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| Technical Specification | |
| 3rd Generation Partnership Project;  Technical Specification Group Services and System Aspects;  Speech/Audio Codec RTP Payload Format Conformance for UE Testing  (Release 18) | |
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# 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".

[x1] 3GPP TS 26.132: "Speech and video telephony terminal acoustic test specification".

[x2] 3GPP TS 26.139: "Real-time Transport Protocol (RTP) / RTP Control Protocol (RTCP) verification procedures".

[x3] 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".

[x4] 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".

[x5] 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)".

[x6] 3GPP TS 34.229-4: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); User Equipment (UE) conformance specification; Part 4: Enabler for IP multimedia applications testing".

[x7] 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)".

[x8] 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".

[x9] 3GPP TS 26.445: "Codec for Enhanced Voice Services (EVS);Detailed algorithmic description".

[x10] GSMA IR.92: "IMS Profile for Voice and SMS".

[x11] GSMA NG.114: "IMS Profile for Voice, Video and Messaging over 5GS".

[x12] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction".

[x13] ITU-T Recommendation P.501 (06/2015): "Test signals for use in telephonometry".

[x14] 3GPP TS 26.101: "Mandatory speech codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec frame structure".

[x15] 3GPP TS 26.190: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions".

# 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

Outgoing CMR : CMR in RTP packet sent by the DUT

Sending: Link from DUT to test simulator

Receiving: Link from test simulator to DUT

## 3.2 Symbols

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

<symbol> <Explanation>

## 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 [x1] (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 [x2] can be reused for RTP /RTCP “data injection” with either active or passive test instrument.

## 4.2 Acoustic interfaces

See clause 5.1 in [x1].

## 4.3 Electrical interfaces

See clause 5.2 in [x1].

# 5 Test setup

## 5.1 General

Similar to clause 5.1 in [x2]:

*-* The "system under test" is the device (DUT). or software to be tested

*-* The "test instrument" is the equipment used *t*o 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.The "data injection" is the device or equipment used to generate RTP/RTCP data sent to the system under test

NOTE1: ‘data injection’ may be collocated or integrated with the test instrument.

NOTE2: 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 [x1] 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 [x1].

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 [x1].

## 5.5 [Test signals]

See clause 5.4 in [x1].

## 5.6 [Environmental conditions]

See clause 6 in [x1].

For LTE, WLAN, and NR connections, an RF shielded room [should/shall] 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.

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.

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 1a for the bandwidth-efficient mode of AMR and Table 1b 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 [x13].
* silence1: test signal forced to silence (same length as speech1)
* speech2: speech1 repeated [x] 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 [22] followed by a speech signal of 160s as in clause 7.10.4.2 of [x1]
* silence3: test signal forced to silence (same length as speech3)

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

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

Table 1a: 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 NOTE1 | speech1 | silence1 |
| amr-cmr1 | mode-set not present (open-offer) | See NOTE1 | speech2  (see NOTE2) | speech2 |
| amr-cmr2 | mode-set=0,2,4,7 | 0,2,4,7 | speech2  (see NOTE2) | speech2 |
| amr-qbit | mode-set=7 | 7 | silence1 | speech1  (see NOTE3) |
| amr-imp | mode-set=7 | 7 | speech3  (see NOTE4) | silence3 |
| NOTE1: 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. | | | | |
| NOTE2: The system simulator inserts CMRs in the RTP stream in this test case | | | | |
| NOTE3: The system simulator forces Q bit to 0 in all packets of the RTP stream in test case ‘amr-qbit’. | | | | |
| NOTE4: The system simulator inserts packet impairments in the RTP stream in this test case. | | | | |

Table 1b: 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 [x8].

The FT entry in the ToC shall be match the SID bit rate index according to RFC 4867 [x8].

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 [x14] 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 [x8]. 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 [x9]. 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 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 Open offer

Requirement:

The AMR bit rate (mode) in sending shall be according to the CMR inserted in receiving after [X] ms.

Test method:

For the test case amr-cmr1 (see clause 6.2.2), the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported.

#### 6.3.3.2 Restricted offer

Requirement:

The AMR bit rate (mode) in sending shall be according to the CMR inserted in receiving after [X] ms if the mode in the CMR is in the allowed mode defined in the mode-set, otherwise the AMR bit rate shall not change.

Test method:

For the test case amr-cmr2 (see clause 6.2.2), the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported.

## 6.4 RTCP tests

### 6.4.1 General

If the DUT is compliant with TS 26.139 [x2], the RTCP tests defined in this clause may be skipped, otherwise the clause applies.

### 6.4.2 Verification of SR and RR reports

For further study

### 6.4.3 RTCP bandwidth verification

For further study

# 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.

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.

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 1a for the bandwidth-efficient mode of AMR-WB and Table 1b 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 [x13].
* silence1: test signal forced to silence (same length as speech1)
* speech2: speech1 repeated [x] 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 [22] followed by a speech signal of 160s as in clause 7.10.4.2 of [x1]
* silence3: test signal forced to silence (same length as speech3)

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

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

Table 2a: 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 NOTE1 | speech1 | silence1 |
| amrwb-cmr1 | mode-set not present (open-offer) | See NOTE1 | speech2  (see NOTE2) | speech2 |
| amrwb-cmr2 | mode-set=0,1,2 | 0,1,2 | speech2  (see NOTE2) | speech2 |
| amrwb-qbit | mode-set=2 | 2 | silence1 | speech1  (see NOTE3) |
| amrwb-imp | mode-set=2 | 2 | speech3  (see NOTE4) | silence3 |
| NOTE1: 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. | | | | |
| NOTE2: The system simulator inserts CMRs in the RTP stream in this test case | | | | |
| NOTE3: The system simulator forces Q bit to 0 in all packets of the RTP stream in test case ‘amr-qbit’. | | | | |
| NOTE4: The system simulator inserts packet impairments in the RTP stream in this test case. | | | | |

Table 2b: 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 [x8].

The FT entry in the ToC shall be match the SID bit rate index according to RFC 4867 [x8].

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 [x15] 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 [x8]. 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 [x9]. 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), analyze 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 Open offer

Requirement:

The AMR-WB bit rate (mode) in sending shall be according to the CMR inserted in receiving after [X] ms.

Test method:

For the test case amrwb-cmr1 (see clause 7.2.2), the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported.

#### 7.3.3.2 Restricted offer

Requirement:

The AMR-WB bit rate (mode) in sending shall be according to the CMR inserted in receiving after [X] ms if the mode in the CMR is in the allowed mode defined in the mode-set, otherwise the AMR-WB bit rate shall not change.

Test method:

For the test case amrwb-cmr2 (see clause 7.2.2), the FT field is extracted from recorded RTP stream from the DUT and the value for active speech frame is reported.

## 7.4 RTCP tests

### 7.4.1 General

If the DUT is compliant with TS 26.139 [x2], the RTCP tests defined in this clause may be skipped, otherwise the clause applies.

### 7.4.2 Verification of SR and RR reports

For further study

### 7.4.3 RTCP bandwidth verification

For further study

# 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.

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.

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 3a for the default packetization mode of EVS (i.e., hf-only present) and Table 3b 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 [x13].
* silence1: test signal forced to silence (same length as speech1)
* speech2: speech1 repeated [x] 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 [22] followed by a speech signal of 160s as in clause 7.10.4.2 of [x1]
* silence3: test signal forced to silence (same length as speech3)

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

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

Table 3a: 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 NOTE1 | speech1 | silence1 |
| evs-imp | br=24.4bw=swb | br=24.4,bw=swb | speech3  (see NOTE2) | silence3 |
| NOTE1 : 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. | | | | |
| NOTE2: The system simulator inserts packet impairments 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 | None (open-offer) | See NOTE1 | speech2  (see NOTE2) | speech2 |
| evs-cmr2 | br=5.9-24.4; bw=swb | See NOTE1 | speech2  (see NOTE2) | speech2 |
| evs-io-cmr | evs-mode-switch=1hf-only=1 | See NOTE1 | speech2  (see NOTE3) | 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 NOTE4) |
| NOTE1 : 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. | | | | |
| NOTE2: The system simulator inserts CMRs in the RTP stream in this test case | | | | |
| NOTE3: 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. | | | | |
| NOTE4: 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 [x9].

The ToC byte in each SID packet shall match the SID indication according to Annex A of TS 26.445 [x9].

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 [x8]. 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 [x9].

SID frames shall be sent:

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

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), analyze 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

#### 8.3.3.1 Open offer

Requirement:

[The EVS operation mode (Primary or AMR-WB IO), bandwidth, and bit rate in sending shall be according to the EVS CMR inserted in receiving after [X] ms

Test method:

For the test case evs-cmr2 (see clause 8.2.2), the ToC byte is extracted from recorded RTP stream from the DUT and the operation mode, bandwidth, and bit rate for active speech frame is reported.

#### 8.3.3.2 Restricted offer

Requirement:

[The EVS operation mode (Primary or AMR-WB IO), bandwidth, and bit rate in sending shall be according to the EVS CMR inserted in receiving after [X] ms, if the requested operation in the CMR is allowed by the accepted SDP answer.

Test method:

For the test case evs-cmr2 (see clause 8.2.2), the ToC byte is extracted from recorded RTP stream from the DUT and the operation mode, bandwidth, and bit rate for active speech frame is reported.

## 8.4 RTCP tests

### 8.4.1 General

If the DUT is compliant with TS 26.139 [x2], the RTCP tests defined in this clause may be skipped, otherwise the clause applies.

### 8.4.2 Verification of SR and RR reports

For further study

### 8.4.3 RTCP bandwidth verification

For further study

Annex A (norrmative):  
Packet impairment profiles

This Annex is a placeholder to document impairment profiles used in RTCP tests.

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. |  |