ETSI TS 126 130 V18.0.0 (2024-05)

TECHNICAL SPECIFICATION

Universal Mobile Telecommunications System (UMTS);
LTE;
5G;
Speech/Audio Codec RTP Payload Format Conformance for UE Testing
(3GPP TS 26.130 version 18.0.0 Release 18)

Reference

DTS/TSGS-0426130vi00

Keywords

5G,LTE,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 - APE 7112B
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° w061004871

Important notice

The present document can be downloaded from:
https://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 prevailing version of an ETSI deliverable is the one made publicly available in PDF format at www.etsi.org/deliver.

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 https://portal.etsi.org/TB/ETSIDeliverableStatus.aspx

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

If you find a security vulnerability in the present document, please report it through our Coordinated Vulnerability Disclosure Program:
https://www.etsi.org/standards/coordinated-vulnerability-disclosure

Intellectual Property Rights

Essential patents

IPRs essential or potentially essential to normative deliverables may have been declared to ETSI. The declarations pertaining to these essential IPRs, if any, are 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 Directives including the ETSI IPR Policy, no investigation regarding the essentiality of IPRs, 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.

Trademarks

The present document may include trademarks and/or tradenames which are asserted and/or registered by their owners. ETSI claims no ownership of these except for any which are indicated as being the property of ETSI, and conveys no right to use or reproduce any trademark and/or tradename. Mention of those trademarks in the present document does not constitute an endorsement by ETSI of products, services or organizations associated with those trademarks.

DECT™, **PLUGTESTS™**, **UMTS™** and the ETSI logo are trademarks of ETSI registered for the benefit of its Members. **3GPP™** and **LTE™** are trademarks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners. **oneM2M™** logo is a trademark of ETSI registered for the benefit of its Members and of the oneM2M Partners. **GSM**® and the GSM logo are trademarks registered and owned by the GSM Association.

Legal Notice

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. These shall be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between 3GPP and ETSI identities can be found under https://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

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.

In the present document, modal verbs have the following meanings:

shall indicates a mandatory requirement to do something

shall not indicates an interdiction (prohibition) to do something

The constructions "shall" and "shall not" are confined to the context of normative provisions, and do not appear in Technical Reports.

The constructions "must" and "must not" are not used as substitutes for "shall" and "shall not". Their use is avoided insofar as possible, and they are not used in a normative context except in a direct citation from an external, referenced, non-3GPP document, or so as to maintain continuity of style when extending or modifying the provisions of such a referenced document.

should indicates a recommendation to do something

should not indicates a recommendation not to do something

may indicates permission to do something

need not indicates permission not to do something

The construction "may not" is ambiguous and is not used in normative elements. The unambiguous constructions "might not" or "shall not" are used instead, depending upon the meaning intended.

can indicates that something is possible

cannot indicates that something is impossible

The constructions "can" and "cannot" are not substitutes for "may" and "need not".

will indicates that something is certain or expected to happen as a result of action taken by an agency the behaviour of which is outside the scope of the present document

will not indicates that something is certain or expected not to happen as a result of action taken by an agency the behaviour of which is outside the scope of the present document

might indicates a likelihood that something will happen as a result of action taken by some agency the behaviour of which is outside the scope of the present document

might not indicates a likelihood that something will not happen as a result of action taken by some agency the behaviour of which is outside the scope of the present document

In addition:

is (or any other verb in the indicative mood) indicates a statement of fact

is not (or any other negative verb in the indicative mood) indicates a statement of fact

The constructions "is" and "is not" do not indicate requirements.

Introduction

The present document specifies requirements and test methods to verify correct implementations of the RTP payload format for 3GPP codecs in UE. The focus is on conversational services in LTE, NR and WLAN terminals when used to provide narrowband, wideband, super-wideband or fullband telephony.

1 Scope

The present document is applicable to any terminal capable of supporting narrowband, wideband, super-wideband or fullband telephony, either as a stand-alone service or as the telephony component of a multimedia service. The present document specifies requirements and test methods to verify correct implementations of the RTP payload format for 3GPP codecs in UE. The focus is on conversational services in LTE, NR and WLAN terminals when used to provide narrowband, wideband, super-wideband or fullband telephony.

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 TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 26.132: "Speech and video telephony terminal acoustic test specification".

[3] 3GPP TS 26.139: "Real-time Transport Protocol (RTP) / RTP Control Protocol (RTCP) verification procedures".

[4] 3GPP TS 34.229-1: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); User Equipment (UE) conformance specification; Part 1: Protocol conformance specification".

[5] 3GPP TS 34.229-2: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); User Equipment (UE) conformance specification; Part 2: Implementation Conformance Statement (ICS) specification".

[6] 3GPP TS 34.229-3: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); User Equipment (UE) conformance specification; Part 3: Abstract test suite (ATS)".

[7] 3GPP TS 34.229-5: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); User Equipment (UE) conformance specification; Part 5: Protocol conformance specification using 5G System (5GS)".

[8] IETF RFC 4867: "RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".

[9] 3GPP TS 26.445: "Codec for Enhanced Voice Services (EVS);Detailed algorithmic description".

[10] ITU-T Recommendation P.501 (06/2015): "Test signals for use in telephonometry".

[11] 3GPP TS 26.101: "Mandatory speech codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec frame structure".

[12] 3GPP TS 26.190: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions".

[13] 3GPP TS 26.114: IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction.

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

3 Definitions of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the terms given in TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in TR 21.905 [1].

Incoming CMR: CMR in RTP packet received by the DUT

Receiving: Link from test simulator to DUT

Sending: Link from DUT to test simulator

3.2 Symbols

Void.

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [1].

DUT	Device Under Test
MO	Mobile Originated
MT	Mobile Terminated
SS	System Simulator
TX	Transmission
VoNR	5G capable DUT supporting voice over NR

4 Interfaces

4.1 General

By default, electrical interfaces should be used for testing. Acoustic interfaces may be used as an alternative. The actual interfaces used in sending and receiving shall be documented.

Test cases may be implemented either by injecting specific packet information (including CMR requests) in a live call or assuming a PCAP player scenario. To define the test setup, one approach is to reuse the test setup defined in TS 26.132 [2] (defining acoustical and electrical interfaces) where only LTE, WLAN and NR apply.

If no acoustic measurement is required, the test setup in TS 26.139 [3] can be reused for RTP /RTCP "data injection" with either active or passive test instrument.

4.2 Acoustic interfaces

See clause 5.1 in TS 26.132 [2].

4.3 Electrical interfaces

See clause 5.2 in TS 26.132 [2].

5 Test setup

5.1 General

Similar to clause 5.1 in TS 26.139 [3]:

- The "system under test" is the device (DUT). or software to be tested
- The "test instrument" is the equipment used to place an IMS call including configurable SIP and SDP parameters and to collect test data (SDP/RTP/RTCP output from the system under test). It can extract, calculate, and store information to check requirements for all the test described in the present document.
- The "data injection" is the device or equipment used to generate RTP/RTCP data sent to the system under test.

NOTE 1: 'data injection' may be collocated or integrated with the test instrument.

NOTE 2: Data collection' 'SDP/RTP / RTCP receiver) may be performed either by "test instrument" or collocated with "data injection" equipment.

5.2 Setup for terminals

See clause 5.2 in TS 26.132 [2] for the setup definition (Handset UE, Headset UE, Desktop-mounted hands-free UE, Hand-held hands-free UE, Softphone UE).

5.3 Setup of the electrical interfaces of test equipment

See clause 5.2 in TS 26.132 [2].

For receive tests, where a user operated volume control is provided, the measurements [should/shall] be carried out at the nominal setting of the volume control.

5.4 Accuracy of test equipment

See clause 5.3 in TS 26.132 [2].

5.5 Test signals

See clause 5.4 in TS 26.132 [2].

5.6 Environmental conditions

See clause 6 in TS 26.132 [2].

For LTE, WLAN, and NR connections, an RF shielded room should be one way to achieve these requirements on block error rate and jitter. Otherwise, care should be taken with potential sources of radio interference and their impact.

5.7 System simulator conditions

Test applicability and test result may depend on the SIM card or eSIM configuration. Depending on MNC and MCC, the DUT could be set in test mode or load specific parameters. Unless otherwise stated, a new call shall be established for each test case.

6 RTP Payload Format Conformance for AMR

6.1 Applicability

The requirements and test methods in this clause shall apply when UE is used to provide narrowband telephony, either as a stand-alone service, or as part of a multimedia service.

6.2 SDP tests

6.2.1 MO call

Requirement:

Requirements on the SDP offer from the DUT are for further study.

NOTE: SDP testing is already considered in [4,5,6,7].

Test method:

A call is established by the DUT. The SDP offer from the DUT shall be documented.

6.2.2 MT calls

Requirement:

Requirements on the SDP answer from the DUT are for further study.

NOTE: Verification of b=AS is for further study. Purpose is to check compliance to [13] Annex K: b=AS is expected to be set according to the highest allowed codec mode and other parameters (IP version, ptime, bandwidth efficient or octet-align mode) in the SDP answer.

Test method:

Every call is established by the system simulator using one AMR payload type in the SDP offer. The system simulator shall configure the SDP offer according to Table 6.2.2-1 for the bandwidth-efficient mode of AMR and Table 6.2.2-2 for the octet-aligned mode of AMR.

For each SDP offer, the SDP answer from the DUT shall be documented and the corresponding RTP and RTCP streams shall be recorded.

The test signal to be used for the measurements shall be the same in both directions as specified below depending on test cases:

- speech1: the British-English single talk sequence described in ITU-T Recommendation P.501 [10].
- silence1: test signal forced to silence (same length as speech1)
- speech2: first sentence of the British-English single talk sequence described in ITU-T Recommendation P.501 [10] shortened to 2.4sec by selecting samples in interval [0.5sec, 2.9sec], repeated 16 times
- silence2: test signal forced to silence (same length as speech2)
- speech3: 3 repeats of the Composite Source Signal (CSS) according to ITU-T Recommendation P.501 [10] followed by a speech signal of 160s as in clause 7.10.4.2 of TS 26.132 [2]
- silence3: test signal forced to silence (same length as speech3)

In sending, for acoustic interfaces, the test signal level should be -4,7 dBPa measured at the MRP; for electrical interfaces, the active speech level of the signal should be calibrated to -60 dBV for analogue and to -16 dBm0 for digital connections.

In receiving, the test signal level should be -16 dBm0 measured at the digital reference point or the equivalent analogue point.

Table 6.2.2-1: List of test cases for MT calls for given SDP offer (bandwidth-efficient)

Test case	Parameter the SDP offer	Mode-set in SDP answer	Input to DUT	Input to system simulator
amr-0	mode-set=0	0	speech1	silence1
amr-1	mode-set=1	1	speech1	silence1
amr-2	mode-set=2	2	speech1	silence1
amr-3	mode-set=3	3	speech1	silence1
amr-4	mode-set=4	4	speech1	silence1
amr-5	mode-set=5	5	speech1	silence1
amr-6	mode-set=6	6	speech1	silence1
amr-7	mode-set=7	7	speech1	silence1
amr-oo	mode-set not present (open offer)	See NOTE 1	speech1	silence1
amr-cmr1	mode-set not present (open-offer)	See NOTE 1	speech2 (see NOTE 2)	speech2
amr-cmr2	mode-set=0,2,4,7	0,2,4,7	speech2 (see NOTE 2)	speech2
amr-qbit	mode-set=7	7	silence1	speech1 (see NOTE 3)
amr-imp	mode-set=7	7	silence3 (see NOTE 4)	speech3
NOTE 1: The DUT may restrict the mode-set in its answer to a restricted set of AMR modes, e.g., to 0,2,4,7 or a further subset due to configuration. NOTE 2: The system simulator inserts CMRs in the RTP stream in this test case. NOTE 3: The system simulator forces Q bit to 0 in all packets of the RTP stream in test case 'amr-qbit'. NOTE 4: The system simulator inserts packet impairments (using profile in Annex A) in the RTP stream in this test case.				

Table 6.2.2-2: List of test cases for MT calls for given SDP offer (octet-aligned)

Test case	Parameter the SDP offer	Mode-set in SDP answer	Input to DUT	Input to system simulator
amr-octet-7	mode-set=7; octet-align=1	7	speech1	silence1

6.3 RTP tests

6.3.1 Test cases in sending

6.3.1.1 FT verification

Requirement:

The FT entry in the ToC shall be match the active speech bit rate index according to RFC 4867 [9].

The FT entry in the ToC shall be match the SID bit rate index according to RFC 4867 [9].

Test method:

- For each test case amr-0 to amr-7 (see clause 6.2.2), the ToC field is extracted for recorded RTP stream from the DUT and compared with the respective AMR mode (0 to 7) for active speech packets or SID bitrate for SID packets.

NOTE: The value of FT is defined in Table 1a in 3GPP TS 26.101 [11] for AMR.

6.3.1.2 Q-bit verification

Requirement:

The Q-bit shall always bit set to 1 in the RTP payload.

NOTE: The Q-bit is the frame quality indicator [8]. If set to 0, it indicates that the corresponding frame is severely damaged, and the receiver should set the RX_TYPE to either SPEECH_BAD or SID_BAD depending on the Frame Type (FT).]

Test method: For each test case amr-0 to amr-7 (see clause 6.2.2), the Q-bit field is extracted from recorded RTP stream from the DUT and the value is compared to 1.

6.3.1.3 SID update periodicity

Requirement:

The DUT shall respect the SID update rules specified in [8]. SID frames shall be sent with the following pattern:

- First SID frame 20 ms after the last speech frame (SID_FIRST)
- Second SID frame (first SID UPDATE): 60 ms after the first SID frame
- Following SID UPDATE: every 160 ms

NOTE: Network equipment may monitor RTP traffic and release the call (false communications cutting detection) if SID update is incorrect.

Test method:

For the test case amr-7 (see clause 6.2.2), analyse and report the RTP sending frames intervals for SID frames according to requirement.

6.3.2 Test cases in receiving

6.3.2.1 Q-bit verification

Requirement:

Quality requirement are ffs.

Test method:

The system simulator shall send an AMR coded RTP stream where the Q bit is set to 0 and record the audio output from the DUT.

6.3.3 Test cases with CMR

6.3.3.1 Response time definition

The UE response time in the send direction (uplink) to an incoming CMR is defined as the delay between the send time of a packet containing the CMR at the network interface of the SS and the arrival time of an active speech packet from the UE in response to this CMR at the network interface of the SS.

6.3.3.2 Open offer

Requirement:

The AMR bit rate (FT field) in sending shall be according to the i-th incoming CMR inserted by the SS with a response time <= 80 ms and shall be valid until the next CMR over a time interval ranging from $i*T_{CMR}+80$ ms to $(i+1)*T_{CMR}$.

NOTE: The expected response time is 60 ms, a margin of 20 ms is added to account for SS delay (assuming SS delay is less than 20 ms).

Test method:

Table 6.3.3.2: List of CMRs to insert

i	0	1	2	3	4	5	6	7
CMR value	0000	0001	0010	0011	0100	0101	0110	0111

For the test case amr-cmr1 (see clause 6.2.2), the SS inserts the i-th CMR (i=0 to 7) at send time $i.T_{CMR}$ according to Table 6.3.3.2, where T_{CMR} is 2.4sec; the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported with the corresponding packet arrival time.

6.3.3.3 Restricted offer

Requirement:

The AMR bit rate (FT field) in sending shall be according to the i-th incoming CMR (i=0 to 7) inserted by the SS with a response time <= 80 ms and and shall be valid until the next CMR over a time interval ranging from $i*T_{CMR}$+80 ms to $(i+1)*T_{CMR}$, if the mode in the CMR is defined in the mode-set, otherwise the AMR bit rate shall not change.

NOTE: The expected response time is 60 ms, a margin of 20 ms is added to account for SS delay (assuming SS delay is less than 20 ms).Test method:

For the test case amr-cmr2 (see clause 6.2.2), he SS inserts the i-th CMR (i=0 to 7) at send time $i.T_{CMR}$ according to Table 6.3.3.2, where T_{CMR} is 2.4sec; the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported with the corresponding packet arrival time.

6.4 RTCP tests

6.4.1 General

If the DUT is compliant with TS 26.139 [3], the RTCP tests defined in this clause may be skipped, otherwise the clause applies.

6.4.2 Verification of SR and RR reports

Characterisation is performed for test case amr-imp. The following information shall be reported: packet loss in terms of ‘fraction lost’ and ‘cumulative number of packets lost’ according to the timing interval of impairments, number of inverted and duplicated packets, interarrival jitter (using a computation similar to [3] clause 6.2.3.2).

NOTE1: Packets that arrive late are not counted as lost (see RFC 3550 [14]).

NOTE2: If the loss is negative due to duplicates, the fraction lost is set to zero (see RFC 3550 [14]).

6.4.3 RTCP bandwidth verification

Characterisation is performed for test case amr-imp. RTCP bandwidth is checked using the computation in [3] clause 6.2.3.2, applied to the whole test duration.

7 RTP Payload Format Conformance for AMR-WB

7.1 Applicability

The requirements and test methods in this clause shall apply when UE is used to provide wideband telephony, either as a stand-alone service, or as part of a multimedia service.

7.2 SDP tests

7.2.1 MO call

Requirement:

Requirements on the SDP offer from the DUT are for further study.

NOTE: SDP testing is already considered in [4,5,6,7].

Test method:

A call is established by the DUT. The SDP offer from the DUT shall be documented.

7.2.2 MT calls

Requirement:

Requirements on the SDP answer from the DUT are for further study.

NOTE: Verification of b=AS is for further study. Purpose is to check compliance to [13] Annex K: b=AS is expected to be set according to the highest allowed codec mode and other parameters (IP version, ptime, bandwidth efficient or octet-align mode) in the SDP answer.

Test method:

Every call is established by the system simulator using one AMR-WB payload type in the SDP offer. The system simulator shall configure the SDP offer according to Table 7.2.2-1 for the bandwidth-efficient mode of AMR-WB and Table 7.2.2-2for the octet-aligned mode of AMR-WB.

For each SDP offer, the SDP answer from the DUT shall be documented and the corresponding RTP and RTCP streams shall be recorded.

The test signal to be used for the measurements shall be the same in both directions as specified below depending on test cases:

- speech1: the British-English single talk sequence described in ITU-T Recommendation P.501 [10].
- silence1: test signal forced to silence (same length as speech1).
- speech2: first sentence of the British-English single talk sequence described in ITU-T Recommendation P.501 [10] shortened to 2.4sec by selecting samples in interval [0.5sec, 2.9sec], repeated 16 times.
- silence2: test signal forced to silence (same length as speech2).
- speech3: 3 repeats of the Composite Source Signal (CSS) according to ITU-T Recommendation P.501 [10] followed by a speech signal of 160 s as in clause 7.10.4.2 of TS 26.132 [2].
- silence3: test signal forced to silence (same length as speech3).

In sending, for acoustic interfaces, the test signal level should be -4,7 dBPa measured at the MRP; for electrical interfaces, the active speech level of the signal should be calibrated to -60 dBV for analogue and to -16 dBm0 for digital connections.

In receiving, the test signal level should be -16 dBm0 measured at the digital reference point or the equivalent analogue point.

Table 7.2.2-1: List of test cases for MT calls for given SDP offer (bandwidth-efficient)

Test case	Parameter the SDP offer	Mode-set in SDP answer	Input to DUT	Input to system simulator
amrwb-0	mode-set=0	0	speech1	silence1
amrwb-1	mode-set=1	1	speech1	silence1
amrwb-2	mode-set=2	2	speech1	silence1
amrwb-3	mode-set=3	3	speech1	silence1
amrwb-4	mode-set=4	4	speech1	silence1
amrwb-5	mode-set=5	5	speech1	silence1
amrwb-6	mode-set=6	6	speech1	silence1
amrwb-7	mode-set=7	7	speech1	silence1
amrwb-8	mode-set=8	8	speech1	silence1
amrwb-oo	mode-set not present (open offer)	See NOTE 1	speech1	silence1
amrwb-cmr1	mode-set not present (open-offer)	See NOTE 1	speech2 (see NOTE 2)	speech2
amrwb-cmr2	mode-set=0,1,2	0,1,2	speech2 (see NOTE 2)	speech2
amrwb-qbit	mode-set=2	2	silence1	speech1 (see NOTE 3)
amrwb-imp	mode-set=2	2	silence3 (see NOTE 4)	speech3
NOTE 1: The DUT may restrict the mode-set in its answer to a restricted set of AMR-WB modes, e.g., to 0,2,4,7 or a further subset due to configuration. NOTE 2: The system simulator inserts CMRs in the RTP stream in this test case. NOTE 3: The system simulator forces Q bit to 0 in all packets of the RTP stream in test case 'amr-qbit'. NOTE 4: The system simulator inserts packet impairments (using profile in Annex A) in the RTP stream in this test case.				

Table 7.2.2-2: List of test cases for MT calls for given SDP offer (octet-aligned)

Test case	Parameter the SDP offer	Mode-set in SDP answer	Input to DUT	Input to system simulator
amrwb-octet-2	mode-set=2; octet-align=1	7	speech1	silence1

7.3 RTP tests

7.3.1 Test cases in sending

7.3.1.1 FT verification

Requirement:

The FT entry in the ToC shall be match the active speech bit rate index according to RFC 4867 [9].

The FT entry in the ToC shall be match the SID bit rate index according to RFC 4867 [9].

Test method:

For each test case amrwb-0 to amrwb-8 (see clause 6.2.2), the ToC field is extracted for recorded RTP stream from the DUT and compared with the respective AMR-WB mode (0 to 8) for active speech packets or SID bitrate for SID packets.

NOTE: The value of FT is defined in Table 1a in TS 26.190 [12] for AMR-WB.

7.3.1.2 Q-bit verification

Requirement:

The Q-bit shall always bit set to 1 in the RTP payload.

NOTE: The Q-bit is the frame quality indicator [8]. If set to 0, it indicates that the corresponding frame is severely damaged, and the receiver should set the RX_TYPE to either SPEECH_BAD or SID_BAD depending on the frame type (FT).

Test method:

For each test case amrwb-0 to amrwb-8 (see clause 7.2.2), the Q-bit field is extracted from recorded RTP stream from the DUT and the value is compared to 1.

7.3.1.3 SID update periodicity

Requirement:

The DUT shall respect the SID update rules specified in [8]. SID frames shall be sent with the following pattern:

- First SID frame 20ms after the last speech frame (SID_FIRST)
- Second SID frame (first SID UPDATE): 60 ms after the first SID frame
- Following SID UPDATE: every 160 ms

NOTE: Network equipment may monitor RTP traffic and release the call (false communications cutting detection) if SID update is incorrect.

Test method:

For the test case amrwb-7 (see clause 7.2.2), analyse and report the RTP sending frames intervals for SID frames according to requirement.

7.3.2 Test cases in receiving

7.3.2.1 Q-bit verification

Requirement:

Quality requirement are ffs.

Test method:

The system simulator shall send an AMR-WB coded RTP stream where the Q bit is set to 0 and record the audio output from the DUT.

7.3.3 Test cases with CMR

7.3.3.1 Response time definition

The UE response time in the send direction (uplink) to an incoming CMR is defined as the delay between the send time of a packet containing the CMR at the network interface of the SS and the arrival time of an active speech packet from the UE in response to this CMR at the network interface of the SS.

7.3.3.2 Open offer

Requirement:

The AMR-WB bit rate (FT field) in sending shall be according to the i-th incoming CMR inserted by the SS with a response time <= 80 ms and shall be valid until the next CMR over a time interval ranging from i*T_{CMR}+80 ms to (i+1)*T_{CMR}.

NOTE: The expected response time is 60 ms, a margin of 20 ms is added to account for SS delay (assuming SS delay is less than 20 ms).

Test method:

Table 7.3.3.2: List of CMRs to insert

i	0	1	2	3	4	5	6	7	8
CMR	0000	0001	0010	0011	0100	0101	0110	0111	1000

For the test case amrwb-cmr1 (see clause 7.2.2), the SS inserts the i-th CMR (i=0 to 8) at send time i.T_{CMR} according to Table 7.3.3.2, where T_{CMR} is 2.4sec;the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported with the corresponding packet arrival time.

7.3.3.2 Restricted offer

Requirement:

The AMR-WB bit rate (FT field) in sending shall be according to the i-th incoming CMR (i=0 to 7) inserted by the SS with a response time <= 80 ms and and shall be valid until the next CMR over a time interval ranging from i*T_{CMR}+80 ms to (i+1)*T_{CMR}, if the mode in the CMR is defined in the mode-set, otherwise the AMR bit rate shall not change.

NOTE: The expected response time is 60 ms, a margin of 20 ms is added to account for SS delay (assuming SS delay is less than 20 ms).

Test method:

For the test case amrwb-cmr2 (see clause 7.2.2), the SS inserts the i-th CMR (i=0 to 7) at send time i.T_{CMR} according to Table 7.3.3.2, where T_{CMR} is 2.4sec; the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported with the corresponding packet arrival time.

7.4 RTCP tests

7.4.1 General

If the DUT is compliant with TS 26.139 [3], the RTCP tests defined in this clause may be skipped, otherwise the clause applies.

7.4.2 Verification of SR and RR reports

Characterisation is performed for test case amr-wb-imp. The following information shall be reported: packet loss in terms of 'fraction lost' and 'cumulative number of packets lost' according to the timing interval of impairments, number of inverted and duplicated packets, interarrival jitter (using a computation similar to [3] clause 6.2.3.2).

NOTE1: Packets that arrive late are not counted as lost (see RFC 3550 [14]).

NOTE2: If the loss is negative due to duplicates, the fraction lost is set to zero (see RFC 3550 [14]).

7.4.3 RTCP bandwidth verification

Characterisation is performed for test case amr-wb-imp. RTCP bandwidth is checked using the computation in [3] clause 6.2.3.2, applied to the whole test duration.

8 RTP Payload Format Conformance for EVS

8.1 Applicability

The requirements and test methods in this clause shall apply when UE is used to provide narrowband, wideband, super-wideband or fullband telephony, either as a stand-alone service, or as part of a multimedia service.

8.2 SDP tests

8.2.1 MO call

Requirement:

Requirements on the SDP offer from the DUT are for further study.

NOTE: SDP testing is already considered in [4,5,6,7].

Test method:

A call is established by the DUT. The SDP offer from the DUT shall be documented.

8.2.2 MT calls

Requirement:

Requirements on the SDP answer from the DUT are for further study.

NOTE: Verification of b=AS is for further study. Purpose is to check compliance to [13] Annex Q: b=AS is expected to be set according to the operation mode (Primary or AMR-WB IO) with highest bitrate and other parameters (IP version, ptime, default or header-full mode) in the SDP answer.

Test method:

Every call is established by the system simulator using one EVS payload type in the SDP offer. The system simulator shall configure the SDP offer according to Table 8.2.2-1 for the default packetization mode of EVS (i.e., hf-only present) and Table 8.2.2-2 for the header-full packetization mode of EVS (i.e., hf-only=1).

For each SDP offer, the SDP answer from the DUT shall be documented and the corresponding RTP and RTCP streams shall be recorded.

The test signal to be used for the measurements shall be the same in both directions as specified below depending on test cases:

- speech1: the British-English single talk sequence described in ITU-T Recommendation P.501 [10].
- silence1: test signal forced to silence (same length as speech1).
- speech2: first sentence of the British-English single talk sequence described in ITU-T Recommendation P.501 [10] shortened to 2.4sec by selecting samples in interval [0.5sec, 2.9sec], repeated 16 times.
- silence2: test signal forced to silence (same length as speech2).
- speech3: 3 repeats of the Composite Source Signal (CSS) according to ITU-T Recommendation P.501 [x13] followed by a speech signal of 160 s as in clause 7.10.4.2 of TS 26.132 [2].
- silence3: test signal forced to silence (same length as speech3).

In sending, for acoustic interfaces, the test signal level should be -4,7 dBPa measured at the MRP; for electrical interfaces, the active speech level of the signal should be calibrated to -60 dBV for analogue and to -16 dBm0 for digital connections.

In receiving, the test signal level should be -16 dBm0 measured at the digital reference point or the equivalent analogue point.

Table 8.2.2-1: List of test cases for MT calls for given SDP offer (default packetization mode)

Test case	Parameter the SDP offer	Parameter the SDP answer	Input to DUT	Input to system simulator
evs-primary-0	br=5.9; bw=nb	Same as in SDP offer	speech1	silence1
evs-primary-1	br=7.2; bw=wb		speech1	silence1
evs-primary-2	br=8; bw=wb		speech1	silence1
evs-primary-3	br=9.6; bw=wb		speech1	silence1
evs-primary-4	br=13.2; bw=swb		speech1	silence1
evs-primary-5	br=16.4; bw=swb		speech1	silence1
evs-primary-6	br=24.4; bw=swb		speech1	silence1
evs-primary-7	br=32; bw=fb		speech1	silence1
evs-primary-8	br=48; bw=fb		speech1	silence1
evs-primary-9	br=64; bw=fb		speech1	silence1
evs-primary-10	br=96; bw=fb		speech1	silence1
evs-primary-11	br=128; bw=fb		speech1	silence1
evs-oo	None (open offer)	See NOTE 1	speech1	silence1
evs-imp	br=24.4;bw=swb	br=24.4; bw=swb	silence3 (see NOTE 2)	speech3
NOTE 1: The DUT may restrict the br parameter or mode-set in its answer to a restricted set of bitrate or AMR-WB-IO modes due to configuration. NOTE 2: The system simulator inserts packet impairments (using profile in Annex A) in the RTP stream in this test case.				

Table 3b: List of test cases for MT calls for given SDP offer (header-full packetization mode).

Test case	Parameter the SDP offer	Parameter the SDP answer	Input to DUT	Input to system simulator
evs-cmr1-1 to evs-cmr1-7	None (open-offer)	See NOTE 1	speech2 (see NOTE 2)	speech2
evs-cmr2-1 to evs-cmr2-7	br=5.9-24.4; bw=swb	See NOTE 1	speech2 (see NOTE 2)	speech2
evs-io-cmr	evs-mode-switch=1hf-only=1	See NOTE 1	speech2 (see NOTE 3)	speech2
evs-io-0	mode-set=0; evs-mode-switch=1; hf-only=1	Same as in SDP offer	speech1	silence1
evs-io-1	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-2	mode-set=1; evs-mode-switch=1		speech1	silence1
evs-io-3	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-4	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-5	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-6	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-7	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-8	mode-set=1; evs-mode-switch=1; hf-only=1		speech1	silence1
evs-io-qbit	mode-set=1; evs-mode-switch=1; hf-only=1		silence1	speech1 (see NOTE 4)
NOTE 1: The DUT may restrict the br parameter or mode-set in its answer to a restricted set of bitrate or AMR-WB-IO modes due to configuration. NOTE 2: The system simulator inserts CMRs in the RTP stream in this test case. NOTE 3: CMR to EVS AMR-BIO modes 14.25 and 19.85 are not supported in Compact mode, see Table A.2 in [x9], therefore this test case is defined in header-full mode. NOTE 4: The system simulator forces Q bit to 0 in all packets of the RTP stream in test case 'evs-io-cmr'.				

8.3 RTP tests

8.3.1 Test cases in sending

8.3.1.1 ToC byte verification

Requirement:

The ToC byte in each RTP active speech packet shall match the active bit rate, bandwidth, and operation mode according to Annex A of TS 26.445 [9].

The ToC byte in each SID packet shall match the SID indication according to Annex A of TS 26.445 [9].

Test method:

For each test case evs-io-0 to evs-io-8 (see clause 8.2.2), the ToC field is extracted for recorded RTP stream from the DUT and compared with the respective operation mode for active speech and SID packets.

8.3.1.2 Q-bit verification

Requirement:

The Q-bit shall always bit set to 1 in the RTP payload when EVS AMR-WB-IO and header-full modes are negotiated.

NOTE: The Q-bit is the frame quality indicator [8]. If set to 0, it indicates that the corresponding frame is severely damaged, and the receiver should set the RX_TYPE to either SPEECH_BAD or SID_BAD depending on the frame type (FT).

Test method:

For each test case evs-io-0 to evs-io-8 (see clause 8.2.2), the Q-bit field is extracted from recorded RTP stream from the DUT and the value is compared to 1.

8.3.1.3 SID update periodicity

Requirement:

The DUT shall respect the SID update rules described in clause 5.6.1.1 of TS 26.445 [9].

SID frames shall be sent:

- Either at a fixed rate: in such case interval shall be 20 ms multiple in range [60 ms; 2 s]
- or at adaptative rate: in such case interval shall be 20 ms multiple in range [160 ms; 1 s]

Test method:

For the test case evs-primary-0 to evs-primary-11 and evs-io-0 to evs-io-7 (see clause 8.2.2), analyse and report the RTP sending frames intervals for SID frames according to requirement.

8.3.2 Test cases in receiving

8.3.2.1 Q-bit verification

Requirement:

Quality requirement are ffs.

Test method:

The system simulator shall send an EVS AMR-WB-IO coded RTP stream where the Q bit is set to 0 (test case 'evs-io-qbit') and record the audio output from the DUT.

8.3.3 Test cases with CMR

6.3.3.1 Response time definition

The UE response time in the send direction (uplink) to an incoming CMR is defined as the delay between the send time of a packet containing the CMR at the network interface of the SS and the arrival time of an active speech packet from the UE in response to this CMR at the network interface of the SS.

8.3.3.2 Open offer

Requirement:

The EVS operation mode (Primary or AMR-WB IO), bandwidth, bit rate, and CAM operation in sending shall be according to the i-th CMR byte inserted by the SS with a response time <= 80ms and shall be valid until the next CMR over a time interval ranging from $i*T_{CMR}$+80 ms to $(i+1)*T_{CMR}$, if a CMR is sent at time T_{CMR}.

NOTE: The expected response time is 60 ms, a margin of 20 ms is added to account for SS delay (assuming SS delay is less than 20 ms).

It the CMR value is 'Not used', the DUT is expected to ignore the value, i.e., it shall not change its encoding mode.

Test method:

Table 8.3.3.2-1: List of CMRs (T value) to insert

j	1	2	3	4	5	6	7
Value of T in CMR byte	000	001	010	011	100	101	110

Table 8.3.3.2-2: List of CMRs (D value) to insert

i	0	1	2	3	4	5	6	7
Value of D in CMR byte	0000	0001	0010	0011	0100	0101	0110	0111
i	8	9	10	11	12	13	14	15
Value of D in CMR byte	1000	1001	1010	1011	1100	1101	1110	1111

For the test case evs-cmr1-j where j=1 to 7 (see clause 8.2.2), the SS inserts the i-th CMR byte (i=0 to 7) at send time $i.T_{CMR}$, where T_{CMR} is 2.4sec; the CMR byte is defined according to TS 26.445 Table A.3 [9] with the T value according to Table 8.3.3.2-1 and the D value according to Table 8.3.3.2-2. If the requested operation in the CMR byte is 'Not used', CMR byte shall be sent. The ToC byte is extracted from recorded RTP stream from the DUT and the operation mode, bandwidth, bit rate, and CAM operation for active speech frame is reported.

8.3.3.3 Restricted offer

Requirement:

The EVS operation mode (Primary or AMR-WB IO), bandwidth, bit rate, and CAM operation in sending shall be according to the i-th CMR byte inserted by the SS with a response time <= 80 ms and shall be valid until the next CMR over a time interval ranging from $i*T_{CMR}$+80 ms to $(i+1)*T_{CMR}$, if a CMR byte is sent at time T_{CMR} and the requested operation in the CMR is allowed by the accepted SDP answer, otherwise it shall not change.

NOTE: The expected response time is 60 ms, a margin of 20 ms is added to account for SS delay (assuming SS delay is less than 20 ms).

Test method:

For the test case evs-cmr2-j where j=1 to 7 (see clause 8.2.2), , the SS inserts the i-th CMR byte (i=0 to 7) at send time $i.T_{CMR}$, where T_{CMR} is 2.4sec; the CMR byte is defined according to TS 26.445 Table A.3 [9] with the T value according to Table 8.3.3.2-1 and the D value according to Table 8.3.3.2-2. The ToC byte is extracted from recorded RTP stream from the DUT and the operation mode, bandwidth, bit rate, and CAM operation for active speech frame is reported.

8.4 RTCP tests

8.4.1 General

If the DUT is compliant with TS 26.139 [3], the RTCP tests defined in this clause may be skipped, otherwise the clause applies.

8.4.2 Verification of SR and RR reports

Characterisation is performed for test case evs-imp. The following information shall be reported: packet loss in terms of 'fraction lost' and 'cumulative number of packets lost' according to the timing interval of impairments, number of inverted and duplicated packets, interarrival jitter (using a computation similar to [3] clause 6.2.3.2).

NOTE1: Packets that arrive late are not counted as lost (see RFC 3550 [14]).

NOTE2: If the loss is negative due to duplicates, the fraction lost is set to zero (see RFC 3550 [14]).

8.4.3 RTCP bandwidth verification

Characterisation is performed for test case evs-imp. RTCP bandwidth is checked using the computation in [3] clause 6.2.3.2, applied to the whole test duration.

Annex A (normative): Packet impairment profile

The impairment profile used in RTCP tests is defined in [2] Annex F.

Annex B (informative):
Change history

Change history							
Date	**Meeting**	**TDoc**	**CR**	**Rev**	**Cat**	**Subject/Comment**	**New version**
2022-08	SA4#120-e	S4-221028				Initial version	0.0.1
2022-08	SA4#120-e	S4-221189				Inclusion of pCR in S4-221029 in brackets	0.1.0
2024-02	SA4#127	S4-240345				Inclusion of proposals in S4-240267 with further offline updates.	0.2.0
2024-02	SA4-SA4-e (AH) Audio SWG post 127	S4aA240006				Fixes to address editorial review from ETSI MCC	0.3.0
2023-03	SA4-SA4-e (AH) Audio SWG post 127	S4aA240007				Inclusion of pCR in S4aA240010 with editorial updates	0.4.0
2024-03	SA#103	SP-240037				Version 1.0.0 created by MCC	1.0.0
2024-03						Version 18.0.0 created by MCC	18.0.0

History

Document history		
V18.0.0	May 2024	Publication