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**Universal Mobile Telecommunications System (UMTS);  
LTE;  
5G;  
Speech/Audio Codec RTP Payload  
Format Conformance for UE Testing  
(3GPP TS 26.130 version 18.1.0 Release 18)**



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# Foreword

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

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- 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.

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- 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.

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## 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.

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# 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.

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# 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 telephony".
- [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.



- [15] ITU-T Recommendation P.863 (09/2014): "Perceptual objective listening quality assessment".
- [16] ITU-T Recommendation P.863.1 (09/2014): "Application guide for Recommendation ITU-T P.863".

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## 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

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## 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.

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# 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  $\leq 80$  ms and shall be valid until the next CMR over a time interval ranging from  $i \cdot T_{\text{CMR}} + 80$  ms to  $(i+1) \cdot T_{\text{CMR}}$ .

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  $i \cdot T_{\text{CMR}}$  according to Table 6.3.3.2, where  $T_{\text{CMR}}$  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  $\leq 80$  ms and shall be valid until the next CMR over a time interval ranging from  $i \cdot T_{\text{CMR}} + 80$  ms to  $(i+1) \cdot T_{\text{CMR}}$ , 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), the SS inserts the  $i$ -th CMR ( $i=0$  to 7) at send time  $i \cdot T_{\text{CMR}}$  according to Table 6.3.3.2, where  $T_{\text{CMR}}$  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 7.2.2-2 for 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  $\leq 80$  ms and shall be valid until the next CMR over a time interval ranging from  $i \cdot T_{\text{CMR}} + 80$  ms to  $(i+1) \cdot T_{\text{CMR}}$ .

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  $i \cdot T_{\text{CMR}}$  according to Table 7.3.3.2, where  $T_{\text{CMR}}$  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  $\leq 80$  ms and shall be valid until the next CMR over a time interval ranging from  $i \cdot T_{\text{CMR}} + 80$  ms to  $(i+1) \cdot T_{\text{CMR}}$ , 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  $i \cdot T_{\text{CMR}}$  according to Table 7.3.3.2, where  $T_{\text{CMR}}$  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-io-0-recv	br=5.9-24.4;bw=nb-swb	Same as in SDP offer	silence1	speech3 (see NOTE1)
evs-io-1-recv			silence1	speech3 (see NOTE1)
evs-io-2-recv			silence1	speech3 (see NOTE1)
evs-io-3-recv			silence1	speech3 (see NOTE1)
evs-io-4-recv			silence1	speech3 (see NOTE1)
evs-io-5-recv			silence1	speech3 (see NOTE1)
evs-io-6-recv			silence1	speech3 (see NOTE1)
evs-io-7-recv			silence1	speech3 (see NOTE1)
evs-io-8-recv			silence1	speech3 (see NOTE1)
evs-oo	None (open offer)	See NOTE 2	speech1	silence1
evs-imp-recv	br=24.4;bw=swb	br=24.4; bw=swb	silence3 (see NOTE 3)	speech3
NOTE 1: The system simulator encodes the input signal with EVS AMR-WB IO at the corresponding mode M for evs-io-M-recv (M=0 to 8).				
NOTE 2: 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 3: 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.

### 8.3.2.2 Quality verification in receiving

Requirement:

The score MOS-LQO<sub>TEST</sub> shall be  $\geq 3.0$  in all receiving tests evs-io-0-recv to evs-io-8-recv.

Test method:

The speech quality of the signal is estimated using the measurement algorithm described in ITU-T Recommendation P.863 [15] in super-wideband mode. Level pre-alignment to -26 dBov of recordings shall be used – see P.863.1 clause 10.2 [16].

A score shall be computed for each 8s speech sentence pair and averaged to produce a mean MOS-LQO value for the test conditions.

$$\text{MOS-LQO}_{\text{TEST}} = (\sum_{i=1}^{20} \text{MOS} - \text{LQO}_{\text{testcondition}}(i)) / 20$$

The synchronization between stimuli and degraded condition shall be done by the test system before applying the P.863 algorithm on each sentence pair according to TS 26.132 [2].

### 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  $\leq 80$  ms and shall be valid until the next CMR over a time interval ranging from  $i \cdot T_{\text{CMR}} + 80$  ms to  $(i+1) \cdot T_{\text{CMR}}$ , if a CMR is sent at time  $T_{\text{CMR}}$ .

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).

If 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  $i \cdot T_{\text{CMR}}$ , where  $T_{\text{CMR}}$  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  $\leq 80$  ms and shall be valid until the next CMR over a time interval ranging from  $i \cdot T_{\text{CMR}} + 80$  ms to  $(i+1) \cdot T_{\text{CMR}}$ , if a CMR byte is sent at time  $T_{\text{CMR}}$  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  $i \cdot T_{\text{CMR}}$ , where  $T_{\text{CMR}}$  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-recv. 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.



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## 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
2024-06	SA#104	SP-240690	0001	2	F	Extra tests for EVS in receiving	18.1.0

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# History

<b>Document history</b>		
V18.0.0	May 2024	Publication
V18.1.0	July 2024	Publication