element for the AMR-WB Codec Types.

Table 5.7-2: Coding of "Single Codec" for the Adaptive Multi-Rate - WideBand Codec Types

Octet	Parameter	MSB 8	7	6	5	4	3	2	1 LSB
1 m	Single Codec	Single Codec (see ITU-T Q.765.5)							
2 m	Length Indication	4							
3 m	Compat. Info	Compatibility Information							
4 m	OID	ETSI OID (See ITU-T Q.765.5 [6])							
5 m	CoID	FR_AMR-WB_CoID or UMTS_AMR-WB_CoID or OHR_AMR-WB_CoID or OFR_AMR-WB_CoID							
6 m	Config-WB	(spare)	(spare)	(spare)	(spare)	Config-WB-Code			

with "m" = mandatory

An AMR-WB speech telephony service is only possible when the whole path allows a digitally transparent transport of the AMR-WB speech parameters end to end.

Normative for GERAN terminals for FR_AMR-WB, OHR_AMR-WB and OFR_AMR-WB.

If a GERAN terminal offers one of these Codec Types in the capability list, then all AMR-WB Configurations that are defined for the offered Codec Type shall be supported by this terminal.

Normative for GERAN infrastructure for FR_AMR-WB, OHR_AMR-WB and OFR_AMR-WB.

If a GERAN infrastructure supports one of these Codec Types, then at least AMR-WB Configuration 0 shall be supported. The other AMR-WB Configurations are not normative, but optional for OFR_AMR-WB.

For information on GERAN A/Gb mode procedures for FR_AMR-WB, OHR_AMR-WB and OFR_AMR-WB (for exact details see GSM Recommendations):

The active Codec Mode is selected from the Active Codec Set (ACS) by the network (Codec Mode Command) with assistance by the mobile station (Codec Mode Request). This Codec Mode Adaptation, also termed Rate Control, can be performed every 40 ms by going one Codec Mode up or down within the ACS. The Codec Modes in uplink and downlink at one radio leg may be different. In Tandem Free Operation both radio legs (A and B) are considered for the optimal selection of the active Codec Mode in each direction (uplink A and then downlink B, respectively vice versa) by the "Distributed Rate Decision" algorithm. The worst of both radio legs determines the highest allowed Codec Mode, respectively the maximally allowed rate ("Maximum Rate Control"). All rate control commands are transmitted inband: on the radio interface, the BTS-TRAU interface and the TRAU-TRAU interface.

The Active Codec Set is configured at call setup or reconfigured during the call. It consists of three or four Codec Modes at a given time, selected from the set of allowed Configurations. The selection of the Configuration may be constrained by the network to consider resources and radio conditions.

The configurations (Active Codec Sets) in uplink and downlink are identical.

First, at start up of Tandem Free Operation both Active Codec Sets are taken into account to determine the common Active Codec Set. The set of allowed AMR-WB configurations guarantees that WB-TFO is always possible. In a later phase the Codec Lists of both radio legs may be taken into account to find the optimum configuration. For exact details see 3GPP TS 28.062. All configuration data and update protocols are transmitted inband.

The DTX scheme of the Adaptive Multi-Rate Wideband Codec Type marks with a specific SID_FIRST frame the end of a speech burst. SID_FIRST does not contain Comfort Noise parameters. This SID_FIRST starts the comfort noise generation with parameters that are calculated at receiver side from the latest received seven speech frames. A DTX hangover period needs to be applied therefore at transmitter side before sending of this SID_FIRST.

Absolutely coded SID_UPDATE frames follow about every eighth frame (160 ms) in speech pauses. SID_UPDATE frames are sent independently of the cell's TDMA frame structure and are related only to the source signal.

An ONSET frame (typically) precedes in uplink direction the beginning of a new speech burst. DTX on or off is defined by the network on a cell basis. The defined Tandem Free Operation allows the reception of FR AMR-WB DTX information for the downlink direction in all cases.

Normative for UTRAN terminals for UMTS_AMR-WB.

If an UTRAN terminal offers Codec Type UMTS_AMR-WB in the capability list, then all allowed AMR-WB Configurations shall be supported by this terminal.

Normative for UTRAN infrastructures for UMTS_AMR-WB.

If an UTRAN infrastructure supports Codec Type UMTS_AMR-WB, then at least AMR-WB Configuration 0 shall be supported. The other AMR-WB Configurations are not normative, but optional.

For information on UMTS procedures for UMTS_AMR-WB (for exact details see 3GPP TS 28.062 (TFO) and 3GPP TS 23.153 (TrFO):

The active Codec Mode is selected from the Active Codec Set (ACS) by the network. This Codec Mode Adaptation, also termed Rate Control, can be performed for the UMTS AMR-WB every 20 ms for the downlink traffic channel, but only every 40ms for the uplink traffic channel by going to another Codec Mode within the ACS. The UE selects at call setup one of the two possible phases for Codec Mode Adaptation (odd or even frames). During the call changes of the Codec Mode in uplink direction are only allowed in this selected phase. Rate Control commands received in downlink direction are considered at the next possible phase. By this definition the UMTS AMR-WB Codec Type is TFO and TrFO compatible to the FR AMR-WB, the OHR_AMR-WB and OFR AMR-WB and the UMTS AMR-WB Codec Types.

The Codec Modes in uplink and downlink at one radio leg may be different. In Tandem Free Operation or Transcoder Free Operation both radio legs (A and B) are considered for the optimal selection of the active Codec Mode in each direction (uplink A and then downlink B, respectively vice versa) by a "Distributed Rate Decision" algorithm. The worst of both radio legs determine the highest allowed Codec Mode, respectively the maximally allowed rate. All rate control commands are transmitted inband on the Iu and Nb interfaces and out of band on the radio interface.

The Active Codec Set is selected at call setup or reselected during the call. It consists of three or four Codec Modes at a given time, selected from the allowed configurations. The selection of the configuration may be constrained by the

network to consider resources and radio conditions.

The Active Codec Sets in uplink and downlink are typically identical.

At call setup with TrFO negotiation the Originating Side sends its preferred AMR-WB configuration and indicates whether it allows a change of this preferred configuration or not (included in the Codec List). The Terminating side then selects a suitable configuration from the given information and sends it back. In case the terminating side does not support TrFO a transcoder is allocated in the path at a suitable position, preferably as close as possible to the terminating side. This transcoder may be inband signalling install a Tandem Free Operation after call setup. The set of allowed AMR-WB configurations guarantees that WB-TFO is always possible. In a later phase the Codec Lists of both radio legs may be taken into account to find the optimum configuration. All configuration data and update protocols are transmitted inband on the TFO interface, but out of band within the UMTS network. For information on Tandem Free Operation see 3GPP TS 28.062 and on Transcoder Free Operation see 3GPP TS 23.153.

The SCR scheme of the Adaptive Multi-Rate WideBand Codec Types mark with a specific SID_FIRST frame the end of a speech burst. SID_FIRST does not contain Comfort Noise parameters. This SID_FIRST starts the comfort noise generation with parameters that are calculated at receiver side from the latest received seven speech frames. A DTX hangover period needs to be applied therefore at transmitter side before sending of this SID_FIRST.

Absolutely coded SID_UPDATE frames follow about every eighth frame (160 ms) in speech pauses. SID_UPDATE frames are sent independently of the cell's timing structure and are related only to the source signal.

An ONSET frame does (typically) not exist in UMTS networks, but may be received in TFO from the distant partner. It marks the beginning of a speech burst. "SCR on" is always defined by the network. The defined Tandem Free Operation and Transcoder Free Operation allows the reception of AMR-WB SCR information for the downlink direction in all cases.

The SCR scheme of the UMTS AMR-WB Codec Type is fully compatible to the DTX schemes of FR AMR-WB, OHR AMR-WB and OFR AMR-WB.

The exact details of these Codec Types and their related procedures (DTX, Rate Control, etc) are described in the respective standard documentation.

5.8 MuMe Dummy Codec (3G.324M)

The Codec Identification (CoID) code is defined to be: MuMe_CoID:= 0x1111.1111.

The MuMe codec has one additional mandatory parameter:

B/W Multiplier, BWM: eight bits.

This defines the required bandwidth for the bearer; the value is a factor of 64K b/s when not equal to 0. When equal to zero then a 32k b/s.

The "Single Codec" information element consists of 6 octets in case of the MuMe Dummy Codec (table 5.8):

Table 5.8: Coding of "Single Codec" for the MuMe Dummy Codec Type

Octet	Parameter	MSB 8	7	6	5	4	3	2	1 LSB
1 m	Single Codec	Single Codec (see ITU-T Q.765.5)							
2 m	Length Indication	4							
3 m	Compat. Info	Compatibility Information							
4 m	OID	ETSI OID (See ITU-T Q.765.5 [6])							
5 m	CoID	MuMe_CoID							
6 m	BWM	BandWidth Multiplier – see note1							

with "m" = mandatory

Note 1:

BWM == 0 => 32Kb/s

BWM == 1-255 => factor n (multiplier of 64Kb/s)

The procedures for use of this codec are defined in TS 23.172 [13].

This MuMe Dummy codec type is only for use in Core Network OoBTC procedures it shall NOT be used across the radio interface.

The MuMe Dummy codec indicates that an Unrestricted multimedia path (UDI) is required, subsequent codec negotiation may occur within this path using MuMe protocols, e.g. H.324M. There are no encoding properties or codec specifications associated to this codec type; it is purely an indication for a MuMe pipe.

5.9 MuMe2 Dummy Codec (3G.324M2)

The Codec Identification (CoID) code is defined to be: MuMe2_CoID:= 0x1111.1110. Otherwise, the Coding is identical to the MuME Dummy Codec described in Clause 5.8.

The Procedural description provided for MuME Dummy Codec in Clause 5.8 is also applicable for the MuMe2 Dummy Codec. The MuMe2 Dummy Codec is used in core network procedures to indicate that a service change to multimedia was indicated by the network. The procedures for use of this codec are defined in TS 23.172 [13].

5.10 Codec Extension

The Codec Identification (CoID) code is defined to be: Codec_Extension_CoID:= 0x0000.1111 in the "long form" and 0x1111 = 0xFh in the "short form".

In TFO, see 3GPP TS 28.062 [7] and in AoIP, see 3GPP TS 48.008 [23] the Codec Lists use in general the short form (4 bits) for the Codec Identifier. In order to allow future extensions of this Codec Lists beyond 16 Codec Types the "Codec_Extension" is defined. These Codec Lists may contain a certain CoID in the range [0x0h, 0xEh] or they may contain the so called "Codec_Extension" (0xFh), in which case the real Codec Type follows in the next octet in its long form (8 bits).

5.11 CSData Dummy Codec (AoIP)

The Codec Identification (CoID) code is defined to be: CSData_CoID:= 0x1111.1101.

The CSData Dummy Codec has one mandatory parameter of one octet length, for details see TS 48.008 [23].

6 Codec List for the Call Control Protocol

For call control on the air interface the Codec Lists need to be specified for each radio access technology separately, because it can not be expected that an UE supports the same Codec Types in different radio access technologies.

3GPP TS 24.008 [9] defines the call control signalling and how to use the "Supported Codec List Information Element" (IE). It contains Codec Lists (in form of Codec Bitmaps) for each supported radio access technology (identified by a SysID).

The coding of this IE is given here. It is also used for TFO in 3GPP TS 28.062 [7].

6.1 System Identifiers for GSM and UMTS

The system identifiers for the radio access technologies supported by this specification are:

SysID for GSM: 0x0000.0000 (bit 8 .. bit 1)

SysID for UMTS: 0x0000.0100 (bit 8 .. bit 1)

These values are selected in accordance with [7] (3GPP TS 28.062).

6.2 Codec Bitmap

The Codec Types are coded in the first and second octet of the Codec List Bitmap as follows:

8	7	6	5	4	3	2	bit 1	
TDMA EFR	UMTS AMR 2	UMTS AMR	HR AMR	FR AMR	GSM EFR	GSM HR	GSM FR	Octet 1
bit 16	15	14	13	12	11	10	bit 9	
(reserved)	(reserved)	OHR AMR-WB	OFR AMR-WB	OHR AMR	UMTS AMR-WB	FR AMR-WB	PDC EFR	Octet 2

A Codec Type is supported, if the corresponding bit is set to "1". All reserved bits shall be set to "0".

6.3 Selected Codec Type

The Selected Codec Type in a BICC-based OoBTC negotiation is coded as shown in Table 6.3-1. The same coding is used also in 3GPP TS 28.062 [7].

Table 6.3-1: Coding of the selected Codec_Type (long form)

Bit 8...Bit 1 CoID	Codec_Type	Name
0000.0000	GSM Full Rate (13.0 kBit/s)	GSM FR
0000.0001	GSM Half Rate (5.6 kBit/s)	GSM HR
0000.0010	GSM Enhanced Full Rate (12.2 kBit/s)	GSM EFR
0000.0011	Full Rate Adaptive Multi-Rate	FR AMR
0000.0100	Half Rate Adaptive Multi-Rate	HR AMR
0000.0101	UMTS Adaptive Multi-Rate	UMTS AMR
0000.0110	UMTS Adaptive Multi-Rate 2	UMTS AMR 2
0000.0111	TDMA Enhanced Full Rate (7.4 kBit/s)	TDMA EFR
0000.1000	PDC Enhanced Full Rate (6.7 kBit/s)	PDC EFR
0000.1001	Full Rate Adaptive Multi-Rate WideBand	FR AMR-WB
0000.1010	UMTS Adaptive Multi-Rate WideBand	UMTS AMR-WB
0000.1011	8PSK Half Rate Adaptive Multi-Rate	OHR AMR
0000.1100	8PSK Full Rate Adaptive Multi-Rate WideBand	OFR AMR-WB
0000.1101	8PSK Half Rate Adaptive Multi-Rate WideBand	OHR AMR-WB
0000.1110	spare, for future use	
0000.1111	Reserved for Codec_Extension	for AoIP and TFO, not for OoBTC
Up to 1111.1100	spare for future use	
1111.1101	Reserved for CSData dummy Codec Type	for AoIP, not for OoBTC
1111.1110	Reserved forMuMe2 dummy Codec Type NOTE: codec not to be used across radio interface.	MuMe2
1111.1111	Reserved forMuMe dummy Codec Type NOTE: codec not to be used across radio interface.	MuMe

7 3GPP Codecs for OoBTC in a SIP-I -based Circuit Switched Core Network

7.1 Overview

In a SIP-I -based Circuit Switched Core Network, as specified in 3GPP TS 23.231 [14], SDP (IETF RFC 4566 [19]) and SDP offer-answer procedures (IETF RFC 3264 [16]) are applied for Out of Band Transcoder Control as specified in 3GPP TS 23.153 [8].

Table 7.1.1 lists the supported 3GPP Speech Codecs for a SIP-I -based Circuit Switched Core Network.

Table 7.1.1 Supported 3GPP Codecs 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	IETF RFC 5993 [22]	Useful if an A-interface over IP is attached. or TFO is used	Optional
audio/PCMA	IETF RFC 3551 [17]	ITU-TG.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

7.2 AMR

AMR (FR_AMR, HR_AMR, OHR_AMR, UMTS_AMR and UMTS_AMR2) shall be encoded in SDP according to the MIME registration in IETF RFC 4867 [21]. The SDP offer-answer related rules in this RFC apply.

The bandwidth efficient mode of RFC 4867 shall be used. To offer the bandwidth-efficient mode, the octet-align parameter should be omitted in SDP.

The AMR Codec Types can be used in conversational speech telephony services in a number of different configurations. Configuration related procedures in Clause 5.4 shall be applied also within a SIP-I based CS CN. The set of preferred configurations is defined in TS 28.062 [7], Table 7.11.3.1.3-2. The configuration is encoded in SDP in the mode-set parameter.

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 (MSC-S and MGW) support all AMR modes for maximum interoperability.

To offer the AMR codec in different configurations, the AMR codec may be included several times with different configurations in an SDP m-line.

A core network node performing a transcoding free interworking towards an A-Interface (TFO towards any A-Interface or TrFO towards an IP-based A-Interface) shall provide the parameters "mode-change-period=2" and "mode-change-neighbour=1" in offer or answer. The parameter "mode-change-capability=2" shall be included by all other CS CN

nodes in the offer to ensure interoperability unless they received an offer from other nodes without this parameter and do not transcode.

7.3 AMR-WB

AMR-WB (FR_AMR-WB, OHR_AMR-WB, OFR_AMR-WB, UMTS_AMR-WB) shall be encoded in SDP according to the MIME registration in IETF RFC 4867 [21]. The SDP offer-answer related rules in this RFC apply.

The bandwidth efficient mode of RFC 4867 shall be used. To offer the bandwidth-efficient mode, the octet-align parameter should be omitted in SDP.

The AMR-WB Codec Types can be used in conversational speech telephony services in a number of different configurations. Configuration related procedures in Clause 5.7 shall be applied also within a SIP-I based CS CN. The set of configurations is defined in Table 5.7-1. The configuration is encoded in SDP in the mode-set parameter.

One of these configurations, **Config-WB-Code 0**, shall be supported by all nodes supporting the AMR-WB codec in a circuit switched core network to ensure interoperability.

However, it is recommended that a node in the core network supports all AMR-WB modes for maximum interoperability.

To offer the AMR-WB codec in different configurations, the AMR-WB codec may be included several times with different configurations in an SDP m-line.

A core network node performing a transcoding free interworking towards an A-Interface (TFO towards any A-Interface or TrFO towards an IP-based A-Interface) shall provide the parameters "mode-change-period=2" and "mode-change-neighbour=1" in offer or answer. The parameter "mode-change-capability=2" shall be included by all other CS CN nodes in the offer to ensure interoperability unless they received an offer from other nodes without this parameter and do not transcode

7.4 GSM_EFR

GSM_EFR shall be encoded in SDP using either the fixed payload type assigned in IETF RFC 3551 [17] or a dynamic payload type described according to the MIME registration in IETF RFC 3551 [17]

The GSM_EFR standard comprises a DTX scheme with VAD, SID frames and Comfort Noise generation that is automatically included in this SDP negotiation. For User Plane details see 3GPP TS 26.102 [24]. No other DTX scheme shall be negotiated in SDP for GSM_EFR.].

7.5 GSM_FR

GSM_FR shall be encoded in SDP using either the fixed payload type assigned in IETF RFC 3551 [17] or a dynamic payload type described according to the MIME registration in IETF RFC 3551 [17]

The GSM_FR standard comprises a DTX scheme with VAD, SID frames and Comfort Noise generation that is automatically included in this SDP negotiation. For User Plane details see 3GPP TS 26.102 [24]. No other DTX scheme shall be negotiated in SDP for GSM_FR.].

7.6 GSM_HR

GSM_HR shall be encoded in SDP according to the MIME registration in [22]. GSM_HR shall be encoded in SDP using a dynamic payload type described according to the MIME registration in [22]. The options specified in [22] are not applied inside the Circuit Switched Core Network and not across the A-Interface, 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.

The GSM_HR standard comprises a DTX scheme with VAD, SID frames and Comfort Noise generation that is automatically included in this SDP negotiation. For User Plane details see [22] and 3GPP TS 26.102 [24]. No other DTX scheme shall be negotiated in SDP for GSM_HR.

7.7 PCM

PCMU and PCMA shall be encoded in SDP using either the fixed payload type assigned in IETF RFC 3551 [ee] or a dynamic payload type described according to the MIME registration in IETF RFC 3551 [ee].

7.8 Telephone-Event

Telephony-Event shall be encoded in SDP according to the MIME registration in IETF RFC 4733 [20].

The MIME type audio/telephone-event in IETF RFC 4733 [20] with default events and default rate shall be used to encode DTMF. Therefore, the rate and event parameters do not need to be supplied.

Annex A (informative): Example Supported Codec List for UMTS

This Annex gives some informative examples how the Codec List for UMTS may look like for the OoBTC protocol in a BICC-based Circuit Switched Core Network.

In this example the UMTS Circuit Switched Core Network does support: UMTS AMR2(set1), (GSM) FR AMR(set1) and (GSM) HR AMR(set1) and GSM EFR. It supports PCM, i.e. ITU-T G.711, here in the Alaw version, with transcoding. It may support also UMTS_AMR(set7), GSM FR, and GSM_HR (not included in the list).

One "Supported Codec List" (with arbitrarily selected Codec Type preference) could look at Originating side like:

Octet	Parameter	MSB 8	7	6	5	4	3	2	1 LSB
1	Codec List	Codec List (see ITU-T Q.765.5)							
2	Length Indication (LI)	30							
3	Compat. Info	Compatibility Information							
4	Single Codec	Single Codec (see ITU-T Q.765.5)							
5	LI	6							
6	Compat. Info	Compatibility Information							
7	OID	ETSI OID (See ITU-T Q.765.5 [6])							
8	CoID	UMTS_AMR2_CoID							
9	ACS (set1)	12.2(1)	10.2(0)	7.95(0)	7.40(1)	6.70(0)	5.90(1)	5.15(0)	4.75(1)
10	SCS (set1)	12.2(1)	10.2(0)	7.95(0)	7.40(1)	6.70(0)	5.90(1)	5.15(0)	4.75(1)
11	MACS	(spare)	(spare)	(spare)	(spare)	OM(0)	MACS(4)		
12	Single Codec	Single Codec (see ITU-T Q.765.5)							
13	LI	3							
14	Compat. Info	Compatibility Information							
15	OID	ITU-T OID (See ITU-T Q.765.5 [6])							
16	CoID	Codec Identifier for PCM Alaw 64kbps							
17	Single Codec	Single Codec (see ITU-T Q.765.5)							
18	LI	6							
19	Compat. Info	Compatibility Information							
20	OID	ETSI OID (See ITU-T Q.765.5 [6])							
21	CoID	FR_AMR_CoID							
22	ACS (set1)	12.2(1)	10.2(0)	7.95(0)	7.40(1)	6.70(0)	5.90(1)	5.15(0)	4.75(1)
23	SCS (set1)	12.2(1)	10.2(0)	7.95(0)	7.40(1)	6.70(0)	5.90(1)	5.15(0)	4.75(1)
24	MACS	(spare)	(spare)	(spare)	(spare)	OM(0)	MACS(4)		
25	Single Codec	Single Codec (see ITU-T Q.765.5)							
26	LI	6							
27	Compat. Info	Compatibility Information							
28	OID	ETSI OID (See ITU-T Q.765.5 [6])							
29	CoID	HR_AMR_CoID							
30	ACS (set1)	(spare)	(spare)	7.95(0)	7.40(1)	6.70(0)	5.90(1)	5.15(0)	4.75(1)
31	SCS (set1)	(spare)	(spare)	7.95(0)	7.40(1)	6.70(0)	5.90(1)	5.15(0)	4.75(1)
32	MACS	(spare)	(spare)	(spare)	(spare)	OM(0)	MACS(3)		
33	Single Codec	Single Codec (see ITU-T Q.765.5)							
34	LI	3							
35	Compat. Info	Compatibility Information							
36	OID	ETSI OID (See ITU-T Q.765.5 [6])							
37	CoID	EFR_CoID							

The Terminating Side selects one of the Codec Types and returns it, together with the selected codec attributes.

The AMR Codec Types may have very similar, if not identical codec attributes at Originating side. The UMTS as Originating side can, however, already decide, which configuration would be preferred in case the Terminating side is UMTS or GSM. A GSM Circuit Switched Core Network as Originating side can not offer UMTS AMR (unless it provides local transcoding) and the Codec attributes for FR AMR and HR AMR may be quite different.

Annex B (informative): Change history

Change history							
Date	TSG SA#	TSG Doc.	CR	Rev	Subject/Comment	Old	New
12-2000	10	SP-000576	004		Introduction of Codec Type Bit-Map for Codec Negotiation	3.0.0	4.0.0
12-2000	10	SP-000576	005		Introduction of Selected Codec Type for Codec Negotiation	3.0.0	4.0.0
12-2000	10	SP-000576	006		Clarification for the use of the Codec List Information Element	3.0.0	4.0.0
03-2001	11	SP-010104	007		Simplification of the Optimisation Mode Field	4.0.0	4.1.0
03-2001	11	SP-010199	008	3	Introduction of UMTS_AMR_2	4.0.0	4.1.0
03-2001	11	SP-010199	009		Introduction of AMR Wideband	4.1.0	5.0.0
03-2002	15	SP-020078	015		Introduction of GERAN-8PSK Codec Types into Codec List	5.0.0	5.1.0
03-2002	15	SP-020078	017		Introduction of codepoint for Dummy Codec for CS Multi Media (3G 324M)	5.0.0	5.1.0
06-2002	16	SP-020223	014	2	UMTS_AMR2 is default Codec Type in all terminals of Rel-4 and onwards	5.1.0	5.2.0
09-2002	17	SP-020437	020	1	TrFO-Signalling for allowed AMR-WB Configurations	5.2.0	5.3.0
12-2002	18	SP-020690	021	1	Correction of uplink SCR activation for UMTS AMR	5.3.0	5.4.0
12-2002	18	SP-020690	022		Correction to the Codec ID Table	5.3.0	5.4.0
09-2004	25	SP-040646	028	1	Correction of Size and Reference of MuMe Codec	5.4.0	5.5.0
09-2004	25	SP-040646	023	2	Harmonisation of AMR Configurations	5.5.0	6.0.0
09-2004	25	SP-040646	025	1	Error Fixes	5.5.0	6.0.0
09-2004	25	SP-040646	029	1	Correction of Size and Reference of MuMe Codec	5.5.0	6.0.0
12-2004	26	SP-040845	032		TFO/TrFO Compatibility of UMTS_AMR and UMTS_AMR2	6.0.0	6.1.0
12-2004	26	SP-040847	036	1	Clarifications for AMR	6.0.0	6.1.0
03-2006	31	SP-060008	0037		3G-324.M2 Codec for Indication of Network-Initiated Service Change	6.1.0	6.2.0
06-2007	36				Version for Release 7	6.2.0	7.0.0
09-2008	41	SP-080475	0038	2	Addition of CS over IP User Plane	7.0.0	8.0.0
12-2008	42	SP-080678	0039	2	Corrections to CS over IP User Plane	8.0.0	8.1.0
12-2009	46				Version for Release 9	8.1.0	9.0.0
03-2011	51	SP-110034	0041		Correction of reference for GSM-HR payload format	9.0.0	9.1.0
03-2011	51				Version for Release 10	9.1.0	10.0.0
09-2012	57				Version for Release 11	10.0.0	11.0.0
09-2014	65				Version for Release 12	11.0.0	12.0.0
12-2015	70				Version for Release 13	12.0.0	13.0.0

History

Document history		
V13.0.0	January 2016	Publication