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**Digital cellular telecommunications system (Phase 2+) (GSM);  
Universal Mobile Telecommunications System (UMTS);  
Speech codec list for GSM and UMTS  
(3GPP TS 26.103 version 16.0.0 Release 16)**



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# Contents

Intellectual Property Rights .....	2
Legal Notice .....	2
Modal verbs terminology.....	2
Foreword.....	4
1 Scope .....	5
2 Normative references .....	5
3 Definitions and Abbreviations.....	7
3.1 Definitions .....	7
3.2 Abbreviations .....	7
4 General .....	8
5 3GPP Codec List for OoBTC in a BICC-based Circuit Switched Core Network and for AoIP .....	10
5.1 GSM Full Rate Codec Type (GSM_FR) .....	10
5.2 GSM Half Rate Codec Type (GSM_HR) .....	10
5.3 GSM Enhanced Full Rate Codec Type (GSM_EFR) .....	10
5.4 Five Adaptive Multi-Rate Codec Types (FR_AMR, HR_AMR, UMTS_AMR, UMTS_AMR2, OHR_AMR) .....	11
5.5 TDMA Enhanced Full Rate Codec Type (TDMA_EFR) .....	14
5.6 PDC Enhanced Full Rate Codec Type (PDC_EFR) .....	14
5.7 Four Adaptive Multi-Rate Wideband Codec Types (FR_AMR-WB, UMTS_AMR-WB, OFR_AMR-WB, OHR_AMR-WB) .....	14
5.7A EVS Codec Type (UMTS_EVS) .....	17
5.8 MuMe Dummy Codec (3G.324M) .....	20
5.9 MuMe2 Dummy Codec (3G.324M2) .....	20
5.10 Codec Extension .....	20
5.11 CSData Dummy Codec (AoIP) .....	21
6 Codec List for the Call Control Protocol .....	21
6.1 System Identifiers for GSM and UMTS .....	21
6.2 Codec Bitmap .....	21
6.3 Selected Codec Type .....	22
7 3GPP Codecs for OoBTC in a SIP-I -based Circuit Switched Core Network .....	23
7.1 Overview .....	23
7.2 AMR .....	23
7.3 AMR-WB .....	24
7.4 GSM_EFR .....	24
7.5 GSM_FR .....	24
7.6 GSM_HR .....	24
7.7 PCM .....	25
7.8 Telephone-Event .....	25
7.9 EVS .....	25
<b>Annex A (informative): Example Supported Codec List for UMTS .....</b>	<b>26</b>
<b>Annex B (informative): Change history .....</b>	<b>28</b>
History .....	29

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# Foreword

This Technical Specification has been produced by the 3<sup>rd</sup> 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:

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# 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.

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# 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".
- [25] 3GPP TS 26.441: "Codec for Enhanced Voice Services (EVS); General overview".
- [26] 3GPP TS 26.442: "Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point)".
- [27] 3GPP TS 26.443: "Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)".
- [28] 3GPP TS 26.444: "Codec for Enhanced Voice Services (EVS); Test Sequences".
- [29] 3GPP TS 26.445: "Codec for Enhanced Voice Services (EVS); Detailed algorithmic description".
- [30] 3GPP TS 26.446: "Codec for Enhanced Voice Services (EVS); Adaptive Multi-Rate - Wideband (AMR-WB) backward compatible functions".
- [31] 3GPP TS 26.447: "Codec for Enhanced Voice Services (EVS); Error concealment of lost packets".
- [32] 3GPP TS 26.448: "Codec for Enhanced Voice Services (EVS); Jitter buffer management".
- [33] 3GPP TS 26.449: "Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) aspects".
- [34] 3GPP TS 26.450: "Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)".
- [35] 3GPP TS 26.451: "Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)".
- [36] 3GPP TR 26.952: "Codec for Enhanced Voice Services (EVS); Performance Characterization".
- [37] 3GPP TS 26.453: "Codec for Enhanced Voice Services (EVS); Speech codec frame structure".
- [38] 3GPP TS 26.454: "Codec for Enhanced Voice Services (EVS); Interface to Iu, Uu, Nb and Mb".
- [39] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".
- [40] 3GPP TS 26.452: "Codec for Enhanced Voice Services (EVS); ANSI C code; Alternative fixed-point using updated basic operators".

## 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
CMR	Codec Mode Request (for AMR and AMR-WB)
CoID	Codec IDentifier
CS	Circuit Switched
CSDData	Circuit Switched Data and Fax dummy Codec
DTX	Discontinuous Transmission (of speech and audio signals, synonym to SCR)
EVS	Enhanced Voice Services
EVS-CMR	Codec Mode Request for EVS
FB	Fullband (audio bandwidth)
GSM	Global System for Mobile communication
MuMe	Multi-Media
NB	Narrowband (audio bandwidth)
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
SWB	Super Wideband (audio bandwidth)
SysID	System Identifier
TDMA	Time Division Multiple Access (synonym for ...)
TFO	<b>T</b> andem <b>F</b> ree <b>O</b> peration (also sometimes called "Transcoder-Through" or "Codec-Bypass")
TrFO	<b>T</b> ranscoder <b>F</b> ree <b>O</b> peration
TX	Transmit
UMTS	Universal Mobile Telecommunications System
WB	Wideband (audio bandwidth)

## 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

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

	<b>Codec Extension</b>	<b>UMTS EVS</b>	<b>OHR AMR-WB</b>	<b>OFR AMR-WB</b>	<b>OHR AMR</b>	<b>UMTS AMR-WB</b>	<b>FR AMR-WB</b>	<b>PDC EFR</b>
<b>CoID</b>	<b>0x0F</b>	<b>0x0E</b>	<b>0x0D</b>	<b>0x0C</b>	<b>0x0B</b>	<b>0x0A</b>	<b>0x09</b>	<b>0x08</b>
<b>GERAN GMSK</b>	reserved	not defined	not defined	not defined	not defined	not possible	yes3 modi	not defined
<b>GERAN 8PSK</b>	reserved	not defined	yes, 3 modi	yes, 3 modi	yes 1..4 modi	not possible	not defined	not defined
<b>UTRAN</b>	reserved	yes	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

<b>Hexadecimal Notation</b>	<b>Binary Notation</b>	<b>Codec Name</b>	<b>Remark</b>
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	UMTS_EVS	
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.