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TECHNICAL SPECIFICATION

5G;
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)
(3GPP TS 34.229-5 version 15.4.0 Release 15)

Reference

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Keywords

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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, certain 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" shall not to be used as 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 is the fifth part of a multi-part conformance specification valid for 3GPP Release 15 and later releases:

3GPP TS 34.229-1 [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 1: Protocol conformance specification".

3GPP TS 34.229-2 [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 2: Implementation Conformance Statement (ICS) proforma specification".

3GPP TS 34.229-3 [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 3: Abstract Test Suites (ATS)".

3GPP TS 34.229-4 [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 4: Enabler for IP multimedia applications testing".

3GPP TS 34.229-5 (the present document): "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)".

NOTE 1: The ATS is written in a standard testing language, TTCN-3, as defined in ETSI ES 201 873, Parts 1 to 3 [8], [9] and [10].

NOTE 2: Further information on testing can be found in ETSI ETS 300 406 [11] and ISO/IEC 9646-1 [12].

For at least a minimum set of services, the prose descriptions of test cases will have a matching detailed test case implemented in TTCN-3 (and provided in 3GPP TS 34.229-3 [4]).

1 Scope

The present document specifies the protocol conformance testing for the User Equipment (UE) supporting the Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) when using the 5G System (5GS).

This is the fifth part of a multi-part test specification. The following information can be found in this part:

- the overall test structure;
- the test configurations;
- the conformance requirement and reference to the core specifications;
- the test purposes; and
- the test procedure.

The following information relevant to testing can be found in accompanying specifications:

- Implementation Conformance Statement (ICS) pro-forma and the applicability of each test case [3].

The present document is valid for UE implemented according to 3GPP Releases starting from Release 15 up to the Release indicated on the cover page of the present document.

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 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".

[3] 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) proforma specification".

[4] 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 Suites (ATS)".

[5] 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".

[6] IETF RFC 3261: "SIP: Session Initiation Protocol".

[7] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[8] ETSI ES 201 873-1: "Methods for Testing and Specification (MTS); The Testing and Test Control Notation version 3; Part 1: TTCN-3 Core Language".

[9] ETSI ES 201 873-2: "Methods for Testing and Specification (MTS); The Testing and Test Control Notation version 3; Part 2: TTCN-3 Tabular Presentation Format (TFT)".

[10] ETSI TR 201 873-3: "Methods for Testing and Specification (MTS); The Testing and Test Control Notation version 3; Part 3: TTCN-3 Graphical Presentation Format (GFT)".

[11] ETSI ETS 300 406: "Methods for testing and Specification (MTS); Protocol and profile conformance testing specifications; Standardization methodology".

[12] ISO/IEC 9646-1: "Information technology - Open systems interconnection - Conformance testing methodology and framework - Part 1: General concepts".

[13] ISO/IEC 9646-7: "Information technology - Open systems interconnection - Conformance testing methodology and framework - Part 7: Implementation Conformance Statements".

[14] 3GPP TS 24.341: "Support of SMS over IP networks; Stage 3".

[15] IETF RFC 3310: "Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)".

[16] 3GPP TS 33.203: "3G security;Access security for IP-based services".

[17] IETF RFC 3329: "Security Mechanism Agreement for the Session Initiation Protocol (SIP)".

[18] IETF RFC 3680: "A Session Initiation Protocol (SIP) Event Package for Registrations".

[19] 3GPP TS 23.501: "System Architecture for the 5G System; Stage 2".

[20] 3GPP TS 24.501: "Non-Access-Stratum (NAS) protocol for 5G System (5GS); Stage 3".

[21] 3GPP TS 38.508-1: "5GS; User Equipment (UE) conformance specification; Part 1: Common test environment".

[22] 3GPP TS 27.007: "AT command set for User Equipment (UE)".

[23] IETF RFC 2617: "HTTP Authentication: Basic and Digest Access Authentication".

[24] 3GPP TS 23.040: "Technical realization of the Short Message Service (SMS)".

[25] 3GPP TS 24.011: "Point-to-Point (PP) Short Message Service (SMS) support on mobile radio interface".

[26] 3GPP TS 24.237: "IP Multimedia (IM) Core Network (CN) subsystem IP Multimedia Subsystem (IMS) Service Continuity".

[27] 3GPP TS 23.003: "Numbering, addressing and identification".

[28] IETF RFC 6665: “SIP-Specific Event Notification”.

[29] IETF RFC 3312: “Integration of Resource Management and SIP”.

[30] IETF RFC 3262: “Reliability of Provisional Responses in the Session Initiation Protocol (SIP)”.

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

[32] 3GPP TS 24.610: "Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem".

[33] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction".

[34] 3GPP TS 24.606: "Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem".

[35] 3GPP TS 24.147: "Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem".

[36] 3GPP TS 24.629: "Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem".

[37] IETF RFC 4028: "Session Timers in the Session Initiation Protocol (SIP)".

[38] IETF RFC 4566: "SDP: Session Description Protocol".

[39] IETF RFC 7462: "URNs for the Alert-Info Header Field of the Session Initiation Protocol (SIP) ".

[40] IETF RFC 3891: "The Session Initiation Protocol (SIP) "Replaces" Header".

[41] IETF RFC 3986: "Uniform Resource Identifier (URI): Generic Syntax".

3 Definitions of terms, symbols and abbreviations

3.1 Terms

Void

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 3GPP TR 21.905 [1].

SS System Simulator

4 Overview

4.1 Test Methodology

4.1.1 Testing of optional functions and procedures

Any function or procedure which is optional, as indicated in the present document may be subject to a conformance test if it is implemented in the UE.

A declaration by the apparatus supplier (Implementation Conformance Statement (ICS)) is used to determine whether an optional function/procedure has been implemented (see ISO/IEC 9646-7 [13] for general information about ICS).

4.2 Implicit Testing

For some 3GPP signalling and protocol features conformance is not verified explicitly in the present document. This does not imply that correct functioning of these features is not essential, but that these are implicitly tested to a sufficient degree in other tests.

4.3 Conformance Requirements

The Conformance Requirements clauses in the present document are copy/paste from the relevant core specification where skipped text has been replaced with "...". References to clauses in the Conformance Requirements clause of the test body refers to clauses in the referred specification, not clauses in the present document.

5 Reference Conditions

5.1 General

The test cases are expected to be executed through the 3GPP radio interface. Details of the radio interfaces are outside the scope of this specification. The reference environments used by tests are specified in the test.

5.2 Generic setup procedures

A set of basic generic procedures for different IMS usage scenarios are described in Annex A of this specification. These procedures are used in numerous test cases throughout the present document. Default Messages are used from and maintained in Annex A of TS 34.229-1 [2].

5.3 Transport protocols applied

For simplicity, UDP (*User Datagram Protocol*) is applied to IMS testing as default DL transport protocol, except for the test cases in clause 6 where TCP (*Transmission Control Protocol*) is applied as DL transport protocol.

NOTE: Which UL transport protocol is used in the test is decided by the UE.

6 Registration

6.1 Initial Registration / 5GS

6.1.1 Test Purpose (TP)

(1)

```
with { UE has an ISIM or USIM inserted, is registered for 5GS, and has acquired P-CSCF address(es) }
ensure that {
  when { UE is made to register for IMS }
   then { UE sends a correctly composed initial REGISTER request to the P-CSCF }
            }
```

(2)

```
with { UE having sent unprotected REGISTER request }
ensure that {
  when { UE receiving a valid 401 (Unauthorized) response for the initial REGISTER request sent }
   then { UE correctly authenticates itself by sending another REGISTER request with a correctly
composed Authorization header using the AKAv1-MD5 algorithm }
            }
```

(3)

```
with { UE having sent unprotected and then protected REGISTER request }
ensure that {
  when { UE receiving a valid 200 OK response from S-CSCF for the REGISTER sent for authentication }
   then { UE subscribes to the reg event package for the public user identity registered, using the
stored service route for routing the SUBSCRIBE request }
            }
```

(4)

```
with { UE having subscribed to reg event }
ensure that {
  when { UE receives NOTIFY request for reg event }
   then { UE responds with a valid 200 OK response }
            }
```

6.1.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause C.2]:

In case the UE is loaded with a UICC that contains a USIM but does not contain an ISIM, the UE shall:

- generate a private user identity;
- generate a temporary public user identity; and
- generate a home network domain name to address the SIP REGISTER request to.

All these three parameters are derived from the IMSI parameter in the USIM, according to the procedures described in TS 23.003 [3]. Also in this case, the UE shall derive new values every time the UICC is changed, and shall discard existing values if the UICC is removed.

NOTE: If there is an ISIM and a USIM on a UICC, the ISIM is used for authentication to the IM CN subsystem, as described in TS 33.203 [19]. See also clause 5.1.1.1A.

[TS 24.229, clause 5.1.1.1A]:

The ISIM shall always be used for authentication to the IM CN subsystem, if it is present, as described in 3GPP TS 33.203 [19].

The ISIM is preconfigured with all the necessary parameters to initiate the registration to the IM CN subsystem. These parameters include:

- the private user identity;
- one or more public user identities; and
- the home network domain name used to address the SIP REGISTER request

The first public user identity in the list stored in the ISIM is used in emergency registration requests.

In case the UE does not contain an ISIM, the UE shall:

- generate a private user identity;
- generate a temporary public user identity; and
- generate a home network domain name to address the SIP REGISTER request to;

in accordance with the procedures in clause C.2.

The temporary public user identity is only used in REGISTER requests, i.e. initial registration, re-registration, UE-initiated deregistration.

The UE shall not reveal to the user the temporary public user identity if the temporary public user identity is barred. The temporary public user identity is not barred if received by the UE in the P-Associated-URI header field.

If the UE is unable to derive the parameters in this clause for any reason, then the UE shall not proceed with the request associated with the use of these parameters and will not be able to register to the IM CN subsystem.

[TS 24.229, clause 5.1.1.2.1]:

The initial registration procedure consists of the UE sending an unprotected REGISTER request and, if challenged depending on the security mechanism supported for this UE, sending the integrity-protected REGISTER request or other appropriate response to the challenge. The UE can register a public user identity with any of its contact addresses at any time after it has acquired an IP address, discovered a P-CSCF, and established an IP-CAN bearer that can be used for SIP signalling. However, the UE shall only initiate a new registration procedure when it has received a final response from the registrar for the ongoing registration, or the previous REGISTER request has timed out.

...

The UE shall send the unprotected REGISTER requests to the port advertised to the UE during the P-CSCF discovery procedure. If the UE does not receive any specific port information during the P-CSCF discovery procedure, or if the UE was pre-configured with the P-CSCF's IP address or domain name and was unable to obtain specific port information, the UE shall send the unprotected REGISTER request to the SIP default port values as specified in RFC 3261 [26].

NOTE 1: The UE will only send further registration and subsequent SIP messages towards the same port of the P-CSCF for security mechanisms that do not require to use negotiated ports for exchanging protected messages.

The UE shall extract or derive a public user identity, the private user identity, and the domain name to be used in the Request-URI in the registration, according to the procedures described in subclause 5.1.1.1A or subclause 5.1.1.1B. A public user identity may be input by the end user.

On sending an unprotected REGISTER request, the UE shall populate the header fields as follows:

a) a From header field set to the SIP URI that contains:

...

2) the public user identity to be registered;

b) a To header field set to the SIP URI that contains:

...

2) the public user identity to be registered;

c) a Contact header field set to include SIP URI(s) containing the IP address or FQDN of the UE in the hostport parameter. If the UE:

1) supports GRUU (see table A.4, item A.4/53);

...

3) has an IMEI available; or

...

the UE shall include a "+sip.instance" header field parameter containing the instance ID. ...

NOTE 2: The requirement placed on the UE to include an instance ID based on the IMEI or the MEID when the UE does not support GRUU and does not support multiple registrations does not imply any additional requirements on the network.

...

The UE shall include all supported ICSI values (coded as specified in subclause 7.2A.8.2) in a g.3gpp.icsi-ref media feature tag as defined in subclause 7.9.2 and RFC 3840 [62] for the IMS communication services it intends to use, and IARI values (coded as specified in subclause 7.2A.9.2), for the IMS applications it intends to use in a g.3gpp.iari-ref media feature tag as defined in subclause 7.9.3 and RFC 3840 [62].

The UE shall include the media feature tags as defined in RFC 3840 [62] for all supported streaming media types.

...

If the UE has no specific reason not to include a user part in the URI of the contact address (e.g. some UE performing the functions of an external attached network), the UE should include a user part in the URI of the contact address such that the user part is globally unique and does not reveal any private information;

NOTE 3: A time-based UUID (Universal Unique Identifier) generated as per subclause 4.2 of RFC 4122 [154] is globally unique and does not reveal any private information.

d) a Via header field set to include the sent-by field containing the IP address or FQDN of the UE and the port number where the UE expects to receive the response to this request when UDP is used. For TCP, the response is received on the TCP connection on which the request was sent. For the UDP, the UE shall also include a "rport" header field parameter with no value in the Via header field. Unless the UE has been configured to not send keep-alives, and unless the UE is directly connected to an IP-CAN for which usage of NAT is not defined, it shall include a "keep" header field parameter with no value in the Via header field, in order to indicate support of sending keep-alives associated with the registration, as described in RFC 6223 [143];

NOTE 4: When sending the unprotected REGISTER request using UDP, the UE transmit the request from the same IP address and port on which it expects to receive the response to this request.

e) a registration expiration interval value of 600 000 seconds as the value desired for the duration of the registration;

NOTE 5: The registrar (S-CSCF) might decrease the duration of the registration in accordance with network policy. Registration attempts with a registration period of less than a predefined minimum value defined in the registrar will be rejected with a 423 (Interval Too Brief) response.

f) a Request-URI set to the SIP URI of the domain name of the home network used to address the REGISTER request;

g) the Supported header field containing the option-tag "path", and

1) if GRUU is supported, the option-tag "gruu"; and

2) if multiple registrations is supported, the option-tag "outbound".

h) if a security association or TLS session exists, and if available to the UE (as defined in the access technology specific annexes for each access technology), a P-Access-Network-Info header field set as specified for the access network technology (see subclause 7.2A.4);

...

On receiving the 200 (OK) response to the REGISTER request, the UE shall:

a) store the expiration time of the registration for the public user identities found in the To header field value and bind it either to the respective contact address of the UE or to the registration flow and the associated contact address (if the multiple registration mechanism is used);

...

b) store as the default public user identity the first URI on the list of URIs present in the P-Associated-URI header field and bind it to the respective contact address of the UE and the associated set of security associations or TLS session;

...

d) store the list of service route values contained in the Service-Route header field and bind the list either to the contact address or to the registration flow and the associated contact address (if the multiple registration mechanism is used), and the associated set of security associations or TLS session over which the REGISTER request was sent;

NOTE 10: When multiple registration mechanism is not used, there will be only one list of service route values bound to a contact address. However, when multiple registration mechanism is used, there will be different list of service route values bound to each registration flow and the associated contact address.

NOTE 11: The UE will use the stored list of service route values to build a proper preloaded Route header field for new dialogs and standalone transactions (other than REGISTER method) when using either the respective contact address or the registration flow and the associated contact address (if the multiple registration mechanism is used), and the associated set of security associations or TLS session.

e) if the UE indicated support for GRUU in the Supported header field of the REGISTER request then:

- if the UE did not use the procedures specified in RFC 6140 [191] for registration, find the Contact header field within the response that matches the one included in the REGISTER request. If this contains a "pub-gruu" header field parameter or a "temp-gruu" header field parameter or both, then store the value of those parameters as the GRUUs for the UE in association with the public user identity and the contact address that was registered; and

...

NOTE 12: When allocating public GRUUs to registering UAs the functionality within the UE that performs the role of registrar will add an "sg" SIP URI parameter that uniquely identifies that UA to the public GRUU it received in the "pub-gruu" header field parameter. The procedures for generating a temporary GRUU using the "temp-gruu-cookie" header field parameter are specified in subclause 7.1.2.2 of RFC 6140 [191].

f) if the REGISTER request contained the "reg-id" and "+sip.instance" Contact header field parameter and the "outbound" option tag in a Supported header field, the UE shall check whether the option-tag "outbound" is present in the Require header field:

- if no option-tag "outbound" is present, the UE shall conclude that the S-CSCF does not support the registration procedure as described in RFC 5626 [92], and the S-CSCF has followed the registration procedure as described in RFC 5627 [93] or RFC 3261 [26], i.e., if there is a previously registered contact address, the S-CSCF replaced the old contact address and associated information with the new contact address and associated information (see bullet e) above). Upon detecting that the S-CSCF does not support the registration procedure as defined in RFC 5626 [92], the UE shall refrain from registering any additional IMS flows for the same private identity as described in RFC 5626 [92]; or

NOTE 13: Upon replaces the old contact address with the new contact address, the S-CSCF performs the network initiated deregistration procedure for the previously registered public user identities and the associated old contact address as described in subclause 5.4.1.5. Hence, the UE will receive a NOTIFY request informing the UE about the deregistration of the old contact address.

- if an option-tag "outbound" is present, the UE may establish additional IMS flows for the same private identity, as defined in RFC 5626 [92];

g) if available, store the announcement of media plane security mechanisms the P-CSCF (IMS-ALG) supports labelled with the "mediasec" header field parameter specified in subclause 7.2A.7 and received in the Security-Server header field, if any. Once the UE chooses a media security mechanism from the list received in the Security-Server header field from the server, it may initiate that mechanism on a media level when it initiates new media in an existing session;

NOTE 14: The "mediasec" header field parameter indicates that security mechanisms are specific to the media plane.

[TS 24.229, clause 5.1.1.2.2]:

On sending a REGISTER request, as defined in subclause 5.1.1.2.1, the UE shall additionally populate the header fields as follows:

a) an Authorization header field, with:

- the "username" header field parameter, set to the value of the private user identity;
- the "realm" header field parameter, set to the domain name of the home network;
- the "uri" header field parameter, set to the SIP URI of the domain name of the home network;
- the "nonce" header field parameter, set to an empty value; and
- the "response" header field parameter, set to an empty value;

NOTE 1: If the UE specifies its FQDN in the hostport parameter in the Contact header field and in the sent-by field in the Via header field, then it has to ensure that the given FQDN will resolve (e.g., by reverse DNS lookup) to the IP address that is bound to the security association.

NOTE 2: The UE associates two ports, a protected client port and a protected server port, with each pair of security association. For details on the selection of the port values see 3GPP TS 33.203 [19].

b) additionally for the Contact header field, if the REGISTER request is protected by a security association, include the protected server port value in the hostport parameter;

c) additionally for the Via header field, for UDP, if the REGISTER request is protected by a security association, include the protected server port value in the sent-by field; and

d) a Security-Client header field set to specify the signalling plane security mechanism the UE supports, the IPsec layer algorithms the UE supports and the parameters needed for the security association setup. The UE shall support the setup of two pairs of security associations as defined in 3GPP TS 33.203 [19]. The syntax of the parameters needed for the security association setup is specified in annex H of 3GPP TS 33.203 [19]. The UE shall support the "ipsec-3gpp" security mechanism, as specified in RFC 3329 [48]. The UE shall support the IPsec layer algorithms for integrity and confidentiality protection as defined in 3GPP TS 33.203 [19], and shall announce support for them according to the procedures defined in RFC 3329 [48].

[TS 24.229, clause 5.1.1.5.1]:

Authentication is performed during initial registration. A UE can be re-authenticated during subsequent reregistrations, deregistrations or registrations of additional public user identities. When the network requires authentication or re-authentication of the UE, the UE will receive a 401 (Unauthorized) response to the REGISTER request.

On receiving a 401 (Unauthorized) response to the REGISTER request, the UE shall:

1) extract the RAND and AUTN parameters;

2) check the validity of a received authentication challenge, as described in 3GPP TS 33.203 [19] i.e. the locally calculated XMAC must match the MAC parameter derived from the AUTN part of the challenge; and the SQN parameter derived from the AUTN part of the challenge must be within the correct range; and

3) check the existence of the Security-Server header field as described in RFC 3329 [48]. If the Security-Server header field is not present or it does not contain the parameters required for the setup of the set of security associations (see annex H of 3GPP TS 33.203 [19]), the UE shall abandon the authentication procedure and send a new REGISTER request with a new Call-ID.

In the case that the 401 (Unauthorized) response to the REGISTER request is deemed to be valid the UE shall:

1) calculate the RES parameter and derive the keys CK and IK from RAND as described in 3GPP TS 33.203 [19];

2) set up a temporary set of security associations for this registration based on the static list and parameters the UE received in the 401 (Unauthorized) response and its capabilities sent in the Security-Client header field in the REGISTER request. The UE sets up the temporary set of security associations using the most preferred mechanism and algorithm returned by the P-CSCF and supported by the UE and using IK and CK (only if encryption enabled) as the shared key. The UE shall use the parameters received in the Security-Server header field to setup the temporary set of security associations. The UE shall set a temporary SIP level lifetime for the temporary set of security associations to the value of reg-await-auth timer;

...

4) send another REGISTER request towards the protected server port indicated in the response using the temporary set of security associations to protect the message. The header fields are populated as defined for the initial REGISTER request that was challenged with the received 401 (Unauthorized) response, with the addition that the UE shall include an Authorization header field containing:

   - the "realm" header field parameter set to the value as received in the "realm" WWW-Authenticate header field parameter;

   - the "username" header field parameter, set to the value of the private user identity;

   - the "response" header field parameter that contains the RES parameter, as described in RFC 3310 [49];

   - the "uri" header field parameter, set to the SIP URI of the domain name of the home network;

   - the "algorithm" header field parameter, set to the value received in the 401 (Unauthorized) response; and

   - the "nonce" header field parameter, set to the value received in the 401 (Unauthorized) response.

   The UE shall also insert the Security-Client header field that is identical to the Security-Client header field that was included in the previous REGISTER request (i.e. the REGISTER request that was challenged with the received 401 (Unauthorized) response). The UE shall also insert the Security-Verify header field into the request, by mirroring in it the content of the Security-Server header field received in the 401 (Unauthorized) response. The UE shall set the Call-ID of the security association protected REGISTER request which carries the authentication challenge response to the same value as the Call-ID of the 401 (Unauthorized) response which carried the challenge.

NOTE 2: The Security-Client header field contains signalling plane security mechanism and if the UE supports media plane security, then media plane security mechanisms are contained, too.

[TS 24.229, clause 5.1.1.5.1]:

On receiving the 200 (OK) response for the security association protected REGISTER request registering a public user identity with the associated contact address, the UE shall:

- change the temporary set of security associations to a newly established set of security associations, i.e. set its SIP level lifetime to the longest of either the previously existing set of security associations SIP level lifetime, or the lifetime of the just completed registration plus 30 seconds; and

- if this is the only set of security associations available toward the P-CSCF, use the newly established set of security associations for further messages sent towards the P-CSCF. If there are additional sets of security associations (e.g. due to registration of multiple contact addresses), the UE can either use them or use the newly established set of security associations for further messages sent towards the P-CSCF as appropriate.

NOTE 3: If the UE has registered multiple contact addresses, the UE can either send requests towards the P-CSCF over the newly established set of security associations, or use different UE's contact address and associated set of security associations when sending the requests towards the P-CSCF. Responses towards the P-CSCF that are sent via UDP will be sent over the same set of security associations that the related request was received on. Responses towards the P-CSCF that are sent via TCP will be sent over the same set of security associations that the related request was received on.

When the first request or response protected with the newly established set of security associations is received from the P-CSCF or when the lifetime of the old set of security associations expires, the UE shall delete the old set of security associations and related keys it may have with the P-CSCF after all SIP transactions that use the old set of security associations are completed.

[TS 24.229, clause 5.1.1.3]:

Upon receipt of a 2xx response to the initial registration, the UE shall subscribe to the reg event package for the public user identity registered at the user's registrar (S-CSCF) as described in RFC 3680 [43] and RFC 6665 [28].

...

The UE shall subscribe to the reg event package upon registering a new contact address via an initial registration procedure. If the UE receives a NOTIFY request via the newly established subscription dialog and via the previously established subscription dialogs (there will be at least one), the UE may terminate the previously established subscription dialogs and keep only the newly established subscription dialog.

The UE shall use the default public user identity for subscription to the registration-state event package.

NOTE 2: The subscription information stored in the HSS ensures that the default public user identity is a SIP URI.

On sending a SUBSCRIBE request, the UE shall populate the header fields as follows:

a) a Request-URI set to the resource to which the UE wants to be subscribed to, i.e. to the SIP URI that is the default public user identity used for subscription;

b) a From header field set to the SIP URI that is the default public user identity used for subscription;

c) a To header field set to the SIP URI that is the default public user identity used for subscription;

d) an Event header field set to the "reg" event package;

e) an Expires header field set to 600 000 seconds as the value desired for the duration of the subscription;

f) void; and

g) void.

[TS 24.229, clause 5.1.2.1]:

Upon receipt of a NOTIFY request for the dialog associated with the subscription to the reg event package the UE shall perform the following actions:

- store the information for the established dialog;

- store the expiration time as indicated in the "expires" header field parameter of the Subscription-State header field, if present, of the NOTIFY request. Otherwise the expiration time is retrieved from the Expires header field of the 2xx response to SUBSCRIBE request;

- if a <registration> element with state attribute "active", i.e. registered, is received for one or more public user identities, the UE shall store the indicated public user identities as registered;

- if a <registration> element with state attribute "active" is received, and the UE supports GRUU (see table A.4, item A.4/53), then for each public user identity indicated in the notification that contains a <pub-gruu> element or a <temp-gruu> element or both (as defined in RFC 5628 [94]), the UE shall store the value of those elements in association with the public user identity;

[TS 24.229, clause 5.1.2A.1.1]:

When the UE sends any request, the UE shall use either a given contact address that has been previously registered or a registration flow and the associated contact address (if the multiple registration mechanism is used) and shall:

- if IMS AKA is in use as a security mechanism:

  a) if the UE has not obtained a GRUU, populate the Contact header field of the request with the protected server port and the respective contact address; and

  b) include the protected server port and the respective contact address in the Via header field entry relating to the UE;

...

The UE shall determine the public user identity to be used for this request as follows:

1) if a P-Preferred-Identity was included, then use that as the public user identity for this request; or

2) if no P-Preferred-Identity was included, then use the default public user identity for the security association or TLS session and the associated contact address as the public user identity for this request;

...

If this is a request for a new dialog, the Contact header field is populated as follows:

1) a contact header value which is one of:

   - if a public GRUU value ("pub-gruu" header field parameter) has been saved associated with the public user identity to be used for this request, and the UE does not indicate privacy of the P-Asserted-Identity, then the UE should insert the public GRUU ("pub-gruu" header field parameter) value as specified in RFC 5627 [93]; or

   - if a temporary GRUU value ("temp-gruu" header field parameter) has been saved associated with the public user identity to be used for this request, and the UE does indicate privacy of the P-Asserted-Identity, then the UE should insert the temporary GRUU ("temp-gruu" header field parameter) value as specified in RFC 5627 [93];

   - otherwise, a SIP URI containing the contact address of the UE that has been previously registered without any contact parameters dedicated to registration procedure;

NOTE 7: The above items are mutually exclusive.

...

If available to the UE (as defined in the access technology specific annexes for each access technology), the UE shall insert a P-Access-Network-Info header field into any request for a dialog, any subsequent request (except CANCEL requests) or response (except CANCEL responses) within a dialog or any request for a standalone method (see subclause 7.2A.4). Insertion of the P-Access-Network-Info header field into the ACK request is optional.

NOTE 13: During the dialog, the points of attachment to the IP-CAN of the UE can change (e.g. UE connects to different cells). The UE will populate the P-Access-Network-Info header field in any request or response within a dialog with the current point of attachment to the IP-CAN (e.g. the current cell information).

NOTE 14: The value of the P-Access-Network-Info header field could be stale if the point of attachment of the UE with the network changes before the message is received by the network.

The UE shall build a proper preloaded Route header field value for all new dialogs and standalone transactions. The UE shall build a list of Route header field values made out of the following, in this order:

a) the P-CSCF URI containing the IP address acquired at the time of the P-CSCF discovery procedures which was used in registration of the contact address (or registration flow); and

NOTE 15: If the UE is provisioned with or receives a FQDN at the time of the P-CSCF discovery procedures, the FQDN is resolved to an IP address at the time of the P-CSCF discovery procedures.

b) the P-CSCF port based on the security mechanism in use:

- if IMS AKA or SIP digest with TLS is in use as a security mechanism, the protected server port learnt during the registration procedure;

...

c) and the values received in the Service-Route header field saved from the 200 (OK) response to the last registration or re-registration of the public user identity with associated contact address.

NOTE 16: When the UE registers multiple contact addresses, there will be a list of Service-Route headers for each contact address. When sending a request using a given contact address and the associated security associations or TLS session, the UE will use the corresponding list of Service-Route headers to construct a list of Route headers.

[TS 24.341, clause 5.3.2.2]

On sending a REGISTER request, the SM-over-IP receiver shall indicate its capability to receive traditional short messages over IMS network by including a "+g.3gpp.smsip" parameter into the Contact header according to RFC 3840 [16].

6.1.3 Test description

6.1.3.1 Pre-test conditions

System Simulator:

- SS is configured with the IMSI within the USIM application, the home domain name, public and private user identities together with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) that is configured on the UICC card equipped into the UE.
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- SS is able to perform IMS AKA authentication for the IMPI, according to 3GPP TS 33.203 [16] clause 6.1.
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UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is switched off.

Preamble:

- None

6.1.3.2 Test procedure sequence

Table 6.1.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is switched on.				
2	Check: does the UE send an initial registration request?	-->	REGISTER	1	P
3	SS sends 401 Unauthorized.	<--	401 Unauthorized		
4	Check: does the UE send a subsequent registration request?	-->	REGISTER	2	P
5	SS sends 200 OK for REGISTER.	<--	200 OK		
	EXCEPTION: In parallel to the events described in steps 6 to 9, the steps specified in Table 6.1.3.2-2 may take place.				
6	Check: does the UE subscribe to reg-event?	-->	SUBSCRIBE	3	P
7	SS sends 200 OK for SUBSCRIBE.	<--	200 OK		
8	SS sends NOTIFY for reg-event package, containing full registration state information for the registered public user identity in the XML body.	<--	NOTIFY		
9	Check: does the UE acknowledge reception of NOTIFY?	-->	200 OK	4	P

Table 6.1.3.2-2: Parallel Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE sends a PUBLISH request.	-->	PUBLISH		
2	SS sends a 503 Service Unavailable response	<--	503 Service Unavailable		

6.1.3.3 Specific message contents

Table 6.1.3.3-1: REGISTER (step 2, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

Table 6.1.3.3-2: 401 Unauthorized for REGISTER (step 3, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.2, Condition A1

Table 6.1.3.3-3: REGISTER (step 4, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2 and A32

Table 6.1.3.3-4: 200 OK for REGISTER (step 5, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.3, Condition A2

Table 6.1.3.3-5: SUBSCRIBE for reg-event package (step 6, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.4, Conditions A1 and A7

Table 6.1.3.3-6: 200 OK for SUBSCRIBE (step 7, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.5, Condition A1

Table 6.1.3.3-7: NOTIFY for reg-event package (step 8, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.6, Condition A1

Table 6.1.3.3-8: 200 OK for requests other than REGISTER or SUBSCRIBE (step 9, table 6.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A8 and A22

Table 6.1.3.3-9: PUBLISH (step 1, table 6.1.3.2-2)

Derivation Path: TS 34.229-1 [2], Table in subclause A.4.3, Conditions A1 and A5

Table 6.1.3.3-10: 503 Service Unavailable (step 2, table 6.1.3.2-2)

Derivation Path: TS 34.229-1 [2], Table in subclause A.4.2

6.2 Initial Registration Failures / 5GS

6.2.1 Test Purpose (TP)

(1)

```
with { UE having sent unprotected REGISTER request }
ensure that {
  when { UE receiving a 503 (Service Unavailable) response without Retry-After header for the
unprotected REGISTER request sent }
   then { UE waits at most 5 minutes and then sends another unprotected REGISTER request }
            }
```

(2)

```
with { UE having sent an unprotected REGISTER request }
ensure that {
  when { UE receiving a 503 (Service Unavailable) response with Retry-After header for the initial
REGISTER request sent }
   then { UE waits until interval given is up and then sends another unprotected REGISTER request }
            }
```

(3)

```
with { UE having sent unprotected REGISTER request }
ensure that {
  when { UE receiving a 423 (Interval Too Brief) response }
   then { UE sends another unprotect REGISTER request with new expiration interval }
            }
```

6.2.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.1.2.1, Release 17]:

For each 4xx, 5xx or 6xx response received without a Retry-After header field to the REGISTER request, the UE shall:

a) use the mechanism defined in subclause 4.5 of RFC 5626 [92] for determination of the retry delay time before each new registration attempt;

b) mark the currently used P-CSCF address as unavailable for the last duration of the retry delay time computed by the algorithm defined in subclause 4.5 of RFC 5626 [92] plus 5 minutes; and

c) initiate an initial registration as specified in subclause 5.1.1.2 after the amount of time of the last retry delay time computed by the algorithm defined in subclause 4.5 of RFC 5626 [92]; and

  - if there is a locally stored P-CSCF address as specified in subclause 5.1.9 which is different from the currently used P-CSCF address and which is not marked as unavailable, may initiate the initial registration using that P-CSCF; and

  - if there is no locally stored P-CSCF address as specified in subclause 5.1.9 which is different from the currently used P-CSCF address and which is not marked as unavailable, may get a new set of P-CSCF addresses as described in subclause 9.2.1 unless otherwise specified in the access specific annexes (as described in annex B, annex L or annex U) and initiate the initial registration as specified in subclause 5.1.1.2.

The values of max-time and base-time (if all failed) may be provided by the network to the UE with the management objects specified in 3GPP TS 24.167 [8G]. If no values of the parameters max-time and base-time (if all failed) have been provided to the UE by the network, the default values defined in subclause 4.5 of RFC 5626 [92] shall be used. Other mechanisms may be used as well and are outside the scope of the present document.

[TS 24.229, clause 5.1.1.2.1]:

On receiving a 503 response with a Retry-After header field to the REGISTER request and the Retry-After header field indicates time bigger than the value for timer F as specified in table 7.7.1, the UE:

a) shall mark the currently used P-CSCF address as unavailable for the time indicated by the Retry-After header field;

b) if there is a locally stored P-CSCF address as specified in subclause 5.1.9 which is different than the currently used P-CSCF address and which is not marked as unavailable, may initiate an initial registration as specified in subclause 5.1.1.2 using that P-CSCF; and

c) if there is no locally stored P-CSCF address as specified in subclause 5.1.9 which is different than the currently used P-CSCF address and which is not marked as unavailable, may get a new set of P-CSCF addresses as described in subclause 9.2.1 unless otherwise specified in the access specific annexes (as described in annex B, annex L or annex U) and initiate an initial registration as specified in subclause 5.1.1.2.

NOTE 19: if the Retry-After header field indicates time smaller than the value for timer F as specified in table 7.7.1, the UE continues using the currently used P-CSCF address.

[TS 24.229, clause 5.1.1.2.1]:

On receiving a 423 (Interval Too Brief) response to the REGISTER request, the UE shall:

- send another REGISTER request populating the registration expiration interval value with an expiration timer of at least the value received in the Min-Expires header field of the 423 (Interval Too Brief) response.

6.2.3 Test description

6.2.3.1 Pre-test conditions

System Simulator:

- SS is configured with the IMSI within the USIM application, the home domain name, public and private user identities together with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) that is configured on the UICC card equipped into the UE.

- SS is listening to SIP default port 5060 for both UDP and TCP protocols.

- SS is able to perform IMS AKA authentication for the IMPI, according to 3GPP TS 33.203 [16] clause 6.1.

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- UE provided with a second P-CSCF in the PDU SESSION ESTABLISHMENT ACCEPT according to Table 4.7.2-2 (conditions Additional_P-CSCF_IPv6 and Additional_P-CSCF_IPv4) of TS 38.508-1 [21].

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.

- The UE is configured to register for IMS after switch on.

- The UE is switched off.

Preamble:

- None

6.2.3.2 Test procedure sequence

Table 6.2.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is switched on			-	-
2	UE sends an initial registration request.	-->	REGISTER	-	-
3	SS sends 503 Service Unavailable without Retry-After header.	<--	503 Service Unavailable	-	-
-	The subsequent messages are sent to second P-CSCF	-	-	-	-
4	Check: does the UE send an initial registration request ?	-->	REGISTER	1	P
5	SS sends 503 Service Unavailable with Retry-After header set to 10 seconds.	<--	503 Service Unavailable	-	-
6	Check: does the UE send an initial registration request, but no earlier than 10 seconds after step 5?	-->	REGISTER	2	P
7	SS sends 423 Interval Too Brief.	<--	423 Interval Too Brief	-	-
8	Check: does the UE send an initial registration request with an expiration value set to the value provided in Step 7?	-->	REGISTER	3	P
9	SS sends 401 Unauthorized.	<--	401 Unauthorized	-	-
10	UE sends a subsequent registration request.	-->	REGISTER	-	-
11	SS sends 200 OK for REGISTER	<--	200 OK	-	-
	EXCEPTION: In parallel to the events described in steps 12 to 15, the steps specified in Table 6.1.3.2-2 may take place.	-	-	-	-
12-15	Steps 5-8 from clause A.2.	-	-	-	-

Table 6.2.3.2-2: Parallel Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE sends a PUBLISH request.	-->	PUBLISH	-	-
2	SS sends a 503 Service Unavailable response	<--	503 Service Unavailable	-	-

6.2.3.3 Specific message contents

Table 6.2.3.3-1: REGISTER (step 2, table 6.2.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

Table 6.2.3.3-2: 503 Service Unavailable (step 3, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.4.2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Retry-After		not present		

Table 6.2.3.3-3: REGISTER (step 4, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

Table 6.2.3.3-4: 503 Service Unavailable (step 5, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.4.2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Retry-After				
delta-seconds		10		

Table 6.2.3.3-5: REGISTER (step 6, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

Table 6.2.3.3-6: 423 Interval Too Brief (step 7, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.7				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Min-Expires				
delta-seconds		800000		

Table 6.2.3.3-7: REGISTER (step 8, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		800000		
Expires				
delta-seconds		800000 Note: value 800000 is given in at least one of Contact or Expires header.		
CSeq				
value		incremented from previous REGISTER		

Table 6.2.3.3-8: 401 Unauthorized (step 9, table 6.2.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.2, Condition A1

Table 6.2.3.3-9: REGISTER (step 10, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2 and A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		800000		
Expires				
delta-seconds		800000 Note: value 800000 is given in at least one of Contact or Expires header.		

Table 6.2.3.3-10: 200 OK (step 11, table 6.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.3, Condition A2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		800000		

6.3 Re-Registration Scenarios / 5GS

6.3.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS with expiration interval at 120 seconds }
ensure that {
  when { 60 seconds passed }
    then {UE re-registers }
            }
```

(2)

```
with { UE starting re-registration procedure by sending REGISTER }
ensure that {
  when { UE receives 500 Server Internal Error response }
    then {UE starts initial registration }
            }
```

(3)

```
with { UE being registered to IMS with expiration interval at 360 seconds }
ensure that {
  when { 180 seconds passed }
    then {UE re-registers }
            }
```

(4)

```
with { UE being registered to IMS with expiration interval at 1600 seconds }
ensure that {
  when { 1000 seconds passed }
    then {UE re-registers }
            }
```

(5)

```
with { UE attempting re-registration }
ensure that {
  when { UE receives 423 Interval Too Brief response }
    then {UE sends another re-registration request with given expiration interval }
            }
```

(6)

```
with { UE being registered }
ensure that {
  when { UE receives notification about shortened expiration interval for one of its registered
public user identities }
    then {UE re-registers after half of the shorted expiration interval elapses }
            }
```

6.3.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.1.4.1]:

The UE can perform the reregistration of a previously registered public user identity bound to any one of its contact addresses and the associated set of security associations or TLS sessions at any time after the initial registration has been completed.

...

Unless either the user or the application within the UE has determined that a continued registration is not required the UE shall reregister an already registered public user identity either 600 seconds before the expiration time if the previous registration was for greater than 1200 seconds, or when half of the time has expired if the previous registration was for 1200 seconds or less,

...

On receiving a 423 (Interval Too Brief) response to the REGISTER request, the UE shall:

- send another REGISTER request populating the registration expiration interval value with an expiration timer of at least the value received in the Min-Expires header field of the 423 (Interval Too Brief) response.

On receiving a 408 (Request Timeout) response or 500 (Server Internal Error) response or 504 (Server Time-Out) response or 403 (Forbidden) response for a reregistration, the UE shall perform the procedures for initial registration as described in subclause 5.1.1.2.

[TS 24.229, clause 5.1.1.4.1]:

At any time, the UE can receive a NOTIFY request carrying information related to the reg event package (as described in subclause 5.1.1.3). If:

- the state attribute in any of the <registration> elements is set to "active";
- the value of the <uri> sub-element inside the <contact> sub-element is set to the Contact address that the UE registered; and
- the event attribute of that <contact> sub-element(s) is set to "shortened";

the UE shall:

1) use the expires attribute of the <contact> sub-element that the UE registered to adjust the expiration time for that public user identity; and
2) start the re-authentication procedures at the appropriate time (as a result of the S-CSCF procedure described in subclause 5.4.1.6) by initiating a reregistration as described in subclause 5.1.1.4, if required.

NOTE: When authenticating a given private user identity, the S-CSCF will only shorten the expiry time within the <contact> sub-element that the UE registered using its private user identity. The <contact> elements for the same public user identity, if registered by another UE using different private user identities remain unchanged. The UE will not initiate a reregistration procedure, if none of its <contact> sub-elements was modified.

6.3.3 Test description

6.3.3.1 Pre-test conditions

System Simulator:

- SS is configured with the IMSI within the USIM application, the home domain name, public and private user identities together with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) that is configured on the UICC card equipped into the UE.
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- SS is able to perform IMS AKA authentication for the IMPI, according to 3GPP TS 33.203 [16] clause 6.1.
- 1 NR Cell

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is switched off.

Preamble:

- None

6.3.3.2 Test procedure sequence

Table 6.3.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is switched on.				
2-9	Steps 1-8 from clause A.2: initial IMS registration happens, with SS giving 120 seconds expiration interval.				
10	UE re-registers 60 seconds later.	-->	REGISTER	1	P
11	SS declines re-registration attempt.	<--	500 Server Internal Error		
12	Step 1 from clause A.2: UE sends initial IMS registration request	-->	REGISTER	2	P
13-19	Steps 2-8 from clause A.2, with SS giving 360 seconds expiration interval.				
20	UE re-registers 180 seconds later.	-->	REGISTER	3	P
21	SS responds with 1600 seconds expiration interval	<--	200 OK		
22	UE re-registers 1000 seconds later	-->	REGISTER	4	P
23	SS responds with 423 Interval Too Brief with Min-Expires value of 800000 seconds	<--	423 Interval Too Brief		
24	UE sends a new another re-registration request using at least 800000 seconds expiration.	-->	REGISTER	5	P
25	SS responds with 200 OK.	<--	200 OK		
26	SS notifies UE about shortened expiration time of 60 seconds for one of the registered public user identities.	<--	NOTIFY		
27	UE responds with 200 OK	-->	200 OK		
28	30 seconds before new expiry time, UE re-registers	-->	REGISTER	6	P
29	SS responds with authentication challenge and security mechanism supported by the network	<--	401 Unauthorized		
30	UE completes security procedures	-->	REGISTER		
31	SS responds with 200 OK	<--	200 OK		

6.3.3.3 Specific message contents

Table 6.3.3.3-1: 200 OK for REGISTER (step 5, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		120		

Table 6.3.3.3-2: REGISTER (step 10, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2, A17, A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Security-Client				
spi-c		new SPI number of the inbound SA at the protected client port, shall be different from previously used number		
spi-s		new SPI number of the inbound SA at the protected server port, shall be different from previously used number		
port-c		new protected client port, shall be different from previously used number		
port-s		same value as in the previous REGISTER		
Authorization				RFC 2617 [23]
nonce-count		2		TS 24.229 [7]

Table 6.3.3.3-3: 500 Server Internal Error (step 11, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.4.7

Table 6.3.3.3-4: 200 OK for REGISTER (step 15, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		360		

Table 6.3.3.3-5: REGISTER (step 20, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2, A17, A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Security-Client				
spi-c		new SPI number of the inbound SA at the protected client port, shall be different from previously used numbers		
spi-s		new SPI number of the inbound SA at the protected server port, shall be different from previously used numbers		
port-c		new protected client port, shall be different from previously used numbers		
port-s		same value as in the previous REGISTER		
Authorization				RFC 2617 [23]
nonce-count		2		TS 24.229 [7]

Table 6.3.3.3-6: 200 OK for REGISTER (step 21, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		1600		

Table 6.3.3.3-7: REGISTER (step 22, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2, A17, A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Security-Client				
spi-c		new SPI number of the inbound SA at the protected client port, shall be different from previously used numbers		
spi-s		new SPI number of the inbound SA at the protected server port, shall be different from previously used numbers		
port-c		new protected client port, shall be different from previously used numbers		
port-s		same value as in the previous REGISTER		
Authorization				RFC 2617 [23]
nonce-count		3		TS 24.229 [7]

Table 6.3.3.3-8: 423 Interval Too Brief (step 23, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.7				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Min-Expires				
delta-seconds		800000		

Table 6.3.3.3-9: REGISTER (step 24, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2, A17, A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		800000 or more (Remark: either the Contact header contains such expires parameter or below Expires header is present. If both are present, Expires header is to be ignored)		
Expires				
delta-seconds		800000 or more (Remark: either the Contact header contains above expires parameter or Expires header is present. If both are present, Expires header is to be ignored)		
Authorization				RFC 2617 [23]
nonce-count		4		TS 24.229 [7]

Table 6.3.3.3-9A: 200 OK for REGISTER (step 25, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
expires		*800000*		

Table 6.3.3.3-10: NOTIFY (step 26, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.6, Conditions A1, and A3 OR A4				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body				
	A3	*<?xml version="1.0" encoding="UTF-8"?>* *<reginfo xmlns="urn:ietf:params:xml:ns:reginfo" version="1" state="partial">* *<registration aor=*" PublicUserIdentity1 (NOTE 1)" *id="a100" state="active">* *<contact id="980" state="active" event="shortened" expires="60">* *<uri>*same value as in Contact header of REGISTER request*</uri>* *</contact>* *</registration>* *</reginfo>*		
	A4	*<?xml version="1.0" encoding="UTF-8"?>* *<reginfo xmlns="urn:ietf:params:xml:ns:reginfo" xmlns:gr="urn:ietf:params:xml:ns:gruuinfo" version="1" state="partial">* *<registration aor=*" PublicUserIdentity1 (NOTE 1)" *id="a100" state="active">* *<contact id="980" state="active" event="shortened" expires="60">* *callid="Call-Id of most recent REGISTER" cseq="CSeq value of most recent REGISTER">* *<uri>*same value as in Contact header of REGISTER request*</uri>* *<unknown-param name="+sip.instance"> "Instance ID of the UE;" </unknown-param>* *<gr:pub-gruu uri="public GRUU associated to this aor"/>* *<gr:temp-gruu uri="temporary GRUU associated to this aor" first-cseq="CSeq of the REGISTER request that caused the temporary GRUU to assigned for the UE"/>* *</contact>* *</registration>* *</reginfo>*		

Table 6.3.3.3-11: 200 OK for other requests than REGISTER or SUBSCRIBE (step 27, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A5, A11, A22

Table 6.3.3.3-12: REGISTER (step 28, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2, A17, A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact expires		 *800000* (if present)		
Expires delta-seconds		present if no expires parameter in Contact header *800000*		
Authorization nonce-count		 5		

Table 6.3.3.3-13: 401 Unauthorized (step 29, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.2, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
WWW-Authenticate nonce		 Base 64 encoding of new RAND and new AUTN (different from the values used in step 3)		RFC 2617 [23] TS 24.229 [7]

Table 6.3.3.3-14: REGISTER (step 30, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A2, A17, A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact expires		*800000* (if present)		
Expires delta-seconds		present if no expires parameter in Contact header *800000*		
Authorization nonce-count		1		

Table 6.3.3.3-15: 200 OK for REGISTER (step 31, Table 6.3.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.3, Condition A2

6.4 De-Registration Scenarios / 5GS

6.4.1 Test Purpose (TP)

(1)

```
Void
```

(2)

```
with { UE being registered to IMS }
ensure that {
  when { UE receiving a NOTIFY request containing de-registration information with contact elements
being "deactivated" }
   then { UE acknowledges de-registration }
            }
```

(3)

```
with { UE being de-registered from IMS by the network with contact elements being "deactivated" }
ensure that {
  when { UE acknowledging de-registration }
   then { UE performs initial registration to IMS }
            }
```

(4)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to de-register its contact address }
   then { UE performs de-registration from IMS }
            }
```

6.4.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.4.1.5]:

For any registered public user identity, the S-CSCF can deregister:

- all contact addresses bound to the indicated public user identity (i.e. deregister the respective public user identity);
- some contact addresses bound to the indicated public user identity;
- a particular contact address bound to the indicated public user identity; or
- one or more registration flows and the associated contact address bound to the indicated public user identity, when the UE supports multiple registration procedure;

by sending a single NOTIFY request.

...

When a network-initiated deregistration event occurs for one or more public user identities that are bound either to one or more contact addresses or registration flows and the associated contact addresses (if the multiple registration mechanism is used), the S-CSCF shall send a NOTIFY request to all subscribers that have subscribed to the respective reg event package. For each NOTIFY request, the S-CSCF shall:

1) set the Request-URI and Route header field to the saved route information during subscription;
2) set the Event header field to the "reg" value;
3) in the body of the NOTIFY request, include as many <registration> elements as many public user identities the S-CSCF is aware of the user owns;
4) set the aor attribute within each <registration> element to one public user identity:

a) set the <uri> sub-element inside each <contact> sub-element of each <registration> element to the respective contact address provided by the UE;

b) if the public user identity:

i) has been deregistered (i.e. all contact addresses and all registration flows and associated contact addresses bound to the indicated public user identity are removed) then:

- set the state attribute within the <registration> element to "terminated";

- set the state attribute within each <contact> element belonging to this UE to "terminated"; and

- set the event attribute within each <contact> element belonging to this UE to either "unregistered", or "deactivated" if the S-CSCF expects the UE to reregister or "rejected" if the S-CSCF does not expect the UE to reregister; or

...

When sending a final NOTIFY request with all <registration> element(s) having their state attribute set to "terminated" (i.e. all public user identities have been deregistered or expired), the S-CSCF shall also terminate the subscription to the registration event package by setting the Subscription-State header field to the value of "terminated".

[TS 24.229, clause 5.1.1.7]:

Upon receipt of a NOTIFY request, on any dialog which was generated during the subscription to the reg event package as described in subclause 5.1.1.3, including one or more <registration> element(s) which were registered by this UE, with:

1) the state attribute within the <registration> element set to "terminated", and within each <contact> element belonging to this UE, the state attribute set to "terminated" and the event attribute set either to "unregistered", or "rejected", or "deactivated", the UE shall remove all registration details relating to the respective public user identity (i.e. consider the public user identity indicated in the aor attribute of the <registration> element as deregistered); or

...

In case of a "deactivated" event attribute, the UE shall start the initial registration procedure as described in subclause 5.1.1.2. In case of a "rejected" event attribute, the UE shall release all dialogs related to those public user identities.

Upon receipt of a NOTIFY request, the UE shall delete all security associations or TLS sessions towards the P-CSCF either:

- if all <registration> element(s) have their state attribute set to "terminated" (i.e. all public user identities are deregistered) and the Subscription-State header field contains the value of "terminated"; or

- if each <registration> element that was registered by this UE has either the state attribute set to "terminated", or the state attribute set to "active" and the state attribute within the <contact> element belonging to this UE set to "terminated".

When all UE's public user identities are registered via a single P-CSCF and the subscription dialog to the reg event package of the UE is set via the respective P-CSCF, the UE shall delete these security associations or TLS sessions towards the respective P-CSCF when all public user identities have been deregistered and after the server transaction (as defined in RFC 3261 [26]) pertaining to the received NOTIFY request terminates.

NOTE 3: Deleting a security association or TLS session is an internal procedure of the UE and does not involve any SIP procedures.

NOTE 4: If all the public user identities (i.e. <contact> elements) registered by this UE are deregistered and the security associations or TLS sessions have been removed, the UE considers the subscription to the reg event package terminated since the NOTIFY request was received with Subscription-State header field containing the value of "terminated".

[TS 24.229, clause 5.1.1.6.1]:

For any public user identity that the UE has previously registered, the UE can deregister via a single registration procedure:

- all contact addresses bound to the indicated public user identity;
- some contact addresses bound to the indicated public user identity;
- a particular contact address bound to the indicated public user identity; or
- when the UE supports multiple registrations (i.e. the "outbound" option tag is included in the Supported header field) one or more flows bound to the indicated public user identity.

The UE can deregister a public user identity that it has previously registered with its contact address at any time. The UE shall protect the REGISTER request using a security association or TLS session that is associated with contact address, see 3GPP TS 33.203 [19], established as a result of an earlier registration, if one is available.

The UE shall extract or derive a public user identity, the private user identity, and the domain name to be used in the Request-URI in the registration, according to the procedures described in subclause 5.1.1.1A or subclause 5.1.1.1B.

Prior to sending a REGISTER request for deregistration, the UE shall release all dialogs that were using the contact addresses or the flow that is going to be deregistered and related to the public user identity that is going to be deregistered or to one of the implicitly registered public user identities. However:

- if the dialog that was established by the UE subscribing to the reg event package used the public user identity that is going to be deregistered; and
- this dialog is the only remaining dialog used for subscription to reg event package of the user, i.e. there are no other contact addresses registered with associated subscription to the reg event package of the user;

then the UE shall not release this dialog.

On sending a REGISTER request that will remove the binding between the public user identity and one of its contact addresses or one of its flows, the UE shall populate the header fields as follows:

a) a From header field set to the SIP URI that contains:

   1) if the UE supports RFC 6140 [191] and performs the functions of an external attached network, the main URI of the UE; else

   2) the public user identity to be deregistered;

b) a To header field set to the SIP URI that contains:

   1) if the UE supports RFC 6140 [191] and performs the functions of an external attached network, the main URI of the UE; else

   2) the public user identity to be deregistered;

c) a Contact header field set to the SIP URI(s) that contain(s) in the hostport parameter the IP address of the UE or FQDN, and:

   1) if the UE is removing the binding between the public user identity indicated in the To header field, (together with the associated implicitly registered public user identities), and the contact address indicated in the Contact header field; and

      - if the UE supports GRUU, or multiple registrations (i.e. the "outbound" option tag is included in the Supported header field), or has an IMEI available, or has an MEID available, the Contact header field also contains the "+sip.instance" header field parameter. Only the IMEI shall be used for generating an instance ID for a multi-mode UE that supports both 3GPP and 3GPP2 defined radio access networks;

      - if the UE supports multiple registrations (i.e. the "outbound" option tag is included in the Supported header field), the Contact header field does not contain the "reg-id" header field parameter;

      - if the UE does not supports GRUU and does not support multiple registrations (i.e. the "outbound" option tag is not included in the Supported header field), and does not have an IMEI available, and does not have an MEID available, the Contact header field does not contain either the "+sip.instance" header field parameter or the "reg-id" header field parameter;

NOTE 1: Since the contact address is deregistered, if there are any flows that were previously registered with the respective contact address, all flows terminating at the respective contact address are removed.

2) if the UE is removing the binding between the public user identity indicated in the To header field, (together with the associated implicitly registered public user identities) and one of its flows, the Contact header field contains the "+sip.instance" header field parameter and the "reg-id" header field parameter that identifies the flow; and

NOTE 2: The requirement placed on the UE to include an instance ID based on the IMEI when the UE does not support GRUU and does not support multiple registrations does not imply any additional requirements on the network.

3) if the UE supports RFC 6140 [191] and performs the functions of an external attached network, for the registration of bulk number contacts the UE shall include a Contact URI without a user portion and containing the "bnc" URI parameter;

d) a Via header field set to include the IP address or FQDN of the UE in the sent-by field;

e) a registration expiration interval value set to the value of zero, appropriate to the deregistration requirements of the user;

f) a Request-URI set to the SIP URI of the domain name of the home network used to address the REGISTER request;

g) if available to the UE (as defined in the access technology specific annexes for each access technology), a P-Access-Network-Info header field set as specified for the access network technology (see subclause 7.2A.4);

h) a Security-Client header field to announce the media plane security mechanisms the UE supports, if any;

NOTE 3: The "mediasec" header field parameter indicates that security mechanisms are specific to the media plane.

i) if the UE supports RFC 6140 [191] and performs the functions of an external attached network, for the registration of bulk number contacts the UE shall include a Require header field containing the option-tag "gin"; and

j) if the UE supports RFC 6140 [191] and performs the functions of an external attached network, for the registration of bulk number contacts the UE shall include a Proxy-Require header field containing the option-tag "gin".

For a public user identity that the UE has registered with multiple contact addresses or multiple flows (e.g. via different P-CSCFs), the UE shall also be able to deregister multiple contact addresses or multiple flows, bound to its public user identity, via single deregistration procedure as specified in RFC 3261 [26]. The UE shall send a single REGISTER request, using one of its contact addresses and the associated set of security associations or TLS session, containing a list of Contact headers. Each Contact header field is populated as specified above in bullets a) through i).

The UE can deregister all contact addresses bound to its public user identity and associated with its private user identity. The UE shall send a single REGISTER request, using one of its contact addresses and the associated set of security associations or TLS session, containing a public user identity that is being deregistered in the To header field, and a single Contact header field with value of "*" and the Expires header field with a value of "0". The UE shall not include the "instance-id" feature tag and the "reg-id" header field parameter in the Contact header field in the REGISTER request.

NOTE 4: All entities subscribed to the reg event package of the user will be informed via NOTIFY request which contact addresses bound to the public user identity have been deregistered.

When a 401 (Unauthorized) response to a REGISTER request is received the UE shall behave as described in subclause 5.1.1.5.1.

On receiving the 200 (OK) response to the REGISTER request, the UE shall:

- remove all registration details relating to this public user identity and the associated contact address.

- store the announcement of the media plane security mechanisms the P-CSCF (IMS-ALG) supports labelled with the "mediasec" header field parameter specified in subclause 7.2A.7 and received in the Security-Server header field, if any.

NOTE 5: The "mediasec" header field parameter indicates that security mechanisms are specific to the media plane.

If there are no more public user identities registered with this contact address, the UE shall delete any stored media plane security mechanisms and related keys and any security associations or TLS sessions and related keys it may have towards the IM CN subsystem.

If all public user identities are deregistered and all security association or TLS session is removed, then the UE shall consider subscription to the reg event package cancelled (i.e. as if the UE had sent a SUBSCRIBE request with an Expires header field containing a value of zero).

[TS 24.229, clause 5.1.1.6.2]:

On sending a REGISTER request, as defined in subclause 5.1.1.6.1, the UE shall additionally populate the header fields as follows:

a) an Authorization header field, with:

   - the "username" header field parameter, set to the value of the private user identity;

   - the "realm" header field parameter, set to the value as received in the "realm" WWW-Authenticate header field parameter;

   - the "uri" header field parameter, set to the SIP URI of the domain name of the home network;

   - the "nonce" header field parameter, set to last received nonce value; and

   - the response directive, set to the last calculated response value;

b) additionally for each Contact header field and associated contact address, include the associated protected server port value in the hostport parameter;

c) additionally for the Via header field, include the protected server port value bound to the security association in the sent-by field;

NOTE 1: If the UE specifies its FQDN in the hostport parameter in the Contact header field and in the sent-by field in the Via header field, then it has to ensure that the given FQDN will resolve (e.g., by reverse DNS lookup) to the IP address that is bound to the security association.

d) a Security-Client header field, set to specify the signalling plane security mechanisms it supports, the IPsec layer algorithms for integrity and confidentiality protection it supports and the new parameter values needed for the setup of two new pairs of security associations. For further details see 3GPP TS 33.203 [19] and RFC 3329 [48]; and

e) a Security-Verify header field that contains the content of the Security-Server header field received in the 401 (Unauthorized) response of the last successful authentication.

NOTE 2: When the UE has received the 200 (OK) response for the REGISTER request of the only public user identity currently registered with this contact address and its associated set of implicitly registered public user identities (i.e. no other public user identity is registered), the UE removes the security association (between the P-CSCF and the UE) that were using this contact address. Therefore further SIP signalling using this security association (e.g. the NOTIFY request containing the deregistration event) will not reach the UE.

6.4.3 Test description

6.4.3.1 Pre-test conditions

System Simulator:

- SS is configured with the IMSI within the USIM application, the home domain name, public and private user identities together with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) that is configured on the UICC card equipped into the UE.

- SS is listening to SIP default port 5060 for both UDP and TCP protocols.

- SS is able to perform IMS AKA authentication for the IMPI, according to 3GPP TS 33.203 [16] clause 6.1.
- 1 NR Cell

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is switched off.

Preamble:

- None

6.4.3.2 Test procedure sequence

Table 6.4.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is switched on.	-	-	-	-
2-9	Steps 1-8 from Annex A.2: initial IMS registration happens.	-	-	-	-
10-12	Void	-		-	-
13	SS de-registers the UE's contact address.	<--	NOTIFY	-	-
14	UE acknowledges.	-->	200 OK	2	P
15-22	Steps 1-8 from Annex A.2: initial IMS registration happens. For the Request-URI, the UE uses the new value of home domain and/or IMS identities name as provided by ISIM after the update in step 10.	-	-	3	P
23	UE is made to de-register its contact address.	-	-	-	-
24-29	Steps 0A-2 defined in Annex A.11			4	P

6.4.3.3 Specific message contents

Table 6.4.3.3-0: Void

Table 6.4.3.3-1: NOTIFY (step 13, Table 6.4.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.6, Conditions A1 AND ((A3 AND A6) OR (A4 AND A6))

Table 6.4.3.3-2: 200 OK for other requests than REGISTER or SUBSCRIBE (step 14, Table 6.4.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A5, A11, A22

6.5 Refresh for ISIM parameters / 5GS

6.5.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE receives indication that contents of USIM/ISIM has been updated }
    then { UE does not de-register from IMS }
            }
```

(2)

```
with { UE waiting for network-initiated de-registration }
ensure that {
  when { UE receives NOTIFY request for reg event }
    then { UE responds with a valid 200 OK response and initiates new IMS registration }
            }
```

6.5.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 Annex C.4]:

3GPP TS 31.102 [15C] and 3GPP TS 31.103 [15B] specify the file structure and contents for the preconfigured parameters stored on the USIM and ISIM, respectively, necessary to initiate the registration to the IM CN subsystem. Any of these parameters can be updated via Data Download or a USAT application, as described in 3GPP TS 31.111 [15D]. If one or more EFs are changed and a REFRESH command is issued by the UICC, then the UE reads the updated parameters from the UICC as specified for the REFRESH command in 3GPP TS 31.111 [15D].

If the UE supports the UICC access to IMS USAT feature defined in 3GPP TS 31.111 [15D] and the $EF_{UICCIARI}$ changes in either the USIM or the ISIM, the UE shall perform the user-initiated reregistration procedure as described in subclause 5.1.1.4 with the new values of the IARI parameter(s) residing on the UICC.

In case of changes to EFs other than the $EF_{UICCIARI}$, the UE is not required to perform deregistration but it shall wait for the network-initiated deregistration procedures to occur as described in subclause 5.4.1.5 unless the user initiates deregistration procedures as described in subclause 5.1.1.6. From this point onwards the normal initial registration procedures can occur.

[TS 24.229 clause 5.1.1.7]:

Upon receipt of a NOTIFY request, on any dialog which was generated during the subscription to the reg event package as described in subclause 5.1.1.3, including one or more <registration> element(s) which were registered by this UE, with:

1) the state attribute within the <registration> element set to "terminated", and within each <contact> element belonging to this UE, the state attribute set to "terminated" and the event attribute set either to "unregistered", or "rejected", or "deactivated", the UE shall remove all registration details relating to the respective public user identity (i.e. consider the public user identity indicated in the aor attribute of the <registration> element as deregistered); or

2) the state attribute within the <registration> element set to "active", and within a given <contact> element belonging to this UE, the state attribute set to "terminated", and the associated event attribute set either to "unregistered", or "rejected" or "deactivated", the UE shall consider the binding between the public user identity and either the contact address or the registration flow and the associated contact address (if the multiple registration mechanism is used) indicated in the respective <contact> element as removed. The UE shall consider its public user identity as deregistered when all bindings between the respective public user identity and all contact addresses and all registration flow and the associated contact address (if the multiple registration mechanism is used) belonging to this UE are removed.

NOTE 1: When multiple registration mechanism is used to register a public user identity and bind it to a registration flow and the associated contact address, there will be one <contact> element for each registration flow and the associated contact address.

NOTE 2: If the state attribute within the <registration> element is set to "active" and the <contact> element belonging to this UE is set to "active", the UE will consider that the binding between the public user identity and either the respective contact address or the registration flow and the associated contact address as left unchanged.

In case of a "deactivated" event attribute, the UE shall start the initial registration procedure as described in subclause 5.1.1.2. In case of a "rejected" event attribute, the UE shall release all dialogs related to those public user identities.

6.5.3 Test description

6.5.3.1 Pre-test conditions

System Simulator:

- SS is configured with the old and new home domain name, public and private user identities (including the public emergency user identity allocated for the user) together with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) that is configured on the UICC card equipped into the UE.
- SS is able to perform AKAv1-MD5 authentication algorithm for that IMPI, according to 3GPP TS 33.203 [16] clause 6.1 and RFC 3310 [15].
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- 1 NR Cell

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is registered to IMS services.
- The Request-URI of SIP REGISTER request sent by the UE contained the old home domain name and IMS identities as found from ISIM.

Preamble:

- None.

6.5.3.2 Test procedure sequence

Table 6.5.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UICC is made to send a REFRESH command to the UE indicating that contents of ISIM has been updated.		REFRESH		
2	Check: does the UE de-register from IMS?			1	F
3	10 seconds after step 1 the SS sends SIP NOTIFY for registration event package, containing full registration state information, with all previously registered IMS public user identities as "terminated" and "deactivated"	<--	NOTIFY		
4	Check: does the UE respond the NOTIFY with 200 OK?	-->	200 OK	2	P
5	Check: does the UE initiate a new IMS registration sequence? For the Request-URI of SIP REGISTER request the UE uses the new value of home domain and/or IMS identities name as provided by ISIM after the update in step 1.	-->	REGISTER	2	P
6-12	Continue with Annex A.2 step 2-8 in order to get the UE in a stable registered state.	-			

6.5.3.3 Specific message contents

Table 6.5.3.3-1: REFRESH (step 1, table 6.5.3.2-1)

Derivation Path: TS 31.111, subclause 6.4.7

Table 6.5.3.3-2: NOTIFY for reg-event package (step 3, table 6.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.6, Conditions A1 AND ((A3 AND A6) OR (A4 AND A6))

Table 6.5.3.3-3: 200 OK for requests other than REGISTER or SUBSCRIBE (step 4, table 6.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A5, A11 and A22

Table 6.5.3.3-4: REGISTER (step 5, table 6.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

6.6 Re-Registration after capability update / 5GS

6.6.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to update its capabilities }
    then { UE re-registers }
            }
```

6.6.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.1.4.1]:

Unless either the user or the application within the UE has determined that a continued registration is not required the UE shall reregister an already registered public user identity either 600 seconds before the expiration time if the previous registration was for greater than 1200 seconds, or when half of the time has expired if the previous registration was for 1200 seconds or less, or when the UE intends to update its capabilities according to RFC 3840 [62] or when the UE needs to modify the ICSI values that the UE intends to use in a g.3gpp.icsi-ref media feature tag or IARI values that the UE intends to use in the g.3gpp.iari-ref media feature tag.

6.6.3 Test description

6.6.3.1 Pre-test conditions

System Simulator:

- SS is configured with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) that is configured on the UICC card equipped into the UE.
- SS is able to perform AKAv1-MD5 authentication algorithm for that IMPI, according to 3GPP TS 33.203 [16] clause 6.1 and RFC 3310 [15].
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- 1 NR Cell

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is registered to IMS services, by executing the generic test procedure in Annex A.2 up to the last step.
- UE is able to be made change its capabilities, manifested through a specific instance which is setting the AT Command +CASIMS (Availability for SMS using IMS, defined in 3GPP TS 27.007 [22] 8.72) to 0.

Preamble:

- None.

6.6.3.2 Test procedure sequence

Table 6.6.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Turning off the UE's SMS over IMS capability through AT command +CASIMS (3GPP TS 27.007 clause 8.72) set to 0.				
2	Check: does the UE initiate a re-registration procedure, and indicating the changed capabilities in the REGISTER message?	-->	REGISTER	1	P
3	Void				
4	Void				
5	SS responds with 200 OK for REGISTER	<--	200 OK		

6.6.3.3 Specific message contents

Table 6.6.3.3-1: REGISTER (step 2, table 6.6.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A2, A17 and A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
feature-param		does not contain "+g.3gpp.smsip"		
Authorization				
nonce-count		2		

Table 6.6.3.3-2: Void

Table 6.6.3.3-3: Void

Table 6.6.3.3-4: 200 OK for REGISTER (step 5, table 6.6.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.3, Condition A2

6.7 Authentication / MAC Parameter Invalid / Only two consecutive invalid challenges / 5GS

6.7.1 Test Purpose (TP)

(1)

```
with { UE starting registration procedure }
ensure that {
  when { UE receiving invalid MAC parameter }
    then { UE sends another REGISTER request without challenge response AUTS and populates a new
Security-Client header }
      }
```

(2)

```
with { UE having responded to invalid MAC parameter }
ensure that {
  when { UE receives another invalid MAC parameter }
    then { UE sends another REGISTER request without challenge response AUTS and populates a new
Security-Client header }
      }
```

(3)

Void

6.7.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.1.5.3]:

If, in a 401 (Unauthorized) response, either the MAC or SQN is incorrect the UE shall respond with a further REGISTER indicating to the S-CSCF that the challenge has been deemed invalid as follows:

- in the case where the UE deems the MAC parameter to be invalid the subsequent REGISTER request shall contain no "auts" Authorization header field parameter and an empty "response" Authorization header field parameter, i.e. no authentication challenge response;

…

Whenever the UE detects any of the above cases, the UE shall:

- send the REGISTER request using an existing set of security associations, if available (see 3GPP TS 33.203 [16]);
- populate a new Security-Client header field within the REGISTER request and associated contact address, set to specify the security mechanisms it supports, the IPsec layer algorithms for integrity and confidentiality protection it supports and the parameters needed for the new security association setup. These parameters shall contain new values for spi_uc, spi_us and port_uc; and
- not create a temporary set of security associations.

[TS 24.229 clause 5.1.1.5.12]:

A UE shall only respond to two consecutive invalid challenges and shall not automatically attempt authentication after receiving two consecutive invalid challenges. The UE may attempt to register with the network again after an implementation specific time.

6.7.3 Test description

6.7.3.1 Pre-test conditions

System Simulator:

- SS is configured with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) configured on the UICC card equipped into the UE.
- SS is able to perform AKAv1-MD5 authentication algorithm for that IMPI, according to 3GPP TS 33.203 [16] clause 6.1 and RFC 3310 [15].
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- 1 NR Cell

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is switched off.

Preamble:

- None.

6.7.3.2 Test procedure sequence

Table 6.7.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is switched on.				
2	UE sends initial registration for IMS services.	-->	REGISTER		
3	SS responds with an invalid AKAv1-MD5 authentication challenge with an invalid MAC value.	<--	401 Unauthorized		
4	Check: does the UE send a REGISTER request: - contains no AUTS directive and an empty response directive, i.e. no authentication challenge response - UE populates a new Security-Client header set to specify the security mechanism it supports, the IPsec layer algorithms it supports and the parameters needed for the new security association setup	-->	REGISTER	1	P
5	SS responds with an invalid AKAv1-MD5 authentication challenge with an invalid MAC value.	<--	401 Unauthorized		
6	Check: does the UE send another REGISTER request: - contains no AUTS directive and an empty response directive, i.e. no authentication challenge response - UE populates a new Security-Client header set to specify the security mechanism it supports, the IPsec layer algorithms it supports and the parameters needed for the new security association setup	-->	REGISTER	2	P
7	Void				
8	Void				
9-16	Steps 2-8 from Clause A.2 are performed.				

6.7.3.3 Specific message contents

Table 6.7.3.3-1: REGISTER (step 2, table 6.7.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

Table 6.7.3.3-2: 401 Unauthorized for REGISTER (step 3/5, table 6.7.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.2, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
WWW-Authenticate				
nonce		Base 64 encoding of RAND and AUTN, generated using invalid MAC value		

Table 6.7.3.3-3: REGISTER (step 4/6, table 6.7.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
CSeq				
value		must be incremented from previous REGISTER		
Call-ID				
callid		same value as in REGISTER at Step 2		
Authorization				
response		present, but empty		
auts		not present		
Security-Client				
spi-c		new SPI number of the inbound SA at the protected client port, shall be different from previously used number(s)		
spi-s		new SPI number of the inbound SA at the protected server port, shall be different from previously used number(s)		
port-c		new protected client port, shall be different from previously used number(s)		

6.8 Authentication / Security-Server missing / SQN out of range / 5GS

6.8.1 Test Purpose (TP)

(1)

```
with { UE starting registration procedure }
ensure that {
  when { UE receiving a challenge response without Security-Server header }
    then { UE abandons the authentication procedure and sends a new REGISTER request with new Call-
           ID }
            }
```

(2)

```
with { UE having sent a new initial REGISTER request }
ensure that {
  when { UE receiving a challenge response with SQN out of range }
    then { UE sends another REGISTER request with challenge response AUTS and populates a new
Security-Client header }
            }
```

6.8.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.1.5.1]:

Authentication is performed during initial registration. A UE can be re-authenticated during subsequent reregistrations, deregistrations or registrations of additional public user identities. When the network requires authentication or re-authentication of the UE, the UE will receive a 401 (Unauthorized) response to the REGISTER request.

On receiving a 401 (Unauthorized) response to the REGISTER request, the UE shall:

1) extract the RAND and AUTN parameters;

2) check the validity of a received authentication challenge, as described in 3GPP TS 33.203 [19] i.e. the locally calculated XMAC must match the MAC parameter derived from the AUTN part of the challenge; and the SQN parameter derived from the AUTN part of the challenge must be within the correct range; and

3) check the existence of the Security-Server header field as described in RFC 3329 [48]. If the Security-Server header field is not present or it does not contain the parameters required for the setup of the set of security associations (see annex H of 3GPP TS 33.203 [19]), the UE shall abandon the authentication procedure and send a new REGISTER request with a new Call-ID.

…

In the case that the 401 (Unauthorized) response is deemed to be invalid then the UE shall behave as defined in subclause 5.1.1.5.3.

[TS 24.229, clause 5.1.1.5.3]

If, in a 401 (Unauthorized) response, either the MAC or SQN is incorrect the UE shall respond with a further REGISTER indicating to the S-CSCF that the challenge has been deemed invalid as follows:

- in the case where the UE deems the MAC parameter to be invalid the subsequent REGISTER request shall contain no "auts" Authorization header field parameter and an empty "response" Authorization header field parameter, i.e. no authentication challenge response;

- in the case where the UE deems the SQN to be out of range, the subsequent REGISTER request shall contain the "auts" Authorization header field parameter (see 3GPP TS 33.102 [18]).

NOTE: In the case of the SQN being out of range, a "response" Authorization header field parameter can be included by the UE, based on the procedures described in RFC 3310 [49].

Whenever the UE detects any of the above cases, the UE shall:

- send the REGISTER request using an existing set of security associations, if available (see 3GPP TS 33.203 [19]);
- populate a new Security-Client header field within the REGISTER request and associated contact address, set to specify the security mechanisms it supports, the IPsec layer algorithms for integrity and confidentiality protection it supports and the parameters needed for the new security association setup. These parameters shall contain new values for spi_uc, spi_us and port_uc; and
- not create a temporary set of security associations.

6.8.3 Test description

6.8.3.1 Pre-test conditions

System Simulator:

- SS is configured with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) configured on the UICC card equipped into the UE.
- SS is able to perform AKAv1-MD5 authentication algorithm for that IMPI, according to 3GPP TS 33.203 [16] clause 6.1 and RFC 3310 [15].
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- 1 NR Cell

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is switched off.

Preamble:

- None.

6.8.3.2 Test procedure sequence

Table 6.8.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is switched on.				
2	UE sends initial registration for IMS services.	-->	REGISTER		
3	SS responds challenge response without Security-Server header.	<--	401 Unauthorized		
4	Check: does the UE sends a new REGISTER request with new Call-ID	-->	REGISTER	1	P
5	SS responds with an invalid AKAv1-MD5 authentication challenge with SQN out of range.	<--	401 Unauthorized		
6	Check: does the UE send another REGISTER request: - contains AUTS directive - UE populates a new Security-Client header set to specify the security mechanism it supports, the IPsec layer algorithms it supports and the parameters needed for the new security association setup.	-->	REGISTER	2	P
7-13	Continue with Annex A.2 step 2-8 in order to get the UE in a stable registered state.	-			

6.8.3.3 Specific message contents

Table 6.8.3.3-1: REGISTER (step 2, table 6.8.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1

Table 6.8.3.3-2: 401 Unauthorized for REGISTER (step 3, table 6.8.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.2, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Security-Server		not present.		

Table 6.8.3.3-3: REGISTER (step 4, table 6.8.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1, Condition A1 and A32				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Call-ID				
callid		Value differs from the one sent in in Step 2 of table 6.8.3.2-1.		

Table 6.8.3.3-4: 401 Unauthorized for REGISTER (step 5, table 6.8.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.2, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
WWW-Authenticate				
nonce		Base 64 encoding of RAND and AUTN, generated with SQN out of range with the AMF information field set to $AMF_{RESYNCH}$ value to trigger SQN re-synchronisation procedure in test ISIM/USIM, see TS 34.108 clause 8.1.2.2.		

Table 6.8.3.3-5: REGISTER (step 6, table 6.8.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.1, Conditions A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
CSeq				
value		must be incremented from previous REGISTER		
Call-ID				
callid		same value as in REGISTER at Step 4		
Authorization				
nonce		same as in previous 401 UNAUTHORIZED message		
opaque		same as in previous 401 UNAUTHORIZED message		
auts		any value		
Security-Client				
spi-c		new SPI number of the inbound SA at the protected client port, shall be different from previously used number		
spi-s		new SPI number of the inbound SA at the protected server port, shall be different from previously used number		
port-c		new protected client port, shall be different from previously used number		

6.9 Subscription / 503 Service Unavailable / 5GS

6.9.1 Test Purpose (TP)

(1)

```
with { UE subscribing to reg event }
ensure that {
  when { UE receives 503 Service unavailable containing a Retry-After header field }
    then { UE does not reattempt the request for the indicated time period }
            }
```

6.9.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2.2]:

If the UE receives a 503 (Service Unavailable) response to an initial SUBSCRIBE request containing a Retry-After header field, then the UE shall not automatically reattempt the request until after the period indicated by the Retry-After header field contents.

6.9.3 Test description

6.9.3.1 Pre-test conditions

System Simulator:

- SS is configured with the shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) configured on the UICC card equipped into the UE.
- SS has performed AKAv1-MD5 authentication with the UE and accepted the registration (IMS security).
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- 1 NR Cell

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is switched off.

Preamble:

- None.

6.9.3.2 Test procedure sequence

Table 6.9.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is switched on.				
2-5	Steps 1-4 of Annex A.2 happen.				
6	UE subscribes to its registration event package.	-->	SUBSCRIBE		
7	SS responds with 503 response containing a Retry-After header with period set to T=128s.	<--	503 Service Unavailable		
8	Check: does the SS receive the UE's re-attempt of SUBSCRIBE within the Time T=128s?			1	F
9	UE reattempts to subscribe to its registration event package.	-->	SUBSCRIBE		
10-12	Continue with Annex A.2 step 6-8 in order to get the UE in a stable registered state.	-			

6.9.3.3 Specific message contents

Table 6.9.3.3-1: SUBSCRIBE for reg-event package (step 6, table 6.9.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.4, Conditions A1 and A7

Table 6.9.3.3-2: 503 Service Unavailable (step 7, table 6.9.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.4.2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Retry-After				RFC 3261 [6]
period		128		

Table 6.9.3.3-3: SUBSCRIBE for reg-event package (step 9, table 6.9.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.1.4, Conditions A1 and A7				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Call-ID				RFC 3261 [6]
callid		value different from the previous SUBSCRIBE request		

7 Call Control

7.1 MTSI MO Voice Call / 503 Service Unavailable / 5GS

7.1.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions and UE having sent an initial
INVITE request for MO Voice call }
ensure that {
  when { UE receiving a 503 Service Unavailable response containing a Retry-After header indicating
a period of 20 seconds }
   then { UE does not reattempt the request until after the indicated period }
            }
```

(2)

```
with { UE waiting for a period of 20 seconds to expire }
ensure that {
  when { period expires }
   then { UE sends initial INVITE request and completes setup of MO Voice call }
            }
```

7.1.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.3.1]:

Upon receiving a 503 (Service Unavailable) response to an initial INVITE request containing a Retry-After header, then the originating UE shall not automatically reattempt the request until after the period indicated by the Retry-After header contents.

[TS 24.229, clause 6.1.2]:

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

…

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

7.1.3 Test description

7.1.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.

- The UE is configured to register for IMS after switch on.

- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.1.3.2 Test procedure sequence

Table 7.1.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call	-	-	-	-
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
-	EXCEPTION: In parallel with Step 8, parallel behaviour defined in table 7.1.3.2-2 takes place	-	-	-	-
-	EXCEPTION: Steps 8a to 8b describe behaviour that depends on UE configuration; the “lower case letter” identifies a step sequence that takes place if such configuration was conducted.	-	-	-	-
8a	IF the UE is configured to use preconditions THEN step1 of Annex A.4.1 takes place.	-	-	-	-
8b	ELSE step 1 of Annex A.4.2 takes place.	-	-	-	-
9	SS sends 503 (Service Unavailable) with Retry-After header indicating a period of 20 seconds.	<--	503 Service Unavailable	-	-
10	UE acknowledges the reception of 503 (Service Unavailable) message.	-->	ACK	-	-
11	The SS starts timer t_Waits=20s.	-	-	-	-
12	Check: Does the UE transmit INVITE request message.	-->	INVITE	1	F
13	The SS waits for expiry of t_Waits.	-	-	-	-
-	EXCEPTION: Steps 14a1 to 14b8 describe behaviour that depends on UE configuration; the “lower case letter” identifies a step sequence that takes place if such configuration was conducted.	-	-	-	-
14a1	IF the UE is configured to use preconditions THEN step 1 of Annex A.4.1 takes place.	-->	INVITE	2	P
14a2-14a12	Steps 2 to 12 of Annex A.4.1 take place	-	-	-	-
14b1	ELSE step 1 of Annex A.4.2 takes place	-->	INVITE	2	P
14b2-14b8	Steps 2 to 8 of Annex A.4.2 take place	-	-	-	-

Table 7.1.3.2-2: Parallel behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE transmits an *RRCReconfigurationComplete* message.	-->	NR RRC: *RRCReconfigurationComplete*	-	-

7.1.3.3 Specific message contents

Table 7.1.3.3-1: 503 Service Unavailable (step 9, table 7.1.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.4.2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Retry-After				
delta-seconds		20		

7.2 MTSI MO Voice Call / 504 Server Time-out / 5GS

7.2.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and UE having sent an INVITE request }
ensure that {
  when { UE receives 504 Server Time-out response }
    then { UE performs initial registration to IMS }
            }
```

7.2.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.2A.1.6]

In the event the UE receives a 504 (Server Time-out) response containing:

1) a P-Asserted-Identity header field set to a value equal to a URI:

   a) from the Service-Route header field value received during registration; or

   b) from the Path header field value received during registration; and

NOTE 1: If there are multiple registration flows associated with the registration, then the UE has received from the P-CSCF during registration multiple sets of Path header field and Service-Route header field values. The Path header field value and Service-Route header field value corresponding to the flow on which the 504 (Server Time-out) response was received are checked.

2) a Content-Type header field set according to subclause 7.6 (i.e. "application/3gpp-ims+xml"), independent of the value or presence of the Content-Disposition header field, independent of the value or presence of Content-Disposition parameters,

then the following treatment is applied:

a) if the 504 (Server Time-out) response includes an IM CN subsystem XML body as described in subclause 7.6 with the <ims-3gpp> element, including a version attribute, with the <alternative-service> child element:

   A) with the <type> child element set to "restoration" (see table 7.6.2); and

   B) with the <action> child element set to "initial-registration" (see table 7.6.3);

   then the UE:

   - shall initiate S-CSCF restoration procedures by performing an initial registration as specified in subclause 5.1.1.2; and

   - may provide an indication to the user based on the text string contained in the <reason> child element of the <alternative-service> child element of the <ims-3gpp> element.

NOTE 2: If the UE has discovered multiple P-CSCF addresses and has information that the P-CSCF was unable to forward the request resulting in sending back the 504 (Server Time-out) response, when starting the initial registration it is appropriate for the UE to select a P-CSCF address different from the one used for the registration binding on which the 504 (Server Time-out) response was received.

7.2.3 Test description

7.2.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- The UE is either configured to use preconditions or to not use preconditions or does not support preconditions.

Preamble:

- UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.2.3.2 Test procedure sequence

Table 7.2.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-	-	-
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
-	EXCEPTION: In parallel to INVITE at Step 8, step described in Table 7.2.3.2-2: Parallel behaviour takes place.	-->	-	-	-
8	Step 1 of Annex A.4.2 happens.	-->	INVITE	-	-
9	SS sends 504 Server Time-out	<--	504 Server Time-out	-	-
9A	UE acknowledges the reception of 504 Server Time-out.	-->	ACK	-	-
10	Check: Does the UE send an initial registration request?	-->	REGISTER	1	P
11-17	Continue with Annex A.2 steps 2-8 in order to get the UE in a stable registered state.	-	-	-	-

Table 7.2.3.2-2: Parallel behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE transmits an *RRCReconfigurationComplete* message.	-->	NR RRC: *RRCReconfigurationComplete*	-	-

Table 7.2.3.3-3: ACK (step 9A, Table 7.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.2.7 Conditions A1 and A4

7.2.3.3 Specific message contents

Table 7.2.3.3-1: 504 Server Time-out (step 3, Table 7.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.4.6

Table 7.2.3.3-2: REGISTER (step 4, Table 7.2.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.1.1 conditions A1 and A32

7.3 Void

7.4 MTSI MO Voice Call with preconditions at both originating and terminating UE / 5GS

7.4.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE is being made to initiate a voice call }
   then { UE sends INVITE for voice call with preconditions }
            }
```

(2)

```
with { UE having sent INVITE with preconditions }
ensure that {
  when { UE receives 100 Trying followed by 183 Session Progress }
   then { UE sends PRACK for 183 Session Progress }
            }
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK and resources are available }
   then { UE sends UPDATE }
            }
```

(4)

```
with { UE having sent UPDATE }
ensure that {
  when { UE receives 200 OK for UPDATE followed by 180 Ringing sent reliably }
   then { UE sends PRACK for 180 Ringing }
            }
```

(5)

```
with { UE having sent PRACK for 180 Ringing }
ensure that {
  when { UE receives 200 OK for PRACK followed by 200 OK for INVITE }
   then { UE sends ACK }
            }
```

7.4.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.2A.1.1]:

The procedures of this subclause are general to all requests and responses, except those for the REGISTER method.

When the UE re-uses a previously registered contact address, the UE shall remove any parameters dedicated to registration from the Contact header field (e.g. "expires").

When the UE sends any request, the UE shall use either a given contact address that has been previously registered or a registration flow and the associated contact address (if the multiple registration mechanism is used) and shall:

- if IMS AKA is in use as a security mechanism:

  a) if the UE has not obtained a GRUU, populate the Contact header field of the request with the protected server port and the respective contact address; and

b) include the protected server port and the respective contact address in the Via header field entry relating to the UE;

If this is a request for a new dialog, the Contact header field is populated as follows:

1) a contact header value which is one of:

- if a public GRUU value ("pub-gruu" header field parameter) has been saved associated with the public user identity to be used for this request, and the UE does not indicate privacy of the P-Asserted-Identity, then the UE should insert the public GRUU ("pub-gruu" header field parameter) value as specified in RFC 5627 [93]; or

- if a temporary GRUU value ("temp-gruu" header field parameter) has been saved associated with the public user identity to be used for this request, and the UE does indicate privacy of the P-Asserted-Identity, then the UE should insert the temporary GRUU ("temp-gruu" header field parameter) value as specified in RFC 5627 [93];

- otherwise, a SIP URI containing the contact address of the UE that has been previously registered without any contact parameters dedicated to registration procedure;

NOTE 7: The above items are mutually exclusive.

2) include an "ob" SIP URI parameter, if the UE supports multiple registrations, and the UE wants all subsequent requests in the dialog to arrive over the same flow identified by the flow token as described in RFC 5626 [92];

3) if the request is related to an IMS communication service that requires the use of an ICSI then the UE shall include in a g.3gpp.icsi-ref media feature tag, as defined in subclause 7.9.2 and RFC 3841 [56B], the ICSI value (coded as specified in subclause 7.2A.8.2) for the IMS communication service. The UE may also include other ICSI values that the UE is prepared to use for all dialogs with the terminating UE(s); and

4) if the request is related to an IMS application that is supported by the UE, then the UE may include in a g.3gpp.iari-ref media feature tag, as defined in subclause 7.9.3 and RFC 3841 [56B], the IARI value (coded as specified in subclause 7.2A.9.2) that is related to the IMS application and that applies for the dialog.

…

If this is a request for a new dialog or standalone transaction and the request is related to an IMS communication service that requires the use of an ICSI then the UE:

1) shall include the ICSI value (coded as specified in subclause 7.2A.8.2), for the IMS communication service that is related to the request in a P-Preferred-Service header field according to RFC 6050 [121]. If a list of network supported ICSI values was received as specified in 3GPP TS 24.167 [8G], the UE shall only include an ICSI value that is in the received list;

NOTE 8: The UE only receives those ICSI values corresponding to the IMS communication services that the network provides to the user.

2) may include an Accept-Contact header field containing an ICSI value (coded as specified in subclause 7.2A.8.2) that is related to the request in a g.3gpp.icsi-ref media feature tag as defined in subclause 7.9.2 if the ICSI for the IMS communication service is known. The UE may remove one or more subclasses from an ICSI when including it in an Accept-Contact header field provided that the included ICSI corresponds to an IMS communication service.

NOTE 9: If the UE includes the same ICSI values into the Accept-Contact header field and the P-Preferred-Service header field, there is a possibility that one of the involved S-CSCFs or an AS changes the ICSI value in the P-Asserted-Service header field, which results in the message including two different ICSI values (one in the P-Asserted-Service header field, changed in the network and one in the Accept-Contact header field).

…

If available to the UE (as defined in the access technology specific annexes for each access technology), the UE shall insert a P-Access-Network-Info header field into any request for a dialog, any subsequent request (except CANCEL requests) or response (except CANCEL responses) within a dialog or any request for a standalone method (see subclause 7.2A.4). Insertion of the P-Access-Network-Info header field into the ACK request is optional.

NOTE 13: During the dialog, the points of attachment to the IP-CAN of the UE can change (e.g. UE connects to different cells). The UE will populate the P-Access-Network-Info header field in any request or response within a dialog with the current point of attachment to the IP-CAN (e.g. the current cell information).

NOTE 14: The value of the P-Access-Network-Info header field could be stale if the point of attachment of the UE with the network changes before the message is received by the network.

The UE shall build a proper preloaded Route header field value for all new dialogs and standalone transactions. The UE shall build a list of Route header field values made out of the following, in this order:

a) the P-CSCF URI containing the IP address acquired at the time of the P-CSCF discovery procedures which was used in registration of the contact address (or registration flow); and

NOTE 15: If the UE is provisioned with or receives a FQDN at the time of the P-CSCF discovery procedures, the FQDN is resolved to an IP address at the time of the P-CSCF discovery procedures.

b) the P-CSCF port based on the security mechanism in use:

- if IMS AKA or SIP digest with TLS is in use as a security mechanism, the protected server port learnt during the registration procedure;

- if SIP digest without TLS, NASS-IMS bundled authentication or GPRS-IMS-Bundled authentication is in use as a security mechanism, the unprotected server port used during the registration procedure;

c) and the values received in the Service-Route header field saved from the 200 (OK) response to the last registration or re-registration of the public user identity with associated contact address.

NOTE 16: When the UE registers multiple contact addresses, there will be a list of Service-Route headers for each contact address. When sending a request using a given contact address and the associated security associations or TLS session, the UE will use the corresponding list of Service-Route headers to construct a list of Route headers.

[TS 24.229, clause 5.1.2A.1.2]:

The UE may include a SIP URI complying with RFC 3261 [26], a tel URI complying with RFC 3966 [22], a pres URI complying with RFC 3859 [179], an im URI complying with RFC 3860 [180] or a mailto URI complying with RFC 2368 [181].

NOTE: This version of the document does not specify how the UE determines the host part of the SIP URI.

The UE may use non-international formats of E.164 numbers or non-E.164 numbers, including geo-local numbers and home-local numbers and other local numbers (e.g. private number), in the Request-URI.

The actual value of the URI depends on whether user equipment performs an analysis of the dial string input by the end user or not, see subclauses 5.1.2A.1.3 and 5.1.2A.1.4.

[TS 24.229, clause 5.1.2A.1.5]:

When the UE uses home-local number, the UE shall include in the "phone-context" tel URI parameter the home network domain name in accordance with RFC 3966 [22].

When the UE uses geo-local number, the UE shall:

- if access technology information available to the UE (i.e., the UE can insert P-Access-Network-Info header field into the request), include the access technology information in the "phone-context" tel URI parameter according to RFC 3966 [22] as defined in subclause 7.2A.10; and

- if access technology information is not available to the UE (i.e., the UE cannot insert P-Access-Network-Info header field into the request), include in the "phone-context" tel URI parameter the home network domain name prefixed by the "geo-local." string according to RFC 3966 [22] as defined in subclause 7.2A.10.

When the UE uses other local numbers, than geo-local number or home local numbers, e.g. private numbers that are different from home-local number or the UE is unable to determine the type of the dialled number, the UE shall include a "phone-context" tel URI parameter set according to RFC 3966 [22], e.g. if private numbers are used a domain name to which the private addressing plan is associated. The "phone-context" value used in the case of other local numbers shall be different from "phone-context" values used with geo-local numbers and home-local numbers.

NOTE 1: The "phone-context" tel URI parameter value can be entered or selected by the subscriber, or can be a "pre-configured" value (e.g. using OMA-DM with the management object specified in 3GPP TS 24.167 [8G]) inserted by the UE.

NOTE 2: The way how the UE determines whether numbers in a non-international format are geo-local, home-local or relating to another network in absence of matching UE configuration in subclause 5.1.2A.1.5A, is implementation specific.

NOTE 3: Home operator's local policy can define a prefix string(s) to enable subscribers to differentiate dialling a geo-local number and/or a home-local number.

[TS 24.229, clause 5.1.3.1]:

Upon generating an initial INVITE request, the UE shall include the Accept header field with "application/sdp", the MIME type associated with the 3GPP IM CN subsystem XML body (see subclause 7.6.1) and any other MIME type the UE is willing and capable to accept.

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

The preconditions mechanism should be supported by the originating UE.

If the precondition mechanism is disabled as specified in subclause 5.1.5A, the UE shall not use the precondition mechanism.

The UE may initiate a session without the precondition mechanism if the originating UE does not require local resource reservation.

NOTE 1: The originating UE can decide if local resource reservation is required based on e.g. application requirements, current access network capabilities, local configuration, etc.

In order to allow the peer entity to reserve its required resources, if the precondition mechanism is enabled as specified in subclause 5.1.5A; the originating UE supporting the precondition mechanism should make use of the precondition mechanism, even if it does not require local resource reservation.

Upon generating an initial INVITE request using the precondition mechanism, the UE shall:

- indicate the support for reliable provisional responses and specify it using the Supported header field; and
- indicate the support for the preconditions mechanism and specify it using the Supported header field.

Upon generating an initial INVITE request using the precondition mechanism, the UE shall not indicate the requirement for the precondition mechanism by using the Require header field.

During the session initiation, if the originating UE indicated the support for the precondition mechanism in the initial INVITE request and:

a) the received response with an SDP body includes a Require header field with "precondition" option-tag, the originating UE shall include a Require header field with the "precondition" option-tag:
    - in subsequent requests that include an SDP body, that the originating UE sends in the same dialog as the response is received from; and
    - in responses with an SDP body to subsequent requests that include an SDP body and include "precondition" option-tag in Supported header field or Require header field received in-dialog; or

b) the received response with an SDP body does not include the "precondition" option-tag in the Require header field,
    - in subsequent requests that include an SDP body, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog;
    - in responses with an SDP body to subsequent requests with an SDP body but without "precondition" option-tag in the Require or Supported header field, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog; and

- in responses with an SDP body to subsequent requests with an SDP body and with "precondition" option-tag in the Require or Supported header field, the originating UE shall include a Require header field with "precondition" option-tag in the same dialog.

NOTE 2: Table A.4 specifies that UE support of forking is required in accordance with RFC 3261 [26]. The UE can accept or reject any of the forked responses, for example, if the UE is capable of supporting a limited number of simultaneous transactions or early dialogs.

Upon successful reservation of local resources the UE shall confirm the successful resource reservation (see subclause 6.1.2) within the next SIP request.

NOTE 3: In case of the precondition mechanism being used on both sides, this confirmation will be sent in either a PRACK request or an UPDATE request. In case of the precondition mechanism not being supported on one or both sides, alternatively a reINVITE request can be used for this confirmation after a 200 (OK) response has been received for the initial INVITE request, in case the terminating UE does not support the PRACK request (as described in RFC 3262 [27]) and does not support the UPDATE request (as described in RFC 3311 [29]).

NOTE 4: The UE can receive a P-Early-Media header field authorizing an early-media flow while the required preconditions, if any, are not met and/or the flow direction is not enabled by the SDP direction parameter. According to RFC 5009 [109], an authorized early-media flow can be established only if the necessary conditions related to the SDP negotiation are met. These conditions can evolve during the session establishment.

NOTE 5: When the UE is confirming the successful resource reservation using an UPDATE request (or a PRACK request) and the UE receives a 180 (Ringing) response or a 200 (OK) response to the initial INVITE request before receiving a 200 (OK) response to the UPDATE request (or a 200 (OK) response to the PRACK request), the UE does not treat this as an error case and does not release the session.

NOTE 6: The UE procedures for rendering of the received early media and of the locally generated communication progress information are specified in 3GPP TS 24.628 [8ZF].

[TS 24.229, clause 6.1.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

In order to authorize the media streams, the P-CSCF and S-CSCF have to be able to inspect SDP message bodies. Hence, the UE shall not encrypt SDP message bodies.

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

NOTE 1: A codec can have multiple payload type numbers associated with it.

In order to support accurate bandwidth calculations, the UE may include the "a=ptime" attribute for all "audio" media lines as described in RFC 4566 [39]. If a UE receives an "audio" media line with "a=ptime" specified, the UE should transmit at the specified packetization rate. If a UE receives an "audio" media line which does not have "a=ptime" specified or the UE does not support the "a=ptime" attribute, the UE should transmit at the default codec packetization rate as defined in RFC 3551 [55A]. The UE will transmit consistent with the resources available from the network.

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

NOTE 2: The above is the minimum requirement for all UEs. Additional requirements can be found in other specifications.

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

If an in-band DTMF codec is supported by the application associated with an audio media stream, then the UE shall include, in addition to the payload type numbers associated with the audio codecs for the media stream, for each clock rate associated with the audio codecs for the media stream, a payload type number associated with the MIME subtype "telephone-event", to indicate support of in-band DTMF as described in RFC 4733 [23].

The UE shall inspect the SDP message body contained in any SIP request or response, looking for possible indications of grouping of media streams according to RFC 3524 [54] and perform the appropriate actions for IP-CAN bearer establishment for media according to IP-CAN specific procedures (see subclause B.2.2.5 for IP-CAN implemented using GPRS, subclause L.2.2.5 for IP-CAN implemented using EPS, and subclause U.2.2.5 for IP-CAN implemented using 5GS).

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

NOTE 4: Based on this resource reservation can, in certain cases, be initiated immediately after the sending or receiving of the initial SDP offer.

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

If the desired QoS resources for one or more media streams have not been reserved at the UE when constructing the SDP offer, the UE:

- shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment, if the UE uses the precondition mechanism (see subclause 5.1.3.1); and

- if the UE uses the precondition mechanism (see subclause 5.1.3.1), shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 1: Previous versions of this document mandated the use of the SDP inactive attribute. This document does not prohibit specific services from using direction attributes to implement their service-specific behaviours.

If the UE uses the precondition mechanism (see subclause 5.1.3.1), and the desired QoS resources for one or more media streams are available at the UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment and shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

Upon confirming successful local resource reservation, the UE shall create an SDP offer in which the related local preconditions are set to met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64].

Upon receiving an SDP answer, which includes more than one codec per media stream, excluding the in-band DTMF codec, as described in subclause 6.1.1, the UE shall:

- send an SDP offer at the first possible time, selecting only one codec per media stream; or

- if the UE is participant in a multi-stream multiparty multimedia conference session using simulcast (indicated by the presence of "a=simulcast" SDP attribute(s) in the SDP answer, as defined in draft-ietf-mmusic-sdp-simulcast [249]), apply the procedures defined in 3GPP TS 26.114 [9B] annex S.

[TS 26.114, clause 5.2.1.1]:

MTSI clients in terminals offering speech communication shall support narrowband, wideband and super-wideband communication.

In addition, MTSI clients in terminals offering speech communication shall support:

- AMR speech codec (3GPP TS 26.071 [11], 3GPP TS 26.090 [12], 3GPP TS 26.073 [13] and 3GPP TS 26.104 [14]) including all 8 modes and source controlled rate operation 3GPP TS 26.093 [15]. The MTSI client in terminal shall be capable of operating with any subset of these 8 codec modes. More detailed codec requirements for the AMR codec are defined in clause 5.2.1.2.

MTSI clients in terminals offering wideband speech communication at 16 kHz sampling frequency shall support:

- AMR-WB codec (3GPP TS 26.171 [17], 3GPP TS 26.190 [18], 3GPP TS 26.173 [19] and 3GPP TS 26.204 [20]) including all 9 modes and source controlled rate operation 3GPP TS 26.193 [21]. The MTSI client in terminal shall be capable of operating with any subset of these 9 codec modes. More detailed codec requirements for the AMR-WB codec are defined in clause 5.2.1.3. When the EVS codec is supported, the EVS AMR-WB IO mode may serve as an alternative implementation of AMR-WB as defined in clause 5.2.1.4.

MTSI clients in terminals offering super-wideband or fullband speech communication shall support:

- EVS codec (TS 26.441 [121], TS 26.444 [124], TS 26.445 [125], TS 26.447 [127], TS 26.451 [131], TS 26.442 [122] and TS 26.443 [123]) as described below including functions for backwards compatibility with AMR-WB (TS 26.446 [126]) and discontinuous transmission (TS 26.449 [129] and TS 26.450 [130]). More detailed codec requirements for the EVS codec are defined in clause 5.2.1.4.

[TS 26.114, clause 6.2.5.1]:

The SDP shall include bandwidth information for each media stream and also for the session in total. The bandwidth information for each media stream and for the session is defined by the Application Specific (AS) bandwidth modifier as defined in RFC 4566 [8].

The bandwidth for RTCP traffic shall be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by RFC 3556 [42]. Therefore, an MTSI client shall include the "b=RS:" and "b=RR:" fields in SDP, and shall be able to interpret them. There shall be an upper limit on the allowed RTCP bandwidth for each RTP session signalled by the MTSI client. This limit is defined as follows:

- 8 000 bps for the RS field (at media level);

- 6 000 bps for the RR field (at media level).

The RS and RR values included in the SDP answer should be treated as the negotiated values for the session and should be used to calculate the total RTCP bandwidth for all terminals in the session.

If the session described in the SDP is a point-to-point speech only session, the MTSI client may request the deactivation of RTCP by setting its RTCP bandwidth modifiers to zero.

If a MTSI client receives SDP bandwidth modifiers for RTCP equal to zero from the originating MTSI client, it should reply (via the SIP protocol) by setting its RTCP bandwidth using SDP bandwidth modifiers with values equal to zero.

7.4.2A Profile requirements (Informative)

[NG.114 Version 1.0, clause 2.3.4.1]:

The ICSI value used must indicate the IMS Multimedia Telephony service, which is urn:urn-7:3gpp-service.ims.icsi.mmtel, as specified in 3GPP TS 24.173 [10].

…

The UE must include the audio and video media feature tags, as defined in IETF RFC 3840 [18], in the Contact header field of the SIP INVITE request, and in the Contact header field of the SIP response to the SIP INVITE request, as specified in 3GPP TS 24.229 [8].

[NG.114 Version 1.0, clause 2.3.5]:

For MMTEL Voice/Conversational Video sessions, the UE must support the preconditions mechanism as specified in sections 5.1.3.1 and 5.1.4.1 of 3GPP TS 24.229 [8]. If the precondition mechanism is enabled by the Precondition_disabling_policy node in Annex C.3, the UE must use the precondition mechanism. If preconditions are used, and the originating UE receives the selected codec in the SDP of a SIP 18x response, then the UE must include only the same codec with its selected configuration parameters in the SDP of the SIP UPDATE request, used for precondition status update.

[NG.114 Version 1.0, clause 3.2.2.1]:

The UE must include in an initial SDP offer at least:

1. one EVS payload type with one of the configurations supporting super-wideband speech as defined in section 3.2.2.3 of this document.

2. one AMR-WB payload type with no mode-set specified as defined in table 6.1 of 3GPP TS 26.114 [16].

3. one AMR payload type with no mode-set specified as defined in table 6.1 of 3GPP TS 26.114 [16].

The codec preference order must be as specified in sections 5.2.1.5 and 5.2.1.6 of 3GPP TS 26.114 [16].

[NG.114 Version 1.0, clause 3.2.2.2 on AMR and AMR-WB]:

The UE must set the b=AS to match the highest codec mode for the offer (maximum codec bit rate if no mode set is included).

[NG.114 Version 1.0, clause 3.2.2.3 on EVS]:

The UE that sends the SDP offer for voice media must include in this SDP offer at least one EVS payload type with one of the following EVS configurations:

1. EVS Configuration A1: br=5.9-13.2; bw=nb-swb.

2. EVS Configuration A2: br=5.9-24.4; bw=nb-swb.

3. EVS Configuration B0: br=13.2; bw=swb.

4. EVS Configuration B1: br=9.6-13.2; bw=swb.

5. EVS Configuration B2: br=9.6-24.4; bw=swb.

…

SDP parameters other than br, bw, max-red and ch-aw-recv must not be included in a media format description associated with the EVS codec within the initial SDP offer (for a list of SDP parameters see table 6.2a in 3GPP TS 26.114 [16]).

[NG.114 Version 1.0, clause 3.2.3]:

The UE and the entities in the IMS core network that terminate the user plane must set the ptime attribute value to receive one speech frame encapsulated in each RTP packet, but must accept any number of frames per RTP packet, up to the maximum limit of 12 speech frames per RTP packet.

Note 1: This means that the ptime attribute must be set to 20 and the maxptime attribute must be set to 240 in the SDP negotiation.

7.4.3 Test description

7.4.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.

- The UE is configured to register for IMS after switch on.

- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS

7.4.3.2 Test procedure sequence

Table 7.4.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
1A-1F	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
2	UE sends INVITE with first SDP offer (Step 1 of Annex A.4.1)	-->	INVITE	1	P
3	SS sends a 100 Trying provisional response (Step 2 of Annex A.4.1)	<--	100 Trying		
4	SS sends an SDP answer (Step 3 of Annex A.4.1)	<--	183 Session Progress		
5	UE acks 183 Session Progress (Step 4 of Annex A.4.1)	-->	PRACK	2	P
6	SS responds to PRACK (Step 5 of Annex A.4.1)	<--	200 OK		
7	UE sends UPDATE with second SDP offer (Step 6 of Annex A.4.1)	-->	UPDATE	3	P
8	SS sends an SDP answer (Step 7 of Annex A.4.1)	<--	200 OK		
9	SS sends 180 Ringing reliably (Step 8 of Annex A.4.1)	<--	180 Ringing		
10	UE acks 180 Ringing (Step 9 of Annex A.4.1)	-->	PRACK	4	P
11	SS responds to PRACK (Step 10 of Annex A.4.1)	<--	200 OK		
12	SS responds to INVITE (Step 11 of Annex A.4.1)	<--	200 OK		
13	UE acks 200 OK for INVITE (Step 12 of Annex A.4.1)	-->	ACK	5	P

7.4.3.3 Specific message contents

None as fully described in Annex A.4.1.

7.5 MTSI MO Voice Call without preconditions at both originating UE and terminating UE / 5GS

7.5.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to not use preconditions  }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call without preconditions }
            }
```

(2)

```
with { UE having sent INVITE without preconditions }
ensure that {
  when { UE receives 183 Session Progress without preconditions }
    then { UE sends PRACK for 183 Session Progress }
            }
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK followed by 180 Ringing followed by 200 OK for INVITE }
    then { UE sends ACK }
            }
```

7.5.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

As described in 7.4.2 except:

[TS 24.229, clause 5.1.3.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

The preconditions mechanism should be supported by the originating UE.

If the precondition mechanism is disabled as specified in subclause 5.1.5A, the UE shall not use the precondition mechanism.

[TS 24.229, clause 5.1.5A]:

The precondition disabling policy indicates whether the UE is allowed to use the precondition mechanism or whether the UE is not allowed to use the precondition mechanism.

If the precondition disabling policy is not configured, the precondition disabling policy is assumed to indicate that the UE is allowed to use the precondition mechanism.

The UE may support the precondition disabling policy.

If the UE supports the precondition disabling policy, the UE may support being configured with the precondition disabling policy using one or more of the following methods:

a) the Precondition_disabling_policy node of the $EF_{IMSConfigDat}$a file described in 3GPP TS 31.102 [15C];

b) the Precondition_disabling_policy node of the $EF_{IMSConfigData}$ file described in 3GPP TS 31.103 [15B]; and

c) the Precondition_disabling_policy node of 3GPP TS 24.167 [8G].

If the UE is configured with both the Precondition_disabling_policy node of 3GPP TS 24.167 [8G] and the Precondition_disabling_policy node of the $EF_{IMSConfigData}$ file described in 3GPP TS 31.102 [15C] or 3GPP TS 31.103 [15B], then the Precondition_disabling_policy node of the $EF_{IMSConfigData}$ file shall take precedence.

NOTE: Precedence for files configured on both the USIM and ISIM is defined in 3GPP TS 31.103 [15B].

The precondition mechanism is disabled, if the UE supports the precondition disabling policy and the precondition disabling policy indicates that the UE is not allowed to use the precondition mechanism.

The precondition mechanism is enabled, if:

1) the UE does not support the precondition disabling policy; or

2) the UE supports the precondition disabling policy and the precondition disabling policy indicates that the UE is allowed to use the precondition mechanism.

[TS 24.229, clause 6.1.2]:

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

…

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

7.5.3 Test description

7.5.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- The UE is configured to not use preconditions.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.5.3.2 Test procedure sequence

Table 7.5.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
1A-1F	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
2	Step 1 of Annex A.4.2 happens	-->	INVITE	1	P
3	Step 2 of Annex A.4.2 happens	<--	100 Trying		
4	Step 3 of Annex A.4.2 happens	<--	183 Session Progress		
5	Step 4 of Annex A.4.2 happens	-->	PRACK	2	P
6	Step 5 of Annex A.4.2 happens	<--	200 OK		
7	Step 6 of Annex A.4.2 happens	<--	180 Ringing		
8	Step 7 of Annex A.4.2 happens	<--	200 OK		
9	Step 8 of Annex A.4.2 happens	-->	ACK	3	P

7.5.3.3 Specific message contents

None as fully described in annex A.4.2.

7.6 MTSI MT Voice Call with preconditions at both originating UE and terminating UE / 5GS

7.6.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE receives INVITE for voice call }
    then { UE responds with 183 Session Progress including SDP }
            }
```

(2)

```
with { UE having sent 183 Session Progress }
ensure that {
  when { UE receives PRACK for 183 Session Progress }
    then { UE sends 200 OK for PRACK }
            }
```

(3)

```
with { UE having sent 200 OK for PRACK }
ensure that {
  when { UE receives UPDATE including SDP }
    then { UE sends 200 OK for UPDATE including SDP and 180 Ringing }
            }
```

(4)

```
with { UE having sent 180 Ringing, possibly reliably }
ensure that {
  when { 180 was sent reliably and consequently UE receives PRACK for 180 Ringing }
    then { UE sends 200 OK for PRACK }
            }
```

(5)

```
with { UE having sent 180 Ringing  }
ensure that {
  when { User accepts the incoming voice call request }
    then { UE sends 200 OK for INVITE }
            }
```

(6)

```
with { UE having sent 200 OK for INVITE }
ensure that {
  when { UE receives ACK followed by BYE }
    then { UE sends 200 OK for BYE }
            }
```

7.6.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

Editor's note: more concrete texts for supporting the TPs need to be investigated.

[TS 24.229, clause 5.1.4.1]

If an initial INVITE request is received the terminating UE shall check whether the terminating UE requires local resource reservation.

NOTE 1: The terminating UE can decide if local resource reservation is required based on e.g. application requirements, current access network capabilities, local configuration, etc.

During the session initiation, if local resource reservation is required at the terminating UE and the terminating UE supports the precondition mechanism, and:

a) the received INVITE request includes the "precondition" option-tag in the Supported header field or Require header field and the precondition mechanism is enabled as specified in subclause 5.1.5A, the terminating UE shall use the precondition mechanism and shall include a Require header field with the "precondition" option-tag:

…

If the terminating UE included an SDP offer or an SDP answer in a reliable provisional response to the INVITE request and both the terminating UE and the originating UE support UPDATE method, then in order to remove one or more media streams negotiated in the session for which a final response to the INVITE request has not been sent yet, the terminating UE shall send an UPDATE request with a new SDP offer and delays sending of 200 (OK) response to the INVITE request till after reception of 200 (OK) response to the UPDATE request.

If the user does not accept a media stream accepted in the SDP answer and the terminating UE, the originating UE or both do not support the UPDATE method, then after reception of ACK request related to 200 (OK) response to the INVITE request, the UE shall modify the session.

The terminating UE shall include the media feature tags as defined in RFC 3840 [62] for all supported streaming media types in the SIP response other than the 100 (Trying) response to the SIP INVITE request.

[TS 24.229, clause 6.1.1]

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

[TS 24.229, clause 6.1.3]

If the terminating UE had previously set one or more media streams to inactive mode and the QoS resources for those media streams are now ready, the UE shall set the media streams to active mode by applying the procedures described in RFC 4566 [39] with respect to setting the direction of media streams.

…

Upon sending an SDP answer to an SDP offer, with the SDP answer including one or more media streams for which the originating side did indicate its local preconditions as not met, if the precondition mechanism is used by the terminating UE (see subclause 5.1.4.1), the terminating UE shall indicate its local preconditions and request the confirmation for the result of the resource reservation at the originating end point.

[TS 26.114, clause 5.2.1]

In addition, MTSI clients in terminals offering speech communication shall support:

- AMR speech codec (3GPP TS 26.071 [11], 3GPP TS 26.090 [12], 3GPP TS 26.073 [13] and 3GPP TS 26.104 [14]) including all 8 modes and source controlled rate operation 3GPP TS 26.093 [15]. The MTSI client in terminal shall be capable of operating with any subset of these 8 codec modes. More detailed codec requirements for the AMR codec are defined in clause 5.2.1.2.

[TS 26.114, clause 6.2.2.1]

An MTSI client offering a speech media session for narrow-band speech and/or wide-band speech should generate an SDP offer according to the examples in Annexes A.1 to A.3. An MTSI client offering EVS should generate an SDP offer according to the examples in Annex A.14.

An MTSI client in terminal supporting EVS should support the RTCP-APP signalling for speech adaptation defined clause 10.2.1, and shall support the RTCP-APP signalling when the MTSI client in terminal supports adaptation for call cases where the RTP-based CMR cannot be used.

[TS 26.114, clause 6.2.5]

The SDP shall include bandwidth information for each media stream and also for the session in total. The bandwidth information for each media stream and for the session is defined by the Application Specific (AS) bandwidth modifier as defined in RFC 4566 [8].

[TS 26.114, clause 7.3.1]

The bandwidth for RTCP traffic shall be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by RFC 3556 [42].

7.6.3 Test description

7.6.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- The UE is configured to use preconditions.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.6.3.2 Test procedure sequence

Table 7.6.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
1	Step 1 of Annex A.5.1 happens	<--	INVITE		
2	Step 2 of Annex A.5.1 happens	-->	100 Trying		
3	Step 3 of Annex A.5.1 happens	-->	183 Session Progress	1	P
4	Step 4 of Annex A.5.1 happens	<--	PRACK		
5	Step 5 of Annex A.5.1 happens	-->	200 OK	2	P
6	Step 6 of Annex A.5.1 happens	<--	UPDATE		
7	Step 7 of Annex A.5.1 happens	-->	200 OK	3	P
8	Step 8 of Annex A.5.1 happens	-->	180 Ringing		
9	Step 9 of Annex A.5.1 happens	<--	PRACK		
10	Step 10 of Annex A.5.1 happens	-->	200 OK	4	P
11	Step 11 of Annex A.5.1 happens	-->	200 OK	5	P
12	Step 12 of Annex A.5.1 happens	<--	ACK	6	P

7.6.3.3 Specific message contents

None as fully described in annex A.5.1.

7.7 MTSI MT Voice Call without preconditions at both originating UE and terminating UE / 5GS

7.7.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and and configured to not use preconditions }
ensure that {
  when { UE receives INVITE for voice call }
    then { UE may respond with 100 Trying and then sends 183 Session Progress with SDP without
preconditions }
            }
```

(2)

```
with { UE having sent 183 Session Progress }
ensure that {
  when { UE receives PRACK for 183 Session Progress }
    then { UE sends 200 OK for PRACK }
            }
```

(3)

```
with { UE having sent 200 OK for PRACK }
ensure that {
  when { UE is ready to start the call }
    then { UE sends 180 Ringing followed by 200 OK for INVITE }
            }
```

7.7.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, annex U.3.1.4]:

Upon receiving an INVITE request not including the "precondition" option-tag in the Supported header field and not including the "precondition" option-tag in the Require header field, and the IP-CAN performs network-initiated resource reservation for the UE, the UE:

1) if the INVITE request contains an SDP offer and the local resources required at the terminating UE for the received SDP offer are not available:

   a) shall not alert the user; and

   b) shall send 183 (Session Progress) response to the INVITE request without waiting for resource reservation and without alerting the user. If the INVITE request includes a Supported header field indicating support of reliable provisional responses, the UE shall send the 183 (Session Progress) response reliably. In the 183 (Session Progress) response, the UE shall include an SDP answer; and

2) if the INVITE request does not contain an SDP offer and the INVITE request includes a Supported header field indicating support of reliable provisional responses:

   a) shall generate an SDP offer;

   b) if the local resources required at the terminating UE for the generated SDP offer are not available:

      A) shall not alert the user; and

      B) shall reliably send 183 (Session Progress) response to the INVITE request without waiting for resource reservation and without alerting the user. In the 183 (Session Progress) response, the UE shall include the generated SDP offer.

Upon successful reservation of local resources, if the precondition mechanism is not used by the terminating UE, the UE can send 180 (Ringing) response to the INVITE request and can alert the user.

7.7.3 Test description

7.7.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- The UE is configured to not use preconditions.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.7.3.2 Test procedure sequence

Table 7.7.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
1	SS sends INVITE with SDP offer (Step 1 of Annex A.5.2)	<--	INVITE		
2	UE may send a 100 Trying provisional response (Step 2 of Annex A.5.2)	-->	100 Trying		
3	UE sends SDP answer (Step 3 of Annex A.5.2)	-->	183 Session Progress	1	P
4	SS acks reception of 183 Session Progress (Step 4 of Annex A.5.2)	<--	PRACK		
5	UE responds to PRACK (Step 5 of Annex A.5.2)	-->	200 OK	2	P
6	UE sends 180 Ringing (Step 6 of Annex A.5.2)	-->	180 Ringing	3	P
7	If 180 Ringing was sent reliably, SS sends PRACK (Step 7 of Annex A.5.2)	<--	PRACK		
8	If 180 Rinnging was sent reliably, UE sends 200 OK for PRACK (Step 8 of Annex A.5.2)	-->	200 OK		
8A	Make the UE accept the voice call.				
9	UE accepts the voice call (Step 9 of Annex A.5.2)	-->	200 OK		
10	SS acknowledges (Step 10 of Annex A.5.2)	<--	ACK		

7.7.3.3 Specific message contents

None as fully described in annex A.5.2.

7.8 MTSI MT Voice Call without preconditions at originating UE and with preconditions at terminating UE / 5GS

7.8.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE receives INVITE for voice call without precondition option-tag in Require or Supported
header }
    then { UE completes setup of voice call without preconditions }
            }
```

7.8.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.4.1]

If an initial INVITE request is received the terminating UE shall check whether the terminating UE requires local resource reservation.

NOTE 1: The terminating UE can decide if local resource reservation is required based on e.g. application requirements, current access network capabilities, local configuration, etc.

During the session initiation, if local resource reservation is required at the terminating UE and the terminating UE supports the precondition mechanism, and:

a) the received INVITE request includes the "precondition" option-tag in the Supported header field or Require header field and the precondition mechanism is enabled as specified in subclause 5.1.5A, the terminating UE shall use the precondition mechanism and shall include a Require header field with the "precondition" option-tag:

   - in responses to that INVITE request if those responses include an SDP body;

   - in responses to subsequent requests received in-dialog that include an SDP body and include "precondition" option-tag in Supported header field or Require header field; and

   - in subsequent requests that include an SDP body, that it sends towards the originating UE during the session initiation;

b) the received INVITE request includes the "precondition" option-tag in the Supported header field, and the precondition mechanism is disabled as specified in subclause 5.1.5A, the terminating UE shall not use the precondition mechanism:

c) the received INVITE request includes the "precondition" option-tag in the Require header field, and the precondition mechanism is disabled as specified in subclause 5.1.5A, the terminating UE shall reject the INVITE request with a 420 (Bad Extension) response; and

d) the received INVITE request does not include the "precondition" option-tag in the Supported header field or Require header field, the terminating UE shall not use the precondition mechanism.

7.8.3 Test description

7.8.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use preconditions.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.
- The UE has registered to IMS and set up an MO voice call, by executing the generic test procedure in Annex A.2 up to the last step and thereafter executing the generic test procedure in Annex A.4.1. The SS then ends the MO voice call by sending BYE.

7.8.3.2 Test procedure sequence

Table 7.8.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9-17	Steps 1-9 of Annex A.5.2 happens	-	-		
18	Step 10 of Annex A.5.2 happens	<--	ACK	1	P

7.8.3.3 Specific message contents

None as fully described in annex A.5.2.

7.9 MTSI MT Voice Call with preconditions at originating UE and without preconditions at terminating UE / 5GS

7.9.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to not use preconditions }
ensure that {
  when { UE receives INVITE for voice call with preconditions }
    then { UE completes setup of voice call without preconditions }
            }
```

7.9.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.4.1]

If an initial INVITE request is received the terminating UE shall check whether the terminating UE requires local resource reservation.

NOTE 1: The terminating UE can decide if local resource reservation is required based on e.g. application requirements, current access network capabilities, local configuration, etc.

During the session initiation, if local resource reservation is required at the terminating UE and the terminating UE supports the precondition mechanism, and:

a) the received INVITE request includes the "precondition" option-tag in the Supported header field or Require header field and the precondition mechanism is enabled as specified in subclause 5.1.5A, the terminating UE shall use the precondition mechanism and shall include a Require header field with the "precondition" option-tag:

  - in responses to that INVITE request if those responses include an SDP body;

  - in responses to subsequent requests received in-dialog that include an SDP body and include "precondition" option-tag in Supported header field or Require header field; and

  - in subsequent requests that include an SDP body, that it sends towards the originating UE during the session initiation;

b) the received INVITE request includes the "precondition" option-tag in the Supported header field, and the precondition mechanism is disabled as specified in subclause 5.1.5A, the terminating UE shall not use the precondition mechanism:

c) the received INVITE request includes the "precondition" option-tag in the Require header field, and the precondition mechanism is disabled as specified in subclause 5.1.5A, the terminating UE shall reject the INVITE request with a 420 (Bad Extension) response; and

d) the received INVITE request does not include the "precondition" option-tag in the Supported header field or Require header field, the terminating UE shall not use the precondition mechanism.

7.9.3 Test description

7.9.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.

- UE is configured to register for IMS after switch on.
- The UE is configured to not use preconditions.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.9.3.2 Test procedure sequence

Table 7.9.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9	Step 1 of Annex A.5.1 happens	<--	INVITE		
10-17	Steps 2-9 of Annex A.5.2 happens	-	-		
18	Step 10 of Annex A.5.2 happens	<--	ACK	1	P

7.9.3.3 Specific message contents

None as fully described in annex A.5.1 and A.5.2.

7.10 MTSI MT Voice call without preconditions and without SDP offer in MT INVITE / 5GS

7.10.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to not use preconditions }
ensure that {
  when { UE receives INVITE for voice call not containing an SDP offer, but indicating support for
reliable provisional responses }
    then { UE sends 183 Session Progress reliably and containing an SDP offer }
            }
```

(2)

```
with { UE having sent 183 Session Progress }
ensure that {
  when { UE receives PRACK for 183 Session Progress }
    then { UE sends 200 OK for PRACK }
            }
```

(3)

```
with { UE having sent 200 OK for PRACK }
ensure that {
  when { UE is ready to start the call }
    then { UE sends 180 Ringing followed by 200 OK for INVITE }
            }
```

7.10.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, annex U.3.1.4]:

Upon receiving an INVITE request not including the "precondition" option-tag in the Supported header field and not including the "precondition" option-tag in the Require header field, and the IP-CAN performs network-initiated resource reservation for the UE, the UE:

…

2) if the INVITE request does not contain an SDP offer and the INVITE request includes a Supported header field indicating support of reliable provisional responses:

   a) shall generate an SDP offer;

   b) if the local resources required at the terminating UE for the generated SDP offer are not available:

      A) shall not alert the user; and

      B) shall reliably send 183 (Session Progress) response to the INVITE request without waiting for resource reservation and without alerting the user. In the 183 (Session Progress) response, the UE shall include the generated SDP offer.

Upon successful reservation of local resources, if the precondition mechanism is not used by the terminating UE, the UE can send 180 (Ringing) response to the INVITE request and can alert the user.

7.10.3 Test description

7.10.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to not use preconditions.

Preamble:

- UE is in state 1N-A (38.508-1[21]) and registered to IMS

7.10.3.2 Test procedure sequence

Table 7.10.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
2	Step 1 of Annex A.5.2 happens (Note: the INVITE message doesn't include an SDP offer, but includes an option-tag indicating reliable provisional responses.)	<--	INVITE	-	-
3	Step 2 of Annex A.5.2 happens (Note: this step is optional.)	-->	(Optional) 100 Trying	-	-
4	Check: Does the UE send 183 Session Progress reliably and containing an SDP offer? (Step 3 of Annex A.5.2 happens)	-->	183 Session Progress	1	P
5	Step 4 of Annex A.5.2 happens (Note: an SDP answer is included.)	<--	PRACK	-	-
6	Step 5 of Annex A.5.2 happens (Check: does the UE send 200 OK for PRACK?)	-->	200 OK	2	P
7	Step 6 of Annex A.5.2 happens (Check: does the UE send 180 Ringing followed by 200 OK for INVITE?)	-->	180 Ringing	3	P
8	Step 7 of Annex A.5.2 happens (Conditional step: if UE sent 180 Ringing reliably, SS acknowledges reception of 180 Ringing)	<--	(Conditional) PRACK	-	-
9	Step 8 of Annex A.5.2 happens (Conditional step: if UE sent 180 Ringing reliably, UE responds to PRACK)	-->	(Conditional) 200 OK	-	-
10	Step 9 of Annex A.5.2 happens	-->	200 OK	-	-
11	Step 10 of Annex A.5.2 happens	<--	ACK	-	-

7.10.3.3 Specific message contents

Table 7.10.3.3-1: INVITE (step 2, table 7.10.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.9, Conditions A1, A3, and A4				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Content-Type		Not present		
Content-Length		0		
Message-body		Not present		

Table 7.10.3.3-2: 183 Session Progress with an SDP offer (step 4, table 7.10.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.3, condition A2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**

Message-body		NOTE: the following SDP offer is identical to the SDP offer shown in Annex A.4.2, Step 1. **Session description:** *v=0* *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS*: (bandwidth-value) **Time description:** *t*= (start-time) (stop-time) **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) [Note 2] *b=RR:* (bandwidth-value) [Note 2] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-24.4; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-24.4; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/16000* *a=fmtp:* (format) *a=rtpmap:* (payload type) *AMR/8000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/8000* *a=fmtp:* (format) *a=ecn-capable-rtp: leap ect=0* [Note 7] *a=rtcp-fb:* nack ecn* [Note 7] *a=rtcp-xr:ecn-sum* [Note 7] *a=rtcp-rsize* [Note 7] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 8] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:WVNf X19zZW1jdGwgKCkgewkyMjA7fQp9CnVubGVz|2^20| 1:4FEC_ORDER=FEC_SRTP"* [Note 8] Note 1: At least one "c=" field shall be present. Note 2: The RR value shall be greater than 0. The RS value can be any value. Note 3: The channel number shall be "/1" or omitted. Note 4: The max-red values from 0 to 220 are allowed. Note 5: The parameters dtx, dtx-recv and evs-mode-switch shall not be present.		TS 24.229 [7]

		Note 6: The parameters mode-set, mode-change-period, mode-change-neighbour, crc, robust-sorting and interleaving shall not be included. Note 7: Attributes for ECN Capability may be present if the UE supports Explicit Congestion Notification. Note 8: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 9: The ordering of payload types shall be as listed, i.e., EVS before AMR-WB before AMR. Note 10: The EVS payload type shall carry at least one of the five EVS configurations		

Table 7.10.3.3-3: PRACK with an SDP answer (step 5, table 7.10.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.4, condition A3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		NOTE: the following SDP offer is identical to the SDP offer shown in Annex A.4.2, Step 3. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 7] Note 1: The values for fmt, payload type and format are copied from step 3. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 4. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 4. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 4. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in previous SDP offer. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in previous SDP offer.		TS 24.229 [7]

7.11 MTSI MT Voice call without preconditions at terminating UE and originating UE requiring them / 5GS

7.11.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and the preconditions mechanism is disabled }
ensure that {
  when { UE receives INVITE for voice call where remote UE requires usage of preconditions }
    then { UE rejects INVITE with 420 Bad Extension response }
            }
```

7.11.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.4.1]

If an initial INVITE request is received the terminating UE shall check whether the terminating UE requires local resource reservation.

NOTE 1: The terminating UE can decide if local resource reservation is required based on e.g. application requirements, current access network capabilities, local configuration, etc.

During the session initiation, if local resource reservation is required at the terminating UE and the terminating UE supports the precondition mechanism, and:

a) the received INVITE request includes the "precondition" option-tag in the Supported header field or Require header field and the precondition mechanism is enabled as specified in subclause 5.1.5A, the terminating UE shall use the precondition mechanism and shall include a Require header field with the "precondition" option-tag:

   - in responses to that INVITE request if those responses include an SDP body;

   - in responses to subsequent requests received in-dialog that include an SDP body and include "precondition" option-tag in Supported header field or Require header field; and

   - in subsequent requests that include an SDP body, that it sends towards the originating UE during the session initiation;

b) the received INVITE request includes the "precondition" option-tag in the Supported header field, and the precondition mechanism is disabled as specified in subclause 5.1.5A, the terminating UE shall not use the precondition mechanism:

c) the received INVITE request includes the "precondition" option-tag in the Require header field, and the precondition mechanism is disabled as specified in subclause 5.1.5A, the terminating UE shall reject the INVITE request with a 420 (Bad Extension) response; and

d) the received INVITE request does not include the "precondition" option-tag in the Supported header field or Require header field, the terminating UE shall not use the precondition mechanism.

7.11.3 Test description

7.11.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.

- UE is configured to register for IMS after switch on.

The UE is configured to use preconditions OR the UE does not support preconditions- .

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.11.3.2 Test procedure sequence

Table 7.11.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9	Step 1 of A.5.1 happens, with one change: SS sends an INVITE request with a Require header field containing the precondition option-tag.	<--	INVITE		
9A	Optional step: UE may send a 100 Trying provisional response.	-->	(Optional) 100 Trying		
10	UE sends a 420 Bad Extension response with an Unsupported header field containing the precondition option-tag.	-->	420 Bad Extension	1	P
11	SS acknowledges the reception of 420 Bad Extension.	<--	ACK		

7.11.3.3 Specific message contents

Table 7.11.3.3-1: INVITE (step 9, table 7.11.3.2-1)

Derivation Path: Annex A.5.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require				RFC 3261 [6]
option-tag		precondition		

Table 7.2.3.3-2: 100 Trying (step 9A, Table 7.2.3.2-1)

Derivation path: TS 34.229-1 [2], Annex A.2.2, Condition A2

Table 7.11.3.3-3: 420 Bad Extension for INVITE (step 10, table 7.11.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.25				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Unsupported				RFC 3261 [6]
option-tag		precondition		

Table 7.2.3.3-4: ACK (step 11, Table 7.2.3.2-1)

Derivation path: TS 34.229-1 [2], Annex A.2.7, Conditions A2 and A4

7.12 MTSI MO Voice Call with preconditions at originating UE and without preconditions at terminating UE / 5GS

7.12.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call with preconditions }
            }
```

(2)

```
with { UE having sent INVITE with preconditions }
ensure that {
  when { UE receives 183 Session Progress without preconditions }
    then { UE sends PRACK for 183 Session Progress }
            }
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK followed by 180 Ringing followed by 200 OK for INVITE }
    then { UE sends ACK }
            }
```

7.12.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.3.1]:

During the session initiation, if the originating UE indicated the support for the precondition mechanism in the initial INVITE request and:

a) the received response with an SDP body includes a Require header field with "precondition" option-tag, the originating UE shall include a Require header field with the "precondition" option-tag:

  - in subsequent requests that include an SDP body, that the originating UE sends in the same dialog as the response is received from; and

  - in responses with an SDP body to subsequent requests that include an SDP body and include "precondition" option-tag in Supported header field or Require header field received in-dialog; or

b) the received response with an SDP body does not include the "precondition" option-tag in the Require header field,

  - in subsequent requests that include an SDP body, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog;

  - in responses with an SDP body to subsequent requests with an SDP body but without "precondition" option-tag in the Require or Supported header field, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog; and

  - in responses with an SDP body to subsequent requests with an SDP body and with "precondition" option-tag in the Require or Supported header field, the originating UE shall include a Require header field with "precondition" option-tag in the same dialog.

7.12.3 Test description

7.12.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.12.3.2 Test procedure sequence

Table 7.12.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-	-	-
2	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
3	Step 1 of Annex A.4.1 happens (Check: does the UE send INVITE for voice call with preconditions?)	-->	INVITE	1	P
4	Step 2 of Annex A.4.1 happens	<--	100 Trying	-	-
5	Step 3 of Annex A.4.2 happens (Note: the SS sends 183 Session Progress without attributes for preconditions in the SDP body.)	<--	183 Session Progress	-	-
6	Step 4 of Annex A.4.2 happens (Check: does the UE send PRACK?)	-->	PRACK	2	P
7	Step 5 of Annex A.4.2 happens	<--	200 OK	-	-
8	Step 6 of Annex A.4.2 happens	<--	180 Ringing	-	-
9	Step 7 of Annex A.4.2 happens	<--	200 OK	-	-
10	Step 8 of Annex A.4.2 happens (Check: does the UE send ACK?)	-->	ACK	3	P

7.12.3.3 Specific message contents

None as fully described in Annex A.4.1 and A.4.2.

7.13 MTSI MT Voice Call with RTCP disabled / 5GS

7.13.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE receives INVITE for voice call with both b=RS and b=RR attributes set to zero }
    then { UE may respond with 100 Trying and then sends 183 Session Progress with SDP with both
b=RS and b=RR set to zero and completes setup of voice call with preconditions }
}
```

7.13.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 26.114, clause 7.3.1]

Point-to-point speech only sessions may not require the above functionalities and may therefore turn off RTCP by setting the SDP bandwidth modifiers (RR and RS) to zero. When RTCP is turned off (for point-to-point speech only sessions) and the media is put on hold, the MTSI client should re-negotiate the RTCP bandwidth with the SDP bandwidth modifier RR value set greater than zero, and send RTCP packets (i.e., Receiver Reports) to the other end. This allows the remote end to detect link aliveness during hold. When media is resumed, the resuming MTSI client should request to turn off the RTCP sending again through a re-negotiation of the RTCP bandwidth with SDP bandwidth modifiers equal to zero.

When RTCP is turned off (for point-to-point speech only sessions) and if sending of an additional associated RTP stream becomes required and both RTP streams need to be synchronized, or if transport feedback due to lack of end-to-end QoS guarantees is needed, a MTSI client should re-negotiate the bandwidth for RTCP by sending an SDP with the RR bandwidth modifier greater than zero. Setting the RR bandwidth modifier greater than zero allows sending of RTCP Receiver Reports even when the session is put on hold and neither terminal is actively sending RTP media.

7.13.3 Profile requirements (Informative)

[GSMA NG.114 V1.0, cl3.6.3]

The RTP implementation must include an RTP Control Protocol (RTCP) implementation according to section 7.3.1 of 3GPP TS 26.114 [16].

The UE and the entities in the IMS core network that terminate the user plane must use symmetric RTCP as defined in IETF RFC 4961 [77], and section 7.3.1 of 3GPP TS 26.114 [16].

The bandwidth for RTCP traffic must be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by IETF RFC 3556 [78], and section 7.3.1 of 3GPP TS 26.114 [16]. Therefore, a UE must include the "b=RS:" and "b=RR:" fields in SDP, and a UE and the entities in the IMS core network that terminate the user plane must be able to interpret them. If the “b=RS:” field or “b=RR:” field or both these fields are not included in a received SDP (offer or answer), then the UE must use the recommended default value for the missing field(s) as defined in IETF RFC 3556 [78].

RTCP is controlled on a per session basis by the SDP offer/answer exchange as defined in section 7.3 of 3GPP TS 26.114 [16] with the following clarifications:

1. If the UE receives an SDP offer that contains “b=RS:” attribute set to zero, then the UE must set the “b=RS:” attribute to zero in an SDP answer to that SDP offer. If the UE receives an SDP offer that contains “b=RR:” attribute set to zero, then the UE must set the “b=RR:” attribute to zero in an SDP answer to that SDP offer. If the UE receives an SDP offer that contains both "b=RR:" and "b=RS:" attributes set to zero, then the UE must not send RTCP packets and must consider RTCP to be disabled for the session.

2. If the UE received an SDP answer containing zero values in both of the “b=RS:” and “b=RR:” attributes, then (regardless of the values assigned to these attributes in the corresponding SDP offer) the UE must not send RTCP packets and must consider RTCP to be disabled for the session.

3. The UE must accept receiving RTCP packets for a session that the UE considers RTCP to be disabled. The UE is not required to process these received RTCP packets.

…

7.13.4 Test description

7.13.4.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.13.4.2 Test procedure sequence

Table 7.13.4.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
1	SS sends INVITE, with both b=RS and b=RR attributes set to zero in SDP. (Step 1 of Annex A.5.1)	<--	INVITE		
2	Optional step: UE may send a 100 Trying provisional response. (Step 2 of Annex A.5.1)	-->	100 Trying		
3	Check: Does the UE send 183 Session Progress with both b=RS and b=RR set to zero in SDP?	-->	183 Session Progress	1	P
4	SS acknowledges reception of 183 Session Progress. (Step 4 of Annex A.5.1)	<--	PRACK		
5	UE responds to PRACK. (Step 5 of Annex A.5.1)	-->	200 OK	1	P
6	SS sends a second SDP offer. (Step 6 of Annex A.5.1)		UPDATE		
7	UE responds to UPDATE, including an SDP answer. (Step 7 of Annex A.5.1)		200 OK		
8	UE sends 180 Ringing. (Step 8 of Annex A.5.1)	-->	180 Ringing	1	P
9	Conditional step: if UE sent 180 Ringing reliably, SS acknowledges reception of 180 Ringing. (Step 9 of Annex A.5.1)	<--	PRACK		
10	Conditional step: if UE sent 180 Ringing reliably, UE responds to PRACK. (Step 10 of Annex A.5.1)	-->	200 OK		
11	Make UE accept the voice call.				
12	UE responds to INVITE. (Step 11 of Annex A.5.1)	-->	200 OK		
13	SS acknowledges. (Step 12 of Annex A.5.1)	<--	ACK		

7.13.4.3 Specific message contents

Table 7.13.4.3-1: INVITE (step 1, table 7.13.4.2-1)

Derivation Path: TS 34.229-5, Step 1 of A.5.1, with following exceptions				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Media description: *b=RR:0*		TS 26.114 [33] GSMA NG.114 [31]

Table 7.13.4.3-2: 183 Session Progress (step 3, table 7.13.4.2-1)

Derivation Path: TS 34.229-5, Step 3 of A.5.1, with following exceptions				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Media description: *b=RS:* 0 *b=RR:* 0		TS 26.114 [33] GSMA NG.114 [31]

Table 7.13.4.3-3: UPDATE (step 6, table 7.13.4.2-1)

Derivation Path: TS 34.229-5, Step 6 of A.5.1, with following exceptions				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Media description: *b=RR:0*		TS 26.114 [33] GSMA NG.114 [31]

Table 7.13.4.3-4: 200 OK (step 7, table 7.13.4.2-1)

Derivation Path: TS 34.229-5, Step 7 of A.5.1, with following exceptions				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Media description: *b=RS:* 0 *b=RR:* 0		TS 26.114 [33] GSMA NG.114 [31]

7.14 MTSI MO Video Call with preconditions at both originating and terminating UE / 5GS

7.14.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE is being made to initiate a video call }
   then { UE sends INVITE for video call with preconditions }
            }
```

(2)

```
with { UE having sent INVITE with preconditions }
ensure that {
  when { UE receives 183 Session Progress }
   then { UE sends PRACK for 183 Session Progress }
            }
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK }
   then { UE sends UPDATE }
            }
```

(4)

```
with { UE having sent UPDATE }
ensure that {
  when { UE receives 200 OK for UPDATE followed by 180 Ringing sent reliably }
   then { UE sends PRACK for 180 Ringing }
            }
```

(5)

```
with { UE having sent PRACK for 180 Ringing }
ensure that {
  when { UE receives 200 OK for PRACK followed by 200 OK for INVITE }
   then { UE sends ACK }
            }
```

7.14.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 6.1.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

In order to authorize the media streams, the P-CSCF and S-CSCF have to be able to inspect SDP message bodies. Hence, the UE shall not encrypt SDP message bodies.

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

...

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

If an in-band DTMF codec is supported by the application associated with an audio media stream, then the UE shall include, in addition to the payload type numbers associated with the audio codecs for the media stream, for each clock rate associated with the audio codecs for the media stream, a payload type number associated with the MIME subtype "telephone-event", to indicate support of in-band DTMF as described in RFC 4733 [23].

The UE shall inspect the SDP message body contained in any SIP request or response, looking for possible indications of grouping of media streams according to RFC 3524 [54] and perform the appropriate actions for IP-CAN bearer establishment for media according to IP-CAN specific procedures (see subclause B.2.2.5 for IP-CAN implemented using GPRS, subclause L.2.2.5 for IP-CAN implemented using EPS, and subclause U.2.2.5 for IP-CAN implemented using 5GS).

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

NOTE 4: Based on this resource reservation can, in certain cases, be initiated immediately after the sending or receiving of the initial SDP offer.

...

[TS 24.229, clause 6.1.2]:

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

If the desired QoS resources for one or more media streams have not been reserved at the UE when constructing the SDP offer, the UE:

- shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment, if the UE uses the precondition mechanism (see subclause 5.1.3.1); and
- if the UE uses the precondition mechanism (see subclause 5.1.3.1), shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 1: Previous versions of this document mandated the use of the SDP inactive attribute. This document does not prohibit specific services from using direction attributes to implement their service-specific behaviours.

If the UE uses the precondition mechanism (see subclause 5.1.3.1), and the desired QoS resources for one or more media streams are available at the UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment and shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

...

Upon generating the SDP offer for an INVITE request generated after receiving a 488 (Not Acceptable Here) response, as described in subclause 5.1.3.1, the SDP offer shall contain a subset of the allowed media types, codecs and other parameters from the SDP message bodies of all 488 (Not Acceptable Here) responses so far received for the same session establishment attempt (i.e. a set of INVITE requests used for the same session establishment). For each media line, the UE shall order the codecs in the SDP offer according to the order of the codecs in the SDP message bodies of the 488 (Not Acceptable Here) responses.

NOTE 6: The UE can attempt a session establishment through multiple networks with different policies and potentially can need to send multiple INVITE requests and receive multiple 488 (Not Acceptable Here) responses from different CSCF nodes. The UE therefore takes into account the SDP message bodies of all the 488 (Not Acceptable Here) responses received related to the same session establishment when building a new INVITE request.

Upon confirming successful local resource reservation, the UE shall create an SDP offer in which the related local preconditions are set to met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64].

Upon receiving an SDP answer, which includes more than one codec per media stream, excluding the in-band DTMF codec, as described in subclause 6.1.1, the UE shall:

- send an SDP offer at the first possible time, selecting only one codec per media stream; or
- if the UE is participant in a multi-stream multiparty multimedia conference session using simulcast (indicated by the presence of "a=simulcast" SDP attribute(s) in the SDP answer, as defined in RFC 8853 [249]), apply the procedures defined in 3GPP TS 26.114 [9B] annex S.

If the UE sends an initial INVITE request that includes only an IPv6 address in the SDP offer, and receives an error response (e.g., 488 (Not Acceptable Here) with 301 Warning header field) indicating "incompatible network address format", the UE shall send an ACK as per standard SIP procedures. Subsequently, the UE may acquire an IPv4 address or use an existing IPv4 address, and send a new initial INVITE request to the same destination containing only the IPv4 address in the SDP offer.

7.14.3 Test description

7.14.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.14.3.2 Test procedure sequence

Table 7.14.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS video call.	-	-	-	-
2	Steps from generic procedure specified in TS 38.508-1 [21] Table 4.9.xx are performed.	-	-	-	-
3	Check: Does UE send INVITE with the first SDP offer? (Step 1 of Annex A.15.1)	-->	INVITE	1	P
4	SS sends a 100 Trying provisional response. (Step 2 of Annex A.15.1)	<--	100 Trying		
5	SS sends an SDP answer. (Step 3 of Annex A.15.1)	<--	183 Session Progress		
6	Check: Does UE acknowledge reception of 183 Session Progress? (Step 4 of Annex A.15.1)	-->	PRACK	2	P
7	SS responds to PRACK.(Step 5 of Annex A.15.1).	<--	200 OK		
8	Check: Does UE send a second SDP offer in an UPDATE request? (Step 6 of Annex A.15.1)	-->	UPDATE	3	P
9	SS responds to UPDATE. (Step 7 of Annex A.15.1)	<--	200 OK		
10	SS sends 180 Ringing reliably. (Step 8 of Annex A.15.1)	<--	180 Ringing		
11	Check: Does UE acknowledge reception of 180 Ringing? (Step 9 of Annex A.15.1)	-->	PRACK	4	P
12	SS responds to PRACK. (Step 10 of Annex A.15.1)	<--	200 OK		
13	SS responds to INVITE. (Step 11 of Annex A.15.1)	<--	200 OK		
14	Check: Does UE acknowledge? (Step 12 of Annex A.15.1)	-->	ACK	5	P

7.14.3.3 Specific message contents

None as fully described in Annex A.15.1

7.15 MTSI MO Video call without preconditions at both originating UE and terminating UE / 5GS

7.15.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to not use preconditions }
ensure that {
  when { UE is being made to initiate a video call }
    then { UE sends INVITE for video call without preconditions }
}
```

(2)

```
with { UE having send INVITE without preconditions }
ensure that {
  when { UE receives 183 Session Progress without preconditions }
    then { UE sends PRACK for 183 Session Progress }
}
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK followed by 180 Ringing followed by 200 OK for INVITE }
    then { UE sends ACK }
}
```

7.15.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 6.1.1]:

For "video" and "audio" media types that utilize the RTP/RTCP, the UE shall specify the proposed bandwidth for each media stream utilizing the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556, then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890.

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in 3GPP TS 29.208.

[TS 24.229, clause 6.1.3]:

Upon sending an SDP answer to an SDP offer, with the SDP answer including one or more media streams for which the originating side did indicate its local preconditions as not met, if the precondition mechanism is used by the terminating UE (see subclause 5.1.4.1), the terminating UE shall indicate its local preconditions and request the confirmation for the result of the resource reservation at the originating end point.

…

Upon receiving an initial INVITE request that includes the SDP offer containing an IP address type (in the "c=" parameter) that is not supported by the UE, the UE shall:

- if the UE is a UE performing the functions of an external attached network and

  1) if the received SDP offer contains an "altc" SDP attribute indicating an alternative and supported IP address; and

  2) the UE supports the "altc" SDP attribute;

  select an IP address type in accordance with RFC 6947 [228]; or

- otherwise respond with a 488 (Not Acceptable Here) response including a 301 Warning header field indicating "incompatible network address format".

NOTE 2: Upon receiving an initial INVITE request that does not include an SDP offer, the UE can accept the request and include an SDP offer in the first reliable response. The SDP offer will reflect the called user's terminal capabilities and user preferences for the session.

If the UE receives an SDP offer that specifies different IP address type for media (i.e. specify it in the "c=" parameter of the SDP offer) that the UE is using for signalling, and if the UE supports both IPv4 and IPv6 addresses simultaneously, the UE shall accept the received SDP offer. Subsequently, the UE shall either acquire an IP address type or use an existing IP address type as specified in the SDP offer, and include it in the "c=" parameter in the SDP answer.

NOTE 3: Upon receiving an initial INVITE request, that includes an SDP offer containing connection addresses (in the "c=" parameter) equal to zero, the UE will select the media streams that is willing to accept for the session, reserve the QoS resources for accepted media streams, and include its valid connection address in the SDP answer.

…

If the terminating UE uses the precondition mechanism (see subclause 5.1.4.1), if the desired QoS resources for one or more media streams have not been reserved at the terminating UE when constructing the SDP offer, the terminating UE shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment.

NOTE 7: It is out of scope of this specification which media streams are to be included in the SDP offer.

If the terminating UE uses the precondition mechanism (see subclause 5.1.4.1) and if the desired QoS resources for one or more media streams are available at the terminating UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment.

If the terminating UE sends an UPDATE request to remove one or more media streams negotiated in the session for which a final response to the INVITE request has not been sent yet, the terminating UE sets the ports of the media streams to be removed from the session to zero in the new SDP offer.

NOTE 8: Upon receiving an initial INVITE request with one or more media streams which the terminating UE supports and one or more media streams which the UE does not support, the UE is not expected to reject the INVITE request just because of the presence of the unsupported media stream.

NOTE 9: Previous versions of this document mandated the use of the SDP inactive attribute in the SDP offer if the desired QoS resources for one or more media streams had not been reserved at the originating UE when constructing the SDP offer unless the originating UE knew that the precondition mechanism was supported by the remote UE. The use can still occur when interoperating with devices based on earlier versions of this document.

[TS 26.114, clause 5.2.2]:

MTSI clients in terminals offering video communication shall support:

- H.264 (AVC) [24] Constrained Baseline Profile (CBP) Level 1.2;

- H.265 (HEVC) [119] Main Profile, Main Tier, Level 3.1.

In addition they should support:

- H.264 (AVC) [24] Constrained High Profile (CHP) Level 3.1.

[TS 26.114, clause 6.2.3.2]:

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

An MTSI client is required to support the AVPF feedback messages trr-int, NACK and PLI [40] and the CCM feedback messages FIR, TMMBR and TMMBN [43], see Clauses 7.3.3 and 10.3. These feedback messages can only be used together with AVPF and shall be negotiated in SDP offer/answer before they can be used in the session [40]. An MTSI client sending an SDP offer for AVPF shall also include these AVPF and CCM feedback messages in the offer. An MTSI client accepting an SDP offer for AVPF for video shall also accept these AVPF and CCM feedback messages if they are offered.

If an MTSI client offers to use ECN for video in RTP streams then the MTSI client shall offer ECN Capable Transport as defined below. If an MTSI client accepts an offer for ECN for video then the MTSI client shall declare ECN Capable Transport in the SDP answer as defined below. The SDP negotiation of ECN Capable Transport is described in [84].

The use of ECN for a video stream in RTP is negotiated with the "ecn-capable-rtp" SDP attribute, [84]. ECN is enabled when both clients agree to use ECN as configured below. An MTSI client using ECN shall therefore also include the following parameters and parameter values for the ECN attribute:

- ‘leap’, to indicate that the leap-of-faith initiation method shall be used;
- ‘ect=0’, to indicate that ECT(0) shall be set for every packet.

An MTSI client offering ECN for video shall indicate support of TMMBR [43] by including the "ccm tmmbr" value within an "rtcp-fb" SDP attribute [40]. An MTSI client offering ECN for video may indicate support for RTCP AVPF ECN feedback messages [84] using the "rtcp-fb" SDP attribute with the "nack" feedback parameter and the "ecn" feedback parameter value. An MTSI client offering ECN for video may indicate support for RTCP XR ECN summary reports [84] using the "rtcp-xr" SDP attribute and the "ecn-sum" parameter.

An MTSI client receiving an offer for ECN for video with an indication of support of TMMBR [43] within an "rtcp-fb" attribute should accept the offer if it supports ECN. It shall then indicate support for TMMBR using an "rtcp-fb" attribute in the SDP answer.

An MTSI client receiving an offer for ECN for video with an indication of support of RTCP AVPF ECN feedback message but without support for TMMBR should accept the offer if it supports ECN and also the RTCP AVPF ECN feedback message. It shall then indicate support of the RTCP AVPF ECN feedback message using the "rtcp-fb" attribute in the SDP answer.

An MTSI client receiving an offer for ECN for video with an indication of support of RTCP XR ECN summary reports [84] without support for TMMBR should accept the offer if it supports ECN and also the RTCP XR ECN summary reports. It shall then indicate support of RTCP XR ECN summary reports in the SDP answer.

The use of ECN is disabled when a client sends an SDP without the "ecn-capable-rtp" SDP attribute.

An MTSI client may initiate a session re-negotiation to disable ECN to resolve ECN-related error cases. An ECN-related error case may be, for example, detecting non-ECT in the received packets when ECT(0) was expected or detecting a very high packet loss rate when ECN is used.

Examples of SDP offers and answers for video can be found in clause A.4. SDP examples for offering and accepting ECT are shown in Annex A.12.2.

NOTE: For H.264 / MPEG-4 (Part 10) AVC, the optional max-rcmd-nalu-size receiver-capability parameter of RFC 6184 [25] should be set to the smaller of the MTU size (if known) minus header size or 1 400 bytes (otherwise).

The "framerate" attribute as specified in [8] indicates the maximum frame rate the offerer wishes to receive. If the "framerate" attribute is present in the SDP offer, its value may be modified in the SDP answer when the answerer wishes to receive video with a different maximum frame rate than what was indicated in the offer.

An MTSI client in terminal setting up asymmetric video streams with H.264 (AVC) should use both the 'level-asymmetry-allowed' parameter and the 'max-recv-level' parameter that are defined in the H.264 payload format, [25]. When the 'max-recv-level' parameter is used then the level offered for the receiving direction using the 'max-recv-level' parameter must be higher than the default level that is offered with the 'profile-level-id' parameter.

An SDP offer-answer example showing the usage of the 'level-asymmetry-allowed' and 'max-recv-level' parameters is included in Annex A.4.5.

An MTSI client in terminal setting up asymmetric video streams with H.265 (HEVC) should use the 'max-recv-level-id' parameter that is defined in the H.265 payload format, [120]. The level offered for the receiving direction using the 'max-recv-level-id' parameter must be higher than the default level that is offered with the 'level-id' parameter.

An SDP offer-answer example showing the usage of the 'max-recv-level-id' parameter is included in Annex A.4.8.

The resolutions in the "imageattr" attribute correspond to the image size information in the encoded video bitstream such that the x-component corresponds to the image width, and the y-component corresponds to the height component. When the bit-rate is being adapted, values of image width or image height smaller than the x- or y-component(s) in the negotiated "imageattr" attribute may be temporarily used.

7.15.3 Profile requirements (Informative)

[GSMA NG.114 V1.0, clause 3.3.1]:

The entities in the IMS core network that terminate the user plane must support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 1.2 implemented as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.264 [83] Constrained High Profile (CHP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.265 [84] Main Profile, Main Tier Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

For backward compatibility, the UE must also support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16], and when H.264 [83] (Advanced Video Coding (AVC)) CHP Level 3.1 is offered, then H.264 [83] CBP Level 3.1 must also be offered.

[GSMA NG.114 V1.0, clause 3.3.2.1]:

The Session Description Protocol (SDP) offer/answer for video media must be formatted as specified in section 6.2.3 of 3GPP TS 26.114 [16], along with the restrictions included in the present document.

Unless preconfigured otherwise by the home operator with the Media_type_restriction_policy parameter as specified in Annex C.3 and when offering video media that is not already part of the session, regardless if it is at the start of the session or at some later point in time, the UE must include in the SDP offer at least:

1. One H.265 (HEVC) Main Profile, Main Tier, Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].
2. One H.264 (AVC) Constrained High Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].
3. One H.264 (AVC) Constrained Baseline Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].

The payload type preference order on the SDP m= line must be as specified by the numbered list above.

Coordination of Video Orientation (CVO) as specified in 3GPP TS 26.114 [16] shall be supported with two (2) bits granularity by the UE and the entities in the IMS core network which terminate the user plane. The support for CVO shall be included in SDP offer and SDP answer as specified in section 6.2.3 of 3GPP TS 26.114 [16].

[GSMA NG.114 V1.0, clause 3.3.2.2]:

If an asymmetric video stream for H.265 (HEVC) is supported, the parameter 'max-recv-level-id' should be included in the SDP offer and SDP answer, and the level offered with it must be higher than the default level offered with the 'level-id' parameter in the SDP offer/answer respectively, as specified in section 7.1 of IETF RFC 7798 [86] and section 6.2.3 of 3GPP TS 26.114 [16].

7.15.4 Test description

7.15.4.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to not use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.15.4.2 Test procedure sequence

Table 7.15.4.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A	UE is made to attempt an IMS video call.	-		-	-
0B-0I	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-		-	-
1	UE sends INVITE with the first SDP offer, without preconditions. (Step 1 of Annex A.15.2)	->	INVITE	1	P
2-3	Step 2-3 of Annex A.15.2		-	-	-
4	UE acknowledges the receipt of 183 response. (Step 4 of Annex A.15.2)	->	PRACK	2	P
5-9	Step 5-9 of Annex A.15.2.	<-	200 OK		
10	UE acknowledges. (Step 10 of Annex A.15.2)	->	ACK	3	P

7.15.4.3 Specific message contents

None as fully specified in Annex A.15.2.

7.16 MTSI MT Video call with preconditions at both originating UE and terminating UE / 5GS

7.16.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE receives INVITE for video call }
    then { UE may respond with 100 Trying }
}
```

(2)

```
with { UE being registered to IMS }
```

```
ensure that {
  when { UE receives INVITE for video call }
    then { UE responds with 183 Session Progress including SDP }
}
```

(3)

```
with { UE having sent 183 Session Progress }
ensure that {
  when { UE receives PRACK for 183 Session Progress }
    then { UE sends 200 OK for PRACK }
}
```

(4)

```
with { UE having sent 200 OK for PRACK }
ensure that {
  when { UE receives UPDATE including SDP }
    then { UE sends 200 OK for UPDATE including SDP and 180 Ringing }
}
```

(5)

```
with { UE having sent 180 Ringing, possibly reliably }
ensure that {
  when { 180 Ringing was sent reliably and consequently UE receives PRACK for 180 Ringing }
    then { UE sends 200 OK for PRACK }
}
```

(6)

```
with { UE having sent 180 Ringing }
ensure that {
  when { User accepts the incoming video call request }
    then { UE sends 200 OK for INVITE }
}
```

(7)

```
with { UE having sent 200 OK for INVITE }
ensure that {
  when { UE receives ACK followed by BYE }
    then { UE sends 200 OK for BYE }
}
```

7.16.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 6.1.1]:

For "video" and "audio" media types that utilize the RTP/RTCP, the UE shall specify the proposed bandwidth for each media stream utilizing the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556, then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890.

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in 3GPP TS 29.208.

[TS 24.229, clause 6.1.3]:

Upon sending an SDP answer to an SDP offer, with the SDP answer including one or more media streams for which the originating side did indicate its local preconditions as not met, if the precondition mechanism is used by the terminating UE (see subclause 5.1.4.1), the terminating UE shall indicate its local preconditions and request the confirmation for the result of the resource reservation at the originating end point.

…

Upon receiving an initial INVITE request that includes the SDP offer containing an IP address type (in the "c=" parameter) that is not supported by the UE, the UE shall:

- if the UE is a UE performing the functions of an external attached network and

  1) if the received SDP offer contains an "altc" SDP attribute indicating an alternative and supported IP address; and

  2) the UE supports the "altc" SDP attribute;

  select an IP address type in accordance with RFC 6947 [228]; or

- otherwise respond with a 488 (Not Acceptable Here) response including a 301 Warning header field indicating "incompatible network address format".

NOTE 2: Upon receiving an initial INVITE request that does not include an SDP offer, the UE can accept the request and include an SDP offer in the first reliable response. The SDP offer will reflect the called user's terminal capabilities and user preferences for the session.

If the UE receives an SDP offer that specifies different IP address type for media (i.e. specify it in the "c=" parameter of the SDP offer) that the UE is using for signalling, and if the UE supports both IPv4 and IPv6 addresses simultaneously, the UE shall accept the received SDP offer. Subsequently, the UE shall either acquire an IP address type or use an existing IP address type as specified in the SDP offer, and include it in the "c=" parameter in the SDP answer.

NOTE 3: Upon receiving an initial INVITE request, that includes an SDP offer containing connection addresses (in the "c=" parameter) equal to zero, the UE will select the media streams that is willing to accept for the session, reserve the QoS resources for accepted media streams, and include its valid connection address in the SDP answer.

…

If the terminating UE uses the precondition mechanism (see subclause 5.1.4.1), if the desired QoS resources for one or more media streams have not been reserved at the terminating UE when constructing the SDP offer, the terminating UE shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment.

NOTE 7: It is out of scope of this specification which media streams are to be included in the SDP offer.

If the terminating UE uses the precondition mechanism (see subclause 5.1.4.1) and if the desired QoS resources for one or more media streams are available at the terminating UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment.

If the terminating UE sends an UPDATE request to remove one or more media streams negotiated in the session for which a final response to the INVITE request has not been sent yet, the terminating UE sets the ports of the media streams to be removed from the session to zero in the new SDP offer.

NOTE 8: Upon receiving an initial INVITE request with one or more media streams which the terminating UE supports and one or more media streams which the UE does not support, the UE is not expected to reject the INVITE request just because of the presence of the unsupported media stream.

NOTE 9: Previous versions of this document mandated the use of the SDP inactive attribute in the SDP offer if the desired QoS resources for one or more media streams had not been reserved at the originating UE when constructing the SDP offer unless the originating UE knew that the precondition mechanism was supported by the remote UE. The use can still occur when interoperating with devices based on earlier versions of this document.

[TS 26.114, clause 5.2.2]:

MTSI clients in terminals offering video communication shall support:

- H.264 (AVC) [24] Constrained Baseline Profile (CBP) Level 1.2;
- H.265 (HEVC) [119] Main Profile, Main Tier, Level 3.1.

In addition they should support:

- H.264 (AVC) [24] Constrained High Profile (CHP) Level 3.1.

[TS 26.114, clause 6.2.3.2]:

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

An MTSI client is required to support the AVPF feedback messages trr-int, NACK and PLI [40] and the CCM feedback messages FIR, TMMBR and TMMBN [43], see Clauses 7.3.3 and 10.3. These feedback messages can only be used together with AVPF and shall be negotiated in SDP offer/answer before they can be used in the session [40]. An MTSI client sending an SDP offer for AVPF shall also include these AVPF and CCM feedback messages in the offer. An MTSI client accepting an SDP offer for AVPF for video shall also accept these AVPF and CCM feedback messages if they are offered.

If an MTSI client offers to use ECN for video in RTP streams then the MTSI client shall offer ECN Capable Transport as defined below. If an MTSI client accepts an offer for ECN for video then the MTSI client shall declare ECN Capable Transport in the SDP answer as defined below. The SDP negotiation of ECN Capable Transport is described in [84].

The use of ECN for a video stream in RTP is negotiated with the "ecn-capable-rtp" SDP attribute, [84]. ECN is enabled when both clients agree to use ECN as configured below. An MTSI client using ECN shall therefore also include the following parameters and parameter values for the ECN attribute:

- ‘leap’, to indicate that the leap-of-faith initiation method shall be used;
- ‘ect=0’, to indicate that ECT(0) shall be set for every packet.

An MTSI client offering ECN for video shall indicate support of TMMBR [43] by including the "ccm tmmbr" value within an "rtcp-fb" SDP attribute [40]. An MTSI client offering ECN for video may indicate support for RTCP AVPF ECN feedback messages [84] using the "rtcp-fb" SDP attribute with the "nack" feedback parameter and the "ecn" feedback parameter value. An MTSI client offering ECN for video may indicate support for RTCP XR ECN summary reports [84] using the "rtcp-xr" SDP attribute and the "ecn-sum" parameter.

An MTSI client receiving an offer for ECN for video with an indication of support of TMMBR [43] within an "rtcp-fb" attribute should accept the offer if it supports ECN. It shall then indicate support for TMMBR using an "rtcp-fb" attribute in the SDP answer.

An MTSI client receiving an offer for ECN for video with an indication of support of RTCP AVPF ECN feedback message but without support for TMMBR should accept the offer if it supports ECN and also the RTCP AVPF ECN feedback message. It shall then indicate support of the RTCP AVPF ECN feedback message using the "rtcp-fb" attribute in the SDP answer.

An MTSI client receiving an offer for ECN for video with an indication of support of RTCP XR ECN summary reports [84] without support for TMMBR should accept the offer if it supports ECN and also the RTCP XR ECN summary reports. It shall then indicate support of RTCP XR ECN summary reports in the SDP answer.

The use of ECN is disabled when a client sends an SDP without the "ecn-capable-rtp" SDP attribute.

An MTSI client may initiate a session re-negotiation to disable ECN to resolve ECN-related error cases. An ECN-related error case may be, for example, detecting non-ECT in the received packets when ECT(0) was expected or detecting a very high packet loss rate when ECN is used.

Examples of SDP offers and answers for video can be found in clause A.4. SDP examples for offering and accepting ECT are shown in Annex A.12.2.

NOTE: For H.264 / MPEG-4 (Part 10) AVC, the optional max-rcmd-nalu-size receiver-capability parameter of RFC 6184 [25] should be set to the smaller of the MTU size (if known) minus header size or 1 400 bytes (otherwise).

The "framerate" attribute as specified in [8] indicates the maximum frame rate the offerer wishes to receive. If the "framerate" attribute is present in the SDP offer, its value may be modified in the SDP answer when the answerer wishes to receive video with a different maximum frame rate than what was indicated in the offer.

An MTSI client in terminal setting up asymmetric video streams with H.264 (AVC) should use both the 'level-asymmetry-allowed' parameter and the 'max-recv-level' parameter that are defined in the H.264 payload format, [25]. When the 'max-recv-level' parameter is used then the level offered for the receiving direction using the 'max-recv-level' parameter must be higher than the default level that is offered with the 'profile-level-id' parameter.

An SDP offer-answer example showing the usage of the 'level-asymmetry-allowed' and 'max-recv-level' parameters is included in Annex A.4.5.

An MTSI client in terminal setting up asymmetric video streams with H.265 (HEVC) should use the 'max-recv-level-id' parameter that is defined in the H.265 payload format, [120]. The level offered for the receiving direction using the 'max-recv-level-id' parameter must be higher than the default level that is offered with the 'level-id' parameter.

An SDP offer-answer example showing the usage of the 'max-recv-level-id' parameter is included in Annex A.4.8.

The resolutions in the "imageattr" attribute correspond to the image size information in the encoded video bitstream such that the x-component corresponds to the image width, and the y-component corresponds to the height component. When the bit-rate is being adapted, values of image width or image height smaller than the x- or y-component(s) in the negotiated "imageattr" attribute may be temporarily used.

7.16.3 Profile requirements (Informative)

[GSMA NG.114 V1.0, cluse 3.3.1]:

The entities in the IMS core network that terminate the user plane must support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 1.2 implemented as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.264 [83] Constrained High Profile (CHP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.265 [84] Main Profile, Main Tier Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

For backward compatibility, the UE must also support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16], and when H.264 [83] (Advanced Video Coding (AVC)) CHP Level 3.1 is offered, then H.264 [83] CBP Level 3.1 must also be offered.

[GSMA NG.114 V1.0, cluse 3.3.2.1]:

The Session Description Protocol (SDP) offer/answer for video media must be formatted as specified in section 6.2.3 of 3GPP TS 26.114 [16], along with the restrictions included in the present document.

Unless preconfigured otherwise by the home operator with the Media_type_restriction_policy parameter as specified in Annex C.3 and when offering video media that is not already part of the session, regardless if it is at the start of the session or at some later point in time, the UE must include in the SDP offer at least:

1. One H.265 (HEVC) Main Profile, Main Tier, Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].

2. One H.264 (AVC) Constrained High Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].

3. One H.264 (AVC) Constrained Baseline Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].

The payload type preference order on the SDP m= line must be as specified by the numbered list above.

Coordination of Video Orientation (CVO) as specified in 3GPP TS 26.114 [16] shall be supported with two (2) bits granularity by the UE and the entities in the IMS core network which terminate the user plane. The support for CVO shall be included in SDP offer and SDP answer as specified in section 6.2.3 of 3GPP TS 26.114 [16].

[GSMA NG.114 V1.0, cluse 3.3.2.2]:

If an asymmetric video stream for H.265 (HEVC) is supported, the parameter ‘max-recv-level-id’ should be included in the SDP offer and SDP answer, and the level offered with it must be higher than the default level offered with the ‘level-id’ parameter in the SDP offer/answer respectively, as specified in section 7.1 of IETF RFC 7798 [86] and section 6.2.3 of 3GPP TS 26.114 [16].

7.16.4 Test description

7.16.4.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.16.4.2 Test procedure sequence

Table 7.16.4.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
1	SS sends INVITE with the first SDP offer. (Step 1 of Annex A.16.1)	<-	INVITE	-	-
2	Check: (Optional) Does the UE respond with a 100 Trying provisional response? (Step 2 of Annex A.16.1)	->	100 Trying	1	-P
3	Check: Does the UE send 183 response reliably with the SDP answer to the offer in INVITE? (Step 3 of Annex A.16.1)	->	183 Session Progress	2	P
4	SS acknowledges the receipt of 183 response from the UE. (Step 4 of Annex A.16.1)	<-	PRACK	-	-
5	Check: Does the UE responds to PRACK with 200 OK. (Step 5 of Annex A.16.1)	->	200 OK	3	P
6	SS sends an UPDATE with SDP offer indicating SS reserved resources. (Step 6 of Annex A.16.1)	<-	UPDATE		
7	Check: Does the UE acknowledges the UPDATE with 200 OK and includes SDP answer to acknowledge its current precondition status. (Step 7 of Annex A.16.1)	->	200 OK	4	P
8	Check: (Optional) Does the UE responds to INVITE with 180 Ringing? (Step 8 of Annex A.16.1)	->	180 Ringing	-	-
9	(Optional) SS shall send PRACK only if the 180 response contains 100rel option tag within the Require header? (Step 9 of Annex A.16.1)	<-	PRACK	-	-
10	Check: (Optional) Does the UE acknowledges the PRACK with 200 OK? (Step 10 of Annex A.16.1)	->	200 OK	5	P
11	UE is made to answer the call.	-	-	-	-
12	Check: Does the UE responds to INVITE with a 200 OK final response after the user answers the call? (Step 12 of Annex A.16.1)	->	200 OK	6	P
13	The SS acknowledges the receipt of 200 OK for INVITE. (Step 13 of Annex A.16.1)	<-	ACK	-	-
14	The SS releases the call with BYE. (Step 1 of Annex A.8)	<-	BYE	-	-
15	Check: Does the UE sends 200 OK for BYE. (Step 2 of Annex A.8)	->	200 OK	7	P

7.16.4.3 Specific message contents

None as fully specified in A.16.1 and A.8.

7.17 MTSI MT Video call without preconditions at both originating UE and terminating UE / 5GS

7.17.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to not use preconditions }
ensure that {
  when { UE receives INVITE for video call }
    then { UE may respond with 100 Trying and then sends 183 Session Progress with SDP without
preconditions }
}
```

(2)

```
with { UE having sent 183 Session Progress }
ensure that {
```

```
  when { UE receives PRACK for 183 Session Progress }
    then { UE sends 200 OK for PRACK }
}
```

(3)

```
with { UE having sent 200 OK for PRACK }
ensure that {
  when { UE is ready to start the call }
    then { UE sends 180 Ringing followed by 200 OK for INVITE }
}
```

7.17.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 6.1.1]:

For "video" and "audio" media types that utilize the RTP/RTCP, the UE shall specify the proposed bandwidth for each media stream utilizing the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556, then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890.

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in 3GPP TS 29.208.

[TS 24.229, clause 6.1.3]:

Upon sending an SDP answer to an SDP offer, with the SDP answer including one or more media streams for which the originating side did indicate its local preconditions as not met, if the precondition mechanism is used by the terminating UE (see subclause 5.1.4.1), the terminating UE shall indicate its local preconditions and request the confirmation for the result of the resource reservation at the originating end point.

…

Upon receiving an initial INVITE request that includes the SDP offer containing an IP address type (in the "c=" parameter) that is not supported by the UE, the UE shall:

- if the UE is a UE performing the functions of an external attached network and

  1) if the received SDP offer contains an "altc" SDP attribute indicating an alternative and supported IP address; and

  2) the UE supports the "altc" SDP attribute;

  select an IP address type in accordance with RFC 6947 [228]; or

- otherwise respond with a 488 (Not Acceptable Here) response including a 301 Warning header field indicating "incompatible network address format".

NOTE 2: Upon receiving an initial INVITE request that does not include an SDP offer, the UE can accept the request and include an SDP offer in the first reliable response. The SDP offer will reflect the called user's terminal capabilities and user preferences for the session.

If the UE receives an SDP offer that specifies different IP address type for media (i.e. specify it in the "c=" parameter of the SDP offer) that the UE is using for signalling, and if the UE supports both IPv4 and IPv6 addresses simultaneously, the UE shall accept the received SDP offer. Subsequently, the UE shall either acquire an IP address type or use an existing IP address type as specified in the SDP offer, and include it in the "c=" parameter in the SDP answer.

NOTE 3: Upon receiving an initial INVITE request, that includes an SDP offer containing connection addresses (in the "c=" parameter) equal to zero, the UE will select the media streams that is willing to accept for the session, reserve the QoS resources for accepted media streams, and include its valid connection address in the SDP answer.

…

If the terminating UE uses the precondition mechanism (see subclause 5.1.4.1), if the desired QoS resources for one or more media streams have not been reserved at the terminating UE when constructing the SDP offer, the terminating UE shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment.

NOTE 7: It is out of scope of this specification which media streams are to be included in the SDP offer.

If the terminating UE uses the precondition mechanism (see subclause 5.1.4.1) and if the desired QoS resources for one or more media streams are available at the terminating UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment.

If the terminating UE sends an UPDATE request to remove one or more media streams negotiated in the session for which a final response to the INVITE request has not been sent yet, the terminating UE sets the ports of the media streams to be removed from the session to zero in the new SDP offer.

NOTE 8: Upon receiving an initial INVITE request with one or more media streams which the terminating UE supports and one or more media streams which the UE does not support, the UE is not expected to reject the INVITE request just because of the presence of the unsupported media stream.

NOTE 9: Previous versions of this document mandated the use of the SDP inactive attribute in the SDP offer if the desired QoS resources for one or more media streams had not been reserved at the originating UE when constructing the SDP offer unless the originating UE knew that the precondition mechanism was supported by the remote UE. The use can still occur when interoperating with devices based on earlier versions of this document.

[TS 26.114, clause 5.2.2]:

MTSI clients in terminals offering video communication shall support:

- H.264 (AVC) [24] Constrained Baseline Profile (CBP) Level 1.2;
- H.265 (HEVC) [119] Main Profile, Main Tier, Level 3.1.

In addition they should support:

- H.264 (AVC) [24] Constrained High Profile (CHP) Level 3.1.

[TS 26.114, clause 6.2.3.2]:

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

An MTSI client is required to support the AVPF feedback messages trr-int, NACK and PLI [40] and the CCM feedback messages FIR, TMMBR and TMMBN [43], see Clauses 7.3.3 and 10.3. These feedback messages can only be used together with AVPF and shall be negotiated in SDP offer/answer before they can be used in the session [40]. An MTSI client sending an SDP offer for AVPF shall also include these AVPF and CCM feedback messages in the offer. An MTSI client accepting an SDP offer for AVPF for video shall also accept these AVPF and CCM feedback messages if they are offered.

If an MTSI client offers to use ECN for video in RTP streams then the MTSI client shall offer ECN Capable Transport as defined below. If an MTSI client accepts an offer for ECN for video then the MTSI client shall declare ECN Capable Transport in the SDP answer as defined below. The SDP negotiation of ECN Capable Transport is described in [84].

The use of ECN for a video stream in RTP is negotiated with the "ecn-capable-rtp" SDP attribute, [84]. ECN is enabled when both clients agree to use ECN as configured below. An MTSI client using ECN shall therefore also include the following parameters and parameter values for the ECN attribute:

- ‘leap’, to indicate that the leap-of-faith initiation method shall be used;
- ‘ect=0’, to indicate that ECT(0) shall be set for every packet.

An MTSI client offering ECN for video shall indicate support of TMMBR [43] by including the "ccm tmmbr" value within an "rtcp-fb" SDP attribute [40]. An MTSI client offering ECN for video may indicate support for RTCP AVPF ECN feedback messages [84] using the "rtcp-fb" SDP attribute with the "nack" feedback parameter and the "ecn" feedback parameter value. An MTSI client offering ECN for video may indicate support for RTCP XR ECN summary reports [84] using the "rtcp-xr" SDP attribute and the "ecn-sum" parameter.

An MTSI client receiving an offer for ECN for video with an indication of support of TMMBR [43] within an "rtcp-fb" attribute should accept the offer if it supports ECN. It shall then indicate support for TMMBR using an "rtcp-fb" attribute in the SDP answer.

An MTSI client receiving an offer for ECN for video with an indication of support of RTCP AVPF ECN feedback message but without support for TMMBR should accept the offer if it supports ECN and also the RTCP AVPF ECN feedback message. It shall then indicate support of the RTCP AVPF ECN feedback message using the "rtcp-fb" attribute in the SDP answer.

An MTSI client receiving an offer for ECN for video with an indication of support of RTCP XR ECN summary reports [84] without support for TMMBR should accept the offer if it supports ECN and also the RTCP XR ECN summary reports. It shall then indicate support of RTCP XR ECN summary reports in the SDP answer.

The use of ECN is disabled when a client sends an SDP without the "ecn-capable-rtp" SDP attribute.

An MTSI client may initiate a session re-negotiation to disable ECN to resolve ECN-related error cases. An ECN-related error case may be, for example, detecting non-ECT in the received packets when ECT(0) was expected or detecting a very high packet loss rate when ECN is used.

Examples of SDP offers and answers for video can be found in clause A.4. SDP examples for offering and accepting ECT are shown in Annex A.12.2.

NOTE: For H.264 / MPEG-4 (Part 10) AVC, the optional max-rcmd-nalu-size receiver-capability parameter of RFC 6184 [25] should be set to the smaller of the MTU size (if known) minus header size or 1 400 bytes (otherwise).

The "framerate" attribute as specified in [8] indicates the maximum frame rate the offerer wishes to receive. If the "framerate" attribute is present in the SDP offer, its value may be modified in the SDP answer when the answerer wishes to receive video with a different maximum frame rate than what was indicated in the offer.

An MTSI client in terminal setting up asymmetric video streams with H.264 (AVC) should use both the ‘level-asymmetry-allowed’ parameter and the ‘max-recv-level’ parameter that are defined in the H.264 payload format, [25]. When the ‘max-recv-level’ parameter is used then the level offered for the receiving direction using the ‘max-recv-level’ parameter must be higher than the default level that is offered with the ‘profile-level-id’ parameter.

An SDP offer-answer example showing the usage of the ‘level-asymmetry-allowed’ and ‘max-recv-level’ parameters is included in Annex A.4.5.

An MTSI client in terminal setting up asymmetric video streams with H.265 (HEVC) should use the ‘max-recv-level-id’ parameter that is defined in the H.265 payload format, [120]. The level offered for the receiving direction using the ‘max-recv-level-id’ parameter must be higher than the default level that is offered with the ‘level-id’ parameter.

An SDP offer-answer example showing the usage of the ‘max-recv-level-id’ parameter is included in Annex A.4.8.

The resolutions in the "imageattr" attribute correspond to the image size information in the encoded video bitstream such that the x-component corresponds to the image width, and the y-component corresponds to the height component. When the bit-rate is being adapted, values of image width or image height smaller than the x- or y-component(s) in the negotiated "imageattr" attribute may be temporarily used.

7.17.3 Profile requirements (Informative)

[GSMA NG.114 V1.0, cluse 3.3.1]:

The entities in the IMS core network that terminate the user plane must support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 1.2 implemented as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.264 [83] Constrained High Profile (CHP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

The UE must support ITU-T Recommendation H.265 [84] Main Profile, Main Tier Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16].

For backward compatibility, the UE must also support ITU-T Recommendation H.264 [83] Constrained Baseline Profile (CBP) Level 3.1 as specified in section 5.2.2 of 3GPP TS 26.114 [16], and when H.264 [83] (Advanced Video Coding (AVC)) CHP Level 3.1 is offered, then H.264 [83] CBP Level 3.1 must also be offered.

[GSMA NG.114 V1.0, cluse 3.3.2.1]:

The Session Description Protocol (SDP) offer/answer for video media must be formatted as specified in section 6.2.3 of 3GPP TS 26.114 [16], along with the restrictions included in the present document.

Unless preconfigured otherwise by the home operator with the Media_type_restriction_policy parameter as specified in Annex C.3 and when offering video media that is not already part of the session, regardless if it is at the start of the session or at some later point in time, the UE must include in the SDP offer at least:

1. One H.265 (HEVC) Main Profile, Main Tier, Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].
2. One H.264 (AVC) Constrained High Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].
3. One H.264 (AVC) Constrained Baseline Profile Level 3.1 payload type as defined in sections 5.2.2 and 7.4.3 of 3GPP TS 26.114 [16].

The payload type preference order on the SDP m= line must be as specified by the numbered list above.

Coordination of Video Orientation (CVO) as specified in 3GPP TS 26.114 [16] shall be supported with two (2) bits granularity by the UE and the entities in the IMS core network which terminate the user plane. The support for CVO shall be included in SDP offer and SDP answer as specified in section 6.2.3 of 3GPP TS 26.114 [16].

[GSMA NG.114 V1.0, cluse 3.3.2.2]:

If an asymmetric video stream for H.265 (HEVC) is supported, the parameter ‘max-recv-level-id’ should be included in the SDP offer and SDP answer, and the level offered with it must be higher than the default level offered with the ‘level-id’ parameter in the SDP offer/answer respectively, as specified in section 7.1 of IETF RFC 7798 [86] and section 6.2.3 of 3GPP TS 26.114 [16].

7.17.4 Test description

7.17.4.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to not use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.17.4.2 Test procedure sequence

Table 7.17.4.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
1	SS sends INVITE with the first SDP offer.	<-	INVITE	-	-
2	Check: (Optional) Does The UE respond with a 100 Trying provisional response?	->	100 Trying	1	-P
3	The UE sends 183 response reliably with the SDP answer to the offer in INVITE	->	183 Session Progress	-	-
4	SS acknowledges the receipt of 183 response from the UE.	<-	PRACK	-	-
5	Check: Does the UE respond to PRACK with 200 OK?	->	200 OK	2	P
6	Check: (Optional) Does the UE responds to INVITE with 180 Ringing?	->	180 Ringing	3	P
7	(Conditional) If the 180 response contains 100rel option tag within the Require header, then the SS shall send PRACK.	<-	PRACK	-	-
8	(Conditional) If the SS sent PRACK, then the UE acknowledges the PRACK with 200 OK.	->	200 OK	-	-
9	Make UE accept the video call.	-	-	-	-
10	The UE responds to INVITE with a 200 OK final response after the user answers the call.	->	200 OK	-	-
11	The SS acknowledges the receipt of 200 OK for INVITE.	<-	ACK	-	-

7.17.4.3 Specific message contents

None as fully specified in A.16.2.

7.18 MTSI MO Voice Call / EVS / AMR-WB / 5GS

7.18.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call with preconditions }
            }
```

(2)

```
with { UE having sent INVITE with preconditions }
ensure that {
  when { UE receives 183 Session Progress indicating AMR-WB }
    then { UE sends PRACK for 183 Session Progress and, after receiving 200 OK for PRACK, agrees to
AMR-WB via UPDATE }
            }
```

7.18.2 Conformance Requirements

[TS 24.229, clause 5.1.3.1]:

Where multiple domains exist for initiating a call/session, before sending an initial INVITE request, the UE shall perform access domain selection in accordance with the appropriate specification for the IP-CAN in use, taking into account the media to be requested. Access domain selection allows the policy of the network operator to be taken into account before the initial INVITE request is sent. Access dependent aspects of access domain selection are defined in the access technology specific annexes for each access technology.

Upon generating an initial INVITE request, the UE shall include the Accept header field with "application/sdp", the MIME type associated with the 3GPP IM CN subsystem XML body (see subclause 7.6.1) and any other MIME type the UE is willing and capable to accept.

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

The preconditions mechanism should be supported by the originating UE.

…

In order to allow the peer entity to reserve its required resources, if the precondition mechanism is enabled as specified in subclause 5.1.5A; the originating UE supporting the precondition mechanism should make use of the precondition mechanism, even if it does not require local resource reservation.

Upon generating an initial INVITE request using the precondition mechanism, the UE shall:

- indicate the support for reliable provisional responses and specify it using the Supported header field; and
- indicate the support for the preconditions mechanism and specify it using the Supported header field.

Upon generating an initial INVITE request using the precondition mechanism, the UE shall not indicate the requirement for the precondition mechanism by using the Require header field.

During the session initiation, if the originating UE indicated the support for the precondition mechanism in the initial INVITE request and:

a) the received response with an SDP body includes a Require header field with "precondition" option-tag, the originating UE shall include a Require header field with the "precondition" option-tag:
    - in subsequent requests that include an SDP body, that the originating UE sends in the same dialog as the response is received from; and
    - in responses with an SDP body to subsequent requests that include an SDP body and include "precondition" option-tag in Supported header field or Require header field received in-dialog; or

b) the received response with an SDP body does not include the "precondition" option-tag in the Require header field,
    - in subsequent requests that include an SDP body, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog;
    - in responses with an SDP body to subsequent requests with an SDP body but without "precondition" option-tag in the Require or Supported header field, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog; and
    - in responses with an SDP body to subsequent requests with an SDP body and with "precondition" option-tag in the Require or Supported header field, the originating UE shall include a Require header field with "precondition" option-tag in the same dialog.

NOTE 2: Table A.4 specifies that UE support of forking is required in accordance with RFC 3261 [26]. The UE can accept or reject any of the forked responses, for example, if the UE is capable of supporting a limited number of simultaneous transactions or early dialogs.

Upon successful reservation of local resources the UE shall confirm the successful resource reservation (see subclause 6.1.2) within the next SIP request.

NOTE 3: In case of the precondition mechanism being used on both sides, this confirmation will be sent in either a PRACK request or an UPDATE request. In case of the precondition mechanism not being supported on one or both sides, alternatively a reINVITE request can be used for this confirmation after a 200 (OK) response has been received for the initial INVITE request, in case the terminating UE does not support the PRACK request (as described in RFC 3262 [27]) and does not support the UPDATE request (as described in RFC 3311 [29]).

NOTE 4: The UE can receive a P-Early-Media header field authorizing an early-media flow while the required preconditions, if any, are not met and/or the flow direction is not enabled by the SDP direction parameter. According to RFC 5009 [109], an authorized early-media flow can be established only if the necessary conditions related to the SDP negotiation are met. These conditions can evolve during the session establishment.

NOTE 5: When the UE is confirming the successful resource reservation using an UPDATE request (or a PRACK request) and the UE receives a 180 (Ringing) response or a 200 (OK) response to the initial INVITE request before receiving a 200 (OK) response to the UPDATE request (or a 200 (OK) response to the PRACK request), the UE does not treat this as an error case and does not release the session.

NOTE 6: The UE procedures for rendering of the received early media and of the locally generated communication progress information are specified in 3GPP TS 24.628 [8ZF].

If the UE wishes to receive early media authorization indications, as described in RFC 5009 [109], the UE shall add the P-Early-Media header field with the "supported" parameter to the initial INVITE request.

A UE supporting the Session Timer extension as described in RFC 4028 [58] may support the extension being configured using Session_Timer_Support node specified in 3GPP TS 24.167 [8G].

If the UE supports the Session Timer extension, the UE shall include the option-tag "timer" in the Supported header field and should either insert a Session-Expires header field with the header field value set to the configured session timer interval value, or should not include the Session-Expires header field in the initial INVITE request. The header field value of the Session-Expires header field may be configured using local configuration or using the Session_Timer_Initial_Interval node specified in 3GPP 24.167 [8G]. If the UE is configured with both the local configuration and the Session_Timer_Initial_Interval node specified in 3GPP 24.167 [8G], then the local configuration shall take precedence.

If the UE inserts the Session-Expires header field in the initial INVITE request, the UE may also include the "refresher" parameter with the "refresher" parameter value set to "uac".

…

The UE may include a "cic" tel URI parameter in a tel URI, or in the userinfo part of a SIP URI with user=phone, in the Request-URI of an initial INVITE request if the UE wants to identify a user-dialled carrier, as described in RFC 4694 [112].

NOTE 8: The method whereby the UE determines when to include a "cic" tel-URI parameter and what value it should contain is outside the scope of this document (e.g. the UE could use a locally configured digit map to look for special prefix digits that indicate the user has dialled a carrier).

NOTE 9: The value of the "cic" tel-URI parameter reported by the UE is not dependent on UE location (e.g. the reported value is not affected by roaming scenarios).

[TS 24.229, clause 6.1.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

…

In order to support accurate bandwidth calculations, the UE may include the "a=ptime" attribute for all "audio" media lines as described in RFC 4566 [39]. If a UE receives an "audio" media line with "a=ptime" specified, the UE should transmit at the specified packetization rate. If a UE receives an "audio" media line which does not have "a=ptime" specified or the UE does not support the "a=ptime" attribute, the UE should transmit at the default codec packetization rate as defined in RFC 3551 [55A]. The UE will transmit consistent with the resources available from the network.

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

NOTE 2: The above is the minimum requirement for all UEs. Additional requirements can be found in other specifications.

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE may include for each RTP payload type "a=bw-info" SDP attribute(s) (defined in clause 19 of 3GPP TS 26.114 [9B]) to indicate the additional bandwidth information. The "a=bw-info" SDP attribute line(s) shall be specified in accordance with 3GPP TS 26.114 [9B]. The value of the "a=bw-info" SDP attribute(s) may affect the assigned QoS which is defined in 3GPP TS 29.213 [13C].

For "video" and "audio" media types that utilize the RTP/RTCP, in addition to the "b=AS" parameter, the UE may specify the "b=TIAS", and "a=maxprate" parameters in accordance with RFC 3890 [152]. The value of the parameter shall be determined as described in RFC 3890 [152]. The value or absence of the "b=" parameter(s) may affect the assigned QoS which is defined in 3GPP TS 29.213 [13C].

If a UE receives a media line which contains both a=ptime and a=maxprate, the UE should use the a=maxprate value, if this attribute is supported.

If multiple codecs are specified on the media line, "a=maxprate" (or "a=ptime" if "a=maxprate" is not available or not supported) should be used to derive the packetization time used for all codecs specified on the media line. Given that not all codecs support identical ranges of packetization, the UE should ensure that the packetization derived by "a=maxprate" (or "a=ptime" if "a=maxprate" is not available or not supported) is a valid packetization time for each codec specified in the list.

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

…

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

NOTE 4: Based on this resource reservation can, in certain cases, be initiated immediately after the sending or receiving of the initial SDP offer.

[TS 26.114, clause 5.2.1.1]:

MTSI clients in terminals offering speech communication shall support narrowband, wideband and super-wideband communication. The only exception to this requirement is for the MTSI client in constrained terminal offering speech communication, in which case the MTSI client in constrained terminal shall support narrowband and wideband, and should support super-wideband communication.

In addition, MTSI clients in terminals offering speech communication shall support:

- .AMR speech codec (3GPP TS 26.071 [11], 3GPP TS 26.090 [12], 3GPP TS 26.073 [13] and 3GPP TS 26.104 [14]) including all 8 modes and source controlled rate operation 3GPP TS 26.093 [15]. The MTSI client in terminal shall be capable of operating with any subset of these 8 codec modes. More detailed codec requirements for the AMR codec are defined in clause 5.2.1.2.

MTSI clients in terminals offering wideband speech communication at 16 kHz sampling frequency shall support:

- AMR-WB codec (3GPP TS 26.171 [17], 3GPP TS 26.190 [18], 3GPP TS 26.173 [19] and 3GPP TS 26.204 [20]) including all 9 modes and source controlled rate operation 3GPP TS 26.193 [21]. The MTSI client in terminal shall be capable of operating with any subset of these 9 codec modes. More detailed codec requirements for the AMR-WB codec are defined in clause 5.2.1.3. When the EVS codec is supported, the EVS AMR-WB IO mode may serve as an alternative implementation of AMR-WB as defined in clause 5.2.1.4.

MTSI clients in terminals offering super-wideband or fullband speech communication shall support:

- EVS codec (TS 26.441 [121], TS 26.444 [124], TS 26.445 [125], TS 26.447 [127], TS 26.451 [131], TS 26.442 [122], TS 26.452 [165] and TS 26.443 [123]) as described below including functions for backwards compatibility with AMR-WB (TS 26.446 [126]) and discontinuous transmission (TS 26.449 [129] and TS 26.450 [130]). More detailed codec requirements for the EVS codec are defined in clause 5.2.1.4.

Encoding of DTMF is described in Annex G.

[TS 26.114, clause 6.2.2.1]:

For AMR or AMR-WB encoded media, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, what RTP profile to use; if all codec modes can be used or if the operation needs to be restricted to a subset; if the bandwidth-efficient payload format can be used or if the octet-aligned payload format must be used; if codec mode changes shall be restricted to be aligned to only every other frame border or if codec mode changes can occur at any frame border; if codec mode changes must be restricted to only neighbouring modes within the negotiated codec mode set or if codec mode changes can be performed to any mode within the codec mode set; the number of speech frames that should be encapsulated in each RTP packet and the maximum number of speech frames that may be encapsulated in each RTP packet. For EVS encoded media, the session setup shall determine the RTP profile to use in the session.

If the session setup negotiation concludes that multiple configuration variants are possible in the session then the default operation should be used as far as the agreed parameters allow, see clause 7.5.2.1. It should be noted that the default configurations are slightly different for different access types.

An MTSI client offering a speech media session for narrow-band speech and/or wide-band speech should generate an SDP offer according to the examples in Annexes A.1 to A.3. An MTSI client offering EVS should generate an SDP offer according to the examples in Annex A.14.

An MTSI client in terminal supporting EVS should support the RTCP-APP signalling for speech adaptation defined clause 10.2.1, and shall support the RTCP-APP signalling when the MTSI client in terminal supports adaptation for call cases where the RTP-based CMR cannot be used.

NOTE 1: Examples of call cases where the RTP-based CMR cannot be used are: when the RTP-based CMR is disabled; or for uni-directional media (sendonly or recvonly).

Some of the request messages are generic for all speech codecs while other request messages are codec-specific. Request messages that can be used in a session are negotiated in SDP, see clause 10.2.3.

[TS 26.114, clause 6.2.2.3]:

An MTSI client in terminal must understand all the payload format options that are defined in RFC 4867 [28], and in [125]. It does not have to support operating according to all these options but must be capable to properly accepting or rejecting all options.

The SDP answer depends on many factors, for example:

- what is included in the SDP offer and in what preference order that is defined. The SDP offer will probably be different if it is generated by another MTSI client in terminal, by an MTSI MGW, a TISPAN client or some other VoIP client that does not follow this specification;

- if terminal and/or network resources are available; and:

- if there are other configurations, for example defined with OMA-DM, that mandate, recommend or prevent some configurations.

Table 6.3 describes requirements and recommendations for handling of the AMR payload format parameters and for how to generate the SDP answer.

NOTE 1: An MTSI client in terminal may support more features than what is required by this specification, e.g. crc, robust sorting and interleaving. Table 6.3 describes the handling of the AMR payload format parameters when the MTSI client implementation supports only those features that are required by this specification. Tables 6.3a-6.3c describe the handling of the EVS payload format parameters.

Table 6.3: Handling of the AMR-NB and AMR-WB SDP parameters in the received SDP offer and in the SDP answer

Parameter in the received SDP offer	Comments	Handling
Codec	Wide-band speech is preferable over narrow-band speech	If both AMR-WB and AMR-NB are offered and if AMR-WB is supported by the answering MTSI client in terminal then it shall select to use the AMR-WB codec and include this codec in the SDP answer, unless another preference order is indicated in the SDP offer. If the MTSI client in terminal only supports AMR-NB then this codec shall be selected to be used and shall be included in the SDP answer. The SDP answer shall only include one RTP Payload Type for speech, see NOTE 1.
octet-align	Both the bandwidth-efficient and the octet-aligned payload formats are supported by the MTSI client in terminal. MTSI MGWs for GERAN or UTRAN are likely to either not include the octet-align parameter or to offer octet-align=0. The bandwidth-efficient payload format is preferable over the octet-aligned payload format.	The offer shall not be rejected purely based on the offered payload format variant. If both bandwidth-efficient and octet-aligned are included in the received SDP offer then the MTSI client in terminal shall select the bandwidth-efficient payload format and include it in the configuration in the SDP answer.
mode-set	The MTSI client in terminal can interoperate properly with whatever mode-set the other end-point offers or if no mode-set is offered. The possibilities to use the higher bit rate codec modes also depend on the offered bandwidth. MTSI MGWs for GERAN or UTRAN inter-working are likely to include the mode-set in the offer if in case the intention is to use TFO or TrFO. Mode sets that give more adaptation possibilities are preferable over mode-sets with fewer or no adaptation possibilities. An MTSI client in terminal may be configured with a preferred mode set. Otherwise, the preferred mode-set for AMR-NB is {12.2, 7.4, 5.9, 4.75} and for AMR-WB it is {12.65, 8.85 and 6.60}.	The offer shall not be rejected purely based on the offered mode-set. If only one mode-set is offered then the MTSI client in terminal shall select to use this and include the same mode-set in the SDP answer. If several different payload types for the same codec with different mode-sets (possibly including one or more payload type without mode set) are included in the received SDP offer, then the MTSI client in terminal should select in the first hand the mode-set that provides the largest degrees of freedom for codec mode adaptation and in the second hand the mode-set that is closest to the preferred mode sets. If only a payload type without mode-set has been offered, or if an MTSI client in terminal selects a payload type without mode-set from among the offered ones, and the MTSI client in terminal intends to use only some modes (e.g. one of the preferred mode sets defined at left), then the MTSI client in terminal should include these modes as the mode-set. There are also dependencies between the mode-set and the SDP b=AS bandwidth parameter; see Clause 6.2.5.2.
mode-change-period	The MTSI client in terminal can interoperate properly with whatever mode-change-period the other end-point offers. MTSI MGWs for GERAN or UTRAN inter-working are likely to include mode-change-period=2 in the offer if in case the intention is to use TFO or TrFO.	The offer shall not be rejected purely based on the offered mode-change-period. If the received SDP offer defines mode-change-period=2 then this information shall be used to determine the mode changes for AMR-NB or AMR-WB encoded media that the MTSI client in terminal sends. The MTSI client in terminal should not include the mode-change-period parameter in the SDP answer since it has no corresponding limitations.

Parameter in the received SDP offer	Comments	Handling
mode-change-capability	The MTSI client in terminal can interoperate with whatever capabilities the other end-point declares.	The offer shall not be rejected purely based on the offered mode-change-capability. The mode-change-capability information should be used to determine a proper value, or prevent using an improper value, for mode-change-period in the SDP answer, see above. If the offer includes mode-change-capability=1, then the MTSI client in terminal shall not offer mode-change-period=2 in the answer. The MTSI client in terminal shall include mode-change-capability=2 in the SDP answer since it is required to support restricting mode changes to every other frame.
mode-change-neighbor	The MTSI client in terminal can interoperate with whatever limitations the other end-point offers.	The offer shall not be rejected purely based on the offered mode-change-neighbor. The MTSI client in terminal shall use this information to determine how mode changes can be performed for AMR-NB or AMR-WB encoded media that the MTSI client in terminal sends. The MTSI client in terminal shall not include the mode-change-neighbor parameter in the SDP answer since it has no corresponding limitations.
maxptime	The MTSI client in terminal can interoperate with whatever value that is offered. The MTSI client in terminal may also use this information to determine a suitable value for max-red in the SDP answer.	The offer shall not be rejected purely based on the offered maxptime. The MTSI client in terminal shall use this information to control the packetization when sending RTP packets to the other end-point, see also clause 7.4.2. The maxptime parameter shall be included in the SDP answer and shall be an integer multiple of 20. If the received SDP offer includes both the max-red and ptime parameter then the MTSI client in terminal may choose to use this information to define a suitable value for maxptime in the SDP answer, see NOTE 2. The MTSI client in terminal may also choose to set the maxptime value to 240, regardless of the ptime and/or max-red parameters in the SDP offer. The maxptime value in the SDP answer shall not be smaller than ptime value in the SDP answer. The maxptime value should be selected to give at least some room for adaptation.
crc	The MTSI client in terminal is not required to support this option.	The MTSI client in terminal may have to reject offered RTP payload types including this option.
robust-sorting	The MTSI client in terminal is not required to support this option.	The MTSI client in terminal may have to reject offered RTP payload types including this option.
interleaving	The MTSI client in terminal is not required to support this option.	The MTSI client in terminal may have to reject offered RTP payload types including this option.

Parameter in the received SDP offer	Comments	Handling
ptime	The MTSI client in terminal can interoperate with whatever value that is offered.	The offer shall not be rejected purely based on the offered ptime. The MTSI client in terminal should use this information and should use the requested packetization when sending RTP packets to the other end-point. The MTSI client should use the ptime value to determine how many non-redundant speech frames that can be packed into the RTP packets. The requirements in clause 7.4.2 shall be followed even if ptime in the SDP offer is larger than 80. The ptime parameter shall be included in the SDP answer and shall be an integer multiple of 20. If the received SDP offer includes the ptime parameters then the MTSI client in terminal may choose to use this information to define a suitable value for ptime in the SDP answer, see NOTE 3. The MTSI client in terminal may also choose to set the ptime value in the SDP answer according to Table 7.1, regardless of the ptime parameter in the SDP offer. The ptime value in the SDP answer shall not be larger than the maxptime value in the SDP answer.
channels	The number of channels may either be explicitly indicated in the SDP by including '/1', '/2', etc. on the a=rtpmap line, but the number of channels may also be omitted. When the number of channels is omitted then the default rule is that one channel is being offered. The MTSI client in terminal is only required to support audio media using one channel. Offered RTP payload types with more than one channel may therefore have to be rejected.	When the MTSI client in terminal accepts an offer for single-channel audio then the SDP answer shall either explicitly indicate '/1' or omit the channels parameter. When the MTSI client in terminal accepts an offer for multi-channel audio then the number of channels shall be included in the SDP answer.
max-red	The MTSI client in terminal may use this information to bound the delay for receiving redundant frames. The MTSI client in terminal may also use this information to determine a suitable value for maxptime in the SDP answer.	The max-red parameter shall be included in the SDP answer and shall be an integer multiple of 20. If the received SDP offer includes both the ptime and maxptime parameters then the MTSI client in terminal may choose to use this information to define a suitable value for max-red in the SDP answer, see NOTE 2. The MTSI client in terminal may also choose to set the max-red value to 220. The max-red value in the SDP answer should be selected to give at least some room for adaptation.
ecn-capable-rtp: leap ect=0	An MTSI client in terminal uses this SDP attribute to offer ECN for RTP-transported media	Shall be included in the SDP answer if accepting an offer to use ECN and if the session setup allows for bit-rate adaptation
NOTE 1: An MTSI client may include both a speech coded, e.g. AMR-NB or AMR-WB, and ‘telephone-events’ for DTMF in the SDP answer, see 3GPP TS 24.229 Clause 6.1, [7]. NOTE 2: It is possible to use the following relationship between maxptime, ptime and max-red: maxptime = ptime + max-red. There is however no mandatory requirement that these parameters must be aligned in this way. NOTE 3: It may be wise to use the same ptime value in the SDP answer as was given in the SDP offer, especially if the ptime in the SDP offer is larger than 20, since a value larger than the frame length indicates that the other end-point is somehow packet rate limited.		

If an SDP offer is received from another MTSI client in terminal using the AMR-NB or AMR-WB codec, then the SDP offer will include configurations as described in Table 6.1 and Table 6.2. If the MTSI client in terminal chooses to accept the offer for using the AMR-NB or AMR-WB codec, as configured in Table 6.1 or Table 6.2 then the MTSI client in terminal shall support a configuration where the MTSI client in terminal creates an SDP answer containing an RTP payload type for the AMR-NB and AMR-WB codec as shown in Table 6.4.

…

Table 6.3b: Handling of the EVS Primary SDP parameters in the received SDP offer and in the SDP answer

Parameter	Comments	Handling
br		An MTSI client in terminal supporting the EVS codec is required to support the entire bit-rate range but may offer a smaller bit-rate range or even a single bit-rate.
br-send		
br-recv		
bw	The session should start with the maximum bandwidth supported by the initial bit-rate up to the maximum negotiated bandwidth. If a range of bandwidth is negotiated, the codec can operate in any bandwidth in the session but the maximum bandwidth in the range should be used after the start of or update of the session. If a single audio bandwidth higher than narrowband is negotiated, the codec operates in the negotiated bandwidth but can use lower bandwidth(s) in the session, depending on the input signal.	Both the offerer and the answerer shall send according to the bandwidth parameter in the answer.
bw-send		
bw-recv		
ch-send		
ch-recv		
cmr	In EVS AMR-WB IO mode, CMR to the bit-rates of EVS AMR-WB IO mode and NO_REQ is always enabled.	If cmr=-1 and the session is in the EVS Primary mode, MTSI client in terminal shall not transmit CMR. If cmr=-1 and the session is in the EVS AMR-WB IO, MTSI client in terminal shall restrict CMR to values of EVS AMR-WB-IO bit-rates and NO_REQ in the session. MTSI client in terminal is required to accept CMR even when cmr=-1. MTSI client in terminal is required to accept RTP payload without CMR even when cmr=1.
ch-aw-recv		If a positive (2, 3, 5, or 7) value of ch-aw-recv is declared for a payload type and the payload type is accepted, the receiver of the parameter shall send partial redundancy (channel-aware mode) at the start of the session using the value as the offset. If ch-aw-recv=0 is declared or not present for a payload type and the payload type is accepted, the receiver of the parameter shall not send partial redundancy (channel-aware mode) at the start of the session. If ch-aw-recv=-1 is declared for a payload type and the payload type is accepted, the receiver of the parameter shall not send partial redundancy (channel-aware mode) in the session. If not present or a non-negative (0, 2, 3, 5, or 7) value of ch-aw-recv is declared for a payload type and the payload type is accepted, partial redundancy (channel-aware mode) can be activated or deactivated during the session based on the expected or estimated channel condition through adaptation signalling, such as CMR (see Annex A.2 of [125]) or RTCP based signalling (see clause 10.2). If not present or a non-negative (0, 2, 3, 5, or 7) value of ch-aw-recv is declared for a payload type and the payload type is accepted, the partial redundancy offset value can also be adjusted during the session based on the expected or estimated channel condition through adaptation signalling.

…

Table 6.4: SDP parameters for AMR-NB or AMR-WB for SDP answer when the SDP offer is received from another MTSI client in terminal

Parameter	Usage
octet-align	Shall not be included
mode-set	See Table 6.3
mode-change-period	Shall not be included
mode-change-capability	May be included. If it is included then it shall be set to 2
mode-change-neighbor	Shall not be included
maxptime	Shall be set to 240, see also Table 7.1
crc	Shall not be included
robust-sorting	Shall not be included
interleaving	Shall not be included
ptime	Shall be set according to Table 7.1
channels	Shall either be set to 1 or be omitted
max-red	Shall be included and shall be set to 220 or less
ecn-capable-rtp: leap ect=0	Shall be included in the SDP answer if accepting an offer to use ECN and if the session setup allows for bit-rate adaptation

7.18.3 Test description

7.18.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use the precondition mechanism.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.18.3.2 Test procedure sequence

Table 7.18.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
8	Step 1 of Annex A.4.1 happens	-->	INVITE	1	P
9	Step 2 of Annex A.4.1 happens	<--	100 Trying		
10	Step 3 of Annex A.4.1 happens	<--	183 Session Progress		
11	Step 4 of Annex A.4.1 happens	-->	PRACK		
12	Step 5 of Annex A.4.1 happens	<--	200 OK		
13	Step 6 of Annex A.4.1 happens	-->	UPDATE	2	P
14	Step 7 of Annex A.4.1 happens	<--	200 OK		
15	Step 8 of Annex A.4.1 happens	<--	180 Ringing		
16	Step 9 of Annex A.4.1 happens	-->	PRACK		
17	Step 10 of Annex A.4.1 happens	<--	200 OK		
18	Step 11 of Annex A.4.1 happens	<--	200 OK		
19	Step 12 of Annex A.4.1 happens	-->	ACK		

7.18.3.3 Specific message contents

183 Session Progress (Step 10)

Use the default message "183 Session Progress" in annex A.4.1 with the following exceptions:

Header/param	Value/Remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:38* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:38* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *AMR-WB/16000/1* [Note 1] *a=fmtp:* (format) *mode-change-capability=2; max-red=220* [Note 1] *a=ecn-capable-rtp: leap ect=0* [Note 4] *a=rtcp-fb:* nack ecn* [Note 4] *a=rtcp-xr:ecn-sum* [Note 4] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 5] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 5] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* *a=conf:qos remote sendrecv* Note 1: The values for fmt, payload type and format are copied from step 2. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 2. Note 4: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 5: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE.

UPDATE (Step 13)

Use the default message "UPDATE" in annex A.4.1 with the following exceptions:

Header/param	Value/Remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values shall be present. **Session description:** *v=0* *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) [Note 2] *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 3] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 3] [Note 5] *a=fmtp:* (format) [Note 3] [Note 4] **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one Note 3: The value for fmt, payload type and format is not checked Note 4: Parameters for the AMR codec are not checked Note 5: The AMR channel number shall be "/1" or omitted.

7.19 MTSI MT Voice Call / EVS / AMR-WB IO mode / 5GS

7.19.1 Test Purpose (TP)

(1)

```
with { UE having set up a voice call using EVS }
ensure that {
  when { UE receives INVITE indicating to switch to EVS AMR-WB IO mode }
    then { UE responds by accepting this switch }
            }
```

7.19.2 Conformance Requirements

[TS 26.114, clause 5.2.1.1]:

MTSI clients in terminals offering speech communication shall support narrowband, wideband and super-wideband communication. The only exception to this requirement is for the MTSI client in constrained terminal offering speech communication, in which case the MTSI client in constrained terminal shall support narrowband and wideband, and should support super-wideband communication.

In addition, MTSI clients in terminals offering speech communication shall support:

- .AMR speech codec (3GPP TS 26.071 [11], 3GPP TS 26.090 [12], 3GPP TS 26.073 [13] and 3GPP TS 26.104 [14]) including all 8 modes and source controlled rate operation 3GPP TS 26.093 [15]. The

MTSI client in terminal shall be capable of operating with any subset of these 8 codec modes. More detailed codec requirements for the AMR codec are defined in clause 5.2.1.2.

MTSI clients in terminals offering wideband speech communication at 16 kHz sampling frequency shall support:

- AMR-WB codec (3GPP TS 26.171 [17], 3GPP TS 26.190 [18], 3GPP TS 26.173 [19] and 3GPP TS 26.204 [20]) including all 9 modes and source controlled rate operation 3GPP TS 26.193 [21]. The MTSI client in terminal shall be capable of operating with any subset of these 9 codec modes. More detailed codec requirements for the AMR-WB codec are defined in clause 5.2.1.3. When the EVS codec is supported, the EVS AMR-WB IO mode may serve as an alternative implementation of AMR-WB as defined in clause 5.2.1.4.

MTSI clients in terminals offering super-wideband or fullband speech communication shall support:

- EVS codec (TS 26.441 [121], TS 26.444 [124], TS 26.445 [125], TS 26.447 [127], TS 26.451 [131], TS 26.442 [122], TS 26.452 [165] and TS 26.443 [123]) as described below including functions for backwards compatibility with AMR-WB (TS 26.446 [126]) and discontinuous transmission (TS 26.449 [129] and TS 26.450 [130]). More detailed codec requirements for the EVS codec are defined in clause 5.2.1.4.

Encoding of DTMF is described in Annex G.

[TS 26.114, clause 5.2.1.4]:

When the EVS codec is supported, the MTSI client in terminal may support dual-mono encoding and decoding.

When the EVS codec is supported, EVS AMR-WB IO may serve as an alternative implementation of the AMR-WB codec, [125]. In this case, the requirements and recommendations defined in this specification for the AMR-WB codec also apply to EVS AMR-WB IO.

NOTE: The DTX operation of EVS Primary and AMR-WB IO can be configured in sending direction with either a fixed SID update interval (from 3 to 100 frames) or an adaptive SID update interval - more details can be found in clauses 4.4.3 and 5.6.1.1 of TS 26.445 [125]. Implementers of MTSI clients are advised to take into account this SID flexibility of EVS.

[TS 26.114, clause 6.2.2.3]:

An MTSI client in terminal must understand all the payload format options that are defined in RFC 4867 [28], and in [125]. It does not have to support operating according to all these options but must be capable to properly accepting or rejecting all options.

…

Table 6.3a: Handling of SDP parameters common to EVS Primary and EVS AMR-WB IO in the received SDP offer and in the SDP answer

Parameter	Comments	Handling
ptime		
maxptime		
dtx		MTSI client in terminal shall not include dtx in the initial SDP offer. MTSI MGW may modify SDP offer to include dtx in order to disable DTX in the session.
dtx-recv		MTSI client in terminal shall not include dtx-recv. MTSI MGW may modify SDP offer or answer in order to disable DTX for the send direction of the receiver of dtx-recv.
hf-only		-
evs-mode-switch	This parameter is used by MTSI MGW either when starting in EVS AMR-WB IO mode instead of EVS Primary mode or when switching between EVS Primary mode and EVS AMR-WB IO mode, e.g., for SRVCC.	MTSI client in terminal shall not include evs-mode-switch in the initial SDP offer. When including evs-mode-switch in the SDP offer during a session, the offerer shall use the requested mode when sending EVS packets. However, if a media stream is already being received, the offerer needs to be prepared to receive packets in both EVS primary and EVS AMR-WB IO modes until receiving the answer. When including evs-mode-switch in the SDP answer during a session, the answerer shall use the requested mode when sending EVS packets. When receiving SDP answer including evs-mode-switch during a session, the offerer shall use the requested mode when sending EVS packets.
max-red	See Table 6.3	
channels	See Table 6.3	

Table 6.3b: Handling of the EVS Primary SDP parameters in the received SDP offer and in the SDP answer

Parameter	Comments	Handling
br		An MTSI client in terminal supporting the EVS codec is required to support the entire bit-rate range but may offer a smaller bit-rate range or even a single bit-rate.
br-send		
br-recv		
bw	The session should start with the maximum bandwidth supported by the initial bit-rate up to the maximum negotiated bandwidth. If a range of bandwidth is negotiated, the codec can operate in any bandwidth in the session but the maximum bandwidth in the range should be used after the start of or update of the session. If a single audio bandwidth higher than narrowband is negotiated, the codec operates in the negotiated bandwidth but can use lower bandwidth(s) in the session, depending on the input signal.	Both the offerer and the answerer shall send according to the bandwidth parameter in the answer.
bw-send		
bw-recv		
ch-send		
ch-recv		
cmr	In EVS AMR-WB IO mode, CMR to the bit-rates of EVS AMR-WB IO mode and NO_REQ is always enabled.	If cmr=-1 and the session is in the EVS Primary mode, MTSI client in terminal shall not transmit CMR. If cmr=-1 and the session is in the EVS AMR-WB IO, MTSI client in terminal shall restrict CMR to values of EVS AMR-WB-IO bit-rates and NO_REQ in the session. MTSI client in terminal is required to accept CMR even when cmr=-1. MTSI client in terminal is required to accept RTP payload without CMR even when cmr=1.
ch-aw-recv		If a positive (2, 3, 5, or 7) value of ch-aw-recv is declared for a payload type and the payload type is accepted, the receiver of the parameter shall send partial redundancy (channel-aware mode) at the start of the session using the value as the offset. If ch-aw-recv=0 is declared or not present for a payload type and the payload type is accepted, the receiver of the parameter shall not send partial redundancy (channel-aware mode) at the start of the session. If ch-aw-recv=-1 is declared for a payload type and the payload type is accepted, the receiver of the parameter shall not send partial redundancy (channel-aware mode) in the session. If not present or a non-negative (0, 2, 3, 5, or 7) value of ch-aw-recv is declared for a payload type and the payload type is accepted, partial redundancy (channel-aware mode) can be activated or deactivated during the session based on the expected or estimated channel condition through adaptation signalling, such as CMR (see Annex A.2 of [125]) or RTCP based signalling (see clause 10.2). If not present or a non-negative (0, 2, 3, 5, or 7) value of ch-aw-recv is declared for a payload type and the payload type is accepted, the partial redundancy offset value can also be adjusted during the session based on the expected or estimated channel condition through adaptation signalling.

Table 6.3c: SDP parameters for the EVS AMR-WB IO parameters in the received SDP offer and in the SDP answer

Parameter	Comments	Handling
mode-set	See Table 6.3	
mode-change-period		
mode-change-neighbor		
mode-change-capability	The default value is re-defined in comparison to that in [28].	As the default and the only allowed value of mode-change-capability is 2 in EVS AMR-WB IO, it is not required to include this parameter in the SDP offer or answer.

NOTE 2: ECN-triggered adaptation is currently undefined for EVS. This does not prevent ECN-triggered adaptation from being negotiated and used for AMR or AMR-WB.

[TS 26.114, clause 6.2.5.2]:

If an MTSI client includes an AMR or AMR-WB mode-set, or EVS Primary mode br or br-recv parameter in the SDP offer or answer, the MTSI client shall set the b=AS parameter to a value matching the maximum codec mode in the mode-set or the highest bit-rate in the br or br-recv, the packetization time (ptime), and the intended redundancy level. For example, b=AS for AMR-WB at IPv6 should be set to 38 if mode-set includes {6.60, 8.85, 12.65}, the packetization time is 20, and if no extra bandwidth is allocated for redundancy. Likewise, b=AS for EVS Primary mode at IPv4 should be set to 42 if br=7.2-24.4, the packetization is header-full payload format, ptime=20, and no extra bandwidth is allocated for redundancy.

If an MTSI client does not include an AMR or AMR-WB mode-set, or EVS Primary mode br or br-recv parameter in the SDP offer or answer, the MTSI client shall set the b=AS parameter in the SDP to a value matching the highest AMR/AMR-WB mode, i.e., AMR 12.2 and AMR-WB 23.85, or the highest bit-rate of EVS Primary mode depending on negotiated bandwidth(s), i.e., EVS 24.4 for NB and EVS 128 for WB, SWB and FB, respectively.

NOTE 1: When no mode-set is defined, then this should be understood as that the offerer or answerer is capable of sending and receiving all codec modes of AMR or AMR-WB. An MTSI client in terminal will not include the mode-set parameter in SDP offer in the initial offer-answer negotiation. See Clause 6.2.2.2, Tables 6.1 and 6.2. It is however expected that the mode-set is defined when an SDP offer is received from an MTSI MGW inter-working with CS GERAN/UTRAN, see Clause 6.2.2.3, Table 6.5.

The bandwidth to use for b=AS for AMR and AMR-WB, and EVS Primary mode should be computed as shown in Annexes K and Q respectively. Tables 6.7 and 6.8 shows the bandwidth for the respective AMR and AMR-WB codec when the packetization time is 20 and no extra bandwidth is allocated for redundancy. The b=AS value is computed without taking statistical variations, e.g., the effects of DTX, into account. Such variations can be considered in the scheduling and call admission control. Detailed procedures to compute b=AS of AMR and AMR-WB, and EVS Primary mode can be found in Annexes K and Q.

NOTE 2: For any payload format, b=AS of EVS Primary mode at 5.9 kbps source controlled variable bit-rate (SC-VBR) coding is computed as the b=AS of its highest component bit-rate, 8 kbps.

NOTE 3: b=AS of EVS AMR-WB IO mode can be computed as in the octet-aligned payload format of AMR-WB as shown in Annex K.

b=AS of EVS shall be equal to the maximum of b=AS of the highest included EVS primary mode and b=AS of the highest included EVS AMR-WB IO mode, regardless of the presence and configuration of evs-mode-switch.

Table 6.7: b=AS for each codec mode of AMR when ptime is 20

Payload format		Codec mode							
		4.75	5.15	5.9	6.7	7.4	7.95	10.2	12.2
Bandwidth-efficient	IPv4	22	22	23	24	24	25	27	29
	IPv6	30	30	31	32	32	33	35	37

Octet-aligned	IPv4	22	22	23	24	25	25	28	30
	IPv6	30	30	31	32	33	33	36	38

Table 6.8: b=AS for each codec mode of AMR-WB when ptime is 20

Payload format		**Codec Mode**								
		6.6	8.85	12.65	14.25	15.85	18.25	19.85	23.05	23.85
Bandwidth-efficient	IPv4	24	26	30	31	33	35	37	40	41
	IPv6	32	34	38	39	41	43	45	48	49
Octet-aligned	IPv4	24	26	30	32	33	36	37	40	41
	IPv6	32	34	38	40	41	44	45	48	49

Table 6.9: b=AS for each bit-rate of EVS Primary mode when ptime is 20

Payload format		**Bit-rate**										
		7.2	8	9.6	13.2	16.4	24.4	32	48	64	96	128
Header-full	IPv4	24	25	27	30	34	42	49	65	81	113	145
	IPv6	32	33	35	38	42	50	57	73	89	121	153

7.19.3 Test description

7.19.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use the precondition mechanism.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.19.3.2 Test procedure sequence

Table 7.19.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed	-	-		
8	Step 1 of Annex A.4.1 happens	-->	INVITE		
9	Step 2 of Annex A.4.1 happens	<--	100 Trying		
10	Step 3 of Annex A.4.1 happens	<--	183 Session Progress		
11	Step 4 of Annex A.4.1 happens	-->	PRACK		
12	Step 5 of Annex A.4.1 happens	<--	200 OK		
13	Step 6 of Annex A.4.1 happens	-->	UPDATE		
14	Step 7 of Annex A.4.1 happens	<--	200 OK		
15	Step 8 of Annex A.4.1 happens	<--	180 Ringing		
16	Step 9 of Annex A.4.1 happens	-->	PRACK		
17	Step 10 of Annex A.4.1 happens	<--	200 OK		
18	Step 11 of Annex A.4.1 happens	<--	200 OK		
19	Step 12 of Annex A.4.1 happens	-->	ACK		
20	Step 1 of Annex A.5.1 happens	<--	INVITE		
21	Step 2 of Annex A.5.1 happens	-->	100 Trying		
22	Step 11 of Annex A.5.1 happens	-->	200 OK	1	P
23	Step 12 of Annex A.5.1 happens	<--	ACK		

7.19.3.3 Specific message contents

INVITE (Step 20)

Use the default message "INVITE (Step 1)" in annex A.5.1 with the following exceptions:

Header/param	Value/remark
Supported	
option-tag	*precondition*
Message-body	SDP body copied from the previous 200 OK (step 12 or 14) and modified as follows: - "o=" line identical to previous SDP sent by SS except that sess-version is incremented. - "a=fmtp" line identical to previous SDP sent by SS except that "evs-mode-switch=1" is added.

200 OK (Step 22)

Use the default message "200 OK (Step 11)" in annex A.5.1 with the following exceptions:

Header/param	Value/remark
Require option-tag	*precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	 length of message-body
Message-body	The following SDP types and values. **Session description:** *v=0* *o*=(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) [Note 4] *s*=(session *name)* *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) *evs-mode-switch=1;* [Note 2, 3] Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote sendrecv* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The values for fmt, payload type and format are not checked. Note 3: The evs-mode-switch is checked, but no other codec parameters. Note 4: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one.

7.20 MTSI MO Voice Call / add video and remove video / with preconditions at both originating UE and terminating UE / 5GS

7.20.1 Test Purpose (TP)

(1)

```
with { UE having set up a voice call with preconditions }
ensure that {
  when { UE is made to add video to the voice call }
    then { UE sends re-INVITE with SDP media for both voice and video }
}
```

(2)

```
with { UE having sent re-INVITE }
ensure that {
  when { UE receives 100 Trying and 183 Session in Progress with SDP answer }
    then { UE acks with PRACK }
}
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK and indication that resources are reserved }
    then { UE sends UPDATE with SDP offer indicating reserved resources }
}
```

(4)

```
with { UE having sent UPDATE }
ensure that {
  when { UE receives 200 OK for UPDATE followed by 200 OK for re-INVITE }
    then { UE sends ACK }
}
```

(5)

```
with { UE having successfully added video }
ensure that {
  when { UE is made to remove video again }
    then { UE sends re-INVITE with SDP indicating that video is being removed from the call }
}
```

(6)

```
with { UE having sent re-INVITE }
ensure that {
  when { UE receives 100 Trying followed by 200 OK and indication that video resources have been
removed }
    then { UE sends ACK }
}
```

7.20.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.173, clause 5.2]:

The IMS multimedia telephony communication service can support different types of media, including media types listed in 3GPP TS 22.173 [2]. The session control procedures for the different media types shall be in accordance with 3GPP TS 24.229 [13] and 3GPP TS 24.247 [14], with the following additions:

a) Multimedia telephony is an IMS communication service and the P-Preferred-Service and P-Asserted-Service headers shall be treated as described in 3GPP TS 24.229 [13]. The coding of the ICSI value in the P-Preferred-Service and P-Asserted-Service headers shall be according to subclause 5.1.

[TS 24.229, clause 5.1.2A.1]:

If this is a request within an existing dialog, and the request includes a Contact header field, then the UE should insert the previously used Contact header field.

...

After the dialog is established the UE may change the dialog capabilities (e.g. add a media or request a supplementary service) if defined for the IMS communication service as identified by the ICSI value using the same dialog. Otherwise, the UE shall initiate a new initial request to the other user.

[TS 24.229, clause 5.1.3]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

The preconditions mechanism should be supported by the originating UE.

If the precondition mechanism is disabled as specified in subclause 5.1.5A, the UE shall not use the precondition mechanism.

The UE may initiate a session without the precondition mechanism if the originating UE does not require local resource reservation.

NOTE 1: The originating UE can decide if local resource reservation is required based on e.g. application requirements, current access network capabilities, local configuration, etc.

In order to allow the peer entity to reserve its required resources, if the precondition mechanism is enabled as specified in subclause 5.1.5A; the originating UE supporting the precondition mechanism should make use of the precondition mechanism, even if it does not require local resource reservation.

Upon generating an initial INVITE request using the precondition mechanism, the UE shall:

- indicate the support for reliable provisional responses and specify it using the Supported header field; and
- indicate the support for the preconditions mechanism and specify it using the Supported header field.

Upon generating an initial INVITE request using the precondition mechanism, the UE shall not indicate the requirement for the precondition mechanism by using the Require header field.

During the session initiation, if the originating UE indicated the support for the precondition mechanism in the initial INVITE request and:

a) the received response with an SDP body includes a Require header field with "precondition" option-tag, the originating UE shall include a Require header field with the "precondition" option-tag:
    - in subsequent requests that include an SDP body, that the originating UE sends in the same dialog as the response is received from; and
    - in responses with an SDP body to subsequent requests that include an SDP body and include "precondition" option-tag in Supported header field or Require header field received in-dialog; or
b) the received response with an SDP body does not include the "precondition" option-tag in the Require header field,
    - in subsequent requests that include an SDP body, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog;
    - in responses with an SDP body to subsequent requests with an SDP body but without "precondition" option-tag in the Require or Supported header field, the originating UE shall not include a Require or Supported header field with "precondition" option-tag in the same dialog; and
    - in responses with an SDP body to subsequent requests with an SDP body and with "precondition" option-tag in the Require or Supported header field, the originating UE shall include a Require header field with "precondition" option-tag in the same dialog.

NOTE 2: Table A.4 specifies that UE support of forking is required in accordance with RFC 3261 [26]. The UE can accept or reject any of the forked responses, for example, if the UE is capable of supporting a limited number of simultaneous transactions or early dialogs.

Upon successful reservation of local resources the UE shall confirm the successful resource reservation (see subclause 6.1.2) within the next SIP request.

NOTE 3: In case of the precondition mechanism being used on both sides, this confirmation will be sent in either a PRACK request or an UPDATE request. In case of the precondition mechanism not being supported on one or both sides, alternatively a reINVITE request can be used for this confirmation after a 200 (OK) response has been received for the initial INVITE request, in case the terminating UE does not support the PRACK request (as described in RFC 3262 [27]) and does not support the UPDATE request (as described in RFC 3311 [29]).

NOTE 4: The UE can receive a P-Early-Media header field authorizing an early-media flow while the required preconditions, if any, are not met and/or the flow direction is not enabled by the SDP direction parameter. According to RFC 5009 [109], an authorized early-media flow can be established only if the necessary conditions related to the SDP negotiation are met. These conditions can evolve during the session establishment.

NOTE 5: When the UE is confirming the successful resource reservation using an UPDATE request (or a PRACK request) and the UE receives a 180 (Ringing) response or a 200 (OK) response to the initial INVITE request before receiving a 200 (OK) response to the UPDATE request (or a 200 (OK) response to the PRACK request), the UE does not treat this as an error case and does not release the session.

NOTE 6: The UE procedures for rendering of the received early media and of the locally generated communication progress information are specified in 3GPP TS 24.628 [8ZF].

[TS 24.229, clause 5.1.4A.1]:

If the precondition mechanism was used during the session establishment, as described in subclause 5.1.3.1 or 5.1.4.1, the UE shall indicate support of the precondition mechanism during a session modification. If the precondition mechanism was not used during the session establishment, the UE shall not indicate support of the precondition mechanism during a session modification.

In order to indicate support of the precondition mechanism during a session modification, upon generating a reINVITE request, an UPDATE request with an SDP body, or a PRACK request with an SDP body, the UE shall:

a) indicate the support for the precondition mechanism using the Supported header field;

b) not indicate the requirement for the precondition mechanism using the Require header field; and

c) if a re-INVITE request is being generated, indicate the support for reliable provisional responses using the Supported header field

[TS 24.229, clause 6.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

In order to authorize the media streams, the P-CSCF and S-CSCF have to be able to inspect SDP message bodies. Hence, the UE shall not encrypt SDP message bodies.

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

...

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

If an in-band DTMF codec is supported by the application associated with an audio media stream, then the UE shall include, in addition to the payload type numbers associated with the audio codecs for the media stream, for each clock rate associated with the audio codecs for the media stream, a payload type number associated with the MIME subtype "telephone-event", to indicate support of in-band DTMF as described in RFC 4733 [23].

The UE shall inspect the SDP message body contained in any SIP request or response, looking for possible indications of grouping of media streams according to RFC 3524 [54] and perform the appropriate actions for IP-CAN bearer establishment for media according to IP-CAN specific procedures (see subclause B.2.2.5 for IP-CAN implemented using GPRS, subclause L.2.2.5 for IP-CAN implemented using EPS, and subclause U.2.2.5 for IP-CAN implemented using 5GS).

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

NOTE 4: Based on this resource reservation can, in certain cases, be initiated immediately after the sending or receiving of the initial SDP offer.

In order to fulfil the QoS requirements of one or more media streams, the UE may re-use previously reserved resources. In this case the UE shall indicate as met the local preconditions related to the media stream, for which resources are re-used.

[TS 24.229, clause 6.1.2]:

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

If the desired QoS resources for one or more media streams have not been reserved at the UE when constructing the SDP offer, the UE:

shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment, if the UE uses the precondition mechanism (see subclause 5.1.3.1); and

- if the UE uses the precondition mechanism (see subclause 5.1.3.1), shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 1: Previous versions of this document mandated the use of the SDP inactive attribute. This document does not prohibit specific services from using direction attributes to implement their service-specific behaviours.

If the UE uses the precondition mechanism (see subclause 5.1.3.1), and the desired QoS resources for one or more media streams are available at the UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment and shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

...

Upon confirming successful local resource reservation, the UE shall create an SDP offer in which the related local preconditions are set to met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64].

Upon receiving an SDP answer, which includes more than one codec per media stream, excluding the in-band DTMF codec, as described in subclause 6.1.1, the UE shall:

send an SDP offer at the first possible time, selecting only one codec per media stream; or

- if the UE is participant in a multi-stream multiparty multimedia conference session using simulcast (indicated by the presence of "a=simulcast" SDP attribute(s) in the SDP answer, as defined in RFC 8853 [249]), apply the procedures defined in 3GPP TS 26.114 [9B] annex S.

If the UE sends an initial INVITE request that includes only an IPv6 address in the SDP offer, and receives an error response (e.g., 488 (Not Acceptable Here) with 301 Warning header field) indicating "incompatible network address format", the UE shall send an ACK as per standard SIP procedures. Subsequently, the UE may acquire an IPv4 address or use an existing IPv4 address, and send a new initial INVITE request to the same destination containing only the IPv4 address in the SDP offer.

[TS 26.114, clause 6.2.1a.1]

MTSI clients should support SDPCapNeg to be able to negotiate RTP profiles for all media types where AVPF is supported. MTSI clients supporting SDPCapNeg shall support the complete SDPCapNeg framework.

SDPCapNeg is described in [69]. This clause only describes the SDPCapNeg attributes that are directly applicable for the RTP profile negotiation, i.e. the tcap, pcfg and acfg attributes. TS 24.229 [7] may outline further requirements needed for supporting SDPCapNeg in SDP messages.

NOTE: This clause describes only how to use the SDPCapNeg framework for RTP profile negotiation using the tcap, pcfg and acfg attributes. Implementers may therefore (incorrectly) assume that it is sufficient to implement only those specific parts of the framework that are needed for RTP profile negotiation. Doing so would however not be future proof since future versions may use other parts of the framework and there are currently no mechanisms for declaring that only a subset of the framework is supported. Hence, MTSI clients are required to support the complete framework.

[TS 26.114, clause 6.2.1a.2]

For voice and real-time text, SDPCapNeg shall be used when offering AVPF the first time for a new media type in the session since the support for AVPF in the answering client is not known at this stage. For video, an MTSI client shall either offer AVPF and AVP together using SDPCapNeg, or the MTSI client shall offer only AVPF without using SDPCapNeg. If an MTSI client has offered only AVPF for video, and then receives as response either an SDP answer where the video media component has been rejected, or an SIP 488 or 606 failure response with an SDP body indicating that only AVP is supported for video media, the MTSI client should send a new SDP offer with AVP as transport for video. Subsequent SDP offers, in a re-INVITE or UPDATE, may offer AVPF without SDPCapNeg if it is known from an earlier re-INVITE or UPDATE that the answering client supports this RTP profile. If the offer includes only AVP then SDPCapNeg does not need to be used, which can occur for: text; speech if RTCP is not used; and in re-INVITEs or UPDATEs where the RTP profile has already been negotiated for the session in a preceding INVITE or UPDATE.

When offering AVP and AVPF using SDPCapNeg, the MTSI client shall offer AVP on the media (m=) line and shall offer AVPF using SDPCapNeg mechanisms. The SDPCapNeg mechanisms are used as follows:

the support for AVPF is indicated in an attribute (a=) line using the transport capability attribute ‘tcap’. AVPF shall be preferred over AVP.

at least one configuration using AVPF shall be listed using the attribute for potential configurations ‘pcfg’.

[TS 26.114, clause 6.2.3.2]:

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

[TS 26.114, clause 6.2.5.1]:

The SDP shall include bandwidth information for each media stream and also for the session in total. The bandwidth information for each media stream and for the session is defined by the Application Specific (AS) bandwidth modifier as defined in RFC 4566 [8].

An MTSI client in terminal should include the ‘a=bw-info’ attribute in the SDP offer. When accepting a media type where the ‘a=bw-info’ attribute is included the MTSI client in terminal shall include the ‘a=bw-info’ attribute in the SDP answer if it supports the attribute. The ‘a=bw-info’ attribute and the below used bandwidth properties are defined in clause 19.

[TS 26.114, clause 6.3]:

During session renegotiation for adding or removing media components, the SDP offerer should continue to use the same media (m=) line(s) from the previously negotiated SDP for the media components that are not being added or removed.

An MTSI client in terminal may support multiple media components including media components of the same media type. An MTSI client in terminal may support adding one or more media components to an on-going session which already contains a media component of the same media type. If an MTSI client in terminal needs to have multiple media components of the same media type in a single MTSI session, then the MTSI client in terminal should use the SDP content attributes as defined in [81] for identifying different media components.

[TS 26.114, clause 7.3.1]

The RS and RR values included in the SDP answer should be treated as the negotiated values for the session and should be used to calculate the total RTCP bandwidth for all terminals in the session.

7.20.3 Test description

7.20.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use preconditions.

Preamble:

- UE has registered to IMS services and set up the MO call, by executing annex A.4.1.

7.20.3.2 Test procedure sequence

Table 7.20.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Make UE add video to the voice call.				
2	Check: Does the UE send re-INVITE with an SDP offer containing media lines for both voice and video?	->	INVITE	1	P
3	The SS responds with a 100 Trying provisional response (Step 2 of Annex A.15.1)	<-	100 Trying		
4	SS responds with an SDP answer indicating that SS has not reserved its resources for video.	<-	183 Session in Progress		
5	Check: Does the UE acknowledge the receipt of 183 response with PRACK? (Step 4 of Annex A.15.1)	->	PRACK	2	P
6	The SS responds PRACK with 200 OK. (Step 5 of Annex A.15.1)	<-	200 OK		
7	Check: Does the UE send an UPDATE after having reserved the resources for video if meeting the preconditions indicated in step 3 or 6?	->	UPDATE	3	P
8	The SS responds UPDATE with 200 OK and indicates having reserved the resources. (Step 7 of Annex A.15.1)	<-	200 OK		
9	The SS responds re-INVITE with 200 OK. (Step 11 of Annex A.15.1)	<-	200 OK		
10	Check: Does the UE acknowledge the receipt of 200 OK for re-INVITE? (Step 12 of Annex A.15.1)	->	ACK	4	P
11	Make UE release video from the media call.				
12	Check: Does the UE send re-INVITE with a SDP offer indicating that the video component is removed from the call?	->	INVITE	5	P
13	The SS responds with a 100 Trying provisional response (Step 2 of Annex A.15.1)	<-	100 Trying		
14	SS releases the QoS flow for video.				
15	The SS responds re-INVITE with 200 OK. (Step 11 of Annex A.15.1)	<-	200 OK		
16	Check: Does the UE acknowledge the receipt of 200 OK for re-INVITE? (Step 12 of Annex A.15.1)	->	ACK	6	P
17	Make the UE release the call.				
18-19	UE releases the voice call. (Steps 1-2 of Annex A.7.)				

7.20.3.3 Specific message contents

Table 7.20.3.3-1: re-INVITE (step 2, table 7.20.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.1, conditions A1, A5 and A28.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported option-tag		*precondition*		
Message-body		Same SDP body as in Step 6 (UPDATE) of Annex A.15.1, with following exceptions: Time description: *t=* (start-time) (stop-time) Attributes for media (using following attributes to replace all the original video attributes): *a=tcap:1 RTP/AVPF* [Note 1] *a=pcfg:1 t=1* [Note 1] *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1;level-id=(att-field) *a=tcap:1 RTP/AVPF* [Note 1] *a=pcfg:1 t=1* [Note 1] *a=rtpmap:* (payload type) *H264/90000* *a=fmtp:* (format) *profile-level-id=* (att-field) Note 1: The tcap/pcfg attributes are present if RTP/AVP is present on the m line.		TS 24.229 [7] TS 26.114 [33]

Table 7.20.3.3-2: 183 Session in Progress (step 4, table 7.20.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.3, Conditions A2, A6 and A12				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported option-tag		*precondition*		
Message-body		Same SDP body as in Step 3 (183 Session in Progress) of Annex A.15.1, with following exceptions: Session description: "o=" line identical to previous SDP sent by SS except that sess-version is incremented by one		TS 24.229 [7] TS 26.114 [33]

Table 7.20.3.3-3: UPDATE (step 7, table 7.20.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, conditions A1 and A6.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported option-tag		*Precondition*		
Message-body		Same contents as specified in step 5 (PRACK).		TS 24.229 [7] TS 26.114 [33]

Table 7.20.3.3-4: re-INVITE (step 12, table 7.20.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.1, conditions A1, A5 and A28.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported option-tag		*precondition*		
Message-body		Same SDP body as in Step 2 (re-INVITE), with following exceptions: Deleting following video media description and attributes: **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1;level-id=(att-field) *a=sendrecv* **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv*		TS 24.229 [7] TS 26.114 [33]

7.21 MTSI MO Voice Call / add video and remove video / without preconditions at both originating UE and terminating UE / 5GS

7.21.1 Test Purpose (TP)

(1)

```
with { UE having set up a voice call without preconditions }
ensure that {
  when { UE receives re-INVITE indicating addition of video to voice call }
    then { UE may send 100 Trying and sends 183 Session Progress with SDP answer }
}
```

(2)

```
with { UE having sent 183 Session Progress }
ensure that {
  when { UE receives PRACK for 183 Session Progress }
    then { UE sends 200 OK for PRACK followed by 200 OK for re-INVITE }
}
```

(3)

```
with { UE receiving ACK }
ensure that {
  when { UE is made to remove video again }
    then { UE sends re-INVITE with SDP indicating that video is being removed from the call }
}
```

(4)

```
with { UE having sent re-INVITE }
ensure that {
```

```
  when { UE receives 100 Trying followed by 200 OK and indication that video resources have been
removed }
    then { UE sends ACK }
}
```

7.21.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.173, clause 5.2]:

The IMS multimedia telephony communication service can support different types of media, including media types listed in 3GPP TS 22.173 [2]. The session control procedures for the different media types shall be in accordance with 3GPP TS 24.229 [13] and 3GPP TS 24.247 [14], with the following additions:

a) Multimedia telephony is an IMS communication service and the P-Preferred-Service and P-Asserted-Service headers shall be treated as described in 3GPP TS 24.229 [13]. The coding of the ICSI value in the P-Preferred-Service and P-Asserted-Service headers shall be according to subclause 5.1.

[TS 24.229, clause 5.1.2A.2]:

After the dialog is established the UE may change the dialog capabilities (e.g. add a media or request a supplementary service) if defined for the IMS communication service as identified by the ICSI value using the same dialog. Otherwise, the UE shall initiate a new initial request to the other user.

[TS 24.229, clause 5.1.4A.1]:

If the precondition mechanism was used during the session establishment, as described in subclause 5.1.3.1 or 5.1.4.1, the UE shall indicate support of the precondition mechanism during a session modification. If the precondition mechanism was not used during the session establishment, the UE shall not indicate support of the precondition mechanism during a session modification.

In order to indicate support of the precondition mechanism during a session modification, upon generating a reINVITE request, an UPDATE request with an SDP body, or a PRACK request with an SDP body, the UE shall:

a) indicate the support for the precondition mechanism using the Supported header field;

b) not indicate the requirement for the precondition mechanism using the Require header field; and

c) if a re-INVITE request is being generated, indicate the support for reliable provisional responses using the Supported header field

and follow the SDP procedures in clause 6 for the precondition mechanism.

[TS 24.229, clause 6.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

In order to authorize the media streams, the P-CSCF and S-CSCF have to be able to inspect SDP message bodies. Hence, the UE shall not encrypt SDP message bodies.

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

...

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b="

lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

If an in-band DTMF codec is supported by the application associated with an audio media stream, then the UE shall include, in addition to the payload type numbers associated with the audio codecs for the media stream, for each clock rate associated with the audio codecs for the media stream, a payload type number associated with the MIME subtype "telephone-event", to indicate support of in-band DTMF as described in RFC 4733 [23].

The UE shall inspect the SDP message body contained in any SIP request or response, looking for possible indications of grouping of media streams according to RFC 3524 [54] and perform the appropriate actions for IP-CAN bearer establishment for media according to IP-CAN specific procedures (see subclause B.2.2.5 for IP-CAN implemented using GPRS, subclause L.2.2.5 for IP-CAN implemented using EPS, and subclause U.2.2.5 for IP-CAN implemented using 5GS).

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

NOTE 4: Based on this resource reservation can, in certain cases, be initiated immediately after the sending or receiving of the initial SDP offer.

In order to fulfil the QoS requirements of one or more media streams, the UE may re-use previously reserved resources. In this case the UE shall indicate as met the local preconditions related to the media stream, for which resources are re-used.

[TS 24.229, clause 6.1.2]:

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

If the desired QoS resources for one or more media streams have not been reserved at the UE when constructing the SDP offer, the UE:

shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment, if the UE uses the precondition mechanism (see subclause 5.1.3.1); and

- if the UE uses the precondition mechanism (see subclause 5.1.3.1), shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 1: Previous versions of this document mandated the use of the SDP inactive attribute. This document does not prohibit specific services from using direction attributes to implement their service-specific behaviours.

If the UE uses the precondition mechanism (see subclause 5.1.3.1), and the desired QoS resources for one or more media streams are available at the UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment and shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

...

Upon confirming successful local resource reservation, the UE shall create an SDP offer in which the related local preconditions are set to met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64].

Upon receiving an SDP answer, which includes more than one codec per media stream, excluding the in-band DTMF codec, as described in subclause 6.1.1, the UE shall:

send an SDP offer at the first possible time, selecting only one codec per media stream; or

- if the UE is participant in a multi-stream multiparty multimedia conference session using simulcast (indicated by the presence of "a=simulcast" SDP attribute(s) in the SDP answer, as defined in RFC 8853 [249]), apply the procedures defined in 3GPP TS 26.114 [9B] annex S.

If the UE sends an initial INVITE request that includes only an IPv6 address in the SDP offer, and receives an error response (e.g., 488 (Not Acceptable Here) with 301 Warning header field) indicating "incompatible network address format", the UE shall send an ACK as per standard SIP procedures. Subsequently, the UE may acquire an IPv4 address or use an existing IPv4 address, and send a new initial INVITE request to the same destination containing only the IPv4 address in the SDP offer.

[TS 26.114, clause 6.2.1a.1]

MTSI clients should support SDPCapNeg to be able to negotiate RTP profiles for all media types where AVPF is supported. MTSI clients supporting SDPCapNeg shall support the complete SDPCapNeg framework.

SDPCapNeg is described in [69]. This clause only describes the SDPCapNeg attributes that are directly applicable for the RTP profile negotiation, i.e. the tcap, pcfg and acfg attributes. TS 24.229 [7] may outline further requirements needed for supporting SDPCapNeg in SDP messages.

NOTE: This clause describes only how to use the SDPCapNeg framework for RTP profile negotiation using the tcap, pcfg and acfg attributes. Implementers may therefore (incorrectly) assume that it is sufficient to implement only those specific parts of the framework that are needed for RTP profile negotiation. Doing so would however not be future proof since future versions may use other parts of the framework and there are currently no mechanisms for declaring that only a subset of the framework is supported. Hence, MTSI clients are required to support the complete framework.

[TS 26.114, clause 6.2.1a.2]

For voice and real-time text, SDPCapNeg shall be used when offering AVPF the first time for a new media type in the session since the support for AVPF in the answering client is not known at this stage. For video, an MTSI client shall either offer AVPF and AVP together using SDPCapNeg, or the MTSI client shall offer only AVPF without using SDPCapNeg. If an MTSI client has offered only AVPF for video, and then receives as response either an SDP answer where the video media component has been rejected, or an SIP 488 or 606 failure response with an SDP body indicating that only AVP is supported for video media, the MTSI client should send a new SDP offer with AVP as transport for video. Subsequent SDP offers, in a re-INVITE or UPDATE, may offer AVPF without SDPCapNeg if it is known from an earlier re-INVITE or UPDATE that the answering client supports this RTP profile. If the offer includes only AVP then SDPCapNeg does not need to be used, which can occur for: text; speech if RTCP is not used; and in re-INVITEs or UPDATEs where the RTP profile has already been negotiated for the session in a preceding INVITE or UPDATE.

When offering AVP and AVPF using SDPCapNeg, the MTSI client shall offer AVP on the media (m=) line and shall offer AVPF using SDPCapNeg mechanisms. The SDPCapNeg mechanisms are used as follows:

the support for AVPF is indicated in an attribute (a=) line using the transport capability attribute ‘tcap’. AVPF shall be preferred over AVP.

at least one configuration using AVPF shall be listed using the attribute for potential configurations ‘pcfg’.

[TS 26.114, clause 6.2.3.2]:

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

[TS 26.114, clause 6.2.5.1]:

The SDP shall include bandwidth information for each media stream and also for the session in total. The bandwidth information for each media stream and for the session is defined by the Application Specific (AS) bandwidth modifier as defined in RFC 4566 [8].

An MTSI client in terminal should include the ‘a=bw-info’ attribute in the SDP offer. When accepting a media type where the ‘a=bw-info’ attribute is included the MTSI client in terminal shall include the ‘a=bw-info’ attribute in the SDP answer if it supports the attribute. The ‘a=bw-info’ attribute and the below used bandwidth properties are defined in clause 19.

[TS 26.114, clause 6.3]:

During session renegotiation for adding or removing media components, the SDP offerer should continue to use the same media (m=) line(s) from the previously negotiated SDP for the media components that are not being added or removed.

An MTSI client in terminal may support multiple media components including media components of the same media type. An MTSI client in terminal may support adding one or more media components to an on-going session which already contains a media component of the same media type. If an MTSI client in terminal needs to have multiple media components of the same media type in a single MTSI session, then the MTSI client in terminal should use the SDP content attributes as defined in [81] for identifying different media components.

[TS 26.114, clause 7.3.1]

The RS and RR values included in the SDP answer should be treated as the negotiated values for the session and should be used to calculate the total RTCP bandwidth for all terminals in the session.

7.21.3 Test description

7.21.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to not use preconditions.

Preamble:

- UE registered to IMS services and set up the MO call, by executing annex A.4.1.

7.21.3.2 Test procedure sequence

Table 7.21.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	SS sends re-INVITE with an SDP offer containing media lines for both voice and video	<-	INVITE		
2	Optional: UE may respond with a 100 Trying provisional response (Step 2 of Annex A.16.2)	->	100 Trying		
3	Check: Does the UE respond 183 Session in Progress with an SDP answer? (Step 3 of Annex A.16.2)	->	183 Session in Progress	1	P
4	SS acknowledges the receipt of 183 response with PRACK and optionally offers second SDP to indicate the changed precondition status. (Step 4 of Annex A.16.2)	<-	PRACK		
5	Check: Does the UE respond PRACK with 200 OK and answers the second SDP (if any) with mirroring its contents? (Step 5 of Annex A.16.2)	->	200 OK	2	P
6	UE responds re-INVITE with 200 OK (Step 9 of Annex A.16.2)	->	200 OK		
7	SS acknowledges the receipt of 200 OK for re-INVITE (Step 10 of Annex A.16.2)	<-	ACK		
8	Make UE release video from the media call	-			
9	Check: Does the UE send re-INVITE with a SDP offer indicating that the video component is removed from the call?	->	INVITE	3	P
10	The SS responds with a 100 Trying provisional response (Step 2 of Annex A.15.1)	<-	100 Trying		
11	SS releases the QoS flow for video	-	-		
12	The SS responds re-INVITE with 200 OK (Step 11 of Annex A.15.1)	<-	200 OK		
13	Check: Does the UE acknowledge the receipt of 200 OK for re-INVITE? (Step 12 of Annex A.15.1)	->	ACK	4	P
14	Make the UE release the call.	-	-		
15-16	UE releases the voice call. (Steps 1-2 of annex A.7)	-	-		

7.21.3.3 Specific message contents

Table 7.21.3.3-1: re-INVITE (step 1, table 7.21.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.1, conditions A1, A5 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported option-tag		*precondition*		
Content-Type media-type		*application/sdp*		
Content-Length value		*length of message-body*		
Message-body		Same SDP body as in Step 1 (INVITE) of Annex A.16.2, with following exceptions: Session description: *o*=line identical to previous SDP sent by SS except that sess-version is incremented by one Time description: *t*= (start-time) (stop-time) Media description (added following line): *c=IN* (addrtype) (connection-address for SS) [Note 1] Note 1: At least one "c=" field shall be present.		TS 24.229 [7] TS 26.114 [33]

Table 7.21.3.3-2: re-INVITE (step 9, table 7.21.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.1, conditions A1, A5 and A28.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Same SDP body as in Step 2 (re-INVITE), with following exceptions: Deleting following video media description and attributes: **Media description:** *m=video (transport port) RTP/AVPF* (fmt) *a=acfg:1 t=1* *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1; level-id=93; \		TS 24.229 [7] TS 26.114 [33]

7.22 MTSI MT Voice Call / add video and remove video / with preconditions at both originating UE and terminating UE / 5GS

7.22.1 Test Purpose (TP)

(1)

```
with { UE being configured to use preconditions and having been invited to voice call and voice call
being set up successfully }
ensure that {
  when { UE is made to add video to the voice call }
    then { UE sends re-INVITE with SDP media for both voice and video }
            }
```

(2)

```
with { UE having sent re-INVITE }
ensure that {
  when { UE receives 100 Trying followed by 183 Session Progress }
    then { UE sends PRACK }
            }
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK and resources being reserved }
    then { UE sends UPDATE with SDP offer indicating reserved resources }
            }
```

(4)

```
with { UE having sent UPDATE }
ensure that {
  when { UE receives 200 OK for UPDATE followed by 200 OK for re-INVITE }
    then { UE sends ACK }
            }
```

(5)

```
with { UE having sent ACK }
ensure that {
  when { UE receives re-INVITE indicating removal of video }
    then { UE may send 100 Trying and sends 200 OK for re-INVITE }
            }
```

7.22.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.2A.1.1]

After the dialog is established the UE may change the dialog capabilities (e.g. add a media or request a supplementary service) if defined for the IMS communication service as identified by the ICSI value using the same dialog. Otherwise, the UE shall initiate a new initial request to the other user.

[TS 24.229, clause 5.1.3.1]

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

The preconditions mechanism should be supported by the originating UE.

…

Upon successful reservation of local resources the UE shall confirm the successful resource reservation (see subclause 6.1.2) within the next SIP request.

NOTE 3: In case of the precondition mechanism being used on both sides, this confirmation will be sent in either a PRACK request or an UPDATE request. In case of the precondition mechanism not being supported on one or both sides, alternatively a reINVITE request can be used for this confirmation after a 200 (OK) response has been received for the initial INVITE request, in case the terminating UE does not support the PRACK request (as described in RFC 3262 [27]) and does not support the UPDATE request (as described in RFC 3311 [29]).

[TS 24.229, clause 5.1.4A.1]

In order to indicate support of the precondition mechanism during a session modification, upon generating a reINVITE request, an UPDATE request with an SDP body, or a PRACK request with an SDP body, the UE shall:

a) indicate the support for the precondition mechanism using the Supported header field;

b) not indicate the requirement for the precondition mechanism using the Require header field; and

c) if a re-INVITE request is being generated, indicate the support for reliable provisional responses using the Supported header field

[TS 24.229, clause 5.1.4A.2]

Upon receiving a reINVITE request, an UPDATE request, or a PRACK request that indicates support for the precondition mechanism by using the Supported header field or requires use of the precondition mechanism by using the Require header field, the UE shall:

a) if the precondition mechanism was used during the session establishment, as described in subclause 5.1.3.1 or 5.1.4.1, use the precondition mechanism for the session modification;

…

If the precondition mechanism is used for the session modification, the UE shall indicate support for the preconditions mechanism, using the Require header field, in responses that include an SDP body, to the session modification request.

[TS 24.229, clause 6.1.1]

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

…

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

…

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

If an in-band DTMF codec is supported by the application associated with an audio media stream, then the UE shall include, in addition to the payload type numbers associated with the audio codecs for the media stream, for each clock rate associated with the audio codecs for the media stream, a payload type number associated with the MIME subtype "telephone-event", to indicate support of in-band DTMF as described in RFC 4733[23].

…

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

[TS 24.229, clause 6.1.4.2]

If the precondition mechanism is used for the session modification, the following applies:

a) if the session modification does not increase the QoS requirement of the already established media stream (e.g., all the media streams in a call hold procedure, audio stream in a call upgrade procedure), in the SDP body of the request (re-INVITE, UPDATE, or PRACK), both local and remote QoS of this media shall be indicated as met; and

b) if the session modification increases the QoS requirement of some already established media stream(s) (e.g., request of using a different audio/video codec that requires higher bandwidth), or if the session modification adds a new media stream (e.g., call upgrade), the setting of the current and desired QoS status of the modified or added media stream shall be the same as specified in subclause 6.1.2. If the network fails to modify or reserve the required resources, the UE shall send a CANCEL request to terminate the session modification.

[TS 26.114, clause 5.2.1.1]

MTSI clients in terminals offering super-wideband or fullband speech communication shall support:

- EVS codec (TS 26.441 [121], TS 26.444 [124], TS 26.445 [125], TS 26.447 [127], TS 26.451 [131], TS 26.442 [122], TS 26.452 [165], and TS 26.443 [123]) as described below including functions for backwards compatibility with AMR-WB (TS 26.446 [126]) and discontinuous transmission (TS 26.449 [129] and TS 26.450 [130]).

[TS 26.114, clause 5.2.2]

MTSI clients in terminals offering video communication shall support:

- H.264 (AVC) [24] Constrained Baseline Profile (CBP) Level 1.2;

- H.265 (HEVC) [119] Main Profile, Main Tier, Level 3.1. The only exception to this requirement is for the MTSI client in constrained terminal offering video communication, in which case the MTSI client in constrained terminal should support H.265 (HEVC) Main Profile, Main Tier, Level 3.1.

In addition they should support:- H.264 (AVC) [24] Constrained High Profile (CHP) Level 3.1.

For backwards compatibility to previous releases, if H.264 (AVC) [24] Constrained High Profile Level 3.1 is supported, then H.264 (AVC) [24] Constrained Baseline Profile (CBP) Level 3.1 should also be offered.

[TS 26.114, clause 6.2.1a.1]

MTSI clients should support SDPCapNeg to be able to negotiate RTP profiles for all media types where AVPF is supported. MTSI clients supporting SDPCapNeg shall support the complete SDPCapNeg framework.

SDPCapNeg is described in [69]. This clause only describes the SDPCapNeg attributes that are directly applicable for the RTP profile negotiation, i.e. the tcap, pcfg and acfg attributes. TS 24.229 [7] may outline further requirements needed for supporting SDPCapNeg in SDP messages.

NOTE: This clause describes only how to use the SDPCapNeg framework for RTP profile negotiation using the tcap, pcfg and acfg attributes. Implementers may therefore (incorrectly) assume that it is sufficient to implement only those specific parts of the framework that are needed for RTP profile negotiation. Doing so would however not be future proof since future versions may use other parts of the framework and there are currently no mechanisms for declaring that only a subset of the framework is supported. Hence, MTSI clients are required to support the complete framework.

[TS 26.114, clause 6.2.1a.2]

For voice and real-time text, SDPCapNeg shall be used when offering AVPF the first time for a new media type in the session since the support for AVPF in the answering client is not known at this stage. For video, an MTSI client shall either offer AVPF and AVP together using SDPCapNeg, or the MTSI client shall offer only AVPF without using SDPCapNeg. If an MTSI client has offered only AVPF for video, and then receives as response either an SDP answer where the video media component has been rejected, or an SIP 488 or 606 failure response with an SDP body indicating that only AVP is supported for video media, the MTSI client should send a new SDP offer with AVP as transport for video. Subsequent SDP offers, in a re-INVITE or UPDATE, may offer AVPF without SDPCapNeg if it is known from an earlier re-INVITE or UPDATE that the answering client supports this RTP profile. If the offer includes only AVP then SDPCapNeg does not need to be used, which can occur for: text; speech if RTCP is not used; and in re-INVITEs or UPDATEs where the RTP profile has already been negotiated for the session in a preceding INVITE or UPDATE.

When offering AVP and AVPF using SDPCapNeg, the MTSI client shall offer AVP on the media (m=) line and shall offer AVPF using SDPCapNeg mechanisms. The SDPCapNeg mechanisms are used as follows:

- The support for AVPF is indicated in an attribute (a=) line using the transport capability attribute ‘tcap’. AVPF shall be preferred over AVP.

- At least one configuration using AVPF shall be listed using the attribute for potential configurations ‘pcfg’.

[TS 26.114, clause 6.2.3.2]

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in IETF RFC 6236 [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

[TS 26.114, clause 6.2.5.1]

The SDP shall include bandwidth information for each media stream and also for the session in total. The bandwidth information for each media stream and for the session is defined by the Application Specific (AS) bandwidth modifier as defined in RFC 4566 [8].

[TS 26.114, clause 6.3]

Addition and removal of media components shall be performed based on the SDP-based offer-answer model as specified in RFC 3264 [58].

During session renegotiation for adding or removing media components, the SDP offeror should continue to use the same media (m=) line(s) from the previously negotiated SDP for the media components that are not being added or removed.

[TS 26.114, clause 7.3.1]

The bandwidth for RTCP traffic shall be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by RFC 3556 [42]. Therefore, an MTSI client shall include the "b=RS:" and "b=RR:" fields in SDP, and shall be able to interpret them.

7.22.3 Test description

7.22.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use preconditions

Preamble:

- The UE has registered to IMS and MT voice call is set up by executing the generic test procedure in Table 4.9.16.2.2-1 of TS 38.508-1 [21] up to the last step

7.22.3.2 Test procedure sequence

Table 7.22.3.2-1: Main Behaviour

Derivation Path: TS 34.229-1 [2], Annex A.2.1, Conditions A1, A3, A5, A28, A30 and A31				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported Option-tag		*precondition*		
Message-body		SDP body of INVITE copied from Annex A.15.1 Step 6 and modified as follows: -Audio media attribute for preconditions: *a=curr:qos remote sendrecv* SDP values for video media as mentioned in A.15.1 Step 1 and modified as follows: -Video media attribute for preconditions *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv*		TS 24.229 [7] TS 26.114 [33]

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
-	UE is triggered to add video to the ongoing voice call	-	-	-	-
1	UE sends re-INVITE with an SDP offer containing media lines for both voice and video	-->	INVITE	1	P
2	SS responds with 100 Trying	<--	100 Trying	-	-
3	SS responds with 183 session Progress including SDP answer	<--	183 Session Progress	-	-
4	UE acknowledges the receipt of 183 response with PRACK	-->	PRACK	2	P
5	SS responds to PRACK with 200 OK	<--	200 OK	-	-
6	UE sends UPDATE after reserving the resources for video	-->	UPDATE	3	P
7	SS responds UPDATE with 200 OK indicating reservation of resources	<--	200 OK	-	-
8	SS responds to re-INVITE with 200 OK	<--	200 OK	-	-
9	The UE acknowledges the receipt of 200 OK for re-INVITE	-->	ACK	4	P
10	SS sends re-INVITE with SDP offer indicating that video component is removed from the call.	<--	INVITE	-	-
11	Optional: UE responds with 100 Trying.	-->	100 Trying	-	-
12	UE responds to re-invite with 200 OK final response.	-->	200 OK	5	P
13	SS deactivates the bearer for video.	-	-	-	-
14	The SS acknowledges the receipt of 200 OK for re-INVITE.	<--	ACK	-	-

7.22.3.3 Specific message contents

Table 7.22.3.3-1: INVITE (step 1, table 7.22.3.2-1)

Table 7.22.3.3-2: 100 Trying (step 2, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.2, Condition A1

Table 7.22.3.3-3: 183 Session Progress (step 3, table 7.22.3.2-1)

Table 7.22.3.3-4: PRACK (step 4, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.4, Conditions A1 and A7

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A10 and A22

Table 7.22.3.3-5: 200 OK for PRACK (step 5, table 7.22.3.2-1)

Table 7.22.3.3-6: UPDATE (step 6, table 7.22.3.2-1)

Derivation Path: Annex A.15.1 Step 6 with following exceptions				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported Option-tag		*precondition*		
Message-body		-Attribute for preconditions for audio media: *a=curr:qos remote sendrecv*		TS 24.229 [7]

Derivation Path: TS 34.229-1 [2], Annex A.2.3, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported Option-tag		*precondition*		
Message-body		SDP body of 183 Session Progress copied from Annex A.15.1 Step 7 and modified as follows: - Video media attribute for preconditions *a=curr:qos local none* and *a=curr:qos remote none* - Video media attribute "*a=acfg:1 t=1"* present if tcap/pcfg attributes were included in INVITE		TS 24.229 [7] TS 26.114 [33]

Table 7.22.3.3-7: 200 OK for UPDATE (step 7, table 7.22.3.2-1)

Derivation Path: Annex A.15.1 Step 7

Table 7.22.3.3-8: 200 OK for re-INVITE (step 8, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A1, A10, A19 and A22

Table 7.22.3.3-9: ACK for re-INVITE (step 9, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.7, Conditions A1, A3 and A5

Table 7.22.3.3-10: INVITE (step 10, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.9, Conditions A1, A3 and A5				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported Option-tag		*precondition*		
Message-body		SDP body of INVITE copied from Annex A.15.1 Step 7 and modified as follows: -transport port on "m=" line for video media is set to 0.		TS 24.229 [7] TS 26.114 [33]

Table 7.22.3.3-11: 100 Trying (step 11, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.2, Condition A2

Table 7.22.3.3-12: 200 OK for INVITE (step 12, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A2, A5, A11, A20 and A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported Option-tag		*precondition*		
Content-Type media-type		*application/sdp*		
Content-Length value		header shall be present if UE uses TCP to send this message and if there is a message body *length of message-body*		
Message-body		SDP body not checked		

Table 7.22.3.3-13: ACK (step 14, table 7.22.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.7, Conditions A2, A3 and A5

7.23 MTSI MT Voice Call / add video and remove video / without preconditions at both originating UE and terminating UE / 5GS

7.23.1 Test Purpose (TP)

(1)

```
with { Ue configured to not use preconditions and having been invited to a voice call and voice call
being set up successfully }
ensure that {
  when { UE is made to add video to the voice call }
    then { UE sends re-INVITE with SDP media for both voice and video }
}
```

(2)

```
with { UE having sent re-INVITE }
ensure that {
  when { UE receives 100 Trying followed by 183 Session Progress }
    then { UE sends PRACK }
}
```

(3)

```
with { UE having sent PRACK }
ensure that {
  when { UE receives 200 OK for PRACK followed by 200 OK for INVITE and resources are reserved }
    then { UE sends ACK }
}
```

(4)

```
with { UE having sent ACK }
ensure that {
  when { UE receives re-INVITE indicating that video is being removed from the call }
    then { UE may send 100 Trying and sends 200 OK for re-INVITE }
}
```

7.23.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.173, clause 5.2]:

The IMS multimedia telephony communication service can support different types of media, including media types listed in 3GPP TS 22.173 [2]. The session control procedures for the different media types shall be in accordance with 3GPP TS 24.229 [13] and 3GPP TS 24.247 [14], with the following additions:

a) Multimedia telephony is an IMS communication service and the P-Preferred-Service and P-Asserted-Service headers shall be treated as described in 3GPP TS 24.229 [13]. The coding of the ICSI value in the P-Preferred-Service and P-Asserted-Service headers shall be according to subclause 5.1.

[TS 24.229, clause 5.1.2A.2]:

After the dialog is established the UE may change the dialog capabilities (e.g. add a media or request a supplementary service) if defined for the IMS communication service as identified by the ICSI value using the same dialog. Otherwise, the UE shall initiate a new initial request to the other user.

[TS 24.229, clause 5.1.4A.1]:

If the precondition mechanism was used during the session establishment, as described in subclause 5.1.3.1 or 5.1.4.1, the UE shall indicate support of the precondition mechanism during a session modification. If the precondition mechanism was not used during the session establishment, the UE shall not indicate support of the precondition mechanism during a session modification.

In order to indicate support of the precondition mechanism during a session modification, upon generating a reINVITE request, an UPDATE request with an SDP body, or a PRACK request with an SDP body, the UE shall:

a) indicate the support for the precondition mechanism using the Supported header field;

b) not indicate the requirement for the precondition mechanism using the Require header field; and

c) if a re-INVITE request is being generated, indicate the support for reliable provisional responses using the Supported header field

and follow the SDP procedures in clause 6 for the precondition mechanism.

[TS 24.229, clause 6.1]:

The "integration of resource management and SIP" extension is hereafter in this subclause referred to as "the precondition mechanism" and is defined in RFC 3312 [30] as updated by RFC 4032 [64].

In order to authorize the media streams, the P-CSCF and S-CSCF have to be able to inspect SDP message bodies. Hence, the UE shall not encrypt SDP message bodies.

During the session establishment procedure, and during session modification procedures, SIP messages shall only contain an SDP message body if that is intended to modify the session description, or when the SDP message body is included in the message because of SIP rules described in RFC 3261 [26].

...

For "video" and "audio" media types that use the RTP/RTCP and where the port number is not zero, the UE shall specify the proposed bandwidth for each media stream using the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

...

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

If an in-band DTMF codec is supported by the application associated with an audio media stream, then the UE shall include, in addition to the payload type numbers associated with the audio codecs for the media stream, for each clock rate associated with the audio codecs for the media stream, a payload type number associated with the MIME subtype "telephone-event", to indicate support of in-band DTMF as described in RFC 4733 [23].

The UE shall inspect the SDP message body contained in any SIP request or response, looking for possible indications of grouping of media streams according to RFC 3524 [54] and perform the appropriate actions for IP-CAN bearer establishment for media according to IP-CAN specific procedures (see subclause B.2.2.5 for IP-CAN implemented using GPRS, subclause L.2.2.5 for IP-CAN implemented using EPS, and subclause U.2.2.5 for IP-CAN implemented using 5GS).

In case of UE initiated resource reservation and if the UE determines resource reservation is needed, the UE shall start reserving its local resources whenever it has sufficient information about the media streams, media authorization and used codecs available.

NOTE 4: Based on this resource reservation can, in certain cases, be initiated immediately after the sending or receiving of the initial SDP offer.

In order to fulfil the QoS requirements of one or more media streams, the UE may re-use previously reserved resources. In this case the UE shall indicate as met the local preconditions related to the media stream, for which resources are re-used.

[TS 24.229, clause 6.1.2]:

An INVITE request generated by a UE shall contain a SDP offer and at least one media description. This SDP offer shall reflect the calling user's terminal capabilities and user preferences for the session.

If the desired QoS resources for one or more media streams have not been reserved at the UE when constructing the SDP offer, the UE:

shall indicate the related local preconditions for QoS as not met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment, if the UE uses the precondition mechanism (see subclause 5.1.3.1); and

- if the UE uses the precondition mechanism (see subclause 5.1.3.1), shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 1: Previous versions of this document mandated the use of the SDP inactive attribute. This document does not prohibit specific services from using direction attributes to implement their service-specific behaviours.

If the UE uses the precondition mechanism (see subclause 5.1.3.1), and the desired QoS resources for one or more media streams are available at the UE when the SDP offer is sent, the UE shall indicate the related local preconditions as met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64], as well as the strength-tag value "mandatory" for the local segment and the strength-tag value either "optional" or as specified in RFC 3312 [30] and RFC 4032 [64] for the remote segment and shall not request confirmation for the result of the resource reservation (as defined in RFC 3312 [30]) at the terminating UE.

NOTE 2: If the originating UE does not use the precondition mechanism (see subclause 5.1.3.1), it will not include any precondition information in the SDP message body.

...

Upon confirming successful local resource reservation, the UE shall create an SDP offer in which the related local preconditions are set to met, using the segmented status type, as defined in RFC 3312 [30] and RFC 4032 [64].

Upon receiving an SDP answer, which includes more than one codec per media stream, excluding the in-band DTMF codec, as described in subclause 6.1.1, the UE shall:

send an SDP offer at the first possible time, selecting only one codec per media stream; or

- if the UE is participant in a multi-stream multiparty multimedia conference session using simulcast (indicated by the presence of "a=simulcast" SDP attribute(s) in the SDP answer, as defined in RFC 8853 [249]), apply the procedures defined in 3GPP TS 26.114 [9B] annex S.

If the UE sends an initial INVITE request that includes only an IPv6 address in the SDP offer, and receives an error response (e.g., 488 (Not Acceptable Here) with 301 Warning header field) indicating "incompatible network address format", the UE shall send an ACK as per standard SIP procedures. Subsequently, the UE may acquire an IPv4 address or use an existing IPv4 address, and send a new initial INVITE request to the same destination containing only the IPv4 address in the SDP offer.

[TS 26.114, clause 6.2.1a.1]

MTSI clients should support SDPCapNeg to be able to negotiate RTP profiles for all media types where AVPF is supported. MTSI clients supporting SDPCapNeg shall support the complete SDPCapNeg framework.

SDPCapNeg is described in [69]. This clause only describes the SDPCapNeg attributes that are directly applicable for the RTP profile negotiation, i.e. the tcap, pcfg and acfg attributes. TS 24.229 [7] may outline further requirements needed for supporting SDPCapNeg in SDP messages.

NOTE: This clause describes only how to use the SDPCapNeg framework for RTP profile negotiation using the tcap, pcfg and acfg attributes. Implementers may therefore (incorrectly) assume that it is sufficient to implement only those specific parts of the framework that are needed for RTP profile negotiation. Doing so would however not be future proof since future versions may use other parts of the framework and there are currently no mechanisms for declaring that only a subset of the framework is supported. Hence, MTSI clients are required to support the complete framework.

[TS 26.114, clause 6.2.1a.2]

For voice and real-time text, SDPCapNeg shall be used when offering AVPF the first time for a new media type in the session since the support for AVPF in the answering client is not known at this stage. For video, an MTSI client shall either offer AVPF and AVP together using SDPCapNeg, or the MTSI client shall offer only AVPF without using SDPCapNeg. If an MTSI client has offered only AVPF for video, and then receives as response either an SDP answer where the video media component has been rejected, or an SIP 488 or 606 failure response with an SDP body indicating that only AVP is supported for video media, the MTSI client should send a new SDP offer with AVP as transport for video. Subsequent SDP offers, in a re-INVITE or UPDATE, may offer AVPF without SDPCapNeg if it is known from an earlier re-INVITE or UPDATE that the answering client supports this RTP profile. If the offer includes only AVP then SDPCapNeg does not need to be used, which can occur for: text; speech if RTCP is not used; and in re-INVITEs or UPDATEs where the RTP profile has already been negotiated for the session in a preceding INVITE or UPDATE.

When offering AVP and AVPF using SDPCapNeg, the MTSI client shall offer AVP on the media (m=) line and shall offer AVPF using SDPCapNeg mechanisms. The SDPCapNeg mechanisms are used as follows:

the support for AVPF is indicated in an attribute (a=) line using the transport capability attribute ‘tcap’. AVPF shall be preferred over AVP.

at least one configuration using AVPF shall be listed using the attribute for potential configurations ‘pcfg’.

[TS 26.114, clause 6.2.3.2]:

If video is used in a session, the session setup shall determine the applicable bandwidth(s) as defined in clause 6.2.5, RTP profile, video codec, profile and level. The "imageattr" attribute as specified in [76] should be supported. The "framesize" attribute as specified in [60] shall not be used in the session setup.

An MTSI client shall offer AVPF for all media streams containing video. RTP profile negotiation shall be done as described in clause 6.2.1a.

[TS 26.114, clause 6.2.5.1]:

The SDP shall include bandwidth information for each media stream and also for the session in total. The bandwidth information for each media stream and for the session is defined by the Application Specific (AS) bandwidth modifier as defined in RFC 4566 [8].

An MTSI client in terminal should include the ‘a=bw-info’ attribute in the SDP offer. When accepting a media type where the ‘a=bw-info’ attribute is included the MTSI client in terminal shall include the ‘a=bw-info’ attribute in the SDP answer if it supports the attribute. The ‘a=bw-info’ attribute and the below used bandwidth properties are defined in clause 19.

[TS 26.114, clause 6.3]:

During session renegotiation for adding or removing media components, the SDP offerer should continue to use the same media (m=) line(s) from the previously negotiated SDP for the media components that are not being added or removed.

An MTSI client in terminal may support multiple media components including media components of the same media type. An MTSI client in terminal may support adding one or more media components to an on-going session which already contains a media component of the same media type. If an MTSI client in terminal needs to have multiple media components of the same media type in a single MTSI session, then the MTSI client in terminal should use the SDP content attributes as defined in [81] for identifying different media components.

[TS 26.114, clause 7.3.1]

The RS and RR values included in the SDP answer should be treated as the negotiated values for the session and should be used to calculate the total RTCP bandwidth for all terminals in the session.

7.23.3 Test description

7.23.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to not use preconditions.

Preamble:

- UE has registered to IMS services and set up the MT call, by executing annex A.4.1.

7.23.3.2 Test procedure sequence

Table 7.23.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Make UE add video to the voice call.				
2	Check: Does the UE send re-INVITE with an SDP offer containing media lines for both voice and video?	->	INVITE	1	P
3	SS responds with a 100 Trying provisional response (Step 2 of Annex A.15.2)	<-	100 Trying		
4	SS responds with an SDP answer indicating that SS has not reserved its resources for video. (Step 3 of Annex A.15.2)	<-	183 Session in Progress		
5	Check: Does the UE acknowledge the receipt of 183 response with PRACK? (Step 4 of Annex A.15.2)	->	PRACK	2	P
6	SS responds PRACK with 200 OK (Step 5 of Annex A.15.2)	<-	200 OK		
7	SS responds re-INVITE with 200 OK (Step 9 of Annex A.15.2)	<-	200 OK		
8	Check: Does the UE acknowledge the receipt of 200 OK for re-INVITE? (Step 10 of Annex A.15.2)	->	ACK	3	P
9	SS sends re-INVITE with a SDP offer indicating that the video component is removed from the call	<-	INVITE		
10	Check: Does the UE optionally respond with a 100 Trying provisional response? (Step 2 of Annex A.16.2)	->	100 Trying	4	P
11	UE releases the QoS flow for video				
12	UE responds re-INVITE with 200 OK (Step 9 of Annex A.16.2)	->	200 OK		
13	SS acknowledges the receipt of 200 OK for re-INVITE (Step 10 of Annex A.16.2)	<-	ACK		
14	Make the UE release the call.				
15-16	UE releases the voice call. (Steps 1-2 of annex A.7)				

7.23.3.3 Specific message contents

Table 7.23.3.3-1: re-INVITE (step 2, table 7.23.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.1, conditions A1, A5 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Same SDP body as in Step 1 (INVITE) of Annex A.15.2, with following exceptions: Session description: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one		TS 24.229 [7] TS 26.114 [33]

Table 7.23.3.3-2: re-INVITE (step 9, table 7.23.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.1, conditions A1 and A5				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Same SDP body as in Step 2 (re-INVITE), with following exceptions: Deleting the following video media description and attributes: **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) or RTP/AVP (fmt) [Note 11] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=tcap:1 RTP/AVPF* [Note 11] *a=pcfg:1 t=1* [Note 11] *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1;level-id=(att-field) *a=tcap:1 RTP/AVPF* [Note 11] *a=pcfg:1 t=1* [Note 11] *a=rtpmap:* (payload type) *H264/90000* *a=fmtp:* (format) *profile-level-id=* (att-field)		TS 24.229 [7] TS 26.114 [33]

7.24 Forking / UE receives two responses and one CANCEL request / 5GS

7.24.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call with preconditions }
}
```

(2)

```
with { UE having sent INVITE }
ensure that {
  when { UE receives two 100 Trying and two 183 Session Progress responses }
    then { UE sends two PRACK requests for the two 183 Session Progress responses }
}
```

(3)

```
with { UE having sent two PRACK requests }
ensure that {
  when { UE receives CANCEL for the first forked early dialog and 200 OK for PRACK for the second
early dialog }
    then { UE completes setup of voice call on the second early dialog }
}
```

7.24.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229, clause 5.1.4.1]:

If the UE sends a CANCEL request to cancel an initial INVITE request, the UE shall when applicable include in the CANCEL request a Reason header field with a protocol value set to "RELEASE_CAUSE" and a "cause" header field parameter as specified in subclause 7.2A.18.11.2. The UE may also include the "text" header field parameter with reason-text as specified in subclause 7.2A.18.11.2.

7.24.3 Profile Requirements (Informative)

[GSMA NG.114 V1.0, clause 2.3.7]:

The IMS core network can support sending and the UE must support receiving a SIP CANCEL request including a Reason header field with values of:

1. SIP; cause=200; text="Call completed elsewhere"
2. SIP; cause=603; text="Declined"
3. SIP; cause=600; text=" Busy Everywhere"

for forked calls as defined in 3GPP TS 24.229 [8].

7.24.4 Test description

7.24.4.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A and registered to IMS

7.24.4.2 Test procedure sequence

Table 7.24.4.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Make UE initiate a voice call.				
2-9	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
10	Check: Does the UE send INVITE with an SDP offer containing preconditions? (Step 1 of A.4.1)	->	INVITE	1	P
11	SS responds with a 100 Trying in forked early dialog 1 (Step 2 of A.4.1)	<-	100 Trying		
12	SS responds with a 100 Trying in forked early dialog 2 (Step 2 of A.4.1)	<-	100 Trying		
13	SS responds with a 183 Session in Progress in forked dialog 1 (Step 3 of A.4.1)	<-	183 Session in Progress		
14	Check: Does the UE acknowledge the receipt of 183 response with PRACK in dialog 1? (Step 4 of A.4.1)	->	PRACK	2	P
15	SS responds with a 183 Session in Progress in forked dialog 2 (Step 3 of A.4.1)	<-	183 Session in Progress		
16	Check: Does the UE acknowledge the receipt of 183 response with PRACK in dialog 2? (Step 4 of A.4.1)	->	PRACK	2	P
17	SS sends CANCEL for the forked early dialog 1	<-	CANCEL		
18	UE sends 200 OK for the CANCEL	->	200 OK		
19	SS sends 200 OK for the forked early dialog 2 (Step 5 of A.4.1)	<-	200 OK		
	Note: resource reservation will be triggered.				
20	Check: Does the UE send an UPDATE after having reserved the resources for the preconditions of the dialog 2? (Step 6 of A.4.1)	->	UPDATE	3	P
21	The SS responds UPDATE with 200 OK and indicates having reserved the resources (Step 7 of A.4.1)	<-	200 OK		
22	SS sends 180 Ringing reliably (Step 8 of A.4.1)	<-	180 Ringing		
23	Check: Does the UE acknowledge reception of 180 Ringing? (Step 9 of A.4.1)	->	PRACK	3	P
24	SS responds to PRACK. (Step 10 of A.4.1)	<-	200 OK		
25	The SS responds INVITE with 200 OK (Step 11 of A.4.1)	<-	200 OK		
26	Check: Does the UE acknowledge the receipt of 200 OK for INVITE? (Step 12 of A.4.1)	->	ACK	3	P
27	Make the UE release the call.				
28-29	Steps 1-2 of annex A.7				

7.24.4.3 Specific message contents

Table 7.24.4.3-1: 100 Trying (step 12, table 7.24.4.2-1)

Derivation Path: Step 2 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		Any value different from what is used in step 11		

Table 7.24.4.3-2: 183 Session Progress (step 13, table 7.24.4.2-1)

Derivation Path: Step 3 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 11		

Table 7.24.4.3-4: PRACK (step 14, table 7.24.4.2-1)

Derivation Path: Step 4 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 11		

Table 7.24.4.3-3: 183 Session Progress (step 15, table 7.24.4.2-1)

Derivation Path: Step 3 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-5: PRACK (step 16, table 7.24.4.2-1)

Derivation Path: Step 4 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-6: CANCEL (step 17, table 7.24.4.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.15				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 11		
Reason		*SIP; cause=603; text="Declined"*		RFC 3326 [xx]

Table 7.24.4.3-7: 200 OK for CANCEL (step 18, table 7.24.4.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, conditions A5 and A11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 11		

Table 7.24.4.3-8: UPDATE (step 20, table 7.24.4.2-1)

Derivation Path: Step 6 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-9: 200 OK for UPDATE (step 21, table 7.24.4.2-1)

Derivation Path: Step 7 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-10: 180 Ringing (step 22, table 7.24.4.2-1)

Derivation Path: Step 8 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-11: PRACK (step 23, table 7.24.4.2-1)

Derivation Path: Step 9 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-12: 200 OK for PRACK (step 24, table 7.24.4.2-1)

Derivation Path: Step 10 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-13: 200 OK for INVITE (step 25, table 7.24.4.2-1)

Derivation Path: Step 11 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

Table 7.24.4.3-14: ACK (step 26, table 7.24.4.2-1)

Derivation Path: Step 12 of A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		The same value with what is used in step 12		

7.25 MTSI MT Voice Call without SDP offer in INVITE / 5GS

7.25.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions }
ensure that {
  when { UE receives INVITE for voice call without SDP offer }
   then { UE responds with 183 Session Progress including SDP offer and completes call initiation }
            }
```

7.25.2 Conformance Requirements

[TS 24.229, Rel-15, clause 6.1.3]

NOTE 2: Upon receiving an initial INVITE request that does not include an SDP offer, the UE can accept the request and include an SDP offer in the first reliable response. The SDP offer will reflect the called user's terminal capabilities and user preferences for the session.

7.25.2A Profile Requirements (Informative)

[GSMA NG.114 V1.0]

The UE must be able to accept a SIP INVITE request without a Session Description Protocol (SDP) offer, and the UE must then include an SDP offer in the first non-failure reliable response to a SIP INVITE request without SDP offer. The SDP offer must contain all codecs (for audio only or for both audio and video) that the UE is currently able and willing to use.
Note 1: Other media than audio can be included in the SDP offer in the first non-failure reliable response.

[GSMA NG.114 V1.0 cl 3.2.2.3]

The UE that sends the SDP offer for voice media must include in this SDP offer at least one EVS payload type with one of the following EVS configurations:

1. EVS Configuration A1: br=5.9-13.2; bw=nb-swb.
2. EVS Configuration A2: br=5.9-24.4; bw=nb-swb.
3. EVS Configuration B0: br=13.2; bw=swb.
4. EVS Configuration B1: br=9.6-13.2; bw=swb.
5. EVS Configuration B2: br=9.6-24.4; bw=swb.

Editor's Note: expand further on NG114 requirements?

7.25.3 Test description

7.25.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.
- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1[21]) and registered to IMS.

7.25.3.2 Test procedure sequence

Table 7.25.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
2	SS sends INVITE.	<--	INVITE	-	-
3	UE may send 100 Trying.	-->	Optional step: 100 Trying	-	-
4	Check: Does the UE send 183 Session Progress reliably and containing an SDP offer?	-->	183 Session Progress	1	P
5	SS sends PRACK containing an SDP answer.	<--	PRACK	-	-
6	UE sends 200 OK response for PRACK.	-->	200 OK	-	-
7	SS sends UPDATE containing an SDP offer.	<--	UPDATE		
8	UE sends 200 OK response for UPDATE, containing an SDP answer.	-->	200 OK		
9	UE sends 180 Ringing response.	-->	180 Ringing	-	-
10	If UE sent 180 Ringing response reliably, the SS sends PRACK.	<--	Conditional step: PRACK	-	-
11	If UE sent 180 Ringing reliably, UE responds to PRACK by sending 200 OK.	-->	Conditional step: 200 OK	-	-
12	Make the UE accept the voice call	-	-	-	-
13	UE sends 200 OK for INVITE.	-->	200 OK	1	P
14	SS sends ACK.	<--	ACK	-	-

7.25.3.3 Specific message contents

Table 7.25.3.3-1: INVITE (step 2, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.9, Conditions A1, A3, and A4				
Content-Type		not present		
Message-body		not present		

Table 7.25.3.3-2: 100 Trying (step 3, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.2, Condition A2

Table 7.25.3.3-3: 183 Session Progress with an SDP offer (step 4, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.3, condition A2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		NOTE: the following SDP offer is identical to the SDP offer shown in Annex A.4.1, Step 1. **Session description:** *v=0* *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS*: (bandwidth-value) **Time description:** *t*= (start-time) (stop-time) **Media description:**		TS 24.229 [7]

		m=audio (transport port) *RTP/AVP* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) [Note 2] *b=RR:* (bandwidth-value) [Note 2] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-24.4; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-24.4; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/16000* *a=fmtp:* (format) *a=rtpmap:* (payload type) *AMR/8000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/8000* *a=fmtp:* (format) *a=ecn-capable-rtp: leap ect=0* [Note 7] *a=rtcp-fb:* nack ecn* [Note 7] *a=rtcp-xr:ecn-sum* [Note 7] *a=rtcp-rsize* [Note 7] *a=ptime:20* *a=maxptime:240*		

		Attributes for media security mechanism: *a=3ge2ae: requested* [Note 8] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:WVNf X19zZW1jdGwgKCkgewkyMjA7fQp9CnVubGVz|2^20|1:4FEC_ORDER=FEC_SRTP"* [Note 8] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The RR value shall be greater than 0. The RS value can be any value. Note 3: The channel number shall be "/1" or omitted. Note 4: The max-red values from 0 to 220 are allowed. Note 5: The parameters dtx, dtx-recv and evs-mode-switch shall not be present. Note 6: The parameters mode-set, mode-change-period, mode-change-neighbor, crc, robust-sorting and interleaving shall not be included. Note 7: Attributes for ECN Capability may be present if the UE supports Explicit Congestion Notification.		

		Note 8: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 9: The ordering of payload types shall be as listed, i.e., EVS before AMR-WB before AMR. Note 10: The EVS payload type shall carry at least one of the five EVS configurations		

Table 7.25.3.3-4: PRACK with an SDP answer (step 5, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.4, condition A3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		NOTE: the following SDP offer is identical to the SDP offer shown in Annex A.4.1, Step 3. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 7] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* *a=conf:qos remote sendrecv*		TS 24.229 [7]

		Note 1: The values for fmt, payload type and format are copied from step 4. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 4. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 4. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 4. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in previous SDP offer. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in previous SDP offer.		

Table 7.25.3.3-5: 200 OK (step 6, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A5, A8, A11, and A22

Table 7.25.3.3-6: UPDATE with an SDP offer (step 7, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.4, condition A3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		**Session description:** *v=0* *o=- 1111111111 1111111112 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 7] Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv*		TS 24.229 [7]
		Note 1: The values for fmt, payload type and format are copied from step 4. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 4. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 4. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 4. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in previous SDP offer. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in previous SDP offer.		

Table 7.25.3.3-7: 200 OK with an SDP answer (step 8, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.3.1, conditions A2, A11, and A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require				
option-tag		*precondition*		
Content-Type				
		application/sdp		
Content-Length		header shall be present if UE uses TCP to send this message and if there is a message body		
value		length of message-body		
Message-body		The following SDP types and values shall be present. Session description: *v=0* *o*=(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) [Note 4] *s*=(session *name)* *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) Attributes for media: *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) [Note 2, 3] Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote sendrecv* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked Note 3: Parameters for the codec are not checked Note 4: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one.		TS 24.229 [7]

Table 7.25.3.3-8: 180 Ringing (step 9, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.6, Conditions A2 and A14

Table 7.25.3.3-9: PRACK (step 10, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.4, Condition A3

Table 7.25.3.3-10: 200 OK (step 11, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A5, A8, A11, and A22

Table 7.25.3.3-11: 200 OK (step 12, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A5, A8, A11, and A22

Table 7.25.3.3-12: ACK (step 13, table 7.25.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.6, Conditions A2 and A3

7.26 Mobile Originating CAT / Forking Model / 5GS

7.26.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and configured to use preconditions and having initiated an MO
voice call with preconditions up to the last step before 180 Ringing }
ensure that {
  when { UE receives 183 Session Progress on a forked dialog indicating Customized Alerting Tones }
    then { UE moves forked dialog forward until up to the last step before 180 Ringing }
            }
```

(2)

```
with { UE having moved both dialogs forward up to the last step before 180 Ringing }
ensure that {
  when { UE receives 200 OK for INVITE for the first dialog }
    then { UE acks reception of 200 OK for INVITE }
            }
```

7.26.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.182, clause 4.5.5.1.1]:

The UE shall follow the procedures specified in 3GPP TS 24.229 [4] for session initiation and termination.

[TS 24.628, clause 4.7.2.1]:

Procedures according to 3GPP TS 24.229 [1] shall apply.

Certain services require the usage of the Alert-Info header field, Call-Info header field and Error-Info header field according to procedures specified by IETF RFC 3261 [4].

If the UE detects that in-band information is received from the network as early media, the in-band information received from the network shall override locally generated communication progress information.

NOTE 1: In-band information received from the network overrides any locally generated communication progress information also when the most recently received P-Early-Media header fields of all early dialogs contain "inactive" or "recvonly".

NOTE 2: When multiple early dialogs exist with authorization as "sendrecv" or "sendonly", the mechanism used by the UE to associate the received early media with the correct early dialog is unspecified in this version of this specification.

The UE shall not generate the locally generated communication progress information if an early dialog exists where the last received P-Early-Media header field as described in IETF RFC 5009 [12] contains "sendrecv" or "sendonly".

If an early dialog exists where a SIP 18x response to the SIP INVITE request other than 183 (Session Progress) response was received, no early dialog exists where the last received P-Early-Media header field as described in IETF RFC 5009 [12] contained "sendrecv" or "sendonly" and in-band information is not received from the network, then the UE is expected to render the locally generated communication progress information.

NOTE 3: According to 3GPP TS 22.173 [23] the UE for an MMTel session generates the communication progress information specified in clause F.2 of 3GPP TS 22.001 [24], with parameters applicable for the home network of the UE.

If the UE supports the P-Early-Media header field as defined in IETF RFC 5009 [12], and at least one P-Early-Media header field has been received on at least one early dialog, then the UE shall send any available user generated media, e.g. speech or DTMF, on media stream(s) associated with the early dialog for which the most recent P-Early-Media header field, as described in IETF RFC 5009 [12], contained a "sendrecv" header field value. If there is more than one such early dialog, the UE shall use the early dialog where the P-Early-Media header field was most recently received.

If the UE receives a re-INVITE request containing no SDP offer, the UE shall send a 200 (OK) response containing an SDP offer according to 3GPP TS 24.229 [1] indicating the directionality used by UE as

- "sendonly" if the re-INVITE request is received on a dialog where the associated communication session has been put on hold by the user or has been put on hold by both users at both ends; and
- "sendrecv" otherwise.

7.26.3 Test description

7.26.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use preconditions.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.26.3.2 Test procedure sequence

Table 7.26.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to initiate a voice call.	-	-	-	-
2-8	Steps 1-7 of generic procedures of MO voice call with preconditions defined in A.4.1.	-	Setup dialog 1	-	-
9	SS sends an SDP answer. (Step 3 of A.4.1)	<--	183 Progress (dialog 2)	-	-
10	Check: Does the UE acknowledge reception of 183 Session Progress? (Step 4 of A.4.1)	-->	PRACK (dialog 2)	1	P
11	SS responds to PRACK. (Step 5 of A.4.1)	<--	200 OK (dialog 2)	-	-
12-13	Void	-		-	-
14	The SS sends 200 OK for INVITE sent in step 1 above	<--	200 OK	-	-
15	Check: Does the UE send the ACK to the 200 OK for the INVITE in step 1?	-->	ACK	2	P
16	The UE is made to release the call	-	-	-	-
17	The UE releases the call with BYE	-->	BYE	-	-
18	The SS sends 200 OK for BYE	<--	200 OK	-	-

7.26.3.3 Specific message contents

Table 7.26.3.3-1: 183 Session Progress with an SDP offer (step 9, table 7.26.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.3, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		 any value different from what is used in steps 1-5		
Contact addr-spec		 *<sip:cat-as.home1.net;+g.3gpp.icsi ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel">*		
P-Early-Media em-param		 *sendonly*		
Require option-tag		 *precondition*		TS 24.229 [7]
Message-body		**Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS for early-media) *s=-* *c=IN* (addrtype) (connection-address for SS for early-media) *b=AS:37* **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* *a=conf:qos remote sendrecv* **Other attributes:** *a=content:g.3gpp.cat*		TS 24.229 [7]

Table 7.26.3.3-2: PRACK (step 10, table 7.26.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.2.4, Conditions A1 and A7				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require option-tag		 *precondition*		
Message-body		Contents is copied from step 6 of annex A.4.1 with the following exceptions: **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote sendrecv* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv*		TS 24.229 [7]

Table 7.26.3.3-3: 200 OK (step 11, table 7.26.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.3.1, Conditions A10 and A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
To tag		 Same value as used in step 9		
Require option-tag		 *precondition*		
Content-Type media-type		 *application/sdp*		
Content-Length value		 length of message-body		
Message-body		SDP body of the 200 OK response copied from the received PRACK and modified as follows: - IP address on "c=" lines and transport port on "m=" lines changed to indicate to which IP address and port the UE should start sending the media (same as used in step 9 above); - "o=" line identical to previous SDP sent by SS except that sess-version is incremented.		TS 24.229 [7]

7.27 Session Timer / MO Call / UE is able to refresh the session / 5GS

7.27.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions and to
be the refresher }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call }
            }
```

(2)

```
with { UE having included Session-Expires in INVITE }
ensure that {
  when { UE receives 100 Trying followed by 422 Session Interval Too Small with Min-SE value of 1860
}
    then { UE sends ACK and new INVITE with Min-SE value and Session-Expires value being 1860 }
            }
```

(3)

```
with { UE having send 2nd INVITE }
ensure that {
  when { UE receives 100 Trying followed by 422 Session Interval Too Small with Min-SE value of 1920
}
    then { UE sends ACK and new INVITE with Min-SE value and Session-Expires value being 1920 }
            }
```

(4)

```
with { UE having sent 3rd INVITE }
ensure that {
  when { UE receiving 100 Trying followed by 183 Session Progress }
    then { UE concludes voice call set up procedure up until sending ACK, with Session-Expires
having value 1920 and refresher being set to uac }
            }
```

(5)

```
with { UE having been chosen as refresher for established voice call }
ensure that {
  when { voice call has been going on for 960 seconds }
    then { UE sends UPDATE to refresh the session }
            }
```

(6)

```
with { UE having been chosen as refresher for established voice call }
ensure that {
  when { voice call has been going on for another 960 seconds }
    then { UE sends UPDATE to refresh the session }
            }
```

7.27.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.27.3 Test description

7.27.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.27.3.2 Test procedure sequence

Table 7.27.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
8	UE sends INVITE with either the Session-Expires value set to 1800 or no Session-Expires header.	-->	INVITE	1	P
-	EXCEPTION: Steps 9a0 to 9a7 describe behaviour that depends on UE capability: the "lower case letter" identifies a step sequence that takes place if the UE included Session-Expires in step 8	-	-		
9a0	SS sends a 100 Trying response. (Step 2 of Annex A.4.1)	<--	100 Trying		
9a1	SS sends 422 Session Interval Too Small response with Min-SE value of 1860.	<--	422 Session Interval Too Small		
9a2	UE sends ACK.	-->	ACK	2	P
9a3	UE sends INVITE with Min-SE value and Session-Expires value being 1860.	-->	INVITE	2	P
9a4	SS sends a 100 Trying response. (Step 2 of Annex A.4.1)	<--	100 Trying		
9a5	SS sends 422 Session Interval Too Small response with Min-SE value of 1920.	<--	422 Session Interval Too Small		
9a6	UE sends ACK.	-->	ACK	3	P
9a7	UE sends INVITE with Min-SE value and Session-Expires value being 1920.	-->	INVITE	3	P
10-18	Steps 2-10 of Annex A.4.1 happen.	-	-		
19	SS sends 200 OK for INVITE with negotiated Session-Expires value set to 1920 and refresher value set to uac.	<--	200 OK	4	P
20	UE sends ACK.	-->	ACK	4	P
21	960 seconds after step 20, UE sends an UPDATE request to refresh the session.	-->	UPDATE	5	P
22	SS sends 200 OK for UPDATE.	<--	200 OK		
23	960 seconds after step 22, UE sends an UPDATE request to refresh the session.	-->	UPDATE	6	P
24	SS sends 200 OK for UPDATE.	<--	200 OK		
25-26	SS releases the call. (Steps 1-2 of Annex A.8)	-	-		

7.27.3.3 Specific message contents

Table 7.27.3.3-1: INVITE (step 8, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A4, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds refresher		(if present) 1800 uac (if present)		RFC 4028 [37]

Table 7.27.3.3-2: 422 Session Interval Too Small (step 9a1, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.24				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Min-SE delta-seconds		 1860		RFC 4028 [37]

Table 7.27.3.3-3: INVITE (step 9a3, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A4, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Call-ID callid		 The same value as in INVITE in Step 8		RFC 4028 [37]
From addr-spec tag		 The same value as in INVITE in Step 8 The same value as in INVITE in Step 8		RFC 4028 [37]
To addr-spec		 The same value as in INVITE in Step 8		RFC 4028 [37]
CSeq value		 The value sent in the INVITE in step 8, incremented by one		RFC 4028 [37]
Session-Expires delta-seconds		 1860		RFC 4028 [37]
Min-SE delta-seconds		 1860		RFC 4028 [37]

Table 7.27.3.3-4: 422 Session Interval Too Small (step 9a5, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.24				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Min-SE delta-seconds		 1920		RFC 4028 [37]

Table 7.27.3.3-5: INVITE (step 9a7, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A4, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Call-ID callid		 The same value as in INVITE in Step 9a3		RFC 4028 [37]
From addr-spec tag		 The same value as in INVITE in Step 9a3 The same value as in INVITE in Step 9a3		RFC 4028 [37]
To addr-spec		 The same value as in INVITE in Step 9a3		RFC 4028 [37]
CSeq value		 The value sent in the INVITE in step 9a3, incremented by one		RFC 4028 [37]
Session-Expires delta-seconds		 1920		RFC 4028 [37]
Min-SE delta-seconds		 1920		RFC 4028 [37]

Table 7.27.3.3-6: 183 Session Progress (step 11, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.3, Conditions A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Allow		INVITE, UPDATE, PRACK, ACK, OPTIONS, CANCEL, BYE		RFC 4028 [37]

Table 7.27.3.3-7: 200 OK (step 19, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Allow		INVITE, UPDATE, PRACK, ACK, OPTIONS, CANCEL, BYE		RFC 4028 [37]
Require		timer		RFC 4028 [37]
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1920 uac		RFC 4028 [37]
Min-SE delta-seconds		 1920		RFC 4028 [37]

Table 7.27.3.3-8: UPDATE (steps 21 and 23, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, Conditions A1 and A6				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1920 uac		RFC 4028 [37]
Min-SE delta-seconds		 1920		RFC 4028 [37]
Content-Type		any value if present		RFC 4028 [37]

Table 7.27.3.3-9: 200 OK (steps 22 and 24, table 7.27.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1920 uac		RFC 4028 [37]
Min-SE delta-seconds		 1920		RFC 4028 [37]

7.28 Session Timer / MO Call / Remote end is refresher / 5GS

7.28.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions and to
not be the refresher }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call without refresher parameter }
            }
```

(2)

```
with { UE having sent INVITE }
ensure that {
  when { UE continues setup of voice call and finally receives 200 OK for INVITE setting refresher
to uas }
    then { UE sends ACK }
            }
```

(3)

```
with { UE having completed call setup }
ensure that {
  when { UE receives refresh request via an UPDATE request }
    then { UE sends 200 OK for UPDATE }
            }
```

(4)

```
with { UE having sent 200 OK for a refresh request }
ensure that {
  when { Session expires }
    then { UE releases the call }
            }
```

7.28.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.28.3 Test description

7.28.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.28.3.2 Test procedure sequence

Table 7.28.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
8	UE sends INVITE with either the Session-Expires value set to 1800 or no Session-Expires header.	-->	INVITE	1	P
9-17	Steps 2-10 of Annex A.4.1 happen.	-	-		
18	SS sends 200 OK for INVITE with Session-Expires value set to 1800 and refresher value set to uas.	<--	200 OK		
19	UE sends ACK.	-->	ACK	2	P
20	900 seconds after step 18, SS sends an UPDATE request to refresh the session.	<--	UPDATE		
21	UE sends 200 OK for UPDATE.	-->	200 OK	3	P
	UE sends BYE to release the call due to session expiry 1800 seconds after step 21. (Step 1 of Annex A.7)	-->	BYE	4	P
23	SS sends 200 OK for BYE. (Step 2 of Annex A.7)	<--	200 OK		

7.28.3.3 Specific message contents

Table 7.28.3.3-1: INVITE (step 8, table 7.28.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A4, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds refresher		(if present) 1800 not present		RFC 4028 [37]

Table 7.28.3.3-2: 200 OK (step 18, table 7.28.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require		timer		RFC 4028 [37]
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uas		RFC 4028 [37]

Table 7.28.3.3-3: UPDATE (step 20, table 7.28.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, Condition A3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]
Content-Type		not present		RFC 4028 [37]

Table 7.28.3.3-4: 200 OK (step 21, table 7.28.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]

7.29 Session Timer / MO Call / Remote end does not support Session Timer / 5GS

7.29.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions and to
be the refresher }
ensure that {
  when { UE is being made to initiate the voice call }
    then { UE sends INVITE for voice call }
            }
```

(2)

```
with { UE having sent INVITE and continuing with the call setup }
ensure that {
  when { UE receives 200 OK for INVITE without timer tag and Session-Expires }
    then { UE sends ACK }
            }
```

(3)

```
with { UE having sent ACK }
ensure that {
  when { 900 seconds have passed }
    then { UE sends UPDATE to refresh the session }
            }
```

(4)

```
with { UE having sent received 200 OK for UPDATE }
ensure that {
  when { Another 900 seconds have passed }
    then { UE sends UPDATE to refresh the session }
            }
```

7.29.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.29.3 Test description

7.29.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.29.3.2 Test procedure sequence

Table 7.29.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
8	UE sends INVITE indicating support for Session Timer, with either the Session-Expires value set to 1800 or no Session-Expires header.	-->	INVITE	1	P
9-17	Steps 2-10 of Annex A.4.1 happen.	-	-		
18	SS sends 200 OK for INVITE, without timer tag in Supported and Require headers and without Session-Expires header.	<--	200 OK		
19	UE sends ACK.	-->	ACK	2	P
20	900 seconds after step 19, UE sends an UPDATE request to refresh the session.	-->	UPDATE	3	P
21	SS sends 200 OK for UPDATE, without timer tag in Supported and Require headers and without Session-Expires header.	<--	200 OK		
22	900 seconds after step 21, UE sends an UPDATE request to refresh the session.	-->	UPDATE	4	P
23	SS sends 200 OK for UPDATE, without timer tag in Supported and Require headers and without Session-Expires header.	<--	200 OK		
24-25	SS releases the call. (Steps 1-2 of Annex A.8)	-	-		

7.29.3.3 Specific message contents

Table 7.29.3.3-1: INVITE (step 8, table 7.29.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A4, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds refresher		(if present) 1800 uac (if present)		RFC 4028 [37]

Table 7.29.3.3-2: 183 Session Progress (step 10, table 7.29.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.3, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Allow		INVITE, UPDATE, PRACK, ACK, OPTIONS, CANCEL, BYE		RFC 4028 [37]

Table 7.29.3.3-3: 200 OK (step 18, table 7.29.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Allow		INVITE, UPDATE, PRACK, ACK, OPTIONS, CANCEL, BYE		RFC 4028 [37]

Table 7.29.3.3-4: UPDATE (steps 20 and 22, table 7.29.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, Conditions A1 and A6				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]
Content-Type		any value if present		RFC 4028 [37]

7.30 Session Timer / MO Call / Remote end supports but does not use Session Timer / 5GS

7.30.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions and to
be the refresher }
ensure that {
  when { UE is being made to initiate a voice call }
    then { UE sends INVITE for voice call }
            }
```

(2)

```
with { UE having sent INVITE for voice call and continuing the call setup }
ensure that {
  when { UE receives 200 OK for INVITE with timer tag and without Session-Expires }
    then { UE sends ACK }
            }
```

(3)

```
with { UE having sent ACK }
ensure that {
  when { 1860 seconds passed without the UE refreshing the session }
    then { UE receives BYE and sends 200 OK for BYE }
            }
```

7.30.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.30.3 Test description

7.30.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.30.3.2 Test procedure sequence

Table 7.30.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt an IMS voice call.	-	-		
2-7	Steps 2-7 of generic procedure specified in Table 4.9.15.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
8	UE sends INVITE indicating support for Session Timer, with either the Session-Expires value set to 1800 or no Session-Expires header.	-->	INVITE	1	P
9-17	Steps 2-10 of Annex A.4.1 happen.	-	-		
18	SS sends 200 OK for INVITE, with timer tag in Supported headers but without Session-Expires header.	<--	200 OK		
19	UE sends ACK.	-->	ACK	2	P
20	SS sends BYE to release the call 1860 seconds after step 19. (Step 1 of Annex A.8)	<--	BYE	3	P
21	UE sends 200 OK for BYE. (Step 2 of Annex A.8)	-->	200 OK	3	P

7.30.3.3 Specific message contents

Table 7.30.3.3-1: INVITE (step 8, table 7.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A4, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds refresher		(if present) 1800 uac (if present)		RFC 4028 [37]

Table 7.30.3.3-2: 200 OK (step 18, table 7.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]

7.31 Session Timer / MT Call / Remote end supports but does not send Session-Expires / 5GS

7.31.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions }
ensure that {
  when { UE receives INVITE for voice call }
    then { UE continues setup of voice call and finally sends 200 OK for INVITE with Session-Expires
being 1800 and setting refresher to uac }
            }
```

(2)

```
with { Call having been set up }
ensure that {
  when { 900 seconds have passed and UE receives UPDATE to refresh the session }
    then { UE sends 200 OK for UPDATE }
            }
```

(3)

```
with { UE having sent 200 OK for UPDATE }
ensure that {
  when { 1800 seconds passed }
    then { UE sends BYE }
            }
```

7.31.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.31.3 Test description

7.31.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.31.3.2 Test procedure sequence

Table 7.31.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9	SS sends INVITE. (Step 1 of Annex A.5.1)	<--	INVITE		
10-18	Steps 2-10 of Annex A.5.1 happen.	-	-		
19	UE sends 200 OK for INVITE with Session-Expires value set to 1800 and refresher value set to uac.	-->	200 OK	1	P
20	Step 12 of Annex A.5.1 happens.	<--	ACK		
21	900 seconds after step 19, SS sends an UPDATE request to refresh the session.	<--	UPDATE		
22	UE sends 200 OK for UPDATE.	-->	200 OK	2	P
23	UE sends BYE to release the call due to session expiry 1800 seconds after step 22. (Step 1 of Annex A.7)	-->	BYE	3	P
24	SS sends 200 OK for BYE. (Step 2 of Annex A.7)	<--	200 OK		

7.31.3.3 Specific message contents

Table 7.31.3.3-1: INVITE (step 9, table 7.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.9, Conditions A1, A3 and A4

Table 7.31.3.3-2: 200 OK (step 19, table 7.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A2 and A11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]

Table 7.31.3.3-3: UPDATE (step 21, table 7.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, Condition A3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]
Content-Type		not present		RFC 4028 [37]

Table 7.31.3.3-4: 200 OK (step 22, table 7.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A2 and A11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]

7.32 Session Timer / MT Call / Remote end sends Session-Expires but does not choose refresher / 5GS

7.32.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions }
ensure that {
  when { UE receives INVITE for voice call containing timer tag and Session-Expires value 1800 }
    then { UE continues setup of voice call and finally sends 200 OK for INVITE with Session-Expires being 1800 and setting refresher to uac }
            }
```

(2)

```
with { Call having been set up }
ensure that {
  when { 900 seconds have passed and UE receives UPDATE to refresh the session }
    then { UE sends 200 OK for UPDATE }
            }
```

7.32.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.32.3 Test description

7.32.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.32.3.2 Test procedure sequence

Table 7.32.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9	SS sends INVITE with timer tag set in Supported header and Session-Expires value set to 1800.	<--	INVITE		
10-18	Steps 2-10 of Annex A.5.1 happen.	-	-		
19	UE sends 200 OK for INVITE with Session-Expires value set to 1800 and refresher value set to uac.	-->	200 OK	1	P
20	SS sends ACK. (Step 12 of Annex A.5.1)	<--	ACK		
21	900 seconds after step 19, SS sends an UPDATE request to refresh the session.	<--	UPDATE		
22	UE sends 200 OK for UPDATE.	-->	200 OK	2	P
23-24	SS releases the call. (Steps 1-2 of Annex A.8)	-	-		

7.32.3.3 Specific message contents

Table 7.32.3.3-1: INVITE (step 9, table 7.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.9, Conditions A1, A3 and A4				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds		 1800		RFC 4028 [37]

Table 7.32.3.3-2: 200 OK (step 19, table 7.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A2 and A11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]

Table 7.32.3.3-3: UPDATE (step 21, table 7.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, Condition A3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]
Content-Type		not present		RFC 4028 [37]

Table 7.32.3.3-4: 200 OK (step 22, table 7.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A2 and A11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Require		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]

7.33 Session Timer / MT Call / Remote end chooses UE as refresher / 5GS

7.33.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and being configured to
not use preconditions }
ensure that {
  when { UE receives INVITE for voice call with Session-Expires value 1800 and refresher set uas and
remote UE not supporting UPDATE }
    then { UE continues setup of voice call and finally sends 200 OK for INVITE with Session-Expires
being 1800 and setting refresher to uas }
            }
```

(2)

```
with { Voice call having been set up }
ensure that {
  when { 900 seconds have passed }
    then { UE sends re-INVITE to refresh the session }
            }
```

(3)

```
with { UE having refreshed the session }
ensure that {
  when { Another 900 seconds have passed }
    then { UE sends another re-INVITE to refresh the session }
            }
```

7.33.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.33.3 Test description

7.33.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer
- UE is configured to not use preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.33.3.2 Test procedure sequence

Table 7.33.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9	SS sends INVITE with Session-Expires value set to 1800 and refresher set to uas.	<--	INVITE		
10-17	Steps 2-8A of Annex A.5.2 happen.	-	-		
18	Void	-	-	-	-
19	UE send 200 OK for INVITE with Session-Expires value set to 1800 and refresher value set to uas.	-->	200 OK	1	P
20	SS sends ACK. (Step 12 of Annex A.5.1)	<--	ACK		
21	900 seconds after step 20, UE sends an INVITE request to refresh the session.	-->	INVITE	2	P
22	SS sends 200 OK for INVITE.	<--	200 OK		
23	UE sends ACK.	-->	ACK		
24	900 seconds after step 23, UE sends an INVITE request to refresh the session.	-->	INVITE	3	P
25	SS sends 200 OK for INVITE.	<--	200 OK		
26	UE sends ACK.	-->	ACK		
27-28	SS releases the call. (Steps 1-2 of Annex A.8)	-	-		

7.33.3.3 Specific message contents

Table 7.33.3.3-1: INVITE (step 9, table 7.33.3.2-1)

Derivation Path: Step 1 of Annex A.5.2				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Allow		*INVITE, ACK, OPTIONS, CANCEL, BYE*		RFC 3261 [6]
Session-Expires delta-seconds refresher		 *1800* *uas*		RFC 4028 [37]

Table 7.33.3.3-2: 200 OK (step 19, table 7.33.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A2, A11, A20, and A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		*timer*		RFC 4028 [37]
Session-Expires delta-seconds refresher		 *1800* *uas*		RFC 4028 [37]

Table 7.33.3.3-3: INVITE (steps 21 and 24, table 7.33.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.1, Conditions A5, A26 and A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds refresher		 *1800* *uac*		RFC 4028 [37]
Content-Type media-type		 *application/sdp*		RFC 3261 [6]
Content-Length value		 length of message-body		RFC 3261 [6]
Message-body		**Session description:** *v=0* *o*=(origin) [Note 1] *s*=(session name) *c*=IN (addrtype) (connection-address for UE) if present b=AS: (bandwidth-value) **Time description:** t= (start-time) (stop-time) **Media description:** Any media, if present. Note 1: Same origin as in last SDP sent by the UE.		RFC 4566 [38]

Table 7.33.3.3-4: 200 OK (steps 22 and 25, table 7.33.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A2, A11, A20 and A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		*timer*		RFC 4028 [37]
Session-Expires delta-seconds refresher		 *1800* *uac*		RFC 4028 [37]

7.34 Session Timer / MT Call / Remote end does not support Session Timer / 5GS

7.34.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and being configured to use Session Timer and preconditions }
ensure that {
  when { UE receives INVITE for voice call without support for Session Timer }
    then { UE continues setup of voice call and finally sends 200 OK for INVITE with Session-Expires
being 1800 and setting refresher to uas }
            }
```

(2)

```
with { Call having been set up }
ensure that {
  when { 900 seconds have passed }
    then { UE sends UPDATE to refresh the session }
            }
```

7.34.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 5.1.2A.1.1]

A UE supporting RFC 4028 [58], when it receives a 422 (Session Interval Too Small) to an INVITE request where the response contains a Min-SE header field, shall retry the request in accordance with RFC 4028 [58] subclause 7.4.

[TS 24.229 clause 5.2.7.2]

When the P-CSCF receives from the UE an INVITE request, the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it.

[TS 24.229 clause 5.2.7.3]

When the P-CSCF receives an INVITE request destined for the UE the P-CSCF may require the periodic refreshment of the session to avoid hung states in the P-CSCF. If the P-CSCF requires the session to be refreshed, then the P-CSCF shall apply the procedures described in RFC 4028 [58] clause 8.

NOTE 1: Requesting the session to be refreshed requires support by at least one of the UEs. This functionality cannot automatically be granted, i.e. at least one of the involved UEs needs to support it in order to make it work.

[TS 24.229 clause 5.4.5.3]

If the S-CSCF requested the session to be refreshed periodically, and the S-CSCF got the indication that the session will be refreshed, when the session timer expires, the S-CSCF shall delete all the stored information related to the dialog.

7.34.3 Test description

7.34.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use Session Timer and preconditions.

Preamble:

- UE is in state 1N-A (TS 38.508-1 [21]) and registered to IMS.

7.34.3.2 Test procedure sequence

Table 7.34.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-8	Steps 1-8 of generic procedure specified in Table 4.9.16.2.2-1 of TS 38.508-1 [21] are performed.	-	-		
9	SS sends INVITE without support for Session-Timer.	<--	INVITE		
10-18	Steps 2-10 of Annex A.5.1 happen.	-	-		
19	UE sends 200 OK for INVITE with Session-Expires value set to 1800 and refresher value set to uas.	-->	200 OK	1	P
20	SS sends ACK. (Step 12 of Annex A.5.1)	<--	ACK		
21	900 seconds after step 20, UE sends an UPDATE request to refresh the session.	-->	UPDATE	2	P
22	SS sends 200 OK for UPDATE.	<--	200 OK		
23-24	SS releases the call. (Steps 1-2 of Annex A.8)	-	-		

7.34.3.3 Specific message contents

Table 7.34.3.3-1: INVITE (step 9, table 7.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.9, Conditions A1, A3 and A4				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Allow		INVITE, UPDATE, PRACK, ACK, OPTIONS, CANCEL, BYE		RFC 4028 [37]

Table 7.34.3.3-2: 200 OK (step 19, table 7.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A2 and A11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Session-Expires delta-seconds refresher		 1800 uas		RFC 4028 [37]

Table 7.34.3.3-3: UPDATE (step 21, table 7.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.5, Conditions A3 and A6				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Supported		timer		RFC 4028 [37]
Session-Expires delta-seconds refresher		 1800 uac		RFC 4028 [37]
Content-Type		any value if present		RFC 4028 [37]

Table 7.34.3.3-4: 200 OK (step 22, table 7.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Conditions A1 and A10

8 Supplementary Services

8.1 Originating Identification Presentation / Configuration / 5GS

8.1.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to activate OIP }
   then { UE authenticates itself using GBA }
            }
```

(2)

```
with { UE having started authentication using GBA }
ensure that {
  when { UE receives 200 OK concluding the authentication }
   then { UE sends HTTP request to activate OIP }
            }
```

(3)

```
with { UE having concluded activation of OIP }
ensure that {
  when { UE is made to de-activate OIP }
   then { UE sends HTTP request to de-activate OIP }
            }
```

8.1.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

Generic requirements for Originating Identification Presentation can be found from TS 34.229-1 Annexes F.1 and F.2.

[TS 24.607, clause 4.2.1]:

The OIP service provides the terminating user with the possibility of receiving trusted (i.e. network provided) identity information in order to identify the originating user.

In addition to the trusted identity information, the identity information from the originating user can include identity information generated by the originating user and in general transparently transported by the network. In the particular case where the "no screening" special arrangement does not apply, the originating network shall verify the content of this user generated identity information. The terminating network cannot be responsible for the content of this user generated identity information.

[TS 24.607 clause 4.10.1]:

The OIP service can be activated/deactivated using the active attribute of the <originating-identity-presentation> service element.

[TS 24.109 clause 4.2]:

The UE shall initiate the bootstrapping procedure when:

a) the UE wants to interact with a NAF and bootstrapping is required;

b) a NAF has requested bootstrapping required indication as described in subclause 5.2.4 or bootstrapping renegotiation indication as described in subclause 5.2.5; or

c) the lifetime of the key has expired in the UE if one or more applications are using that key.

A UE and the BSF shall establish bootstrapped security association between them by running bootstrapping procedure. Bootstrapping security association consists of a bootstrapping transaction identifier (B-TID) and key material Ks. Bootstrapping session on the BSF also includes security related information about subscriber (e.g. user's private identity). Bootstrapping session is valid for a certain time period, and shall be deleted in the BSF when the session becomes invalid.

Bootstrapping procedure shall be based on HTTP Digest AKA as described in 3GPP TS 33.220 [1] and in RFC 3310 [6] with the modifications described below.

The BSF address is derived from the IMPI or IMSI according to 3GPP TS 23.003 [7].

A UE shall indicate to the BSF that it supports the use of TMPI as defined in 3GPP 33.220 [1] by including a "product" token in the "User-Agent" header field (cf. RFC 2616 [14]) that is set to a static string "3gpp-gba-tmpi" in HTTP requests sent to the BSF.

A BSF shall indicate to the UE that it supports the use of TMPI as defined in 3GPP 33.220 [1] by including a "product" token in the "Server" header field (cf. RFC 2616 [14]) that is set to a static string "3gpp-gba-tmpi" in HTTP responses sent to the UE.

In the bootstrapping procedure, Authorization, WWW-Authenticate, and Authentication-Info HTTP headers shall be used as described in RFC 3310 [6] with following exceptions:

a) the "realm" parameter shall contain the network name where the username is authenticated;

b) the quality of protection ("qop") parameter shall be "auth-int"; and

c) the "username" parameter shall contain user's private identity (IMPI).

NOTE: If the UE does not have an ISIM application with an IMPI, the IMPI will be constructed from IMSI, according to 3GPP TS 23.003 [7].

In addition to RFC 3310 [6], the following apply:

a) In the initial request from the UE to the BSF, the UE shall include Authorization header with following parameters:

  - the username directive, set to

    1) the value of the TMPI if one has been associated with the private user identity as described in 3GPP 33.220 [1]; or

    2) the value of the private user identity;

  - the realm directive, set to the BSF address derived from the IMPI or IMSI according to 3GPP TS 23.003 [7];

  - the uri directive, set to either absoluteURL "http://<BSF address>/" or abs_path "/", and which one is used is specified in RFC 2617 [9];

  - the nonce directive, set to an empty value; and

  - the response directive, set to an empty value;

b) In the challenge response from the BSF to the UE, the BSF shall include parameters to WWW-Authenticate header as specified in RFC 3310 [6] with following clarifications:

  - the realm directive, set to the BSF address derived from the IMPI or IMSI according to 3GPP TS 23.003 [7];

c) In the message from the BSF to the UE, the BSF shall include bootstrapping transaction identifier (B-TID) and the key lifetime to an XML document in the HTTP response payload. The BSF may also include additional server specific data to the XML document. The XML schema definition of this XML document is given in Annex C.

d) When responding to a challenge from the BSF, the UE shall include an Authorization header containing a realm directive set to the value as received in the realm directive in the WWW-Authenticate header.

e) Authentication-Info header shall be included into the subsequent HTTP response after the BSF concluded that the UE has been authenticated. Authentication-Info header shall include the "rspauth" parameter.

After successful bootstrapping procedure the UE and the BSF shall contain the key material (Ks) and the B-TID. The key material shall be derived from AKA parameters as specified in 3GPP TS 33.220 [1]. In addition, BSF shall also contain a set of security specific attributes related to the UE.

An example flow of successful bootstrapping procedure can be found in clause A.3.

8.1.3 Test description

8.1.3.1 Pre-test conditions

System Simulator:

- SS is configured with shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) configured on the UICC card equipped into the UE.
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- At the SS, a HTTP Server is established at port 80 to simulate the XCAP server
- 1 NR Cell

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured with the name of the XCAP root directory on the XCAP server and the user's directory name.
- UE has activated an IPCAN bearer with SS.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1) and registered to IMS.
- The UE has established a PDN connectivity for IMS XCAP signalling. The UE may either be configured to re-use the Internet APN for XCAP signalling or the UE uses a specific XCAP-only APN
- During these procedures the UE may request a DNS server address via NAS signalling and as parallel behaviour the UE may resolve the IP address of the XCAP server via DNS.

8.1.3.2 Test procedure sequence

Table 8.1.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is triggered for activation of OIP	-	-	-	-
2	Step 1 of TS 34.229-1 subclause C.29.2 happens	-->	HTTP Request	-	-
3	Step 2 of TS 34.229-1 subclause C.29.2 happens	<--	HTTP Response: 401 UNAUTHORIZED	-	-
4	Step 3 of TS 34.229-1 subclause C.29.2 happens Check: Does the UE send HTTP request with valid authorization credentials?	-->	HTTP Request	1	P
5	Step 4 of TS 34.229-1 subclause C.29.2 happens	<--	HTTP Response: 200 OK	-	-
6	Step 5 of TS 34.229-1 subclause C.29.1 happens	-	-		
7	Check: Does the Simservs document stored in the SS contain the following information supplied by UE? -<originating-identity-presentation> element with "active" attribute being set "true"	-	-	2	P
8	Make the UE attempt deactivation of OIP	-	-	-	-
9	Step 8 of TS 34.229-1 subclause C.29.1 happens	-	-	-	-
10	Check: Does the Simservs document stored in the SS contain the following information supplied by UE? -<originating-identity-presentation> element with "active" attribute being set "false"	-	-	3	P

8.1.3.3 Specific message contents

Table 8.1.3.3-1: HTTP Request and Responses (Table 8.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause C.29.1 and C.29.2

Editor's Note: XML content needs to be specified and refer to the HTTP steps once a generic procedure is defined.

8.2 to 8.17 FFS

8.18 Barring of All Incoming Calls / except for a specific user / 5GS

8.18.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to activate incoming communication barring except for a specific user (ICBESU) }
   then { UE authenticates itself using Digest }
            }
```

(2)

```
with { UE having started authentication using Digest }
ensure that {
  when { UE receives 200 OK concluding the authentication }
   then { UE sends HTTP request to activate ICBESU }
            }
```

(3)

```
with { UE having concluded activation of ICBESU }
ensure that {
  when { UE is made to de-activate ICBESU }
   then { UE sends HTTP request to de-activate ICBESU }
            }
```

8.18.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

References: Conformance requirements for activating and deactivating Communication Barring are specified in TS 34.229-1 Annexes F.1 and F.5; TS 24.611, clause 4.9.1.4; TS 24.109, clause 4.2

[TS 24.611, clause 4.9.1.4]:

cp:identity: This condition evaluates to true when the remote user's identity matches with the value of the identity element. The interpretation of all the elements of this condition is described in the in the common policy draft (see RFC 4745). In all other cases the condition evaluates to false.

...

ocp:other-identity: If present in any rule, the "other-identity" element, which is empty, matches all identities that are not referenced in any rule. It allows for specifying a default policy. The exact interpretation of this condition is specified in OMA-TS-XDM_Core.

[TS 24.109 clause 4.2]:

The UE shall initiate the bootstrapping procedure when:

a) the UE wants to interact with a NAF and bootstrapping is required;

b) a NAF has requested bootstrapping required indication as described in subclause 5.2.4 or bootstrapping renegotiation indication as described in subclause 5.2.5; or

c) the lifetime of the key has expired in the UE if one or more applications are using that key.

A UE and the BSF shall establish bootstrapped security association between them by running bootstrapping procedure. Bootstrapping security association consists of a bootstrapping transaction identifier (B-TID) and key material Ks. Bootstrapping session on the BSF also includes security related information about subscriber (e.g. user's private identity). Bootstrapping session is valid for a certain time period, and shall be deleted in the BSF when the session becomes invalid.

Bootstrapping procedure shall be based on HTTP Digest AKA as described in 3GPP TS 33.220 [1] and in RFC 3310 [6] with the modifications described below.

The BSF address is derived from the IMPI or IMSI according to 3GPP TS 23.003 [7].

A UE shall indicate to the BSF that it supports the use of TMPI as defined in 3GPP 33.220 [1] by including a "product" token in the "User-Agent" header field (cf. RFC 2616 [14]) that is set to a static string "3gpp-gba-tmpi" in HTTP requests sent to the BSF.

A BSF shall indicate to the UE that it supports the use of TMPI as defined in 3GPP 33.220 [1] by including a "product" token in the "Server" header field (cf. RFC 2616 [14]) that is set to a static string "3gpp-gba-tmpi" in HTTP responses sent to the UE.

In the bootstrapping procedure, Authorization, WWW-Authenticate, and Authentication-Info HTTP headers shall be used as described in RFC 3310 [6] with following exceptions:

a) the "realm" parameter shall contain the network name where the username is authenticated;

b) the quality of protection ("qop") parameter shall be "auth-int"; and

c) the "username" parameter shall contain user's private identity (IMPI).

NOTE: If the UE does not have an ISIM application with an IMPI, the IMPI will be constructed from IMSI, according to 3GPP TS 23.003 [7].

In addition to RFC 3310 [6], the following apply:

a) In the initial request from the UE to the BSF, the UE shall include Authorization header with following parameters:

   - the username directive, set to

      1) the value of the TMPI if one has been associated with the private user identity as described in 3GPP 33.220 [1]; or

      2) the value of the private user identity;

   - the realm directive, set to the BSF address derived from the IMPI or IMSI according to 3GPP TS 23.003 [7];

   - the uri directive, set to either absoluteURL "http://<BSF address>/" or abs_path "/", and which one is used is specified in RFC 2617 [9];

   - the nonce directive, set to an empty value; and

   - the response directive, set to an empty value;

b) In the challenge response from the BSF to the UE, the BSF shall include parameters to WWW-Authenticate header as specified in RFC 3310 [6] with following clarifications:

   - the realm directive, set to the BSF address derived from the IMPI or IMSI according to 3GPP TS 23.003 [7];

c) In the message from the BSF to the UE, the BSF shall include bootstrapping transaction identifier (B-TID) and the key lifetime to an XML document in the HTTP response payload. The BSF may also include additional server specific data to the XML document. The XML schema definition of this XML document is given in Annex C.

d) When responding to a challenge from the BSF, the UE shall include an Authorization header containing a realm directive set to the value as received in the realm directive in the WWW-Authenticate header.

e) Authentication-Info header shall be included into the subsequent HTTP response after the BSF concluded that the UE has been authenticated. Authentication-Info header shall include the "rspauth" parameter.

After successful bootstrapping procedure the UE and the BSF shall contain the key material (Ks) and the B-TID. The key material shall be derived from AKA parameters as specified in 3GPP TS 33.220 [1]. In addition, BSF shall also contain a set of security specific attributes related to the UE.

An example flow of successful bootstrapping procedure can be found in clause A.3.

8.18.3 Test description

8.18.3.1 Pre-test conditions

System Simulator:

- SS is configured shared secret key of IMS AKA algorithm, related to the IMS private user identity (IMPI) configured on the UICC card equipped into the UE.
- SS is listening to SIP default port 5060 for both UDP and TCP protocols.
- At the SS, a HTTP Server is established at port 80 to simulate the XCAP server
- 1 NR Cell

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured with the name of the XCAP root directory on the XCAP server and the user's directory name.
- UE has activated an IPCAN bearer with SS.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1) and registered to IMS
- The UE has established a PDN connectivity for IMS XCAP signalling. The UE may either be configured to re-use the Internet APN for XCAP signalling or the UE uses a specific XCAP-only APN
- During these procedures the UE may request a DNS server address via NAS signalling and as parallel behaviour the UE may resolve the IP address of the XCAP server via DNS.

8.18.3.2 Test procedure sequence

Table 8.18.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is triggered for activation of supplementary service ICBESU	-	-	-	-
2	UE sends Initial HTTP Request (Note 1)	-->	HTTP Request		
	EXCEPTION: steps 3 and 4 describe behaviour in case of HTTP Digest XCAP authentication when the UE does not provide correct authorization credentials within its initial request	-	-	-	-
3	SS sends HTTP Response: “401 Unauthorized”	<--	HTTP Response: 401 Unauthorized	-	-
4	Check: Does the UE send HTTP Request with valid authorization credentials?	-->	HTTP Request	1	P
5	SS sends HTTP Response: “200 OK”	<--	HTTP Response: 200 OK	-	-
	EXCEPTION: Steps 6 and steps 7 can be repeated several times; this exchange of information is considered to be finished when there is no further HTTP request sent by the UE within 20 seconds after the previous request	-	-	-	-
6	Check: Does the UE send HTTP Request? (Note 1)	-->	HTTP Request	2	P
7	SS sends HTTP Response: “200 OK” or “404 File Not Found” (Note 2)	<--	HTTP Response	-	-
8	Check: Does the simservs document stored in the SS contain the information supplied by the UE as required by the test requirements of the specific test case?	-	This is done by fetching the whole simservs document from the XCAP server and checking its content against the respective XML file (according to the XSD definitions for the respective supplementary service)	-	-
9	UE is triggered for deactivation of supplementary service ICBESU	-	-	-	-
	EXCEPTION: steps 10 and 11 describe the message exchange between the UE and the SS which can be repeated several times; this exchange of information is considered to be finished when there is no further HTTP request sent by the UE within 10 seconds after the previous request	-	-	-	-
10	Check: Does the UE send HTTP Request? (Note 1)	-->	HTTP Request	3	P
11	SS sends HTTP Response: “200 OK” or “404 File Not Found” (Note 2)	<--	HTTP Response	-	-
12	Check: Does the simservs document stored in the SS contain the information supplied by the UE as required by the test requirements of the specific test case?	-	This is done by fetching the whole simservs document from the XCAP server and checking its content against the respective XML file (according to the XSD definitions for the respective supplementary service)	-	-
Note 1: The HTTP requests sent by the UE are processed by an XCAP server implementation at the SS to modify the contents of the simservs document. Note 2: “404 File Not Found” is sent as response for a GET request to a non-existing node.					

8.18.3.3 Specific message contents

Table 8.18.3.3-1: HTTP Requests and Responses (Table 8.18.3.2-1)

Derivation Path: TS 34.229-1 [2], Tables in subclause C.29.1 and C.29.2

Editor's Note: XML content needs to be specified and refer to the HTTP steps once a generic procedure is defined.

8.19 to 8.25

8.26 MO Call Hold without announcement / 5GS

8.26.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and having set up a voice call }
ensure that {
  when { UE is being made to hold the call }
    then { UE sends re-INVITE or UPDATE, and completes the call hold procedure }
            }
```

(2)

```
with { UE having put the voice call on hold }
ensure that {
  when { UE is being made to resume the call }
    then { UE sends re-INVITE or UPDATE, and completes the call resume procedure }
            }
```

8.26.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.610 clause 4.5.2.1]:

In addition to the application of procedures according to 3GPP TS 24.229 [1], the following procedures shall be applied at the invoking UE in accordance with RFC 3264 [4].

A UE shall not invoke the HOLD service on a dialog associated with an emergency call the UE has initiated.

If not all the media streams are affected, the invoking UE shall generate a new SDP offer where:

1) for each media stream that is to be held, the SDP offer contains:

   - an "inactive" SDP attribute if the stream was previously set to "recvonly"; or

   - a "sendonly" SDP attribute if the stream was previously set to "sendrecv";

NOTE 1: If the directionality attribute of the media stream is currently "sendonly" or "inactive", then that media stream is not put on hold and, in the SDP offer, the directionality for that media stream remains unchanged.

2) for each held media stream that is to be resumed, the SDP offer contains:

   - a "recvonly" SDP attribute if the stream was previously an inactive media stream; or

   - a "sendrecv" SDP attribute if the stream was previously a sendonly media stream, or the attribute may be omitted, since sendrecv is the default; and

3) for each media stream that is unaffected, the media parameters in the SDP offer remain unchanged from the previous SDP.

If all the media streams are to be held:

- if they all have identical directionality, the invoking UE shall generate an SDP offer containing a session level direction attribute, or separate media level direction attributes, in the SDP that is set to:

  1) "inactive" if the streams were previously set to "recvonly"; or

  2) "sendonly" if the streams were previously set to "sendrecv"; and

NOTE 2: If the directionality attribute of all the media streams is currently "sendonly" or "inactive", then all these media streams are not put on hold and, in the SDP offer, the directionality for these media streams will remain unchanged.

- if they all do not have identical directionality, then for each media stream in the session, the invoking UE shall follow the procedure listed above for individual media streams.

If all the media streams were previously on hold and are to be resumed:

- if they all have identical directionality, the invoking UE shall generate a session level direction attribute, or separate media level direction attributes, in the SDP that is set to:

  1) "recvonly" if the streams were previously inactive media streams; or

  2) "sendrecv" if the streams were previously sendonly media streams, or the attribute may be omitted, since sendrecv is the default; and

- if they all do not have identical directionality, then for each media stream in the session, the invoking UE shall follow the procedure listed above for individual media streams.

If, in the generated SDP offer, there is at least one media stream whose directionality has changed from the previous SDP, the UE shall send the generated SDP offer in a re-INVITE request (or UPDATE request) to the remote UE.

[TS 26.114 clause 7.3.1]:

RTCP packets should be sent for all types of multimedia sessions to enable synchronization with other RTP transported media, remote end-point aliveness information, monitoring of the transmission quality, and carriage of feedback messages such as TMMBR for video and RTCP APP for speech. The RR value should be set greater than zero to enable RTCP packets to be sent when media is put on hold and during active RTP media transmission, including real-time text sessions which may have infrequent RTP media transmissions.

[TS 24.229 clause 6.1.1]:

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

8.26.3 Test description

8.26.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.

- UE is configured to register for IMS after switch on.

- UE is configured to use preconditions

Preamble:

- The UE has registered to IMS and set up the MO call, by executing the generic test procedure in Annex A.2 up to the last step and thereafter executing the generic test procedure in A4.1.

8.26.3.2 Test procedure sequence

Table 8.26.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	**Message**		
1	The UE is made to hold the call	-	-		
2	Check: Does the UE send INVITE or UPDATE with a SDP offer to hold the call? (Step 1 of Annex A.17)	-->	INVITE or UPDATE	1	P
3	The SS responds to the INVITE with a 100 Trying provisional response. (Step 2 of Annex A.17)	<--	100 Trying		
4	The SS responds to INVITE or UPDATE with 200 OK to indicate that the remote UE is no more sending any media (call hold) or resumes sending media (call resume) (Step 3 of Annex A.17)	<--	200 OK		
5	If the UE sent INVITE in step 1 then UE acknowledges the receipt of 200 OK for INVITE. (Step 4 of Annex A.17)	-->	ACK		
6	The UE is made to resume the call	-	-		
7	Check: Does the UE send INVITE or UPDATE with a SDP offer to resume the call? (Step 1 of Annex A.17)	-->	INVITE or UPDATE	2	P
8	The SS responds to the INVITE with a 100 Trying provisional response. (Step 2 of Annex A.17)	<--	100 Trying		
9	The SS responds to INVITE or UPDATE with 200 OK to indicate that the remote UE is no more sending any media (call hold) or resumes sending media (call resume) (Step 3 of Annex A.17)	<--	200 OK		
10	Optional: If the UE sent INVITE in step 1 then UE acknowledges the receipt of 200 OK for INVITE. (Step 4 of Annex A.17)	-->	ACK		
11	The SS releases the call with BYE. (Step 1 of Annex A.8)	<--	BYE	-	-
12	The UE sends 200 OK for BYE. (Step 2 of Annex A.8)	-->	200 OK	-	-

8.26.3.3 Specific message contents

None as fully specified in Annex A.17 and Annex A.8.

8.27

8.28 MT Call Hold without announcement / 5GS

8.28.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and having set up an MO voice call }
ensure that {
  when { UE receives re-INVITE including call hold instructions }
    then { UE may send 100 Trying and sends 200 OK for re-INVITE }
            }
```

(2)

```
with { UE having responded to re-INVITE for call hold }
ensure that {
  when { UE receives ACK followed by re-INVITE including call resume instructions }
    then { UE may send 100 Trying and sends 200 OK for re-INVITE }
            }
```

(3)

```
with { UE having concluded the call resume procedure }
ensure that {
  when { UE is being made to release the call }
    then { UE sends BYE }
            }
```

8.28.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.610 clause 4.5.2.9]:

3GPP TS 24.229 [1] shall apply.

[TS 26.114 clause 7.3.1]:

RTCP packets should be sent for all types of multimedia sessions to enable synchronization with other RTP transported media, remote end-point aliveness information, monitoring of the transmission quality, and carriage of feedback messages such as TMMBR for video and RTCP APP for speech. The RR value should be set greater than zero to enable RTCP packets to be sent when media is put on hold and during active RTP media transmission, including real-time text sessions which may have infrequent RTP media transmissions.

[TS 24.229 clause 6.1.1]:

If the media line in the SDP message body indicates the usage of RTP/RTCP, and if the UE is configured to request an RTCP bandwidth level for the session is different than the default RTCP bandwidth as specified in RFC 3556 [56], then in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP. The bandwidth-value in the b=RS: and b=RR: lines may include transport overhead as described in subclause 6.1 of RFC 3890 [152].

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in or 3GPP 29.213 [13C].

NOTE 3: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifier will typically get the value of zero.

8.28.3 Test description

8.28.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1) and registered to IMS.

8.28.3.2 Test procedure sequence

Table 8.28.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	SS sends INVITE with a SDP offer to hold the call (Step 1 of Annex A.18)	<--	INVITE	-	-
2	Optional: The UE responds with a 100 Trying provisional response (Step 2 of Annex A.18)	-->	100 Trying	-	-
3	Check: Does the UE respond to INVITE with 200 OK to indicate that the UE is no more expecting to receive any media? (Step 3 of Annex A.18)	-->	200 OK	1	P
4	The SS acknowledges the receipt of 200 OK for INVITE (Step 4 of Annex A.18)	<--	ACK	-	-
5	SS sends INVITE with a SDP offer to resume the call (Step 1 of Annex A.18)	<--	INVITE	-	-
6	Optional: The UE responds with a 100 Trying provisional response (Step 2 of Annex A.18)	-->	100 Trying	-	-
7	Check: Does the UE respond to INVITE with 200 OK to indicate that the UE is no more expecting to receive any media? (Step 3 of Annex A.18)	-->	200 OK	2	P
8	The SS acknowledges the receipt of 200 OK for INVITE (Step 4 of Annex A.18)	<--	ACK	-	-
9	UE is made to release the call	-	-	-	-
10	Check: Does the UE send BYE to release the call? (Step 1 of Annex A.7)	-->	BYE	3	P
11	The SS sends 200 OK for BYE (Step 2 of Annex A.7)	<--	200 OK	-	-

8.28.3.3 Specific message contents

None as fully specified in Annex A.18 and A.7.

8.29

8.30 Subscription to the MWI event package / 5GS

8.30.1 Test Purpose (TP)

(1)

```
with { UE being configured to subscribe to MWI }
ensure that {
  when { UE registers to IMS }
    then { UE subscribes to MWI and to reg event }
            }
```

(2)

```
with { UE being registered to MWI and reg event }
ensure that {
  when { UE receives initial NOTIFY for MWI }
    then { UE sends 200 OK }
            }
```

(3)

```
with { UE being registered to MWI and reg event }
ensure that {
  when { UE receives second NOTIFY for MWI indicating one voice message waiting }
    then { UE sends 200 OK }
            }
```

(4)

```
with { UE being registered to MWI and reg event }
ensure that {
  when { UE receives NOTIFY for reg event }
    then { UE sends 200 OK }
            }
```

8.30.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.606, clause 4.1]:

The Message Waiting Indication (MWI) service enables the network, upon the request of a controlling user to indicate to the receiving user, that there is at least one message waiting.

[TS 24.606, clause 4.6]:

The application/simple-message-summary MIME type used to provide Message Summary and Message Waiting Indication Information is defined in subclause 5 of RFC 3842 [3].

The coding of the message types in the message-context-class values is defined in the specifications listed in the "reference" column of table 1.

Table 1: Coding requirements

Value	Reference
voice-message	RFC 3458 [5]
video-message	RFC 3938 [6]
fax-message	RFC 3458 [5]
pager-message	RFC 3458 [5]
multimedia-message	RFC 3458 [5]
text-message	RFC 3458 [5]
none	RFC 3458 [5]

The coding of the additional information about deposited messages in the application/simple-message-summary MIME body is defined in subclause 25 of RFC 3261 [11] for SIP extension-header (subclause 3.5 of RFC 3842 [3]) and follow the rules defined in the specifications listed in the "reference" column of table 2.

Table 2: Additional information

Header	Description	Reference
To:	Indicates the subscriber's public user identity used by correspondent to deposit a message.	subclause 3.6.3 of RFC 2822 [7]
From:	Indicates the correspondent's public user identity, if available.	subclause 3.6.2 of RFC 2822 [7]
Subject:	Indicates the topic of the deposited message as provided by correspondent.	subclause 3.6.5 of RFC 2822 [7]
Date:	Indicates the time and date information about message deposit.	subclause 3.6.1 of RFC 2822 [7]
Priority:	Indicates the message priority as provided by correspondent.	RFC 2156 [8]
Message-ID:	Indicates a single unique message identity.	subclause 3.6.4 of RFC 2822 [7]
Message-Context:	Indicates a type or context of message.	RFC 3458 [5]

[TS 24.606, clause 4.7.1]:

The MWI service is immediately activated after the SUBSCRIBE request from the MSUA is successfully processed, see subclause 4.7.2.

The MWI service is deactivated after subscription expiry or after unsuccessful attempt to deliver a notification about message waiting.

[TS 24.606, clause 4.7.2.1]:

When the MSUA intends to subscribe for status information changes of a message account, the MSUA shall generate a SUBSCRIBE request in accordance with RFC 6665 [4] and RFC 3842 [3] and in alignment with the procedures described in 3GPP TS 24.229 [2]. If the UE receives a 489 (Bad Event) response or a 405 (Method Not Allowed) response to the SUBSCRIBE request, the UE shall not re-try the SUBSCRIBE request until de-registration of the public user identity from IMS.

NOTE: 489 (Bad Event) response or 405 (Method Not Allowed) response to the SUBSCRIBE request indicates that MWI is not supported in the network.

The MSUA will address the SUBSCRIBE request to one of the subscriber's public user identities (see subclause 4.5.1).

The MSUA shall implement the "application/simple-message-summary" content type as described in RFC 3842 [3].

Reference(s)

3GPP TS 24.606 [34] TS 24.606 clause 4.1, 4.6, 4.7.1 and 4.7.2.1.

8.30.3 Test description

8.30.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- The UE is pre-configured to autonomously subscribe to the Message Waiting Indication package.

- The UE is configured with the public service identity of the message account. (Otherwise the phone is expected to use the public identity of the user when subscribing to the Message Waiting Indication package.)

Preamble:

- The UE is in test state 1N-A (TS 38.508-1) and registered to IMS.

8.30.3.2 Test procedure sequence

Table 8.30.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Check: Does the UE subscribe to the Message Waiting Indication event package?	-->	SUBSCRIBE	1	P
2	The SS responds SUBSCRIBE with 200 OK	<--	200 OK	-	-
3	The UE subscribes to the registration event package	-->	SUBSCRIBE	-	-
4	The SS responds with 200 OK	<--	200 OK	-	-
5	The SS sends initial NOTIFY for Message Waiting Indication event package	<--	NOTIFY	-	-
6	Check: Does the UE respond the NOTIFY with 200 OK?	-->	200 OK	2	P
7	The SS sends another NOTIFY for Message Waiting Indication event package, now referring to one voice message waiting	<--	NOTIFY	-	-
8	Check: Does the UE respond the NOTIFY with 200 OK?	-->	200 OK	3	P
9	The SS sends initial NOTIFY for registration event package, containing full registration state information for the registered public user identity in the XML body	<--	NOTIFY	-	-
10	Check: Does the UE respond with 200 OK?	-->	200 OK	4	P

8.30.3.3 Specific message contents

Table 8.30.3.3-1: SUBSCRIBE (step 1, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.6.1, Conditions A1, A6

Table 8.30.3.3-2: 200 OK for SUBSCRIBE (step 2/4, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.1.5, Condition A1

Table 8.30.3.3-3: SUBSCRIBE (step 3, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.1.4, Conditions A1, A7

Table 8.30.3.3-4: NOTIFY for Message Waiting Indication package (step 5, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.6.2, Condition A1

Table 8.30.3.3-5: 200 OK (step 6/8/10, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.3.1, Conditions A5, A11, A22

Table 8.30.3.3-6: NOTIFY for Message Waiting Indication package (step 7/9, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.6.2, Condition A1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		*Messages-Waiting: yes* *Message-Account:* same IMPU as in From header *Voice-Message: 1/0 (0/0)* *To:* <same IMPU as sent by the UE in the From header of the SUBSCRIBE in step 1> *From: <user2_public1@home1.net>* *Subject: call me back!* *Date: Fri 05 Feb 2021 14:24 +0100* *Priority: urgent* *Message-ID: 27775334485*@home domain name *Message-Context: voice-message*		

Table 8.30.3.3-7: NOTIFY for reg-event package (step 8, table 8.30.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.1.6, Condition A1

8.31 Creating and leaving a conference / 5GS

8.31.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to start a conference call }
    then { UE sends INVITE to the conference factory and completes the conference call initiation
and subscribes to conference event }
            }
```

(2)

```
with { Conference call going on }
ensure that {
  when { UE is made to leave the call }
    then { UE sends BYE and processes notification for conf event if any }
            }
```

8.31.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.147, clause 5.3.1.3.1]:

A conference can be created by means of SIP, as described in subclause 5.3.1.3.2 or subclause 5.3.1.3.3.

NOTE: Additionally, creation of a conference can be provided by other means.

The conference participant shall make use of the procedures for session establishment as described in subclauses 5.1.2A and 5.1.3 of 3GPP TS 24.229 [5] when creating conferences by means of SIP.

[TS 24.147, clause 5.3.1.3.2]:

Upon a request to create a conference with a conference factory URI, the conference participant shall:

1) generate an initial INVITE request in accordance with subclause 5.1.3.1 of 3GPP TS 24.229 [5]; and

2) set the request URI of the INVITE request to the conference factory URI.

On receiving a 200 (OK) response to the INVITE request with the "isfocus" feature parameter indicated in Contact header, the conference participant shall store the content of the received Contact header as the conference URI. In addition to this, the conference participant may subscribe to the conference event package as described in RFC 4575 [11] by using the stored conference URI.

NOTE 1: A conference participant can decide not to subscribe to the conference event package for conferences with a large number of attendees, due to, e.g. the signalling traffic caused by the notifications about users joining or leaving the conference.

NOTE 2: A conference can also be created with a conference URI. The procedures for this case at the conference participant are identical to those for joining a conference, as described in subclause 5.3.1.4.1. It is not assumed that the conference participant is aware that the conference gets created in this case.

NOTE 3: The UE can discover the conference factory URI from the Management Object as defined in 3GPP TS 24.166 [38]. Further discovery mechanisms for the conference factory URI are outside the scope of the present document.

8.31.3 Test description

8.31.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1) and registered to IMS.

8.31.3.2 Test procedure sequence

Table 8.31.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	Make the UE attempt an IMS Conference call	-	-	-	-
2	Check: Does the UE send INVITE with the first SDP offer?	-->	INVITE	1	P
3-8	Steps 2-7 of annex A4.1 happen	-	-	-	-
9	The SS responds INVITE with 200 OK and gives the final conference URI within the response	<--	200 OK	-	-
10	The UE acknowledges the receipt of 200 OK for INVITE	-->	ACK	-	-
	EXCEPTION: steps 11 – 14 describe optional behaviour depending on UE configuration. The SS shall wait up to 3s for the SUBSCRIBE of step 11	-	-	-	-
11	UE subscribes the conference event	-->	SUBSCRIBE	-	-
12	SS responds to the subscription	<--	200 OK	-	-
13	SS sends the initial state of the conference event to the UE	<--	NOTIFY	-	-
14	UE responds to the NOTIFY	-->	200 OK	-	-
15	The UE leaves the conference with BYE	-->	BYE	2	P
16	The SS sends 200 OK for BYE	<--	200 OK	-	-
17	If the UE had subscribed to the conference event package, the SS notifies the UE that its subscription to conference event package is terminated	-->	NOTIFY	-	-
18	The UE sends 200 OK for NOTIFY (if sent by SS)	-->	200 OK	-	-

8.31.3.3 Specific message contents

Table 8.31.3.3-1: INVITE (step 2, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Step 2 of C.21, Condition A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line Request-URI		 *sip:mmtel@conf-factory* appended with px_IMS_HomeDomainName		
To addr-spec		 *sip:mmtel@conf-factory* appended with px_IMS_HomeDomainName		

Table 8.31.3.3-2: 183 Session in Progress for INVITE (step 4, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Step 4 of C.21				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact addr-spec feature-param		 *sip:temporary@conf-factory* appended with px_IMS_HomeDomainName *isfocus*		
Record-Route rec-route		 <*sip:orig@scscf.3gpp.org;lr*>, <sip:SS P-CSCF address: protected server port of SS;*lr*>		

Table 8.31.3.3-3: 200 OK for INVITE (step 9, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A3.1, Condition A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Record-Route rec-route		 Same value as in the 183 response		
Contact addr-spec feature-param		 *sip:final@conf-factory* appended with px_IMS_HomeDomainName *isfocus*		

Table 8.31.3.3-4: ACK (step 10, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Step 2 of C.21, Condition A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line Request-URI		 *sip:final@conf-factory* appended with px_IMS_HomeDomainName		

Table 8.31.3.3-5: SUBSCRIBE for conference event package (step 11, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.7, Condition A1

Table 8.31.3.3-6: 200 OK for SUBSCRIBE (step 12, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.5.2

Table 8.31.3.3-7: NOTIFY for conference event package (step 13, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.5.3, Condition A3

Table 8.31.3.3-8: 200 OK for other requests than REGISTER or SUBSCRIBE (step 14/16/18, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Condition A22

Table 8.31.3.3-9: BYE (step 15, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.8, Condition A8				
Request-Line Request-URI		*sip:final@conf-factory* appended with px_IMS_HomeDomainName		

Table 8.31.3.3-10: NOTIFY for conference event package (step 17, table 8.31.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.5.3, Condition A4

8.32 Inviting user to conference by sending a REFER request to the conference focus / 5GS

8.32.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to start a conference call }
    then { UE sends INVITE to the conference factory and completes the conference call initiation
and subscribes to conference event }
            }
```

(2)

```
with { Conference call going on }
ensure that {
  when { UE is made to invite another user to the conference call }
    then { UE sends REFER to the conference focus }
            }
```

(3)

```
with { UE having invited another user to the conference call }
ensure that {
  when { UE receives 202 Accepted followed by notification messages for the REFER request, the
confirmation on the other user and conditional conference event package }
    then { UE sends 200 OK for each received NOTIFY request }
            }
```

(4)

```
with { Conference call going on }
ensure that {
  when { UE is made to leave the call }
    then { UE sends BYE and processes notification for conf event if any }
            }
```

8.32.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.147, clause 5.3.1.5.3]:

Upon generating a REFER request in accordance with the procedures specified in 3GPP TS 24.229 [5], IETF RFC 3515 [17] as updated by IETF RFC 6665 [10] and IETF RFC 7647 [39] that is destined to the conference focus in order to invite another user to a specific conference, the conference participant shall:

1) set the request URI of the REFER request to the conference URI to which the user is invited to;

2) set the Refer-To header of the REFER request to the SIP URI or tel URL of the user who is invited to the conference;

3) either include the "method" URI parameter with the value "INVITE" or omit the "method" URI parameter in the Refer-To header; and

NOTE: Other headers of the REFER request will be set in accordance with 3GPP TS 24.229 [5].

4) send the REFER request towards the conference focus that is hosting the conference.

The UE may additionally include the Referred-By header to the REFER request and set it to the URI of the conference participant that is sending the REFER request.

In case of an active session the UE may additionally include the Replaces header in the header portion of the SIP URI of the Refer-to header field of the REFER request. If the user involved in the active session is identified by a tel URI, the UE shall convert the tel URI to an SIP URI as described in RFC 3261 [7] before including the Replaces header field. The included Replaces header field shall refer to the active dialog that is replaced by the ad-hoc conference. The Replaces header field shall comply with RFC 3891 [33].

Afterwards the UE shall treat incoming NOTIFY requests that are related to the previously sent REFER request in accordance with RFC 3515 [17] as updated by RFC 6665 [10] and may indicate the received information to the user.

8.32.3 Test description

8.32.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.
- SS has performed AKAv1-MD5 authentication with the UE and accepted the registration.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1) and registered to IMS.

8.32.3.2 Test procedure sequence

Table 8.32.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is made to attempt an IMS Conference call	-	-	-	-
2	Check: Does the UE send INVITE with the first SDP offer?	-->	INVITE	1	P
3-8	Steps 2-7 of annex A.4.1 happen	-	-	-	-
9	The SS responds INVITE with 200 OK and gives the final conference URI within the response	<--	200 OK	-	-
10	The UE acknowledges the receipt of 200 OK for INVITE	-->	ACK	-	-
11	EXCEPTION: steps 11 – 14 describe optional behaviour depending on UE configuration. The SS shall wait up to 3s for the SUBSCRIBE of step 11	-	-	-	-
12	UE subscribes the conference event	-->	SUBSCRIBE	-	-
13	SS responds to the subscription	<--	200 OK	-	-
14	The UE is made to invite another user to the conference	-	-	-	-
15	The UE sends REFER to SS referring to the conference	-->	REFER	2	P
16	The SS responds with a 202 final response	<--	202 Accepted	-	-
17	The SS sends initial NOTIFY for the implicit subscription created by the REFER request	<--	NOTIFY	-	-
18	The UE responds the NOTIFY with 200 OK	-->	200 OK	3	P
19	The SS sends a NOTIFY related to REFER request to confirm that the invited user was able to join the conference	<--	NOTIFY		
20	The UE responds the NOTIFY with 200 OK	-->	200 OK	3	P
21	Optional: If the UE has subscribed the conference event package, the SS sends a NOTIFY for conference event package to inform that the invited user was able to join the conference	<--	NOTIFY	-	-
22	Optional: The UE responds the NOTIFY with 200 OK	-->	200 OK	3	P
23	The UE is made to leave the conference	-	-	-	-
24	The UE sends BYE to ler leaving the conference		BYE	4	P
25	The SS sends 200 OK for BYE	<--	200 OK	-	-
26	If the UE had subscribed to the conference event package, the SS notifies the UE that its subscription to conference event package is terminated	-->	NOTIFY	-	-
27	The UE sends 200 OK for NOTIFY (if sent by SS)	-->	200 OK	-	-

8.32.3.3 Specific message contents

Table 8.32.3.3-1: INVITE (step 2, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Step 2 of C.21, Condition A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line				
Request-URI		*sip:mmtel@conf-factory* appended with px_IMS_HomeDomainName		
To				
addr-spec		*sip:mmtel@conf-factory* appended with px_IMS_HomeDomainName		

Table 8.32.3.3-2: 200 OK for INVITE (step 9, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Condition A22				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Record-Route				
rec-route		Same value as in the 183 response		
Contact				
addr-spec		*sip:final@conf-factory* appended with px_IMS_HomeDomainName		
feature-param		*isfocus*		

Table 8.32.3.3-3: ACK (step 10, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Step 2 of C.21, Condition A28				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line Request-URI		 *sip:final@conf-factory* appended with px_IMS_HomeDomainName		

Table 8.32.3.3-4: SUBSCRIBE for conference event package (step 11, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.7, Condition A1

Table 8.32.3.3-5: 200 OK for SUBSCRIBE (step 12, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A5.2

Table 8.32.3.3-6: REFER (step 15, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.10, Condition A5				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line Request-URI		 *sip:final@conf-factory* appended with px_IMS_HomeDomainName		
Refer-To addr-spec		 SIP URI or tel URI of the user invited to the conference. If an active session exists, the Replaces header in the header portion of the SIP URI shall be included (mandatory inclusion is stated in IR.92) and set to the dialog ID of the active session according to RFC 3891. In this case, if the user has been invited with a tel URI, the UE shall convert the tel URI to a SIP URI according to RFC 3261 [6] clause 19.1.6. (NOTE: the dialog ID is percent encoded according to RFC 3986).		
To addr-spec tag		 remote SIP URI as used in To header in step 12 remote tag of the dialog with the conference focus created in step 12		
Route Route-param		 URIs of the Record-Route header of 183 response sent in step 4 (step 3 of A.4.1)		

Table 8.32.3.3-7: 202 Accepted (step 16, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.3

Table 8.32.3.3-8: NOTIFY (step 17, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		*SIP/2.0 100 Trying*		

Table 8.32.3.3-9: 200 OK (step 18/20/22/27, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Condition A22

Table 8.32.3.3-10: NOTIFY (step 19, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Subscription-State substate-value expires reason		 *terminated* omitted from the request *noresource*		
Message-body		*SIP/2.0 100 Trying*		

Table 8.32.3.3-11: NOTIFY (step 21, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		*<?xml version="1.0" encoding="UTF-8"?>* *<conference-info xmlns="urn:ietf:params:xml:ns:conference-info">* *entity="sip:final@conf-factory.* appended with px_IMS_HomeDomainName*"* *state="partial"* *version="* value as in previous notification for conference event package but incremented by one*"* *<users>* *<user entity="* **SIP URI or tel URI of the invited user***">* *<endpoint entity="* **Contact URI of the invited user***">* *<status>connected</status>* *<joining-method>dialed-in</joining-method>* *<media id="1">* *<type>audio</type>* *<label>* **unique identifier for the media stream between the focus and the endpoint of the invited user (e.g. 11223)***</label>* *<src-id>***random SSRC value***</src-id>* *<status>sendrecv</status>* *</media>* *</endpoint>* *</users>* ***</conference-info>***		

Table 8.32.3.3-12: BYE (step 24, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.8, Condition A8				
Request-Line Request-URI		 *sip:final@conf-factory* appended with px_IMS_HomeDomainName		

Table 8.32.3.3-13: NOTIFY for conference event package (step 17, table 8.32.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.5.3, Condition A4

8.33

8.34 Three way session creation / 5GS

8.34.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and having set up an MO voice call with A }
ensure that {
  when { UE is made to start a three way voice call }
    then { UE sends re-INVITE or UPDATE, and completes the call hold procedure with A }
            }
```

(2)

```
with { UE being in the process of starting a three way voice call }
ensure that {
  when { UE having put A on hold  }
    then { UE initiates a voice call with B }
            }
```

(3)

```
with { UE being in the process of starting a three way voice call }
ensure that {
  when { UE having initiated a voice call with B }
    then { UE sends re-INVITE or UPDATE, and completes the call hold procedure with B }
            }
```

(4)

```
with { UE being in the process of starting a three way voice call }
ensure that {
  when { UE having put both A and B on hold }
    then { UE sends INVITE to the conference factory and completes the conference call initiation 
and subscribes to conference event }
            }
```

(5)

```
with { UE being in the process of starting a three way voice call }
ensure that {
  when { UE having created a call at the conference factory }
    then { UE sends REFER to the conference focus in order to invite A }
            }
```

(6)

```
with { UE having invited A to the conference call }
ensure that {
  when { UE receives 202 Accepted followed by notification messages for the REFER request, the 
confirmation on A and conditional conference event package }
    then { UE sends 200 OK for each received NOTIFY request }
            }
```

(7)

```
with { UE being in the process of starting a three way voice call }
ensure that {
  when { UE having completed the invitation of A }
    then { UE sends REFER to the conference focus in order to invite B }
            }
```

(8)

```
with { UE having invited B to the conference call }
ensure that {
  when { UE receives 202 Accepted followed by notification messages for the REFER request, the 
confirmation on B and conditional conference event package }
    then { UE sends 200 OK for each received NOTIFY request }
            }
```

8.34.2 Conformance Requirements

When a user is participating in two or more SIP sessions and wants to join together two of these active sessions to a so-called three-way session, the user shall perform the following steps.

1) create a conference at the conference focus by sending an INVITE request with the conference factory URI for the three-way session towards the conference focus, as described in subclause 5.3.1.3.2;

2) decide and perform for each of the active sessions that are requested to be joined to the three-way session, how the remote user shall be invited to the three-way session, which can either be:

   a) by performing the procedures for inviting a user to a conference by sending an REFER request to the user, as described in subclause 5.3.1.5.2; or

   b) by performing the procedures for inviting a user to a conference by sending a REFER request to the conference focus, as described in subclause 5.3.1.5.3;

3) release the active session with the user, by applying the procedures for session release in accordance with RFC 3261 [7], provided that a BYE request has not already been received, after a NOTIFY request has been received, indicating that the user has successfully joined the three-way session, i.e. including:

   a) a body of content-type "message/sipfrag" that indicates a "200 OK" response; and,

   b) a Subscription-State header set to the value "terminated"; and,

4) treat the created three-way session as a normal conference, i.e. the conference participant shall apply the applicable procedures of subclause 5.3.1 for it.

Reference(s)

3GPP TS 24.147 [35] clause 5.3.1.3.3.

8.34.3 Test description

8.34.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.

- UE is configured to register for IMS after switch on.

Preamble:

- The UE has registered to IMS and set up the MO voice call, by executing the generic test procedure in Annex A.2 up to the last step and thereafter executing the generic test procedure in A.4.1.

8.34.3.2 Test procedure sequence

Table 8.34.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-4	Steps 1-4 of A.17 are used to put the first call on hold.		Messages in Annex A.17	1	P
5	UE is made to initiate a MTSI voice call.				
6-17	Steps 1-12 of A.4.1 are used to start a second call.		Messages in in Annex A.4.1	2	P
18	UE is made to start a Multiparty Call				
19-22	Steps 1-4 of A.17 are used to put the second call on hold.		Messages in Annex A.17	3	P
23-35	Steps 1-13 of A.19 are used to create a conference.		Messages in Annex A.19	4	P
36	UE sends REFER to invite user A to the conference. (Step 1 of A.20)	->	REFER	5	P
37-38	SS responds with a 202 final response and NOTIFY for the subscription created by the REFER. (Steps 2-3 of A.20)	-	Messages in Annex A.20		
39	The UE responds the NOTIFY request with 200 OK. (Step 4 of A.20)	->	200 OK	6	P
40	SS responds with a NOTIFY to confirm the user the invited user was able to join the conference. (Steps 5 of A.20)	-	NOTIFY		
41	UE responds the NOTIFY request with 200 OK (Steps 6 of A.20)	->	200 OK	6	P
42-43	Conditional: If the UE has subscribed the conference event package, then the SS sends a NOTIFY for conference event package and UE responds with 200 OK. (Steps 7-8 of A.20)	-	Messages in Annex A.20		
44	UE sends REFER to invite user B to the conference. (Step 1 of A.20)	->	REFER	7	P
45-46	SS responds with a 202 final response and NOTIFY for the subscription created by the REFER. (Steps 2-3 of A.20)	-	Messages in Annex A.20		
47	The UE responds the NOTIFY request with 200 OK. (Step 4 of A.20)	->	200 OK	8	P
48	SS responds with a NOTIFY to confirm the user the invited user was able to join the conference. (Steps 5 of A.20)	-	NOTIFY		
49	UE responds the NOTIFY request with 200 OK (Steps 6 of A.20)	->	200 OK	8	P
50-51	Conditional: If the UE has subscribed the conference event package, then the SS sends a NOTIFY for conference event package and UE responds with 200 OK. (Steps 7-8 of A.20)	-	Messages in Annex A.20		
52-53	Steps 1-2 of A.7 are used to release the first call.		Messages in Annex A.7		
54-55	Steps 1-2 of A.7 are used to release the second call.		Messages in Annex A.7		
56-57	Steps 1-2 of A.8 are used to release the active session.		Messages in Annex A.7		
58	Conditional: If the UE has subscribed the conference event package, then the SS notifies the UE that its subscription to conference event package is terminated	<-	NOTIFY		
59	Conditional: If the SS sent NOTIFY, then the UE sends 200 OK for NOTIFY. (Steps 8 of A.20)	->	200 OK		

8.34.3.3 Specific message contents

Table 8.34.3.3-1: INVITE (Step 6, table 8.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.1, Conditions A1, A3, A5 and A28.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line				
Request-URI		px_IMS_CalleeUri2 px_IMS_CalleeUri2 is used to invite another user to the session. px_IMS_CalleeUri2 may be either SIP or Tel URI. It may contain a dialstring and phone-context parameter, when calling to dialstring. When calling to dialstring SIP URI must also contain user=phone or user=dialstring parameter. The dialstring, if used, may be global, home local number or geo-local number. For home local numbers the value of phone-context parameter must equal the home domain name i.e. px_IMS_HomeDomainName. For geo-local numbers the home domain name must be prefixed by string “geo-local.” or access technology specific prefix, if the UE supports that option. Note: The way how the UE determines whether numbers in a non-international format are geo-local, home-local or relating to another network, is UE implementation specific. For instance the UE might have a UI setting.		
To				
addr-spec		px_IMS_CalleeUri2		

Table 8.34.3.3-2: 183 Session in Progress for INVITE (Step 8, table 8.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.3, Condition A1.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

Table 8.34.3.3-3: 200 OK for INVITE (Step 11, table 8.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A1, A10 and A19.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

Table 8.34.3.3-4: 180 Ringing for INVITE (Step 13, table 8.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.2.6, Condition A1.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

Table 8.34.3.3-5: NOTIFY for conference event package (Step 58, table 8.34.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.5.3, Conditions A1 ans A4.				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

8.35

8.36 MO Explicit Communication Transfer / Consultative Call Transfer / 5GS

8.36.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and having established a voice call with A (the transferee) }
ensure that {
  when { UE is being made to attempt Consultative Call Transfer }
    then { UE puts A on hold and sets up voice call with B (the transfer target) and puts B on hold
and sends REFER to the transferee }
            }
```

(2)

```
with { UE having initiated consultative call transfer }
ensure that {
  when { UE receives NOTIFY }
    then { UE sends 200 OK response for NOTIFY }
            }
```

(3)

```
with { UE having processed the NOTIFY exchange }
ensure that {
  when { UE receives instruction to be put on hold by A }
    then { UE processes call hold instruction and responds to it }
            }
```

(4)

```
with { UE having been put on hold by A }
ensure that {
  when { UE receives BYE from B }
    then { UE sends 200 OK for BYE }
            }
```

(5)

```
with { Call with B having ended }
ensure that {
  when { UE receives NOTIFY from A }
    then { UE sends 200 OK for NOTIFY and may send BYE }
            }
```

8.36.2 Conformance Requirements

[TS 24.629, clause 4.5.2.1]:

A UE that has initiated an emergency call, shall not perform any transfer operation involving the dialog associated with the emergency call.

A UE that initiates a transfer operation shall if the Contact address of the transferee is a GRUU:

- issue a REFER outside an existing dialog as specified in RFC 3515 [2] as updated by IETF RFC 6665 [14] and IETF RFC 7647 [16], where:

  a) the request URI shall contain the SIP URI of the transferee as received in the Contact header field:

  b) the Refer-To header field shall indicate the public address of the transfer target;

  c) in case of Consultative transfer, the transferor UE has a consultation communication with the transfer target, a Replaces header field parameter shall be added to the Refer-To URI together with a Require=replaces header field parameter;

d) the Referred-By header field can be used to indicate the identity of the transferor. When privacy was required in the original communications dialog and a Referred-By header field is included, the UE shall include a Privacy header field set to "user"; and

e) the Target-Dialog header field identifies the dialog to be transferred;

otherwise the UE shall:

- issue a REFER request in the original communications dialog as specified in RFC 3515 [2], where:

a) the request URI shall contain the SIP URI of the transferee as received in the Contact header field;

b) the Refer-To header field shall indicate the public address of the transfer target;

c) in case of consultative transfer, the transferor UE has a consultation communication with the transfer target, a Replaces header field parameter shall be added to the Refer-To URI together with a Require=replaces header field parameter; and

d) the Referred-By header field can be used to indicate the identity of the transferor. When privacy was required in the original communications dialog and a Referred-By header field is included, the UE shall include a Privacy header field set to "user".

If assured transfer is requested, the UE may include an Expires header field in the Refer-To URI of the REFER request.

NOTE 1: The value of the Expires header field indicates the maximum duration of the transfer attempt. If the transfer does not succeed within this duration, the UE will receive a NOTIFY request indicating the transfer failure.

After the REFER request is accepted by the other end with a 2xx response, the transferor UE gets notifications of how the transferee's communication setup towards the transfer Target is progressing.

When a NOTIFY request is received on the REFER dialog that indicates that the transferee and the transfer Target have successfully setup a communication, the transferor UE may terminate the original communication with the transferee UE, by sending a BYE request on the original dialog.

If an assured transfer attempt is not completed (i.e. the UE has not received a NOTIFY request with a "message/sipfrag" body’s status line containing a final response code indicating the end of the transfer operation), the UE may request to terminate the transfer attempt by:

- sending a REFER request in the same communications dialog as the previous REFER request as specified in RFC 3515 [2] as updated by IETF RFC 6665 [14] and IETF RFC 7647 [16], where:

a) the request URI shall contain the SIP URI of the transferee as received in the Contact header field; and

b) the Refer-To header field shall indicate the public address of the transfer target and shall contain the method parameter set to "CANCEL"; and

c) if applicable include a Target-Dialog header field that identifies the dialog under transfer.

If the UE receives a NOTIFY request indicating that the assured transfer attempt failed, followed by a re-INVITE or an UPDATE request taking the UE off HOLD the UE may decide to retrieve the original communication by sending a re-INVITE request in the original SIP dialog.

NOTE 2: If the user requests the retrieval of the original communication while the transfer attempt has not been completed, the UE needs to first request the termination of the transfer attempt before retrieving the original communication via a re-INVITE request.

Reference(s)

3GPP TS 24.629 [36], clause 4.5.2.1.

8.36.3 Test description

8.36.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- UE is configured to use preconditions.

Preamble:

- The UE has registered to IMS and set up the MO call, by executing the generic test procedure in Annex A.2 up to the last step and thereafter executing the generic test procedure in A4.1.
- The SS has accepted the UE's MO call.

8.36.3.2 Test procedure sequence

Table 8.36.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	The UE is made to attempt Consultative Call Transfer	-	-	-	-
2	Check: Does the UE send INVITE or UPDATE with a SDP offer to hold the call with A by running step 1 of A.17 for MO Call Hold?	-->	INVITE or UPDATE	1	P
3-5	Remaining steps 2-4 of A.17 for MO Call Hold happen.	-	-	-	-
6	Void	-	-	-	-
7	Check: Does the UE initiate a voice call with B by exercising step 1 of Annex A.4.1?		INVITE	1	P
8-18	Steps 2-12 of A.4.1 happen	-	-	-	-
19	Void	-	-		
20	Check: Does the UE send INVITE or UPDATE with a SDP offer to hold the call with B by running step 1 of A.17 for MO Call Hold?	-->	INVITE or UPDATE	1	P
21-23	Remaining steps 2-4 of A.17 for MO Call Hold happen	-	-	-	-
24	Check: Does the UE send REFER to SS, simulating the transferee, referring to the transfer target	-->	REFER	1	P
25	The SS responds to REFER with 200 OK	<--	200 OK		
26	The SS, simulating the transferee, sends initial NOTIFY for the implicit subscription created by the REFER request	<--	NOTIFY		
27	The UE responds to NOTIFY with 200 OK	-->	200 OK	2	P
28-31	The SS, simulating the transferee, puts the UE on hold by executing the MT Call Hold procedure of Annex A.18, but setting the direction attribute to inactive.	-	-	3	P
32	The SS, simulating the transfer target, releases the call between UE and the transfer target with BYE	<--	BYE	-	-
33	The UE responds to BYE with 200 OK	-->	200 OK	4	P
34	The SS, simulating the transferee, sends a NOTIFY request to confirm that the call transfer has been completed	<--	NOTIFY	-	-
35	The UE responds to NOTIFY with 200 OK	-->	200 OK	-	-
36	Optional: UE may send a BYE request to release call with the transferee	-->	BYE	-	-
37	If the UE has sent BYE in step 33 then SS sends 200 OK for BYE	-->	200 OK	5	P

8.36.3.3 Specific message contents

Table 8.36.3.3-1: INVITE (step 7, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 1 in A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line				
Request-URI		px_IMS_CalleeUri2		
To				
addr-spec		px_IMS_CalleeUri2		

Table 8.36.3.3-2: 183 Session Progress (step 9, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 3 in A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

Table 8.36.3.3-3: 180 Ringing (step 14, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 8 in A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

Table 8.36.3.3-4: 200 OK for INVITE (step 17, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 11 in A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Contact				
addr-spec		px_IMS_CalleeContactUri2		

Table 8.36.3.3-5: INVITE/UPDATE (step 20, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 1 in A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Request-Line				
Request-URI		px_IMS_CalleeUri2		

Table 8.36.3.3-6: REFER (step 24, table 8.36.3.2-1)

Derivation Path: TS 34.229-1 [2], Step 1 in A.2.10				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Refer-To				
value		<public address of transfer target?Replaces=(dialog id of the dialog between the UE and the transfer target)&Require=replaces>		
Referred-By				
value		same value as addr-spec field in From header in the first INVITE during initial call setup, if header present		
Privacy				
value		user (shall be included if privacy was required during original communication dialog and Referred-By header field is included)		

Table 8.36.3.3-7: NOTIFY (step 26, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 1 in A.4.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		*SIP/2.0 100 Trying*		

Table 8.36.3.3-8: 200 OK for NOTIFY (step 27, table 8.36.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A11 and A22

Table 8.36.3.3-9: INVITE (step 28, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 1 of A.19				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Each media line carries direction attribute “a=inactive”		

Table 8.36.3.3-10: 200 OK for re-INVITE (step 30, table 8.36.3.2-1)

Derivation Path: TS 34.229-5, Step 4 of A.19				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		Each media line carries direction attribute “a=inactive”		

Table 8.36.3.3-11: NOTIFY (step 34, table 8.36.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.2.11				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Subscription-State				
Substate-value		*terminated*		
expires		omitted from request		
reason		*noresource*		
Message-body		*SIP/2.0 200 OK*		

Table 8.36.3.3-12: 200 OK for NOTIFY (step 35, table 8.36.3.2-1)

Derivation Path: TS 34.229-1 [2], Annex A.3.1, Conditions A11 and A22

8.37

8.38 Communication Waiting and cancelling the call / 5GS

8.38.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS and having initiated an MO voice call with preconditions }
ensure that {
  when { UE receives INVITE for MT voice call with preconditions }
   then { UE continues voice call initiation until 180 Ringing (including conditional PRACK/200 OK)
}
            }
```

(2)

```
with { UE having continued initiation of incoming voice call until 180 Ringing }
ensure that {
  when { UE receives CANCEL for incoming voice call }
   then { UE responds with 200 OK and 487 Request Terminated responses }
            }
```

8.38.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.615 clause 1]:

The **Communication Waiting (CW)** service enables a user to be informed, that very limited resources are available for an incoming communication. The user then has the choice of accepting, rejecting or ignoring the waiting call (as per basic call procedures).

[TS 24.615 clause 4.2.1]:

When a communication arrives at the destination user, the UE validates the status of the user. If the user is already involved in one or more communications, the terminal notifies the served user of a communication waiting situation.

[TS 24.615 clause 4.5.5.3.2]:

The UE shall insert an Alert-Info header field set to "<urn:alert:service:call-waiting>", specified in RFC 7462 [8] in the 180 (Ringing) response, in accordance with the provisional response procedures described in 3GPP TS 24.229.

[TS 24.615 clause 4.5.5.3.4]:

If user B's UE receives a CANCEL request or BYE request from User C during a CW condition, user B's UE shall:

- stop timer $T_{UE\text{-}CW}$ (if necessary);
- stop providing the CW indication to User B; and
- apply the terminating UE procedures upon receipt of CANCEL or BYE as described in 3GPP TS 24.229.

8.38.3 Test description

8.38.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- The UE contains either ISIM and USIM applications or only USIM application on UICC.
- The UE is configured to register for IMS after switch on.

- The UE is configured to use preconditions.

Preamble:

- UE is in state 1N-A, registered to IMS and has set up an MO call with preconditions, by executing the generic test procedure in Table 4.9.15.2.2-1 of TS 38.508-1.

8.38.3.2 Test procedure sequence

Table 8.38.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1-7	MT Voice call setup takes place according to steps 1-7 of Annex A.5.1	-	-	-	-
8	Step 8 of Annex A.5.1 happens	-->	180 Ringing	1	P
9	Conditional Step: if UE sent 180 Ringing reliably, Step 9 of Annex A.5.1 happens	<--	PRACK	-	-
10	Conditional Step: if UE sent 180 Ringing reliably, Step 10 of Annex A.5.1 happens	-->	200 OK	-	-
11	SS sends CANCEL request to terminate INVITE transaction	<--	CANCEL	-	-
12	UE acknowledges CANCEL with 200 OK	-->	200 OK	-	-
13	The UE responds to INVITE with a 487 Request Terminated final response after transaction was terminated.	-->	487 Request Terminated	2	P
14	SS acknowledges the receipt of 487 Request Terminated	<--	ACK	-	-

8.38.3.3 Specific message contents

Table 8.38.3.3-1: 180 Ringing (step 8, table 8.38.3.2-1)

Derivation path: Step 8 of Annex A.5.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Alert-Info		<urn:alert:service:call-waiting>		RFC 7462 [39]

Table 8.38.3.3-2: CANCEL (step 11, table 8.38.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.2.15

Table 8.38.3.3-3: 200 OK (step 12, table 8.38.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.3.1 applying conditions A5 and A11

Table 8.38.3.3-4: 487 Request Terminated (step 13, table 8.38.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.2.16

Table 8.38.3.3-5: ACK (step 14, table 8.38.3.2-1)

Derivation path: TS 34.229-1 [2], Table in subclause A.2.7 applying conditions A2 and A4

9 SMS

9.1 Mobile Originating SMS / 5GS

9.1.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to send an SMS over IP }
    then { UE sends a SIP MESSAGE request containing a short message }
            }
```

(2)

```
with { UE having sent a SIP MESSAGE request containing a short message }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing a
submission report  }
    then { UE sends a 200 OK response }
            }
```

(3)

```
with { UE having sent a 200 OK response for submission report }
ensure that {
  when { UE receives a SIP MESSAGE request containing a status report }
    then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing a delivery
report for status report }
            }
```

9.1.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.341, clause 5.3.1.2]:

When an SM-over-IP sender wants to submit an SM over IP, the SM-over-IP sender shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the PSI of the SC of the SM-over-IP sender;

NOTE 1: The PSI of the SC can be SIP URI or tel URI based on operator policy. The PSI of the SC can be obtained using one of the following methods in the priority order listed below:

1) provided by the user;

2) if UICC is used, then:

- if present in the ISIM, then the PSI of the SC is obtained from the $EF_{PSISMSC}$ in DF_TELECOM of the ISIM as per 3GPP TS 31.103 [18];

- if not present on the ISIM, then the PSI of the SC is obtained from the $EF_{PSISMSC}$ in DF_TELECOM of the USIM as per 3GPP TS 31.102 [19]; or

- if neither present on the ISIM nor on the USIM, then the PSI of the SC contains the TS-Service-Centre-Address stored in the EF_{SMSP} in DF_TELECOM as per 3GPP TS 31.102 [19]. If the PSI of the SC is based on the E.164 number from the TS-Service-Centre-Address stored in the EF_{SMSP} in DF_TELECOM then the URI constructed can be either a tel URI or a SIP URI (using the "user=phone" SIP URI parameter format).

3) if SIM is used instead of UICC, then the PSI of the SC contains the TS-Service Centre Address stored in the EF_{SMSP} in DF_TELECOM as per 3GPP TS 51.011 [20]. If the PSI of the SC is based on the

E.164 number from the TS-Service-Centre-Address stored in the EF_{SMSP} in DF_TELECOM then the URI constructed can be either a tel URI or a SIP URI (using the "user=phone" SIP URI parameter format); or

4) if neither the UICC nor SIM is used, then how the PSI of the SC is configured and obtained is through means outside the scope of this specification.

b) the From header, which shall contain a public user identity of the SM-over-IP sender;

NOTE 2: The IP-SM-GW will have to use an address of the SM-over-IP sender that the SC can process (i.e. an E.164 number). This address will come from a tel URI in a P-Asserted-Identity header (as defined in RFC 3325 [13]) placed in the SIP MESSAGE request by the P-CSCF or S-CSCF.

NOTE 3: The SM-over-IP sender has to store the Call-ID of the SIP MESSAGE request, so it can associate the appropriate SIP MESSAGE request including a submit report with it.

c) the To header, which shall contain the SC of the SM-over-IP sender;

d) the Content-Type header, which shall contain "application/vnd.3gpp.sms"; and

e) the body of the request shall contain an RP-DATA message as defined in 3GPP TS 24.011 [8], including the SMS headers and the SMS user information encoded as specified in 3GPP TS 23.040 [3].

NOTE 4: The address of the SC is included in the RP-DATA message content. The address of the SC included in the RP-DATA message content is stored in the EF_{SMSP} in DF_TELECOM of the (U)SIM of the SM-over-IP sender.

NOTE 5: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

NOTE 6: Both the address of the SC and the PSI of the SC can be configured in the $EF_{PSISMSC}$ in DF_TELECOM of the USIM and ISIM respectively using the USAT as per 3GPP TS 31.111 [21].

The SM-over-IP sender may request the SC to return the status of the submitted message. The support of status report capabilities is optional for the SC.

When a SIP MESSAGE request including a submit report in the "vnd.3gpp.sms" payload is received, the SM-over-IP sender shall:

- if SM-over-IP sender supports In-Reply-To header usage and the In-Reply-To header indicates that the request corresponds to a short message submitted by the SM-over-IP sender, generate a 200 (OK) SIP response according to RFC 3428 [14].

  if SM-over-IP sender supports In-Reply-To header usage and the In-Reply-To header indicates that the request does not correspond to a short message submitted by the SM-over-IP sender, a 488 (Not Acceptable here) SIP response according to RFC 3428 [14].

- if SM-over-IP sender does not support In-Reply-To header usage, generate a 200 (OK) SIP response according to RFC 3428 [14]; and extract the payload encoded according to 3GPP TS 24.011 [8] for RP-ACK or RP-ERROR.

[TS 24.341 clause 5.3.1.3]:

When a SIP MESSAGE request including a status report in the "vnd.3gpp.sms" payload is delivered, the SM-over-IP sender shall:

- generate a SIP response according to RFC 3428 [14];
- extract the payload encoded according to 3GPP TS 24.011 [8] for RP-DATA; and
- create a delivery report for the status report as described in subclause 5.3.2.4. The content of the delivery report is defined in 3GPP TS 24.011 [8].

[TS 24.341 clause 5.3.2.4]:

When an SM-over-IP receiver wants to send an SM delivery report over IP, the SM-over-IP receiver shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the IP-SM-GW;

NOTE 1: The address of the IP-SM-GW is received in the P-Asserted-Identity header in the SIP MESSAGE request including the delivered short message.

b) the From header, which shall contain a public user identity of the SM-over-IP receiver.

c) the To header, which shall contain the IP-SM-GW;

b) the Content-Type header shall contain "application/vnd.3gpp.sms"; and

c) the body of the request shall contain the RP-ACK or RP-ERROR message for the SM delivery report, as defined in 3GPP TS 24.011 [8].

NOTE 2: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

9.1.3 Test description

9.1.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

9.1.3.2 Test procedure sequence

Table 9.1.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to attempt a Mobile Originating SMS over IMS	-	-	-	-
1A-1F	Steps 2-7 of generic procedure specified in Table 4.9.19.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
2	Check: Does UE send a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a short message?	-->	SIP MESSAGE	1	P
3	SS responds with 202 Accepted	<--	202 Accepted	-	-
4	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the short message submission report indicating a positive acknowledgement of the short message sent by the UE at Step 2	<--	SIP MESSAGE	-	-
5	Check: Does UE respond with 200 OK?	-->	200 OK	2	P
6	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a status report	<--	SIP MESSAGE	-	-
7	Check: Does UE respond with 200 OK?	-->	200 OK	3	P
8	Check: Does UE send a SIP MESSAGE request including a vnd.3gpp.sms payload that contains an acknowledgement for the status report received at Step 6?	-->	SIP MESSAGE	3	P
9	SS responds with 202 Accepted	<--	202 Accepted	-	-

9.1.3.3 Specific message contents

Table 9.1.3.3-1: SIP MESSAGE (step 2, table 9.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.3, Condition A5

Table 9.1.3.3-2: 202 Accepted (step 3 and 9, table 9.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

Table 9.1.3.3-3: SIP MESSAGE (step 4, table 9.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.4

Table 9.1.3.3-4: 200 OK (step 5 and 7, table 9.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A5 and A22

Table 9.1.3.3-5: SIP MESSAGE (step6, table 9.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.5

Table 9.1.3.3-6: SIP MESSAGE (step 8, table 9.1.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.6

9.2 Mobile Terminating SMS / 5GS

9.2.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE receives a SIP MESSAGE request containing a short message }
    then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing a delivery
report }
            }
```

9.2.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.341, clause 5.3.2.3]

When a SIP MESSAGE request including a short message in the "vnd.3gpp.sms" payload is delivered, the SM-over-IP receiver shall:

- generate a SIP response according to RFC 3428;
- extract the payload encoded according to 3GPP TS 24.011 for RP-DATA; and
- create a delivery report as described in subclause 5.3.2.4. The content of the report is defined in 3GPP TS 24.011.

[TS 24.341, clause 5.3.2.4]

When an SM-over-IP receiver wants to send an SM delivery report over IP, the SM-over-IP receiver shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the IP-SM-GW;

NOTE 1: The address of the IP-SM-GW is received in the P-Asserted-Identity header in the SIP MESSAGE request including the delivered short message.

b) the From header, which shall contain a public user identity of the SM-over-IP receiver.

c) the To header, which shall contain the IP-SM-GW;

b) the Content-Type header shall contain "application/vnd.3gpp.sms"; and

c) the body of the request shall contain the RP-ACK or RP-ERROR message for the SM delivery report, as defined in 3GPP TS 24.011 [8].

NOTE 2: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

9.2.3 Test description

9.2.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

9.2.3.2 Test procedure sequence

Table 9.2.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.20.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
1	The SS sends a Short Message.	<--	SIP MESSAGE	-	-
2	Check: Does the UE send a 200 OK response?	-->	200 OK	1	P
3	Check: Does the UE respond with a delivery report?	-->	SIP MESSAGE	1	P
4	The SS sends a 202 ACCEPTED response.	<--	202 ACCEPTED	-	-

9.2.3.3 Specific message contents

Table 9.2.3.3-1: SIP MESSAGE (step 1, table 9.2.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.1

Table 9.2.3.3-2: 200 OK (step 2 table 9.2.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A5 and A22

Table 9.2.3.3-3: SIP MESSAGE (step 3, table 9.2.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.2

Table 9.2.3.3-4: 202 Accepted (step 4, table 9.2.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

9.3 Mobile Originating Concatenated SMS / 5GS

9.3.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to send a concatenated SMS over IP }
   then { UE sends a SIP MESSAGE request containing the first segment of the concatenated SMS }
            }
```

(2)

```
with { UE having sent a SIP MESSAGE request containing the first segment of the concatenated SMS }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing a
submission report for the first segment }
   then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing the second
segment of the concatenated SMS }
            }
```

(3)

```
with { UE having sent a SIP MESSAGE request containing the second segment of the concatenated SMS }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing a
submission report for the second segment }
   then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing the third segment
of the concatenated SMS }
            }
```

(4)

```
with { UE having sent a SIP MESSAGE request containing the third segment of the concatenated SMS }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing a
submission report for the third segment }
   then { UE sends a 200 OK response }
            }
```

9.3.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 23.040, clause 9.2.3.23]:

The TP-User-Data-Header-Indicator is a 1 bit field within bit 6 of the first octet of the following six PDUs:

- SMS-SUBMIT,
- SMS-SUBMIT-REPORT,
- SMS-DELIVER,
- SMS-DELIVER-REPORT,
- SMS-STATUS-REPORT,
- SMS-COMMAND.

TP-UDHI has the following values.

Bit no. 6 0 The TP-UD field contains only the short message

1 The beginning of the TP-UD field contains a Header in addition to the short message.

[TS 23.040, clause 9.2.3.24]:

The length of the TP-User-Data field is defined in the PDU's of the SM-TL (see clause 9.2.2).

The TP-User-Data field may comprise just the short message itself or a Header in addition to the short message depending upon the setting of TP-UDHI.

Where the TP-UDHI value is set to 0 the TP-User-Data field comprises the short message only, where the user data can be 7 bit (default alphabet) data, 8 bit data, or 16 bit (UCS2 [24]) data.

Where the TP-UDHI value is set to 1 the first octets of the TP-User-Data field contains a Header in the following order starting at the first octet of the TP-User-Data field.

Irrespective of whether any part of the User Data Header is ignored or discarded, the MS shall always store the entire TPDU exactly as received.

FIELD	LENGTH
Length of User Data Header	1 octet
Information-Element-Identifier "A"	1 octet
Length of Information-Element "A"	1 octet
Information-Element "A" Data	0 to "n" octets
Information-Element-Identifier "B"	1 octet
Length of Information-Element "B"	1 octet
Information-Element "B" Data	0 to "n" octets
Information-Element-Identifier "X"	1 octet
Length of Information-Element "X"	1 octet
Information-Element "X" Data	0 to "n" octets

The diagram below shows the layout of the TP-User-Data-Length and the TP-User-Data for uncompressed GSM 7 bit default alphabet data. The UDHL field is the first octet of the TP-User-Data content of the Short Message.

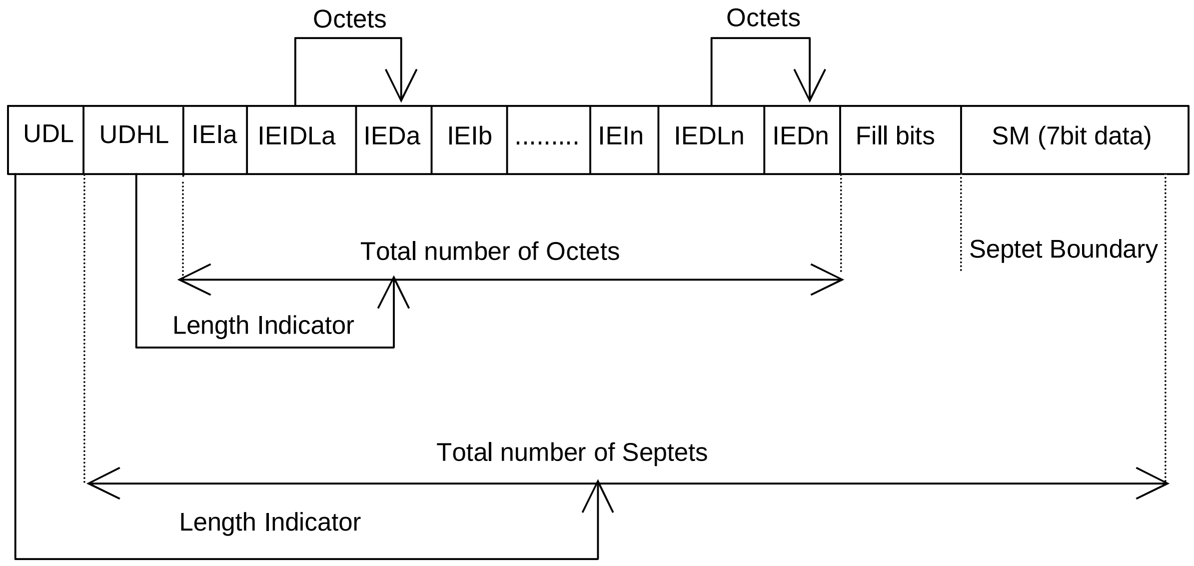

Figure 9.2.3.24 (a)

The diagram below shows the layout of the TP-User-Data-Length and the TP-User-Data for uncompressed 8 bit data or uncompressed UCS2 data. The UDHL field is the first octet of the TP-User-Data content of the Short Message.

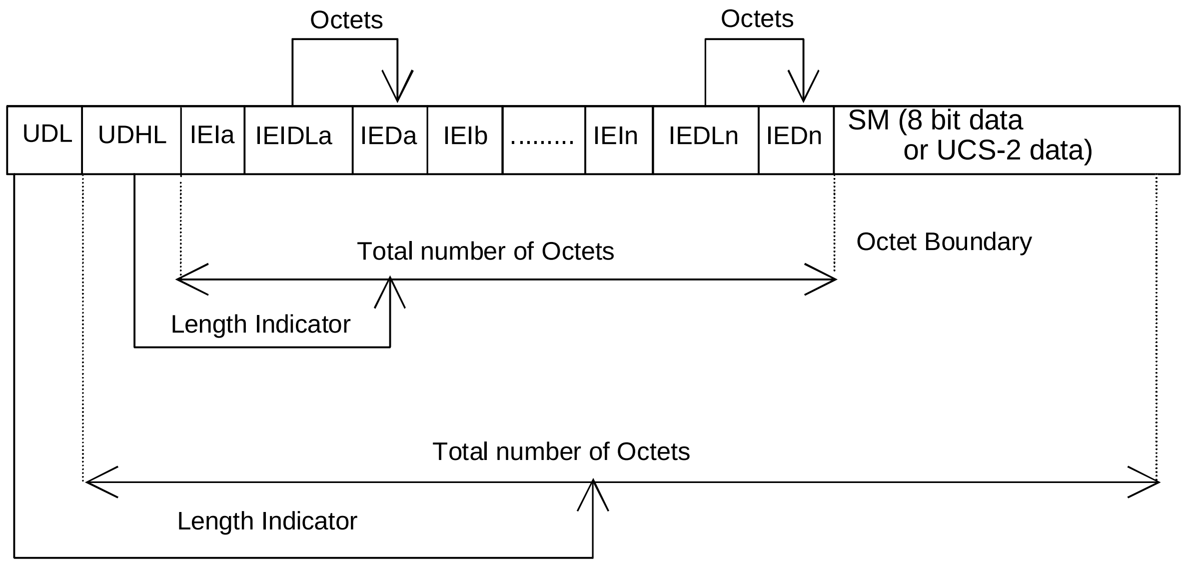

Figure 9.2.3.24 (b)

The diagram below shows the layout of the TP-User-Data-Length and the TP-User-Data for compressed GSM 7 bit default alphabet data, compressed 8 bit data or compressed UCS2 data. The UDHL field is the first octet of the TP-User-Data content of the Short Message.

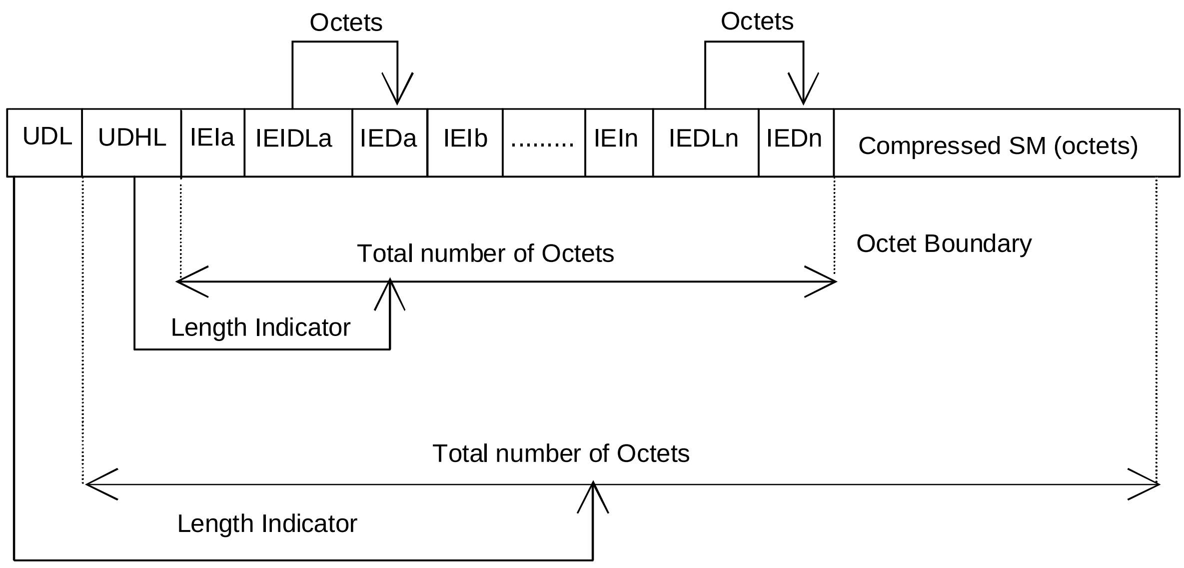

Figure 9.2.3.24 (c)

The definition of the TP-User-Data-Length field which immediately precedes the "Length of User Data Header" is unchanged and shall therefore be the total length of the TP-User-Data field including the Header, if present. (see 9.2.3.16).

The "Length-of-Information-Element" fields shall be the integer representation of the number of octets within its associated "Information-Element-Data" field which follows and shall not include itself in its count value.

The "Length-of-User-Data-Header" field shall be the integer representation of the number of octets within the "User-Data-Header" information fields which follow and shall not include itself in its count or any fill bits which may be present (see text below).

Information Elements may appear in any order and need not follow the order used in the present document. Information Elements are classified into 3 categories as described below.

- SMS Control – identifies those IEIs which have the capability of dictating SMS functionality.
- EMS Control – identifies those IEIs which manage EMS Content IEIs.
- EMS Content – identifies those IEIs containing data of a unique media format.

It is permissible for certain IEs to be repeated within a short message, or within a concatenated message. There is no restriction on the repeatability of IEs in the EMS Content classification. The repeatability of SMS Control and EMS Control IEs is determined on an individual basis. See the IE table below for the repeatability of each IE.

In the event that IEs determined as not repeatable are duplicated, the last occurrence of the IE shall be used. In the event that two or more IEs occur which have mutually exclusive meanings (e.g. an 8bit port address and a 16bit port address), then the last occurring IE shall be used.

If the length of the User Data Header is such that there are too few or too many octets in the final Information Element then the whole User Data Header shall be ignored.

If any reserved values are received within the content of any Information Element then that part of the Information Element shall be ignored.

The support of any Information Element Identifier is optional unless otherwise stated.

The Information Element Identifier octet shall be coded as follows:

VALUE (hex)	MEANING	Classification	Repeatability
00	Concatenated short messages, 8-bit reference number	SMS Control	No
01	Special SMS Message Indication	SMS Control	Yes
02	Reserved	N/A	N/A
03	Value not used to avoid misinterpretation as <LF> character	N/A	N/A
04	Application port addressing scheme, 8 bit address	SMS Control	No
05	Application port addressing scheme, 16 bit address	SMS Control	No
06	SMSC Control Parameters	SMS Control	No
07	UDH Source Indicator	SMS Control	Yes
08	Concatenated short message, 16-bit reference number	SMS Control	No
09	Wireless Control Message Protocol	SMS Control	Note 3
0A	Text Formatting	EMS Control	Yes
0B	Predefined Sound	EMS Content	Yes
0C	User Defined Sound (iMelody max 128 bytes)	EMS Content	Yes
0D	Predefined Animation	EMS Content	Yes
0E	Large Animation (16*16 times 4 = 32*4 =128 bytes)	EMS Content	Yes
0F	Small Animation (8*8 times 4 = 8*4 =32 bytes)	EMS Content	Yes
10	Large Picture (32*32 = 128 bytes)	EMS Content	Yes
11	Small Picture (16*16 = 32 bytes)	EMS Content	Yes
12	Variable Picture	EMS Content	Yes
13	User prompt indicator	EMS Control	Yes
14	Extended Object	EMS Content	Yes
15	Reused Extended Object	EMS Control	Yes
16	Compression Control	EMS Control	No
17	Object Distribution Indicator	EMS Control	Yes
18	Standard WVG object	EMS Content	Yes
19	Character Size WVG object	EMS Content	Yes
1A	Extended Object Data Request Command	EMS Control	No
1B-1F	Reserved for future EMS features (see subclause 3.10)	N/A	N/A
20	RFC 5322 E-Mail Header	SMS Control	No
21	Hyperlink format element	SMS Control	Yes
22	Reply Address Element	SMS Control	No
23	Enhanced Voice Mail Information	SMS Control	No
24	National Language Single Shift	SMS Control	No
25	National Language Locking Shift	SMS Control	No
26 – 6F	Reserved for future use	N/A	N/A
70 – 7F	(U)SIM Toolkit Security Headers	SMS Control	Note 1
80 – 9F	SME to SME specific use	SMS Control	Note 2
A0 – BF	Reserved for future use	N/A	N/A
C0 – DF	SC specific use	SMS Control	Note 2
E0 – FF	Reserved for future use	N/A	N/A
Note 1: The functionality of these IEIs is defined in 3GPP TSG 31.115 [28], and therefore, the repeatability is not within the scope of this document and will not be determined here. Note 2: The functionality of these IEIs is used in a proprietary fashion by different SMSC vendors, and therefore, are not within the scope of this technical specification. Note 3: The functionality of these IEIs is defined by the WAP Forum and therefore the repeatability is not within the scope of this document and will not be determined here.			

A receiving entity shall ignore (i.e. skip over and commence processing at the next information element) any information element where the IEI is Reserved or not supported. The receiving entity calculates the start of the next information element by looking at the length of the current information element and skipping that number of octets.

The SM itself may be coded as 7, 8 or 16 bit data.

If 7 bit data is used and the TP-UD-Header does not finish on a septet boundary then fill bits are inserted after the last Information Element Data octet up to the next septet boundary so that there is an integral number of septets for the entire TP-UD header. This is to ensure that the SM itself starts on an septet boundary so that an earlier Phase mobile shall be capable of displaying the SM itself although the TP-UD Header in the TP-UD field may not be understood.

It is optional to make the first character of the SM itself a Carriage Return character encoded according to the default 7 bit alphabet so that earlier Phase mobiles, which do not understand the TP-UD-Header, shall over-write the displayed TP-UD-Header with the SM itself.

If 16 bit (USC2) data is used then padding octets are not necessary. The SM itself shall start on an octet boundary.

If 8 bit data is used then padding is not necessary. An earlier Phase mobile shall be able to display the SM itself although the TP-UD header may not be understood.

It is also possible for mobiles not wishing to support the TP-UD header to check the value of the TP-UDHI bit in the SMS-Deliver PDU and the first octet of the TP-UD field and skip to the start of the SM and ignore the TP-UD header.

[TS 23.040, clause 9.2.3.24.1]:

This facility allows short messages to be concatenated to form a longer message.

In the case of uncompressed 8-bit data, the maximum length of the short message within the TP-UD field is 134 (140-6) octets.

In the case of uncompressed GSM 7 bit default alphabet data, the maximum length of the short message within the TP-UD field is 153 (160-7) characters. A character represented by an escape-sequence shall not be split in the middle.

In the case of 16 bit uncompressed USC2 data, the maximum length of the short message within the TP-UD field is 67 ((140-6)/2) characters. A UCS2 character shall not be split in the middle; if the length of the User Data Header is odd, the maximum length of the whole TP-UD field is 139 octets.

In the case of compressed GSM 7 bit default alphabet data, 8 bit data or UCS2 the maximum length of the compressed short message within the TP-UD field is 134 (140-6) octets including the Compression Header and Compression Footer, both or either of which may be present (see clause 3.9).

The maximum length of an uncompressed concatenated short message is 39015 (255*153) default alphabet characters, 34170 (255*134) octets or 17085 (255*67) UCS2 characters.

The maximum length of a compressed concatenated message is 34170 (255*134) octets including the Compression Header and Compression Footer (see clause 3.9 and figure 9.2.3.24.1(a) below).

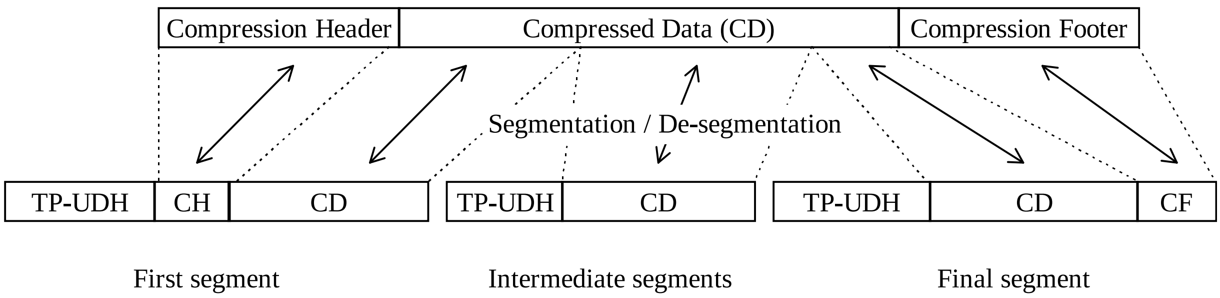

Figure 9.2.3.24.1 (a): Concatenation of a Compressed short message

The Information-Element-Data field contains information set by the application in the SMS-SUBMIT so that the receiving entity is able to re-assemble the short messages in the correct order. Each concatenated short message contains a reference number which together with the originating address and Service Centre address allows the receiving entity to discriminate between concatenated short messages sent from different originating SMEs and/or SCs. In a network which has multiple SCs, it is possible for different segments of a concatenated SM to be sent via different SCs and so it is recommended that the SC address should not be checked by the MS unless the application specifically requires such a check.

The TP elements in the SMS-SUBMIT PDU, apart from TP-MR, TP-SRR, TP-UDL and TP-UD, should remain unchanged for each SM which forms part of a concatenated SM, otherwise this may lead to irrational behaviour. TP-MR must be incremented for every segment of a concatenated message as defined in clause 9.2.3.6. A SC shall handle segments of a concatenated message like any other short message. The relation between segments of a concatenated message is made only at the originator, where the message is segmented, and at the recipient, where the message is reassembled. SMS-COMMANDs identify messages by TP-MR and therefore apply to only one segment of a concatenated message. It is up to the originating SME to issue SMS-COMMANDs for all the required segments of a concatenated message.

The Information-Element-Data octets shall be coded as follows.

Octet 1 Concatenated short message reference number.

This octet shall contain a modulo 256 counter indicating the reference number for a particular concatenated short message. This reference number shall remain constant for every short message which makes up a particular concatenated short message.

Octet 2 Maximum number of short messages in the concatenated short message.

This octet shall contain a value in the range 0 to 255 indicating the total number of short messages within the concatenated short message. The value shall start at 1 and remain constant for every short message which makes up the concatenated short message. If the value is zero then the receiving entity shall ignore the whole Information Element.

Octet 3 Sequence number of the current short message.

This octet shall contain a value in the range 0 to 255 indicating the sequence number of a particular short message within the concatenated short message. The value shall start at 1 and increment by one for every short message sent within the concatenated short message. If the value is zero or the value is greater than the value in octet 2 then the receiving entity shall ignore the whole Information Element.

The IEI and associated IEI length and IEI data shall be present in every segment of the concatenated SM.

[TS 24.341, clause 5.3.2.3]

When a SIP MESSAGE request including a short message in the "vnd.3gpp.sms" payload is delivered, the SM-over-IP receiver shall:

- generate a SIP response according to RFC 3428;
- extract the payload encoded according to 3GPP TS 24.011 for RP-DATA; and
- create a delivery report as described in subclause 5.3.2.4. The content of the report is defined in 3GPP TS 24.011.

[TS 24.341, clause 5.3.2.4]

When an SM-over-IP receiver wants to send an SM delivery report over IP, the SM-over-IP receiver shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the IP-SM-GW;

NOTE 1: The address of the IP-SM-GW is received in the P-Asserted-Identity header in the SIP MESSAGE request including the delivered short message.

b) the From header, which shall contain a public user identity of the SM-over-IP receiver.

c) the To header, which shall contain the IP-SM-GW;

b) the Content-Type header shall contain "application/vnd.3gpp.sms"; and

c) the body of the request shall contain the RP-ACK or RP-ERROR message for the SM delivery report, as defined in 3GPP TS 24.011 [8].

NOTE 2: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

9.3.3 Test description

9.3.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- SMS over IP is enabled.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

9.3.3.2 Test procedure sequence

Table 9.3.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to send a Concatenated SMS over IP (The length of SMS text is determined so that the amount of segments of the concatenated SMS is three).	-	-	-	-
1A-1F	Steps 2-7 of generic procedure specified in Table 4.9.19.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
2	Check: Does the UE send a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the first segment of the concatenated SMS?	-->	SIP MESSAGE request	1	P
3	SS responds with 202 Accepted.	<--	202 Accepted	-	-
4	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the short message submission report indicating a positive acknowledgement of the first segment of the concatenated SMS sent by the UE at Step 2.	<--	SIP MESSAGE request	-	-
5	Check: Does the UE respond with 200 OK?	-->	200 OK	2	P
6	Check: Does the UE send a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the second segment of the concatenated SMS?	-->	SIP MESSAGE request	2	P
7	SS responds with 202 Accepted.	<--	202 Accepted	-	-
8	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the short message submission report indicating a positive acknowledgement of the second segment of the concatenated SMS sent by the UE at Step 6.	<--	SIP MESSAGE request	-	-
9	Check: Does the UE respond with 200 OK?	-->	200 OK	3	P
10	Check: Does the UE send a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the final segment of the concatenated SMS?	-->	SIP MESSAGE request	3	P
11	SS responds with 202 Accepted.	<--	202 Accepted	-	-
12	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the short message submission report indicating a positive acknowledgement of the final segment of the concatenated SMS sent by the UE at Step 10.	<--	SIP MESSAGE request	-	-
13	Check: Does the UE respond with 200 OK?	-->	200 OK	4	P

9.3.3.3 Specific message contents

Table 9.3.3.3-1: MESSAGE for MO SMS (step 2, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		- TP-UDHI='1'B (The beginning of the TP UD field contains a Header in addition to the short message.) - TP-MR=any allowed value - TP-UD - Length of User Data Header (UDHL)=5 - Information Element Identifier (IEI)=0x00 (Concatenated short messages, 8-bit reference number) - Length of Information Element (IEIDL)=3 - Concatenated short message reference number=any allowed value - Maximum number of short messages in the concatenated short message=3 - Sequence number of the current short message=1		TS 24.011 [25] TS 23.040 [24]

Table 9.3.3.3-2: 202 ACCEPTED (step 3, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

Table 9.3.3.3-3: Short message submission report for MO SMS (step 4, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.4

Table 9.3.3.3-4: 200 OK for other requests than REGISTER or SUBSCRIBE (step 5, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A5, A22

Table 9.3.3.3-5: MESSAGE for MO SMS (step 6, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		- TP-UDHI='1'B (The beginning of the TP UD field contains a Header in addition to the short message.) - TP-MR= The value sent in the step1 + 1 (incremented) - TP-UD - Length of User Data Header (UDHL)=5 - Information Element Identifier (IEI)=0x00 (Concatenated short messages, 8-bit reference number) - Length of Information Element (IEIDL)=3 - Concatenated short message reference number= The same value sent in the step1 - Maximum number of short messages in the concatenated short message=3 - Sequence number of the current short message=2		TS 24.011 [25] TS 23.040 [24]

Table 9.3.3.3-6: 202 ACCEPTED (step 7, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

Table 9.3.3.3-7: Short message submission report for MO SMS (step 8, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.4

Table 9.3.3.3-8: 200 OK for other requests than REGISTER or SUBSCRIBE (step 9, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A5, A22

Table 9.3.3.3-9: MESSAGE for MO SMS (step 10, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.3				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		- TP-UDHI='1'B (The beginning of the TP UD field contains a Header in addition to the short message.) - TP-MR= The value sent in the step5 + 1 (incremented) - TP-UD - Length of User Data Header (UDHL)=5 - Information Element Identifier (IEI)=0x00 (Concatenated short messages, 8-bit reference number) - Length of Information Element (IEIDL)=3 - Concatenated short message reference number= The same value sent in the step5 - Maximum number of short messages in the concatenated short message=3 - Sequence number of the current short message=3		TS 24.011 [25] TS 23.040 [24]

Table 9.3.3.3-10: 202 ACCEPTED (step 11, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

Table 9.3.3.3-11: Short message submission report for MO SMS (step 12, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.4

Table 9.3.3.3-12: 200 OK for other requests than REGISTER or SUBSCRIBE (step 13, table 9.3.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A5, A22

9.4 Mobile Terminating Concatenated SMS / 5GS

9.4.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE receives a SIP MESSAGE request containing a first segment of a concatenated SMS }
then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing a delivery report
for the first segment }
    }
```

(2)

```
with { UE having sent a SIP MESSAGE request containing a delivery report for the first segment }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing a second
segment of a concatenated SMS }
    then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing a delivery
report for the second segment }
    }
```

(3)

```
with { UE having sent a SIP MESSAGE request containing a delivery report for the second segment }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing a third
segment of a concatenated SMS }
    then { UE sends a 200 OK response, followed by a SIP MESSAGE request containing a delivery
report for the third segment }
    }
```

9.4.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 23.040, clause 9.2.3.23]:

The TP-User-Data-Header-Indicator is a 1 bit field within bit 6 of the first octet of the following six PDUs:

- SMS-SUBMIT,
- SMS-SUBMIT-REPORT,
- SMS-DELIVER,
- SMS-DELIVER-REPORT,
- SMS-STATUS-REPORT,
- SMS-COMMAND.

TP-UDHI has the following values.

Bit no. 6 0 The TP-UD field contains only the short message

1 The beginning of the TP-UD field contains a Header in addition to the short message.

[TS 23.040, clause 9.2.3.24]:

The length of the TP-User-Data field is defined in the PDU's of the SM-TL (see clause 9.2.2).

The TP-User-Data field may comprise just the short message itself or a Header in addition to the short message depending upon the setting of TP-UDHI.

Where the TP-UDHI value is set to 0 the TP-User-Data field comprises the short message only, where the user data can be 7 bit (default alphabet) data, 8 bit data, or 16 bit (UCS2 [24]) data.

Where the TP-UDHI value is set to 1 the first octets of the TP-User-Data field contains a Header in the following order starting at the first octet of the TP-User-Data field.

Irrespective of whether any part of the User Data Header is ignored or discarded, the MS shall always store the entire TPDU exactly as received.

FIELD	LENGTH
Length of User Data Header	1 octet
Information-Element-Identifier "A"	1 octet
Length of Information-Element "A"	1 octet
Information-Element "A" Data	0 to "n" octets
Information-Element-Identifier "B"	1 octet
Length of Information-Element "B"	1 octet
Information-Element "B" Data	0 to "n" octets
Information-Element-Identifier "X"	1 octet
Length of Information-Element "X"	1 octet
Information-Element "X" Data	0 to "n" octets

The diagram below shows the layout of the TP-User-Data-Length and the TP-User-Data for uncompressed GSM 7 bit default alphabet data. The UDHL field is the first octet of the TP-User-Data content of the Short Message.

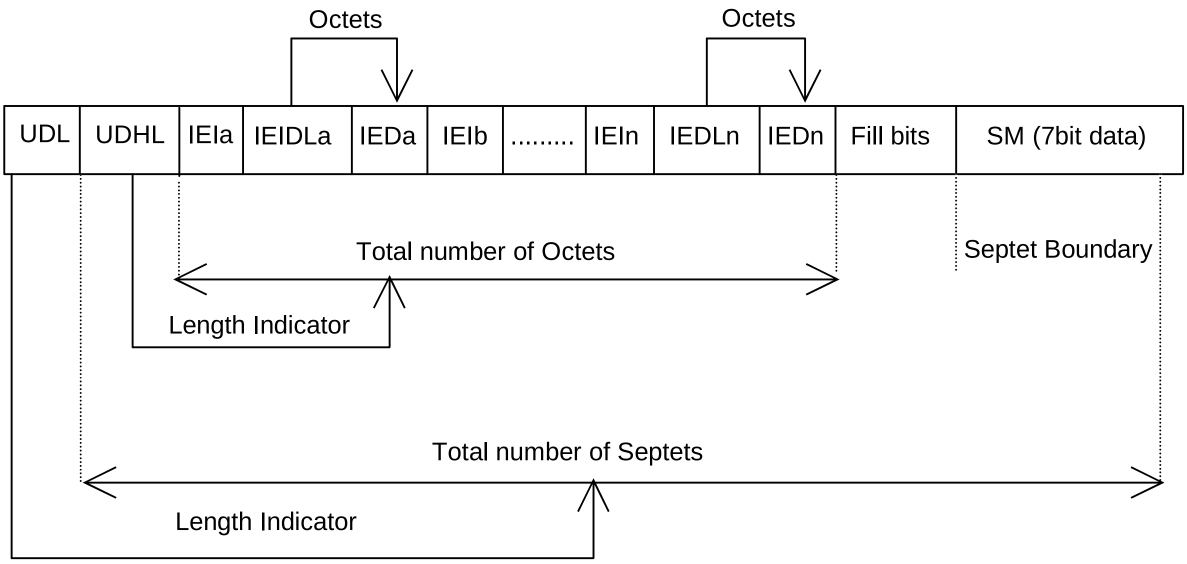

Figure 9.2.3.24 (a)

The diagram below shows the layout of the TP-User-Data-Length and the TP-User-Data for uncompressed 8 bit data or uncompressed UCS2 data. The UDHL field is the first octet of the TP-User-Data content of the Short Message.

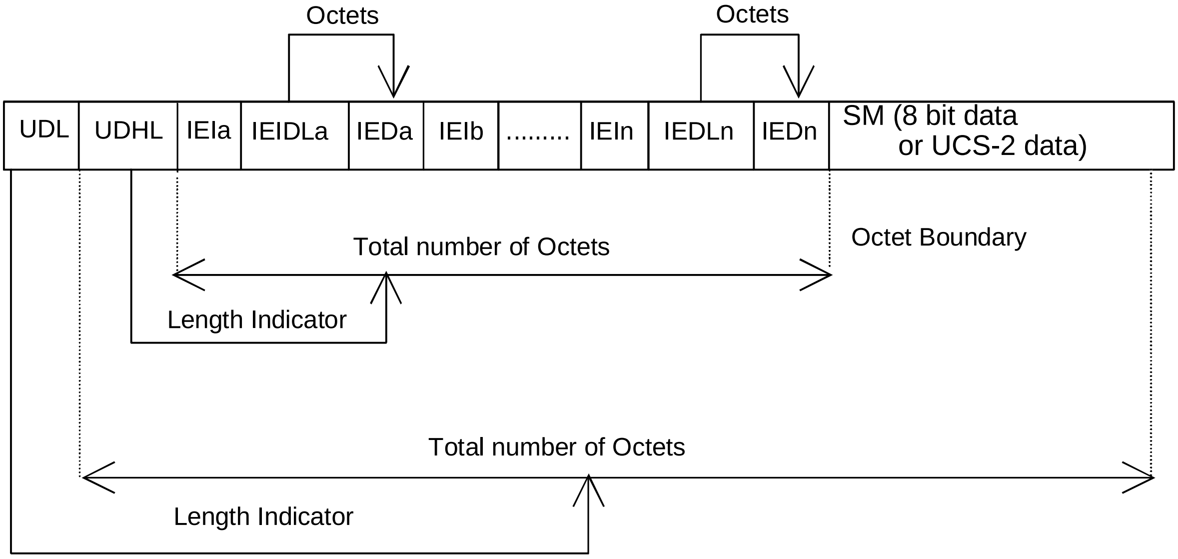

Figure 9.2.3.24 (b)

The diagram below shows the layout of the TP-User-Data-Length and the TP-User-Data for compressed GSM 7 bit default alphabet data, compressed 8 bit data or compressed UCS2 data. The UDHL field is the first octet of the TP-User-Data content of the Short Message.

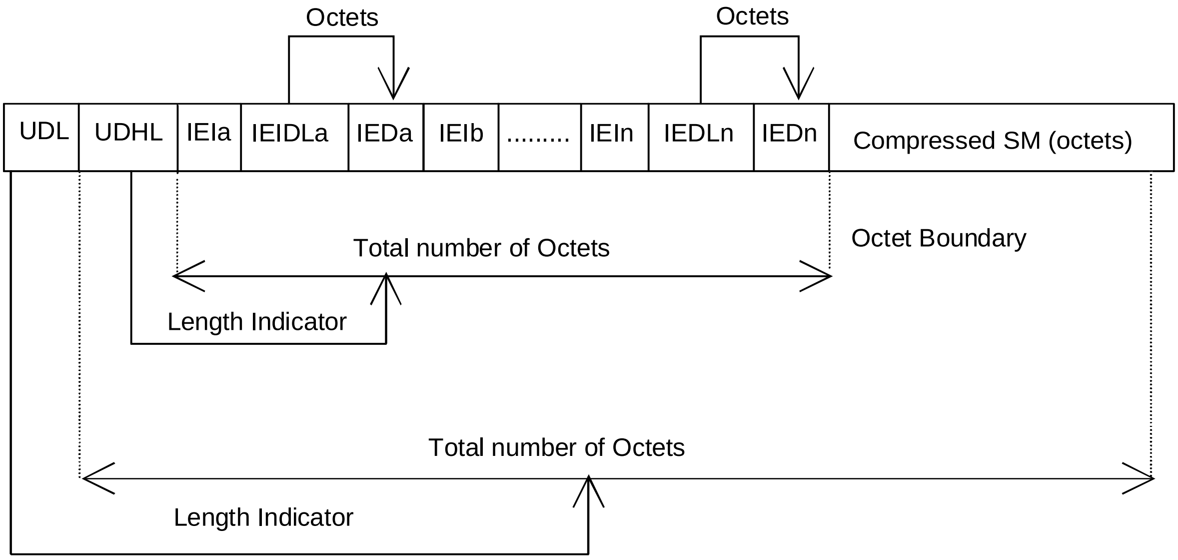

Figure 9.2.3.24 (c)

The definition of the TP-User-Data-Length field which immediately precedes the "Length of User Data Header" is unchanged and shall therefore be the total length of the TP-User-Data field including the Header, if present. (see 9.2.3.16).

The "Length-of-Information-Element" fields shall be the integer representation of the number of octets within its associated "Information-Element-Data" field which follows and shall not include itself in its count value.

The "Length-of-User-Data-Header" field shall be the integer representation of the number of octets within the "User-Data-Header" information fields which follow and shall not include itself in its count or any fill bits which may be present (see text below).

Information Elements may appear in any order and need not follow the order used in the present document. Information Elements are classified into 3 categories as described below.

- SMS Control – identifies those IEIs which have the capability of dictating SMS functionality.
- EMS Control – identifies those IEIs which manage EMS Content IEIs.
- EMS Content – identifies those IEIs containing data of a unique media format.

It is permissible for certain IEs to be repeated within a short message, or within a concatenated message. There is no restriction on the repeatability of IEs in the EMS Content classification. The repeatability of SMS Control and EMS Control IEs is determined on an individual basis. See the IE table below for the repeatability of each IE.

In the event that IEs determined as not repeatable are duplicated, the last occurrence of the IE shall be used. In the event that two or more IEs occur which have mutually exclusive meanings (e.g. an 8bit port address and a 16bit port address), then the last occurring IE shall be used.

If the length of the User Data Header is such that there are too few or too many octets in the final Information Element then the whole User Data Header shall be ignored.

If any reserved values are received within the content of any Information Element then that part of the Information Element shall be ignored.

The support of any Information Element Identifier is optional unless otherwise stated.

The Information Element Identifier octet shall be coded as follows:

VALUE (hex)	MEANING	Classification	Repeatability
00	Concatenated short messages, 8-bit reference number	SMS Control	No
01	Special SMS Message Indication	SMS Control	Yes
02	Reserved	N/A	N/A
03	Value not used to avoid misinterpretation as <LF> character	N/A	N/A
04	Application port addressing scheme, 8 bit address	SMS Control	No
05	Application port addressing scheme, 16 bit address	SMS Control	No
06	SMSC Control Parameters	SMS Control	No
07	UDH Source Indicator	SMS Control	Yes
08	Concatenated short message, 16-bit reference number	SMS Control	No
09	Wireless Control Message Protocol	SMS Control	Note 3
0A	Text Formatting	EMS Control	Yes
0B	Predefined Sound	EMS Content	Yes
0C	User Defined Sound (iMelody max 128 bytes)	EMS Content	Yes
0D	Predefined Animation	EMS Content	Yes
0E	Large Animation (16*16 times 4 = 32*4 =128 bytes)	EMS Content	Yes
0F	Small Animation (8*8 times 4 = 8*4 =32 bytes)	EMS Content	Yes
10	Large Picture (32*32 = 128 bytes)	EMS Content	Yes
11	Small Picture (16*16 = 32 bytes)	EMS Content	Yes
12	Variable Picture	EMS Content	Yes
13	User prompt indicator	EMS Control	Yes
14	Extended Object	EMS Content	Yes
15	Reused Extended Object	EMS Control	Yes
16	Compression Control	EMS Control	No
17	Object Distribution Indicator	EMS Control	Yes
18	Standard WVG object	EMS Content	Yes
19	Character Size WVG object	EMS Content	Yes
1A	Extended Object Data Request Command	EMS Control	No
1B-1F	Reserved for future EMS features (see subclause 3.10)	N/A	N/A
20	RFC 5322 E-Mail Header	SMS Control	No
21	Hyperlink format element	SMS Control	Yes
22	Reply Address Element	SMS Control	No
23	Enhanced Voice Mail Information	SMS Control	No
24	National Language Single Shift	SMS Control	No
25	National Language Locking Shift	SMS Control	No
26 – 6F	Reserved for future use	N/A	N/A
70 – 7F	(U)SIM Toolkit Security Headers	SMS Control	Note 1
80 – 9F	SME to SME specific use	SMS Control	Note 2
A0 – BF	Reserved for future use	N/A	N/A
C0 – DF	SC specific use	SMS Control	Note 2
E0 – FF	Reserved for future use	N/A	N/A
Note 1: The functionality of these IEIs is defined in 3GPP TSG 31.115 [28], and therefore, the repeatability is not within the scope of this document and will not be determined here. Note 2: The functionality of these IEIs is used in a proprietary fashion by different SMSC vendors, and therefore, are not within the scope of this technical specification. Note 3: The functionality of these IEIs is defined by the WAP Forum and therefore the repeatability is not within the scope of this document and will not be determined here.			

A receiving entity shall ignore (i.e. skip over and commence processing at the next information element) any information element where the IEI is Reserved or not supported. The receiving entity calculates the start of the next information element by looking at the length of the current information element and skipping that number of octets.

The SM itself may be coded as 7, 8 or 16 bit data.

If 7 bit data is used and the TP-UD-Header does not finish on a septet boundary then fill bits are inserted after the last Information Element Data octet up to the next septet boundary so that there is an integral number of septets for the entire TP-UD header. This is to ensure that the SM itself starts on an septet boundary so that an earlier Phase mobile shall be capable of displaying the SM itself although the TP-UD Header in the TP-UD field may not be understood.

It is optional to make the first character of the SM itself a Carriage Return character encoded according to the default 7 bit alphabet so that earlier Phase mobiles, which do not understand the TP-UD-Header, shall over-write the displayed TP-UD-Header with the SM itself.

If 16 bit (USC2) data is used then padding octets are not necessary. The SM itself shall start on an octet boundary.

If 8 bit data is used then padding is not necessary. An earlier Phase mobile shall be able to display the SM itself although the TP-UD header may not be understood.

It is also possible for mobiles not wishing to support the TP-UD header to check the value of the TP-UDHI bit in the SMS-Deliver PDU and the first octet of the TP-UD field and skip to the start of the SM and ignore the TP-UD header.

[TS 23.040, clause 9.2.3.24.1]:

This facility allows short messages to be concatenated to form a longer message.

In the case of uncompressed 8-bit data, the maximum length of the short message within the TP-UD field is 134 (140-6) octets.

In the case of uncompressed GSM 7 bit default alphabet data, the maximum length of the short message within the TP-UD field is 153 (160-7) characters. A character represented by an escape-sequence shall not be split in the middle.

In the case of 16 bit uncompressed USC2 data, the maximum length of the short message within the TP-UD field is 67 ((140-6)/2) characters. A UCS2 character shall not be split in the middle; if the length of the User Data Header is odd, the maximum length of the whole TP-UD field is 139 octets.

In the case of compressed GSM 7 bit default alphabet data, 8 bit data or UCS2 the maximum length of the compressed short message within the TP-UD field is 134 (140-6) octets including the Compression Header and Compression Footer, both or either of which may be present (see clause 3.9).

The maximum length of an uncompressed concatenated short message is 39015 (255*153) default alphabet characters, 34170 (255*134) octets or 17085 (255*67) UCS2 characters.

The maximum length of a compressed concatenated message is 34170 (255*134) octets including the Compression Header and Compression Footer (see clause 3.9 and figure 9.2.3.24.1(a) below).

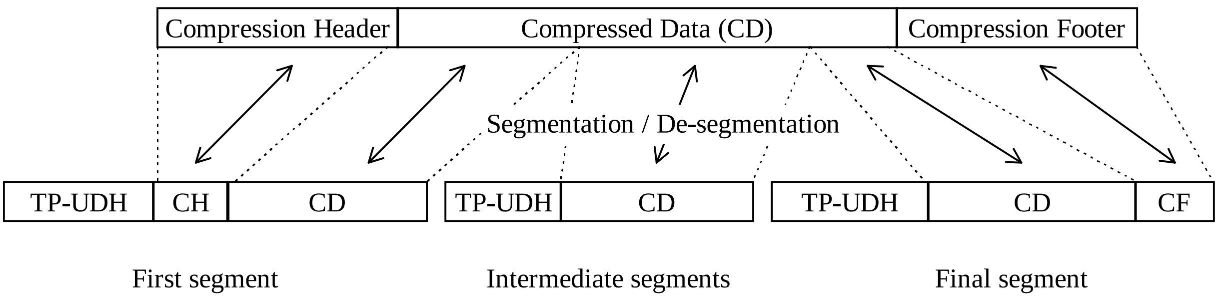

Figure 9.2.3.24.1 (a): Concatenation of a Compressed short message

The gNB-DU controlling a UE-associated logical F1-connection initiates the procedure by generating a UE The Information-Element-Data field contains information set by the application in the SMS-SUBMIT so that the receiving entity is able to re-assemble the short messages in the correct order. Each concatenated short message contains a reference number which together with the originating address and Service Centre address allows the receiving entity to discriminate between concatenated short messages sent from different originating SMEs and/or SCs. In a network which has multiple SCs, it is possible for different segments of a concatenated SM to be sent via different SCs and so it is recommended that the SC address should not be checked by the MS unless the application specifically requires such a check.

The TP elements in the SMS-SUBMIT PDU, apart from TP-MR, TP-SRR, TP-UDL and TP-UD, should remain unchanged for each SM which forms part of a concatenated SM, otherwise this may lead to irrational behaviour. TP-MR must be incremented for every segment of a concatenated message as defined in clause 9.2.3.6. A SC shall handle segments of a concatenated message like any other short message. The relation between segments of a concatenated message is made only at the originator, where the message is segmented, and at the recipient, where the message is reassembled. SMS-COMMANDs identify messages by TP-MR and therefore apply to only one segment of a concatenated message. It is up to the originating SME to issue SMS-COMMANDs for all the required segments of a concatenated message.

The Information-Element-Data octets shall be coded as follows.

Octet 1 Concatenated short message reference number.

This octet shall contain a modulo 256 counter indicating the reference number for a particular concatenated short message. This reference number shall remain constant for every short message which makes up a particular concatenated short message.

Octet 2 Maximum number of short messages in the concatenated short message.

This octet shall contain a value in the range 0 to 255 indicating the total number of short messages within the concatenated short message. The value shall start at 1 and remain constant for every short message which makes up the concatenated short message. If the value is zero then the receiving entity shall ignore the whole Information Element.

Octet 3 Sequence number of the current short message.

This octet shall contain a value in the range 0 to 255 indicating the sequence number of a particular short message within the concatenated short message. The value shall start at 1 and increment by one for every short message sent within the concatenated short message. If the value is zero or the value is greater than the value in octet 2 then the receiving entity shall ignore the whole Information Element.

The IEI and associated IEI length and IEI data shall be present in every segment of the concatenated SM.

[TS 24.341, clause 5.3.2.3]

When a SIP MESSAGE request including a short message in the "vnd.3gpp.sms" payload is delivered, the SM-over-IP receiver shall:

- generate a SIP response according to RFC 3428 [14];
- extract the payload encoded according to 3GPP TS 24.011 [8] for RP-DATA; and
- create a delivery report as described in subclause 5.3.2.4. The content of the report is defined in 3GPP TS 24.011 [8].

[TS 24.341, clause 5.3.2.4]

When an SM-over-IP receiver wants to send an SM delivery report over IP, the SM-over-IP receiver shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the IP-SM-GW;

NOTE 1: The address of the IP-SM-GW is received in the P-Asserted-Identity header in the SIP MESSAGE request including the delivered short message.

b) the From header, which shall contain a public user identity of the SM-over-IP receiver.

c) the To header, which shall contain the IP-SM-GW;

d) the In-Reply-To header which shall contain the Call-Id of the SIP MESSAGE request that was received in the received short message;

e) the Content-Type header shall contain "application/vnd.3gpp.sms"; and

f) the body of the request shall contain the RP-ACK or RP-ERROR message for the SM delivery report, as defined in 3GPP TS 24.011 [8].

NOTE 2: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

9.4.3 Test description

9.4.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

9.4.3.2 Test procedure sequence

Table 9.4.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
0A-0H	Steps 1-8 of generic procedure specified in Table 4.9.20.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
1	SS sends a first segment of a concatenated SMS in the message-body of SIP_MESSAGE.	<--	SIP MESSAGE request		
2	Check: Does the UE respond with a 200 OK?	-->	200 OK	1	P
3	Check: When the payload is extracted, does the UE respond with a delivery report included in the message-body of MESSAGE?	-->	SIP MESSAGE request	1	P
4	SS responds with a 202 ACCEPTED.	<--	202 ACCEPTED		
5	SS sends a second segment of a concatenated SMS in the message-body of SIP_MESSAGE.	<--	SIP MESSAGE request		
6	Check: Does the UE respond with a 200 OK?	-->	200 OK	2	P
7	Check: When the payload is extracted, does the UE respond with a delivery report included in the message-body of MESSAGE?	-->	SIP MESSAGE request	2	P
8	SS responds with a 202 ACCEPTED.	<--	202 ACCEPTED		
9	SS sends a final segment of a concatenated SMS in the message-body of SIP_MESSAGE.	<--	SIP MESSAGE request		
10	Check: Does the UE respond with a 200 OK?	-->	200 OK	3	P
11	Check: When the payload is extracted, does the UE respond with a delivery report included in the message-body of MESSAGE?	-->	SIP MESSAGE request	3	P
12	SS responds with a 202 ACCEPTED.	<--	202 ACCEPTED		

9.4.3.3 Specific message contents

Table 9.4.3.3-1: MESSAGE for MT SMS (step 1, table 9.4.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		- TP-RP='0'B (TP Reply Path parameter is not set in this SMS SUBMIT/DELIVER) - TP-MMS='0'B (More messages are waiting for the MS in this SC) - TP-UDHI='1'B (The beginning of the TP UD field contains a Header in addition to the short message.) - TP-PID='00000000'B - TP-UD - Length of User Data Header (UDHL)=5 - Information Element Identifier (IEI)=0x00 (Concatenated short messages, 8-bit reference number) - Length of Information Element (IEIDL)=3 - Concatenated short message reference number=any allowed value - Maximum number of short messages in the concatenated short message=3 - Sequence number of the current short message=1		TS 24.011 [25] TS 23.040 [24]

Table 9.4.3.3-2: 200 OK for other requests than REGISTER or SUBSCRIBE (step 2/6/10, table 9.4.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A22

Table 9.4.3.3-3: MESSAGE for delivery report (step 3/7/11, table 9.4.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.7.2

Table 9.4.3.3-4: 202 ACCEPTED (step 4/8, table 9.4.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

Table 9.4.3.3-5: MESSAGE for MT SMS (step 5, table 9.4.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		- TP-RP='0'B (TP Reply Path parameter is not set in this SMS SUBMIT/DELIVER) - TP-MMS='0'B (More messages are waiting for the MS in this SC) - TP-UDHI='1'B (The beginning of the TP UD field contains a Header in addition to the short message.) - TP-PID='00000000'B - TP-UD - Length of User Data Header (UDHL)=5 - Information Element Identifier (IEI)=0x00 (Concatenated short messages, 8-bit reference number) - Length of Information Element (IEIDL)=3 - Concatenated short message reference number=The same value sent in the step1 - Maximum number of short messages in the concatenated short message=3 - Sequence number of the current short message=2		TS 24.011 [25] TS 23.040 [24]

Table 9.4.3.3-6: MESSAGE for MT SMS (step 9, table 9.4.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.1				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**

Message-body		- TP-RP='0'B (TP Reply Path parameter is not set in this SMS SUBMIT/DELIVER)
		- TP-MMS='0'B (More messages are waiting for the MS in this SC)
		- TP-UDHI='1'B (The beginning of the TP UD field contains a Header in addition to the short message.)
		- TP-PID='00000000'B
		- TP-UD
		  - Length of User Data Header (UDHL)=5
		  - Information Element Identifier (IEI)=0x00 (Concatenated short messages, 8-bit reference number)
		  - Length of Information Element (IEIDL)=3
		  - Concatenated short message reference number=The same value sent in the step1
		  - Maximum number of short messages in the concatenated short message=3
		  - Sequence number of the current short message=3
				TS 24.011 [25]
				TS 23.040 [24]

9.5 Mobile Originating SMS / RP-ERROR / 5GS

9.5.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is made to send an SMS over IP }
    then { UE sends a SIP MESSAGE request containing a short message }
    }
```

(2)

```
with { UE having sent a SIP MESSAGE request containing a short message }
ensure that {
  when { UE receives a 202 Accepted response, followed by a SIP MESSAGE request containing an RP-
ERROR message }
    then { UE sends a 200 OK response }
    }
```

9.5.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.341, clause 5.3.1.1]:

In addition to the procedures specified in subclause 5.3.1, the SM-over-IP sender shall support the procedures specified in 3GPP TS 24.229 [10] appropriate to the functional entity in which the SM-over-IP sender is implemented. The SM-over-IP sender shall build and populate RP-DATA message, containing all the information that a mobile station submitting an SM according to 3GPP TS 24.011 [8] would place, for successful delivery. The SM-over-IP sender shall parse and interpret RP- DATA, RP-ACK and RP-ERROR messages, containing all the information that a mobile station receiving an SM according to 3GPP TS 24.011 [8] would see, in a SM submission or status report.

NOTE 1: If the SM-over-IP sender uses SMR entity timers as specified in 3GPP TS 24.011 [8], then TR1M is set to a value greater than timer F (see 3GPP TS 24.229 [10]).

NOTE 2: If the SM-over-IP sender expects to receive a SM submit report will include the "+g.3gpp.smsip" parameter in the Contact header field when sending a REGISTER request.

[TS 24.341, clause 5.3.1.2]:

When an SM-over-IP sender wants to submit an SM over IP, the SM-over-IP sender shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the PSI of the SC of the SM-over-IP sender;

NOTE 1: The PSI of the SC can be SIP URI or tel URI based on operator policy. The PSI of the SC can be obtained using one of the following methods in the priority order listed below:

1) provided by the user;

2) if UICC is used, then:

- if an ISIM is present, then the PSI of the SC is obtained from the $EF_{PSISMSC}$ in DF_TELECOM as per 3GPP TS 31.103 [18];

- if an ISIM is not present, then the PSI of the SC is obtained from the $EF_{PSISMSC}$ in DF_TELECOM as per 3GPP TS 31.102 [19]; or

- if the PSI of the SC is not available in $EF_{PSISMSC}$ in DF_TELECOM, then the PSI of the SC contains the TS-Service-Centre-Address stored in the EF_{SMSP} in DF_TELECOM as per 3GPP TS 31.102 [19]. If the PSI of the SC is based on the E.164 number from the TS-Service-Centre-Address stored in the EF_{SMSP} in DF_TELECOM then the URI constructed can be either a tel URI or a SIP URI (using the "user=phone" SIP URI parameter format).

3) if SIM is used instead of UICC, then the PSI of the SC contains the TS-Service Centre Address stored in the EF_{SMSP} in DF_TELECOM as per 3GPP TS 51.011 [20]. If the PSI of the SC is based on the E.164 number from the TS-Service-Centre-Address stored in the EF_{SMSP} in DF_TELECOM then the URI constructed can be either a tel URI or a SIP URI (using the "user=phone" SIP URI parameter format); or

4) if neither the UICC nor SIM is used, then how the PSI of the SC is configured and obtained is through means outside the scope of this specification.

b) the From header, which shall contain a public user identity of the SM-over-IP sender;

NOTE 2: The IP-SM-GW will have to use an address of the SM-over-IP sender that the SC can process (i.e. an E.164 number). This address will come from a tel URI in a P-Asserted-Identity header (as defined in RFC 3325 [13]) placed in the SIP MESSAGE request by the P-CSCF or S-CSCF.

NOTE 3: The SM-over-IP sender has to store the Call-ID of the SIP MESSAGE request, so it can associate the appropriate SIP MESSAGE request including a submit report with it.

c) the To header, which shall contain the PSI of the SC of the SM-over-IP sender;

d) the Content-Type header, which shall contain "application/vnd.3gpp.sms"; and

e) the body of the request shall contain an RP-DATA message as defined in 3GPP TS 24.011 [8], including the SMS headers and the SMS user information encoded as specified in 3GPP TS 23.040 [3].

NOTE 4: The address of the SC is included in the RP-DATA message content. The address of the SC included in the RP-DATA message content is stored in the EF_{SMSP} in DF_TELECOM of the (U)SIM of the SM-over-IP sender.

NOTE 5: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

NOTE 6: Both the address of the SC and the PSI of the SC can be configured in the $EF_{PSISMSC}$ in DF_TELECOM of the USIM and ISIM respectively using the USAT as per 3GPP TS 31.111 [21].

The SM-over-IP sender may request the SC to return the status of the submitted message. The support of status report capabilities is optional for the SC.

When a SIP MESSAGE request including a submit report in the "vnd.3gpp.sms" payload is received, the SM-over-IP sender shall:

- if SM-over-IP sender supports In-Reply-To header usage and the In-Reply-To header indicates that the request corresponds to a short message submitted by the SM-over-IP sender, generate a 200 (OK) SIP response according to RFC 3428 [14].

  if SM-over-IP sender supports In-Reply-To header usage and the In-Reply-To header indicates that the request does not correspond to a short message submitted by the SM-over-IP sender, a 488 (Not Acceptable here) SIP response according to RFC 3428 [14].

- if SM-over-IP sender does not support In-Reply-To header usage, generate a 200 (OK) SIP response according to RFC 3428 [14]; and extract the payload encoded according to 3GPP TS 24.011 [8] for RP-ACK or RP-ERROR.

[TS 24.341 clause 5.3.1.3]:

When a SIP MESSAGE request including a status report in the "vnd.3gpp.sms" payload is delivered, the SM-over-IP sender shall:

- generate a SIP response according to RFC 3428 [14];

- extract the payload encoded according to 3GPP TS 24.011 [8] for RP-DATA; and

- create a delivery report for the status report as described in subclause 5.3.2.4. The content of the delivery report is defined in 3GPP TS 24.011 [8].

[TS 24.341 clause 5.3.2.4]:

When an SM-over-IP receiver wants to send an SM delivery report over IP, the SM-over-IP receiver shall send a SIP MESSAGE request with the following information:

a) the Request-URI, which shall contain the IP-SM-GW;

NOTE 1: The address of the IP-SM-GW is received in the P-Asserted-Identity header in the SIP MESSAGE request including the delivered short message.

b) the From header, which shall contain a public user identity of the SM-over-IP receiver.

c) the To header, which shall contain the IP-SM-GW;

d) the In-Reply-To header which shall contain the Call-Id of the SIP MESSAGE request that was received in the received short message;

e) the Content-Type header shall contain "application/vnd.3gpp.sms"; and

f) the body of the request shall contain the RP-ACK or RP-ERROR message for the SM delivery report, as defined in 3GPP TS 24.011 [8].

NOTE 2: The SM-over-IP sender will use content transfer encoding of type "binary" for the encoding of the SM in the body of the SIP MESSAGE request.

[TS 24.011 clause 8.2.5.4]:

This element is a variable length element always included in the RP-ERROR message, conveying a negative result of a RP-DATA message transfer attempt or RP-SMMA notification attempt. The element contains a cause value and optionally a diagnostic field giving further details of the error cause.

The coding of the cause value is given in table 8.4/3GPP TS 24.011. The mapping between error causes in 3GPP TS 24.011 and 3GPP TS 29.002 (MAP) is specified in 3GPP TS 23.040. Parameters included in the return error from MAP (e.g. System Failure) are mapped directly into the diagnostic field.

8	7 6 5 4 3 2 1	
0	1 0 0 0 0 1 0 RP-Cause IEI	1 octet
Length indicator		1 octet
0 ext	Cause value	1 octet
Diagnostic field		1 octet *

Figure 8.8/3GPP TS 24.011: RP-Cause element layout

Table 8.4/3GPP TS 24.011 (part 1): Cause values that may be contained in an RP-ERROR message in a mobile originating SM-transfer attempt

Cause value	Cause number	Cause
7 6 5 4 3 2 1	#	
0 0 0 0 0 0 1	1	Unassigned (unallocated) number
0 0 0 1 0 0 0	8	Operator determined barring
0 0 0 1 0 1 0	10	Call barred
0 0 0 1 0 1 1	11	Reserved
0 0 1 0 1 0 1	21	Short message transfer rejected
0 0 1 1 0 1 1	27	Destination out of order
0 0 1 1 1 0 0	28	Unidentified subscriber
0 0 1 1 1 0 1	29	Facility rejected
0 0 1 1 1 1 0	30	Unknown subscriber
0 1 0 0 1 1 0	38	Network out of order
0 1 0 1 0 0 1	41	Temporary failure
0 1 0 1 0 1 0	42	Congestion
0 1 0 1 1 1 1	47	Resources unavailable, unspecified
0 1 1 0 0 1 0	50	Requested facility not subscribed
1 0 0 0 1 0 1	69	Requested facility not implemented
1 0 1 0 0 0 1	81	Invalid short message transfer reference value
1 0 1 1 1 1 1	95	Semantically incorrect message
1 1 0 0 0 0 0	96	Invalid mandatory information
1 1 0 0 0 0 1	97	Message type non-existent or not implemented
1 1 0 0 0 1 0	98	Message not compatible with short message protocol state
1 1 0 0 0 1 1	99	Information element non-existent or not implemented
1 1 0 1 1 1 1	111	Protocol error, unspecified
1 1 1 1 1 1 1	127	Interworking, unspecified
Note: All other cause values shall be treated as cause number 41, "Temporary Failure"		

9.5.3 Test description

9.5.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.
- SMS over IP is enabled.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

9.5.3.2 Test procedure sequence

Table 9.5.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to send an SMS over IP.				
1A-1F	Steps 2-7 of generic procedure specified in Table 4.9.19.2.2-1 of TS 38.508-1 [21] are performed.	-	-	-	-
2	Check: Does the UE send a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a short message?	-->	SIP MESSAGE request	1	P
3	SS responds with 202 Accepted.	<--	202 ACCEPTED		
4	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload and RP-ERROR message.	<--	SIP MESSAGE request		
5	Check: Does the UE respond with 200 OK?	-->	200 OK	2	P

9.5.3.3 Specific message contents

Table 9.5.3.3-1: Message for MO SMS (step 2, table 9.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.3, Condition A2, A5

Table 9.5.3.3-2: 202 ACCEPTED (step 3, table 9.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.3

Table 9.5.3.3-3: Short message submission report for MO SMS (step 4, table 9.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in annex A.7.4				
Header/param	**Cond**	**Value/remark**	**Rel**	**Reference**
Message-body		RP-ERROR message with RP-Cause Data: Length: 2, Length indicator = 1 Extension: not extended Cause value: 38 (Network out of order)		TS 24.011 [25] TS 23.040 [24]

Table 9.5.3.3-4: 200 OK for other requests than REGISTER or SUBSCRIBE (step 5, table 9.5.3.2-1)

Derivation Path: TS 34.229-1 [2], Table in subclause A.3.1, Condition A5, A22

10 Emergency Calls

10.1 Emergency Call with emergency registration / Success / Location information available / 5GS

10.1.1 Test Purpose (TP)

(1)

```
with { UE being registered to IMS }
ensure that {
  when { UE is being made to initiate an emergency call }
    then { UE sends a correctly composed initial REGISTER request for IMS emergency registration }
```

(2)

```
with { UE having sent an unprotected REGISTER request }
ensure that {
  when { UE receiving a valid 401 (Unauthorized) response for the initial REGISTER request sent }
    then { UE correctly authenticates itself by sending another REGISTER request with a correctly
composed Authorization header using the AKAv1-MD5 algorithm }
    }
```

(3)

```
with { UE having sent unprotected and then protected REGISTER request }
ensure that {
  when { UE receiving a valid 200 OK response for the REGISTER sent for authentication }
    then { UE sends a correctly composed INVITE request }
    }
```

(4)

```
with { UE having sent INVITE }
ensure that {
  when { UE receiving 100 Trying, followed by 180 Ringing, followed by 200 OK }
    then { UE sends ACK }
      }
```

(5)

```
with { Emergency call being established }
ensure that {
  when { UE receives BYE  }
    then { UE sends a 200 OK response }
    }
```

10.1.2 Conformance Requirements

The conformance requirements covered in the present test case are, unless otherwise stated, Rel-15 requirements.

[TS 24.229 clause 4.7.5]:

A number of mechanisms also exist for providing location in support of emergency calls, both for routeing to a PSAP, and for use by the PSAP itself, in the IM CN subsystem:

a) by the inclusion by the UE of the Geolocation header field containing a location by reference or by value (see RFC 6442 [89]);

b) by the inclusion by the UE of a P-Access-Network-Info header field, which contains a cell identifier or location identifier, which is subsequently mapped, potentially by the recipient, into a real location;

...

Which means of providing location is used depends on local regulatory and operator requirements. One or more mechanisms can be used. Location can be subject to privacy constraints.

[TS 24.229 clause 5.1.6.2]:

When the user initiates an emergency call, if emergency registration is needed (including cases described in subclause 5.1.6.2A), the UE shall perform an emergency registration prior to sending the SIP request related to the emergency call.

...

When a UE performs an initial emergency registration the UE shall perform the actions as specified in subclause 5.1.1.2 with the following additions and modifications:

a) the UE shall include a "sos" SIP URI parameter in the Contact header field as described in subclause 7.2A.13, indicating that this is an emergency registration and that the associated contact address is allowed only for emergency service; and

b) the UE shall populate the From and To header fields of the REGISTER request with:

  - the first entry in the list of public user identities provisioned in the UE;

  - the default public user identity obtained during the normal registration, if the UE is not provisioned with a list of public user identities, but the UE is currently registered to the IM CN subsystem; and

  - the derived temporary public user identity, in all other cases.

[TS 24.229 clause 5.1.6.3]

Upon receiving the 200 (OK) to the REGISTER request that completes the emergency registration, the UE shall not subscribe to the reg event package of the public user identity specified in the REGISTER request.

[TS 24.229 clause 5.1.6.5]

When a UE performs authentication a UE shall perform the procedures as specified in subclause 5.1.1.5.

[TS 24.229 clause 5.1.6.8.3]

After a successful initial emergency registration, the UE shall apply the procedures as specified in subclause 5.1.2A and 5.1.3 with the following additions:

1) the UE shall insert in the INVITE request, a From header field that includes the public user identity registered via emergency registration or the tel URI associated with the public user identity registered via emergency registration, as described in subclause 4.2;

2) the UE shall include a service URN in the Request-URI of the INVITE request in accordance with subclause 5.1.6.8.1;

3) the UE shall insert in the INVITE request, a To header field with the same emergency service URN as in the Request-URI;

4) if available to the UE, and if defined for the access type as specified in subclause 7.2A.4, the P-Access-Network-Info header field shall contain a location identifier such as the cell id, line id or the identity of the WLAN access node, which is relevant for routeing the IMS emergency call;

NOTE 1: The IMS emergency specification in 3GPP TS 23.167 [4B] describes several methods how the UE can get its location information from the access network or from a server. Such methods are not in the scope of this specification.

5) the UE shall insert in the INVITE request, one or two P-Preferred-Identity header field(s) that include the public user identity registered via emergency registration or the tel URI associated with the public user identity registered via emergency registration as described in subclause 4.2;

NOTE 2: Providing two P-Preferred-Identity header fields is usually supported by UE acting as enterprise network.

6) void;

7) if the UE has its location information available, or a URI that points to the location information, then the UE shall include a Geolocation header field in the INVITE request in the following way:

   - if the UE is aware of the URI that points to where the UE's location is stored, include the URI as the Geolocation header field value, as described in RFC 6442 [89]; or

   - if the UE is aware of its location information, include the location information in a PIDF location object, in accordance with RFC 4119 [90], include the location object in a message body with the content type application/pidf+xml, and include a Content ID URL, referring to the message body, as the Geolocation header field value, as described RFC 6442 [89], and include a Content-Disposition header field with a disposition type "render" value and a "handling" header field parameter with an "optional" value, as described in RFC 3261 [26];

8) if the UE includes a Geolocation header field, the UE shall also include a Geolocation-Routing header field with a "yes" header field value, which indicates that the location of the UE can be used by other entities to make routing decisions, as described in RFC 6442 [89];

NOTE 3: It is suggested that UE's only use the option of providing a URI when the domain part belongs to the current P-CSCF or S-CSCF provider. This is an issue on which the network operator needs to provide guidance to the end user. A URI that is only resolvable to the UE which is making the emergency call is not desirable.

9) if the UE has neither geographical location information available, nor a URI that points to the location information, the UE shall not insert a Geolocation header field in the INVITE request; and

10) if support of the current location discovery during an emergency call is allowed in the IP-CAN specific annex and the UE supports the current location discovery during an emergency call, the UE shall include a Recv-Info header field as described in RFC 6086 [25], indicating the g.3gpp.current-location-discovery info package name and shall include an Accept header field indicating the "application/vnd.3gpp.current-location-discovery+xml" MIME type.

NOTE 4: RFC 3261 [26] provides for the use of the Priority header field with a suggested value of "emergency". It is not precluded that emergency sessions contain this value, but such usage will have no impact on the processing within the IM CN subsystem.

[TS 24.237 clause 7.2]:

When originating an emergency call as specified in 3GPP TS 24.229 [2] and if the SC UE has an IMEI, then the SC UE shall include the sip.instance media feature tag as specified in IETF RFC 5626 [22] with value based on the IMEI as defined in 3GPP TS 23.003 [12] in the Contact header field of the SIP INVITE request according to IETF RFC 3840 [53].

[TS 23.003 clause 13.8]:

An instance-id is a SIP Contact header parameter that uniquely identifies the SIP UA performing a registration.

When an IMEI is available, the instance-id shall take the form of a IMEI URN (see RFC 7254 [79]). The format of the instance-id shall take the form "urn:gsma:imei:<imeival>" where by the imeival shall contain the IMEI encoded as defined in RFC 7254 [79]. The optional <sw-version-param> and <imei-version-param> parameters shall not be included in the instance-id. RFC 7255 [104] specifies additional considerations for using the IMEI as an instance-id. An example of such an instance-id is as follows:

EXAMPLE: urn:gsma:imei:90420156-025763-0

If no IMEI is available, the instance-id shall take the form of a string representation of a UUID as a URN as defined in IETF RFC 4122 [80]. An example of such an instance-id is as follows:

EXAMPLE: urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6

For more information on the instance-id and when it is used, see 3GPP TS 24.229 [81].

10.1.3 Test description

10.1.3.1 Pre-test conditions

System Simulator:

- 1 NR Cell connected to 5GC, default parameters.

UE:

- UE contains either ISIM and USIM applications or only USIM application on UICC.
- UE is configured to register for IMS after switch on.

Preamble:

- The UE is in test state 1N-A (TS 38.508-1 [21]) and registered to IMS.

10.1.3.2 Test procedure sequence

Table 10.1.3.2-1: Main Behaviour

St	Procedure	Message Sequence		TP	Verdict
		U - S	Message		
1	UE is made to make an emergency call				
2	Step 1 of annex A.3 (emergency registration) Check: Does the UE send a correctly composed initial REGISTER request for IMS emergency registration?	-->	REGISTER	1	P
3	Step 2 of annex A.3 (emergency registration)	<--	401 Unauthorized		
4	Step 3 of annex A.3 (emergency registration) Check: Does the UE correctly authenticate itself by sending another REGISTER request with a correctly composed Authorization header using the AKAv1-MD5 algorithm?	-->	REGISTER	2	P
5	Step 4 of annex A.3 (emergency registration)	<--	200 OK		
6	Step 1 of annex A.6 (emergency call) Check: Does the UE send a correctly composed INVITE request?	-->	INVITE	3	P
7	Step 2 of annex A.6 (emergency call)	<--	100 Trying		
8	Step 3 of annex A.6 (emergency call)	<--	180 Ringing		
9	Step 4 of annex A.6 (emergency call)	<--	200 OK		
10	Step 5 of annex A.6 (emergency call) Check: Does the UE send ACK?	-->	ACK	4	P
11	Step 1 of annex A.8 (MT Release of Voice Call)	<--	BYE		
12	Step 2 of annex A.8 (MT Release of Voice Call) Check: Does the UE send 200 OK for the BYE request and ends the call?	-->	200 OK	5	P

10.1.3.3 Specific message contents

None as fully described in annex A.3, A.6 and A.8.

Annex A (normative): Generic Test Procedures

A.1 Introduction

This annex specifies general procedures for IMS usages as well as application specific procedures, e.g. for a MTSI client.

A.2 IMS Registration / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		REGISTER	The UE sends initial registration for IMS services.
2	←		401 Unauthorized	The SS responds with a valid AKAv1-MD5 authentication challenge and security mechanisms supported by the network.
3	→		REGISTER	The UE completes the security negotiation procedures, sets up a temporary set of SAs and uses those for sending another REGISTER with AKAv1-MD5 credentials.
4	←		200 OK	The SS responds with 200 OK.
-			EXCEPTION: In parallel to the events described in steps 5-8, the steps specified in Annex A.10 on PUBLISH may happen.	
5	→		SUBSCRIBE	The UE subscribes to its registration event package.
6	←		200 OK	The SS responds with 200 OK.
7	←		NOTIFY	The SS sends initial NOTIFY for registration event package, containing full registration state information for the registered public user identity in the XML body.
8	→		200 OK	The UE responds with 200 OK.

Specific Message Contents

REGISTER (Step 1)

Use the default message "REGISTER" in Annex A.1.1 of TS 34.229-1 [2] applying condition A1.

401 Unauthorized (Step 2)

Use the default message "401 Unauthorized for REGISTER" in Annex A.1.2 of TS 34.229-1 [2] applying condition A1.

REGISTER (Step 3)

Use the default message "REGISTER" in Annex A.1.1 of TS 34.229-1 [2] applying conditions A2 and A32.

200 OK (Step 4)

Use the default message "200 OK for REGISTER" in Annex A.1.3 of TS 34.229-1 [2] applying condition A2.

SUBSCRIBE (Step 5)

Use the default message "SUBSCRIBE for reg-event package" in Annex A.1.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK (Step 6)

Use the default message "200 OK for SUBSCRIBE" in Annex A.1.4 of TS 34.229-1 [2] applying condition A1.

NOTIFY (Step 7)

Use the default message "NOTIFY for reg-event package" in Annex A.1.6 of TS 34.229-1 [2] applying condition A1.

200 OK (Step 8)

Use the default message "200 OK for requests other than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, and A22.

A.3 IMS Emergency Registration / 5GS

Test procedure:

1) SS waits for the UE to send an initial REGISTER request.

2) The SS responds to the initial REGISTER request with a valid 401 Unauthorized response.

3) The SS waits for the UE to set up a temporary set of security associations and to send another REGISTER request over those security associations.

4) The SS responds to the second REGISTER request with valid 200 OK response, sent over the same temporary set of security associations that the UE used for sending the REGISTER request.

Expected sequence:

Step	Direction		Message	Comment
	UE	SS		
1	→		REGISTER	The UE sends initial IMS emergency registration
2	←		401 Unauthorized	The SS responds with a valid AKAv1-MD5 authentication challenge and security mechanisms supported by the network.
3	→		REGISTER	The UE completes the security negotiation procedures, sets up a temporary set of SAs and uses those for sending another REGISTER with AKAv1-MD5 credentials.
4	←		200 OK	The SS responds with 200 OK.

Specific Message Contents:

REGISTER (Step 1)

Use the default message “REGISTER” in Annex A.1.1 of TS 34.229-1 [2] with conditions A1 and A7.

401 Unauthorized (Step 2)

Use the default message “401 Unauthorized for REGISTER” in Annex A.1.2 of TS 34.229-1 [2] with condition A1.

REGISTER (Step 3)

Use the default message “REGISTER” in Annex A.1.1 of TS 34.229-1 [2] with conditions A2, A7, and A32.

200 OK for REGISTER (Step 4)

Use the default message “200 OK for REGISTER” in Annex A.1.3 of TS 34.229-1 [2] with condition A3.

A.4 MTSI MO Voice Call / 5GS

A.4.1 MTSI MO Voice Call / with preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		INVITE	UE sends INVITE with the first SDP offer.
2	←		100 Trying	SS sends a 100 Trying provisional response.
3	←		183 Session Progress	SS sends an SDP answer.
4	→		PRACK	UE acknowledges reception of 183 Session Progress.
5	←		200 OK	SS responds to PRACK.
6	→		UPDATE	UE sends a second SDP offer in an UPDATE request.
7	←		200 OK	SS responds to UPDATE.
8	←		180 Ringing	SS sends 180 Ringing reliably.
9	→		PRACK	UE acknowledges reception of 180 Ringing.
10	←		200 OK	SS responds to PRACK.
11	←		200 OK	SS responds to INVITE.
12	→		ACK	UE acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MO Call Setup" in Annex A.2.1 of TS 34.229-1 [2] applying conditions A1, A3, A4, A28, A29, A30, and A31, and with the following exceptions:

Header/param	Value/Remark
Supported	
option-tag	*precondition*

<table>
<tr><td>Message-body</td><td>The following SDP types and values.

Session description:
v=0
o=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE)
s=(session name)
c=IN (addrtype) (connection-address for UE) [Note 1]
b=AS: (bandwidth-value)

Time description:
t= (start-time) (stop-time)

Media description:
m=audio (transport port) RTP/AVP (fmt)
c=IN (addrtype) (connection-address for UE) [Note 1]
b=AS: (bandwidth-value)
b=RS: (bandwidth-value) [Note 2]
b=RR: (bandwidth-value) [Note 2]

Attributes for media:
a=rtpmap: (payload type) EVS/16000 [Note 3, 9, 10]
a=fmtp: (format) br=5.9-13.2; bw=nb-swb; max-red= (att-field) [Note 4, 5, 10]
a=rtpmap: (payload type) EVS/16000 [Note 3, 9, 10]
a=fmtp: (format) br=5.9-24.4; bw=nb-swb; max-red= (att-field) [Note 4, 5, 10]
a=rtpmap: (payload type) EVS/16000 [Note 3, 9, 10]
a=fmtp: (format) br=13.2; bw=swb; max-red= (att-field) [Note 4, 5, 10]
a=rtpmap: (payload type) EVS/16000 [Note 3, 9, 10]
a=fmtp: (format) br=9.6-13.2; bw=swb; max-red= (att-field) [Note 4, 5, 10]
a=rtpmap: (payload type) EVS/16000 [Note 3, 9, 10]
a=fmtp: (format) br=9.6-24.4; bw=swb; max-red= (att-field) [Note 4, 5, 10]
a=rtpmap: (payload type) AMR-WB/16000 [Note 3, 9]
a=fmtp: (format) mode-change-capability=2; max-red= (att-field) [Note 4, 6]
a=rtpmap: (payload type) telephone-event/16000
a=fmtp: (format)
a=rtpmap: (payload type) AMR/8000 [Note 3, 9]
a=fmtp: (format) mode-change-capability=2; max-red= (att-field) [Note 4, 6]
a=rtpmap: (payload type) telephone-event/8000
a=fmtp: (format)
a=ecn-capable-rtp: leap ect=0 [Note 7]
a=rtcp-fb:* nack ecn [Note 7]
a=rtcp-xr:ecn-sum [Note 7]
a=rtcp-rsize [Note 7]
a=ptime:20
a=maxptime:240

Attributes for media security mechanism:
a=3ge2ae: requested [Note 8]
a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:WVNfX19zZW1jdGwgKCkgewkyMjA7fQp9 CnVubGVz|2^20|
1:4FEC_ORDER=FEC_SRTP" [Note 8]

Attributes for preconditions:
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv

Note 1: At least one "c=" field shall be present.
Note 2: The RR value shall be greater than 0. The RS value can be any value.
Note 3: The channel number shall be "/1" or omitted.
Note 4: The max-red values from 0 to 220 are allowed.
Note 5: The parameters dtx, dtx-recv and evs-mode-switch shall not be present.
Note 6: The parameters mode-set, mode-change-period, mode-change-neighbor, crc, robust-sorting and interleaving shall not be included.
Note 7: Attributes for ECN Capability may be present if the UE supports Explicit Congestion Notification.
Note 8: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE.</td></tr>
</table>

	Note 9: The ordering of payload types shall be as listed, i.e., EVS before AMR-WB before AMR according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3]. Note 10: The EVS payload type shall carry at least one of the five EVS configurations according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3].

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A1.

183 Session Progress (Step 3)

Use the default message "183 Session Progress for INVITE" in Annex A.2.3 of TS 34.229-1 [2] applying condition A1, and with the following exceptions:

Header/param	Value/Remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 7] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* *a=conf:qos remote sendrecv* Note 1: The values for fmt, payload type and format are copied from step 1. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 1. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 1. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 1. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in INVITE. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in INVITE.

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK for PRACK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A10 and A22.

UPDATE (Step 6)

Use the default message "UPDATE" in Annex A.2.5 of TS 34.229-1 [2] applying conditions A1 and A6, and with the following exceptions:

Header/param	Value/Remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values shall be present. **Session description:** *v=0* *o=*(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) [Note 2] *s=*(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 3] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3] [Note 5] [Note 6] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=*(att-field) [Note 7] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=*(att-field) [Note 8] **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one Note 3: The value for fmt, payload type and format is not checked Note 4: Void Note 5: The channel number shall be "/1" or omitted Note 6: EVS shall be the only codec in UPDATE and shall come with the same configuration parameters as sent in 183 Session Progress Note 7: Sent by UE if it sent it as first of its EVS configurations in INVITE. Note 8: Sent by UE if it did not send "br=13.2; bw=swb" as the first of its EVS configurations in INVITE.

200 OK for UPDATE (Step 7)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A1, A10 and A22, and with the following exceptions:

Header/param	Value/remark
Require option-tag	*precondition*
Content-Type media-type	*application/sdp*
Content-Length value	length of message-body
Message-body	SDP body of the 200 response copied from the received UPDATE and modified as follows: - IP address on "c=" lines and transport port on "m=" lines changed to indicate to which IP address and port the UE should start sending the media; - "o=" line identical to previous SDP sent by SS except that sess-version is incremented; - Attributes for preconditions: *a=curr:qos remote sendrecv*

180 Ringing (Step 8)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A3.

PRACK (Step 9)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK for PRACK (Step 10)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying condition A10.

200 OK for INVITE (Step 11)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A1, A10, and A19.

ACK (Step 12)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A3.

A.4.2 MTSI MO Voice Call / without preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		INVITE	UE sends INVITE with the first SDP offer.
2	←		100 Trying	SS sends a 100 Trying provisional response.
3	←		183 Session Progress	SS sends an SDP answer.
4	→		PRACK	UE acknowledges reception of 183 Session Progress.
5	←		200 OK	SS responds to PRACK.
6	←		180 Ringing	SS sends 180 Ringing.
7	←		200 OK	SS responds to INVITE.
8	→		ACK	UE acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MO Call" in Annex A.2.1 of TS 34.229-1 [2] applying conditions A1, A3, A4, A28, A29, A30, and A31, and with the following exceptions:

Header/param	Value/Remark
Message-body	The following SDP types and values. **Session description:** *v=0* *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS*: (bandwidth-value) **Time description:** *t*= (start-time) (stop-time) **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) [Note 2] *b=RR:* (bandwidth-value) [Note 2] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-24.4; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-24.4; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/16000* *a=fmtp:* (format) *a=rtpmap:* (payload type) *AMR/8000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/8000* *a=fmtp:* (format) *a=ecn-capable-rtp: leap ect=0* [Note 7] *a=rtcp-fb:* nack ecn* [Note 7] *a=rtcp-xr:ecn-sum* [Note 7] *a=rtcp-rsize* [Note 7] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 8] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:WVNfX19zZW1jdGwgKCkgewkyMjA7fQp9CnVubGVz|2^20|* *1:4FEC_ORDER=FEC_SRTP"* [Note 8] Note 1: At least one "c=" field shall be present. Note 2: The RR value shall be greater than 0. The RS value can be any value. Note 3: The channel number shall be "/1" or omitted. Note 4: The max-red values from 0 to 220 are allowed. Note 5: The parameters dtx, dtx-recv and evs-mode-switch shall not be present. Note 6: The parameters mode-set, mode-change-period, mode-change-neighbor, crc, robust-sorting and interleaving shall not be included. Note 7: Attributes for ECN Capability may be present if the UE supports Explicit Congestion Notification. Note 8: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 9: The ordering of payload types shall be as listed, i.e., EVS before AMR-WB before AMR according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3]. Note 10: The EVS payload type shall carry at least one of the five EVS configurations according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3].

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A1.

183 Session Progress (Step 3)

Use the default message "183 Session Progress" in Annex A.2.3 of TS 34.229-1 [2] applying condition A1, and with the following exceptions:

Header/param	Value/Remark
Message-body	The following SDP types and values. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0 mXXtxaVBR|2^20|1:4* [Note 7] Note 1: The values for fmt, payload type and format are copied from step 1. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 1. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 1. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 1. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in INVITE. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in INVITE.

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK for PRACK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A10 and A22.

180 Ringing (Step 6)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A14.

200 OK for INVITE (Step 7)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A1, A10, and A19.

ACK (Step 8)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A3.

A.5 MTSI MT Voice Call / 5GS

A.5.1 MTSI MT Voice Call / with preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	←		INVITE	SS sends INVITE with the first SDP offer.
2	→		100 Trying	Optional step: UE may send a 100 Trying provisional response.
3	→		183 Session Progress	UE sends 183 Session Progress response reliably, including an SDP answer.
4	←		PRACK	SS acknowledges reception of 183 Session Progress.
5	→		200 OK	UE responds to PRACK.
6	←		UPDATE	SS sends a second SDP offer
7	→		200 OK	UE responds to UPDATE, including an SDP answer.
8	→		180 Ringing	UE sends 180 Ringing.
9	←		PRACK	Conditional step: if UE sent 180 Ringing reliably, SS acknowledges reception of 180 Ringing
10	→		200 OK	Conditional step: if UE sent 180 Ringing reliably, UE responds to PRACK.
10A				Make UE accept the voice call.
11	→		200 OK	UE responds to INVITE.
12	←		ACK	SS acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MT Call" in Annex A.2.9 of TS 34.229-1 [2] applying conditions A1, A3, and A4, and with the following exceptions:

Header/param	Value/remark
Supported	
option-tag	*precondition*
Message-body	The following SDP types and values. Session description: *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP 96 97 98 99 100* *b=AS:65* *b=RS:0* *b=RR:2000* Attributes for media: *a=rtpmap: 96 EVS/16000/1* *a=fmtp: 96 br=13.2; bw=swb; max-red=220* *a=rtpmap:97 AMR-WB/16000/1* *a=fmtp:97 mode-change-capability=2; max-red=220* *a=rtpmap: 98 telephone-event/16000* *a=fmtp: 98 0-15* *a=rtpmap:99 AMR/8000/1* *a=fmtp:99 mode-change-capability=2; max-red=220* *a=rtpmap: 100 telephone-event/8000* *a=fmtp: 100 0-15* *a=ptime:20* *a=maxptime:240* Attributes for preconditions: *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv*

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A2.

183 Session Progress (Step 3)

Use the default message "183 Session Progress" in Annex A.2.3 of TS 34.229-1 [2] applying condition A2, and with the following exceptions:

Header/param	Value/remark
Status-Line Reason-Phrase	 Not checked
Require option-tag	 *precondition*
Message-body	The following SDP types and values shall be present. Session description: *v=0* *o*=(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) Attributes for media: *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) *br=13.2; bw=swb; max-red*=(att-field) Attributes for preconditions: *a=curr:qos local none* or *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, A11, and A22.

UPDATE (step 6)

Use the default message "UPDATE" in Annex A.2.5 of TS 34.229-1 [2] applying condition A3, and with the following exceptions:

Header/param	Value/remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values. Session description: *v=0* *o=- 1111111111 1111111112 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP 96* *b=AS:65* *b=RS:0* *b=RR:2000* Attributes for media: *a=rtpmap:96 EVS/16000/1* *a=fmtp:96 br=(*att-field*); bw=(*att-field*); max-red=220* [Note 2] *a=ptime:20* *a=maxptime:240* Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote none* or *curr:qos remote sendrecv* [Note 1] *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: Use the value (none/sendrecv) received from 183 Session Progress and attribute a=curr:qos local. Note 2: The br and bw values are taken from step 3.

200 OK (step 7)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A2, A11, and A22, and with the following exceptions:

Header/param	Value/remark
Require option-tag	 *precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body length of message-body
Message-body	The following SDP types and values shall be present. Session description: *v=0* *o*=(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) [Note 4] *s*=(session *name)* *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) Attributes for media: *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) [Note 2, 3] Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote sendrecv* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked Note 3: Parameters for the AMR codec are not checked Note 4: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one.

180 Ringing (Step 8)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A14, and with the following exceptions:

Header/param	Value/remark
Content-Type media-type	Header not present
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body 0
Message-body	Not present

PRACK (Step 9)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK (Step 10)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, A11, and A22.

200 OK (Step 11)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, A11, and A22.

ACK (Step 12)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A3.

A.5.2 MTSI MT Voice Call / without preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	←		INVITE	SS sends INVITE with the first SDP offer.
2	→		100 Trying	Optional step: UE may send a 100 Trying provisional response.
3	→		183 Session Progress	UE sends 183 Session Progress response reliably, including an SDP answer.
4	←		PRACK	SS acknowledges reception of 183 Session Progress.
5	→		200 OK	UE responds to PRACK.
6	→		180 Ringing	UE sends 180 Ringing.
7	←		PRACK	Conditional step: if UE sent 180 Ringing reliably, SS acknowledges reception of 180 Ringing
8	→		200 OK	Conditional step: if UE sent 180 Ringing reliably, UE responds to PRACK.
8A				Make UE accept the voice call.
9	→		200 OK	UE responds to INVITE.
10	←		ACK	SS acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MT Call" in Annex A.2.9 of TS 34.229-1 [2] applying conditions A1, A3, and A4, and with the following exceptions:

Header/param	Value/remark
Message-body	The following SDP types and values. Session description: *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP 96 97 98 99 100* *b=AS:65* *b=RS:0* *b=RR:2000* Attributes for media: *a=rtpmap: 96 EVS/16000/1* *a=fmtp: 96 br=13.2; bw=swb; max-red=220* *a=rtpmap:97 AMR-WB/16000/1* *a=fmtp:97 mode-change-capability=2; max-red=220* *a=rtpmap: 98 telephone-event/16000* *a=fmtp: 98 0-15* *a=rtpmap:99 AMR/8000/1* *a=fmtp:99 mode-change-capability=2; max-red=220* *a=rtpmap: 100 telephone-event/8000* *a=fmtp: 100 0-15* *a=ptime:20* *a=maxptime:240*

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A2.

183 Session Progress (Step 3)

Use the default message "183 Session Progress" in Annex A.2.3 of TS 34.229-1 [2] applying condition A2, and with the following exceptions:

Header/param	Value/remark
Status-Line Reason-Phrase	 Not checked
Message-body	The following SDP types and values shall be present. Session description: *v=0* *o=*(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) *s=*(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) Attributes for media: *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) *br=13.2; bw=swb; max-red=*(att-field) Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, A11, and A22.

180 Ringing (Step 6)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A14, and with the following exceptions:

Header/param	Value/remark
Content-Type media-type	Header not present
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body 0
Message-body	Not present

PRACK (Step 7)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK for PRACK (Step 8)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, A11, and A22.

200 OK for INVITE (Step 9)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A5, A8, A11, and A22.

ACK (Step 10)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A3.

A.6 IMS Emergency Voice Call / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		INVITE	UE sends INVITE with the first SDP offer.
2	←		100 Trying	SS sends a 100 Trying provisional response.
3	←		180 Ringing	SS sends a 180 Ringing.
4	←		200 OK	SS responds INVITE with 200 OK.
5	→		ACK	UE acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MO Call" in Annex A.2.1 of TS 34.229-1 [2] with conditions A7, A8, and A28 and the following exceptions:

Header/param	Value/remark
Message-body	The following SDP types and values. **Session description:** *v=0* *o=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE)* *s=(session name)* *c=IN (addrtype*) (connection-address for UE) [Note 1] **Time description:** *t=* (start-*time*) (stop-time) **Media description:** *m=audio* (transport port) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) Note 1: At least one "c=" field shall be present. Note 2: EVS codec shall be present in the media attributes, optionally including channel number "/1".

180 Ringing for INVITE (Step 3)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] with conditions A4 and A14.

200 OK for INVITE (Step 4)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] with conditions A6 and A22 and the following exceptions:

Header/param	Value/remark
Content-Type	
media-type	*application/sdp*
Content-Length	
value	length of message-body
Message-body	The following SDP types and values. **Session description:** *v=0* *o=- 1111111111 1111111111 IN (addrtype) (unicast-address for SS)* *s=-* *c=IN (addrtype) (connection-address for SS)* *b=AS:37* **Time description:** *t=0 0* **Media description:** *m=audio (transport port) RTP/AVP (fmt) [Note 1]* *b=AS:37* *b=RS:0* *b=RR:0* **Attributes for media:** *a=rtpmap: (payload type) EVS/16000/1 [Note 1]* *a=fmtp: (format) mode-change-capability=2; max-red=220* *a=ptime:20* *a=maxptime:240* Note 1: The value for fmt, payload type and format is copied from step 1.

A.7 MO Release of Voice Call / 5GS

Expected sequence

Step	Direction		Message/Procedure	Comment
	UE	SS		
1	→		BYE	The UE releases the call with BYE
2	←		200 OK	The SS sends 200 OK for BYE

Specific message contents

BYE (Step 1)

Use the default message "BYE" in Annex A.2.8 of TS 34.229-1 [2] with condition s A1 and A8.

200 OK (Step 2)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in annex A.3.1 of TS 34.229-1 [2] with condition A10.

A.8 MT Release of Voice Call / 5GS

Expected sequence

Step	Direction		Message/Procedure	Comment
	UE	SS		
1	←		BYE	The SS releases the call with BYE
2	→		200 OK	The UE sends 200 OK for BYE

Specific message contents

BYE (Step 1)

Use the default message "BYE" in Annex A.2.8 of TS 34.229-1 [2] with conditions A3 and A8.

200 OK (Step 2)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in annex A.3.1 of TS 34.229-1 [2] with conditions A5, A8, and A22.

A.9 EPS Fallback for Voice Call / 5GS

A.9.1 EPS Fallback for Voice Call / steps before fallback / 5GS

Expected sequence

Step	Direction		Message/Procedure	Comment
	UE	SS		
1	→		INVITE	UE sends INVITE including an SDP offer.
2	←		100 Trying	SS sends a 100 Trying provisional response.
3	←		183 Session Progress	SS sends 183 Session Progress including an SDP answer.
4	→		PRACK	UE acknowledges reception of 183 Session Progress.
5	←		200 OK	SS sends 200 OK for PRACK.

Specific message contents

INVITE (Step 1)

Use the default message "INVITE for MO Call Setup" in Annex A.2.1 of TS 34.229-1 [2] with conditions A1, A3, A4, and A28 and the following exceptions:

Header/param	Value/Remark
Message-body	SDP body present but contents not checked

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] with condition A1.

183 Session Progress (Step 3)

Use the default message "183 Session Progress for INVITE" in Annex A.2.3 of TS 34.229-1 [2] with condition A1 and the following exceptions:

Header/param	Value/Remark
Message-body	The following SDP types and values. Session description: - *v=0* - *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) - *s=-* - *c=IN* (addrtype) (connection-address for SS) - *b=AS:37* Time description: - *t=0 0* Media description: - *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 4] - *b=AS:37* - *b=RS:0* - *b=RR:2000* Attributes for media: - *a=rtpmap:* (payload type) *AMR-WB/16000/1* [Note 1] - *a=fmtp:* (format) *mode-change-capability=2; max-red=220* [Note 1] - *a=ecn-capable-rtp: leap ect=0* [Note 2] - *a=rtcp-fb:* nack ecn* [Note 2] - *a=rtcp-xr:ecn-sum* [Note 2] - *a=ptime:20* - *a=maxptime:240* Attributes for media security mechanism: - *a=3ge2ae: requested* [Note 3] - *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 3] Note 1: The value for fmt, payload type (AMR) and format is copied from Step 1. Note 2: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 3: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 4: transport port is the port number of the SS (see RFC 3264 clause 6).

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] with conditions A1 and A7.

200 OK (Step 5)

Use the default message "200 OK for requests other than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] with conditions A10 and A22.

A.9.2 EPS Fallback for Voice Call / steps after fallback / 5GS

Expected sequence

Step	Direction		Message/Procedure	Comment
	UE	SS		
0A	<--		180 Ringing	-
0B	-		SS starts a timer (5 seconds) to wait for optional UPDATE or REGISTER from the UE.	-
-	-		EXCEPTION: Step 1a1-1c1 describes behaviour that depends on UE implementation. The “lower case letter” identifies a step sequence that takes place if such implementation was applied.	-
1a1	-->		UPDATE	Optional: The UE sends an UPDATE request containing a second SDP offer.
1a2	<--		200 OK for UPDATE	If the UE sent UPDATE, the SS sends a 200 OK response for UPDATE containing an SDP answer.
1b1	-->		REGISTER	Optional: The UE sends a REGISTER request
1b2	<--		200 OK for REGISTER	If the UE sent REGISTER, the SS sends a 200 OK response for REGISTER containing an SDP answer.
1c1			The timer started in step 0B is expired.	
2	-		-Void	
3	<--		Void	
4	<--		200 OK	SS responds to INVITE with 200 OK.
5	-->		ACK	UE acknowledges.

Specific message contents

180 Ringing (Step 0A)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] with conditions A1 and A13.

UPDATE (Step 1a1)

Use the default message "UPDATE" in Annex A.2.5 of TS 34.229-1 [2] with conditions A1 and A5 and the following exceptions:

Header/param	Value/Remark
Message-body	The following SDP types and values shall be present. Session description: - *v=0* - *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) [Note 2] - *s*=(session name) - *c=IN* (addrtype) (connection-address for UE) [Note 1] - *b=AS:* (bandwidth-value) Time description: - *t=0 0* Media description: - *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] - *c=IN* (addrtype) (connection-address for UE) [Note 1] - *b=AS:* (bandwidth-value) - *b=RS:* (bandwidth-value) - *b=RR:* (bandwidth-value) Attributes for media: - *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 2] [Note 4] - *a=fmtp:* (format) [Note 2, 3] Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked Note 3: Parameters for the AMR codec are not checked Note 4: The AMR channel number shall be “/1” or omitted.

200 OK (Step 1a2)

Use the default message "200 OK for requests other than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] with conditions A1, A10 and A21 and the following exceptions:

Header/param	Value/remark
Content-Type	
media-type	*application/sdp*
Content-Length	
Value	length of message-body
Message-body	SDP body of the 200 OK response copied from the received UPDATE and modified as follows: - IP address on "c=" lines and transport port on "m=" lines changed to indicate to which IP address and port the UE should start sending the media; - "o=" line identical to previous SDP sent by SS except that sess-version is incremented.

REGISTER (Step 1b1)

Use the default message "REGISTER" in Annex A.1.1 of TS 34.229-1 [2] applying conditions A2 and A31.

200 OK (Step 1b2)

Use the default message "200 OK for REGISTER" in Annex A.1.3 of TS 34.229-1 [2] with condition A2.

200 OK (Step 4)

Use the default message "200 OK for requests other than REGISTER or SUBSRIBE" in Annex A.3.1 of TS 34.229-1 [2] with conditions A1, A10, A19, and A21.

ACK (Step 5)

Use the default message "ACK" in Annex A.2.7 of TS 34.229-1 [2] with condition A1.

A.10 Default handling of PUBLISH requests

This procedure may occur within 3 seconds after a successful IMS registration.

NOTE: For sake of testability and to mitigate detrimental effect on non-IMS test cases, it is assumed that such PUBLISH request arrives at SS within 3 seconds of sending 200 OK for REGISTER.

The generic test procedure:

1 SS receives from the UE a PUBLISH request.

2 The SS responds to the PUBLISH request with a 503 Service Unavailable response carrying a Retry-after header field big enough to quench further publication traffic during test case execution.

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		PUBLISH	The UE sends a PUBLISH request (A.4.3).
2	←		503 Service Unavailable	The SS responds with 503 Service Unavailable (A.4.2).

Specific Message Contents

PUBLISH (Step 1)

Use the default message "PUBLISH" in Annex A.4.3 of TS 34.229-1 [2] applying conditions A1 and A5.

503 Service Unavailable (Step 2)

Use the default message “503 Service Unavailable” in Annex A.4.2 of TS 34.229-1 [2] and with the following exceptions:

Header/param	Value/remark	Rel	Reference
Retry-after			RFC 3261 [6]
period	*7200*		
duration	Not present		
comment	Not present		

A.11 Mobile Initiated De-Registration / 5GS

IMS de-registration is initiated on the UE. The SS waits for the UE to send a REGISTER request, in accordance with 3GPP TS 24.229 [7], clause 5.1.1.6.

Expected sequence:

Step	Direction		Message	Comment
	UE	SS		
0A	→		SUBSCRIBE	Optional: The UE unsubscribes from one of its subscribed to event packages.
0B	←		200 OK	If the UE sent SUBSCRIBE, the SS responds to SUBSCRIBE with 200 OK.
0C	←		NOTIFY	If the UE sent SUBSCRIBE, the SS sends a final NOTIFY
0D	→		200 OK	If the UE sent SUBSCRIBE, the UE responds to NOTIFY with 200 OK.
1	→		REGISTER	The UE sends a de-registration request for IMS services.
2	←		200 OK	The SS responds to REGISTER with 200 OK.
Note 1: Steps 0A-0D may be repeated for any or all event packages subscribed to by the UE. It is the UE's decision which unsubscriptions to perform. Note 2: The UE can send the 200 OK for NOTIFY (step 0D) after the REGISTER request (step 1) or even not send it at all.				

Specific message contents

SUBSCRIBE (step 0A)

Use the default message "SUBSCRIBE for reg-event package" in Annex A.1.4 of TS 34.229-1 [2] or "SUBSCRIBE for conference event package" in Annex A.5.1 of TS 34.229-1 [2] or "SUBSCRIBE for message-summary event package" in Annex A.6.1 of TS 34.229-1 [2], and with the following exceptions:

Header/param	Cond	Value/remark	Rel	Reference
From				
addr-spec		Same as in original SUBSCRIBE that set up the corresponding subscription		
tag		Same as in original SUBSCRIBE that set up the corresponding subscription		
To				
addr-spec		As specified in TS 34.229-1 [2] Annex A.1.4/A.5.1/A.6.1		
tag		Same as in 200 OK for original SUBSCRIBE that set up the corresponding subscription		
CSeq				
value		value of the previous SUBSCRIBE sent by the UE for this dialog incremented by one		
method		*SUBSCRIBE*		
Expires				
delta-seconds		*0*		

200 OK for SUBSCRIBE (step 0B)

Use the default message "200 OK for SUBSCRIBE" in Annex A.1.5, A.5.2 or A.6.3 of TS 34.229-1 [2], whatever appropriate, with the following exceptions:

Header/param	Cond	Value/remark	Rel	Reference
To				RFC 3261 [6]
addr-spec		As specified in TS 34.229-1 [2] Annex A.1.4/A.5.1/A.6.1		
tag		Same as in step 0A		
Expires				RFC 3261 [6]
delta-seconds		*0*		

NOTIFY (step 0C)

Header/param	Cond	Value/remark	Rel	Reference
Request-Line Method Request-URI SIP-Version		 *NOTIFY* UE's contact address in SIP URI form, as provided in the Contact header within the SUBSCRIBE creating the dialog *SIP/2.0*		RFC 3261 [6]
Via **via-param1:** sent-protocol sent-by via-branch **via-param2:** sent-protocol sent-by via-branch		order of the parameters in this header must be like in this table *SIP/2.0/UDP* when using UDP or *SIP/2.0/TCP* when using TCP IP address and protected server port of SS value starting with '*z9hG4bK'* (NOTE 1) *SIP/2.0/UDP* when using UDP or *SIP/2.0/TCP* when using TCP *scscf.3gpp.org* value starting with '*z9hG4bK'* (NOTE 1)		RFC 3261 [6]
From addr-spec tag		 same URI as received in the To header of the corresponding SUBSCRIBE message same as to-tag in step 0A		RFC 3261 [6]
To addr-spec tag		 same URI as received in the From header of the corresponding SUBSCRIBE message same as from-tag in step 0A		RFC 3261 [6]
Call-ID callid		 same as value received in SUBSCRIBE message		RFC 3261 [6]
CSeq value method		 1 *NOTIFY*		RFC 3261 [6]
Contact addr-spec	 A1	 *<sip:scscf.3gpp.org>*		RFC 3261 [6]
addr-spec addr-spec	A2 A3	*sip:final@conf-factory.* appended with px_IMS_HomeDomainName *<scscf.3gpp.org>*		
Event event-type event-type event-type	 A1 A2 A3	 *reg* *conference* *message-summary*		RFC 6665 [28] RFC 3680 [18]
Max-Forwards value		 *69*		RFC 3261 [6]
Subscription-State substate-value		 *terminated*		RFC 6665 [28]
Content-Length value		 *0*		

Condition	Explanation
A1	Final NOTIFY sent for reg-event
A2	Final NOTIFY sent for conf-event
A3	Final NOTIFY sent for message-summary

NOTE 1: Branch parameter values sent by SS are different within a test case execution.

200 OK (step 0D)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2].

REGISTER (step 1)

Use the default message “REGISTER” in Annex A.1.1 of TS 34.229-1 [1] with conditions A2 and A17 "UE initiated IMS re-registration or de-registration" with the following exceptions:

Header/param	Cond	Value/remark	Rel	Reference
Contact				RFC 3261 [6]
addr-spec		SIP URI with IP address or FQDN and protected server port of the UE, AND, if the UE supports GRUU, the following parameter: +*sip.instance*="<*urn:gsma:imei:* (gsma-specifier-defined-substring)>”, OR *		
expires		*0* (if present)		
Expires		(must be present if addr-spec is *)		RFC 3261 [6]
delta-seconds		*0* (if present)		
Supported		header may be missing or it may contain any value		
Authorization		value not checked		

NOTE: In contrast to Annex A.1.1 of TS 34.229-1 [2], the Contact header does not have any further mandatory feature parameters.

200 OK (step 2)

Use the default message “200 OK for REGISTER” in Annex A.1.3 of TS 34.229-1 [2] with the following exceptions:

Header/param	Cond	Value/remark	Rel	Reference
Contact				RFC 3261 [6]
addr-spec		same value as in REGISTER request if "*" is not included in the Contact header field of the REGISTER request in step 1 same value as in the Contact header field of the "200 OK" response to the initial registration if "*" is included in the Contact header field of the REGISTER request in step 1 (NOTE)		
expires		*0*		
NOTE: According to 3GPP TS 24.229 [7] clause 5.4.1.4.1 when the S-CSCF gets a wild-carded contact address for de-registration it shall include all de-registered contact addresses in the contact header of the 200 OK response ⇒ there is no “*” in DL				

A.12 IMS Re-Registration / 5GS

The generic test procedure for IMS re-registration

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		REGISTER	The UE sends a re-registration request.
2	←		200 OK	The SS responds with 200 OK.

REGISTER (Step 1)

Use the default message "REGISTER" in Annex A.1.1 of TS 34.229-1 [2] with conditions A2 and A32.

A.13 IMS MO SMS / 5GS

Expected sequence

Step	Direction		Message/Procedure	Comment
	UE	SS		
1	→		SIP MESSAGE	UE sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a short message
2	←		202 Accepted	The SS responds with 202 Accepted
3	←		SIP MESSAGE	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains the short message submission report indicating a positive acknowledgement of the short message sent by the UE at Step 1
4	→		200 OK	UE responds with 200 OK
5	←		SIP MESSAGE	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a status report
6	→		200 OK	UE responds with 200 OK
7	→		SIP MESSAGE	UE sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains an acknowledgement for the status report received at Step 6
8	←		202 Accepted	The SS responds with 202 Accepted

Specific message contents:

SIP MESSAGE (Step 1)

Use the default “MESSAGE for MO SMS” in Annex A.7.3 of TS 34.229-1 [2] with conditions A2 and A5.

202 Accepted (Step 2 and 8)

Use the default “202 Accepted” in Annex A.3.3 of TS 34.229-1 [2].

SIP MESSAGE (Step 3)

Use the default “MESSAGE for submission report for MO SMS” in Annex A.7.4 of TS 34.229-1 [2].

200 OK (step 4 and 6)

Use the default message “200 OK for requests other than REGISTER or SUBSCRIBE” in Annex A.3.1 of TS 34.229-1 [2] with conditions A5 and A22.

SIP MESSAGE (Step 5)

Use the default “MESSAGE for status report for MO SMS” in Annex A.7.5 of TS 34.229-1 [2].

SIP MESSAGE (Step 7)

Use the default “MESSAGE for delivery report for MO SMS” in Annex A.7.6 of TS 34.229-1 [2].

A.14 IMS MT SMS / 5GS

Expected sequence

Step	Direction		Message/Procedure	Comment
	UE	SS		
1	←		SIP MESSAGE	SS sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a short message
2	→		200 OK	UE responds with 200 OK
3	←		SIP MESSAGE	UE sends a SIP MESSAGE request including a vnd.3gpp.sms payload that contains a delivery report
4	→		202 Accepted	The SS responds with 202 Accepted

Specific message contents:

SIP MESSAGE (Step 1)

Use the default "MESSAGE for status report for MO SMS" in Annex A.7.1 of TS 34.229-1 [2].

200 OK (Step 2)

Use the default message "200 OK for requests other than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] with conditions A22.

SIP MESSAGE (Step 3)

Use the default "MESSAGE for delivery report for MT SMS" in Annex A.7.2 of TS 34.229-1 [2].

202 Accepted (Step 4)

Use the default "202 Accepted" in Annex A.3.3 of TS 34.229-1 [2].

A.15 MTSI MO Video Call / 5GS

A.15.1 MTSI MO Video Call / with preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		INVITE	UE sends INVITE with the first SDP offer.
2	←		100 Trying	SS sends a 100 Trying provisional response.
3	←		183 Session Progress	SS sends an SDP answer.
4	→		PRACK	UE acknowledges reception of 183 Session Progress.
5	←		200 OK	SS responds to PRACK.
6	→		UPDATE	UE sends a second SDP offer in an UPDATE request.
7	←		200 OK	SS responds to UPDATE.
8	←		180 Ringing	SS sends 180 Ringing reliably.
9	→		PRACK	UE acknowledges reception of 180 Ringing.
10	←		200 OK	SS responds to PRACK.
11	←		200 OK	SS responds to INVITE.
12	→		ACK	UE acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MO Call Setup" in Annex A.2.1 of TS 34.229-1 [2] applying conditions A1, A3, A4, A28, A29, A30, and A31, and with the following exceptions:

Header/param	Value/Remark
Supported	
option-tag	*precondition*

Message-body	The following SDP types and values. **Session description:** *v=0* *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS*: (bandwidth-value) **Time description:** *t*= (start-time) (stop-time) **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) [Note 2] *b=RR:* (bandwidth-value) [Note 2] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-24.4; bw=nb-swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-13.2; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-24.4; bw=swb; max-red=* (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/16000* *a=fmtp:* (format) *a=rtpmap:* (payload type) *AMR/8000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red=* (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/8000* *a=fmtp:* (format) *a=ecn-capable-rtp: leap ect=0* [Note 7] *a=rtcp-fb:* nack ecn* [Note 7] *a=rtcp-xr:ecn-sum* [Note 7] *a=rtcp-rsize* [Note 7] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 8] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:WVNfX19zZW1jdGwgKCkgewkyMjA7fQp9 CnVubGVz|2^20|* *1:4FEC_ORDER=FEC_SRTP"* [Note 8] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) or RTP/AVP (fmt) [Note 11] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=tcap:1 RTP/AVPF* [Note 11] *a=pcfg:1 t=1* [Note 11] *a=rtpmap:* (payload type) *H265/90000*

	a=fmtp: (format) profile-id=1;level-id=(att-field) *a=rtpmap:* (payload type) *H264/90000* *a=fmtp:* (format) *profile-level-id*= (att-field) **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The RR value shall be greater than 0. The RS value can be any value. Note 3: The channel number shall be "/1" or omitted. Note 4: The max-red values from 0 to 220 are allowed. Note 5: The parameters dtx, dtx-recv and evs-mode-switch shall not be present. Note 6: The parameters mode-set, mode-change-period, mode-change-neighbor, crc, robust-sorting and interleaving shall not be included. Note 7: Attributes for ECN Capability may be present if the UE supports Explicit Congestion Notification. Note 8: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 9: The ordering of payload types shall be as listed, i.e., EVS before AMR-WB before AMR according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3]. Note 10: The EVS payload type shall carry at least one of the five EVS configurations according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3]. Note 11: The tcap/pcfg attributes are present if RTP/AVP is present on the m line.

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A1.

183 Session Progress (Step 3)

Use the default message "183 Session Progress for INVITE" in Annex A.2.3 of TS 34.229-1 [2] applying condition A1, and with the following exceptions:

Header/param	Value/Remark
Require	
option-tag	*precondition*

Message-body	The following SDP types and values. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mX XtxaVBR|2^20|1:4* [Note 7] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* *a=conf:qos remote sendrecv* **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) [Note 1] *b=AS:* (bandwidth-value) [Note 1] *b=RS:* (bandwidth-value) [Note 1] *b=RR:* (bandwidth-value) [Note 1] **Attributes for media:** *a=acfg:1 t=1* [Note 10] *a=rtpmap:* (payload type) *H265/90000* [Note 1] *a=fmtp:* (format) (format specific parameters) [Note 1] **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remotel sendrecv* *a=des:qos remote sendrecv* Note 1: The values for fmt, bandwidth, payload type, format and format specific parameters are copied from step 1. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 1. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 1. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 1. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification.

	Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in INVITE. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in INVITE. Note 10: Present if tcap/pcfg attributes were included in step 1

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK for PRACK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A10 and A22.

UPDATE (Step 6)

Use the default message "UPDATE" in Annex A.2.5 of TS 34.229-1 [2] applying conditions A1 and A6, and with the following exceptions:

Header/param	Value/Remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values shall be present. **Session description:** *v=0* *o=*(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) [Note 2] *s=*(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 3] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3] [Note 5] *a=fmtp:* (format) [Note 3] [Note 4] **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv* **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1;level-id=(att-field) **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* or *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one Note 3: The value for fmt, payload type and format is not checked Note 4: Parameters for the codec are not checked Note 5: The channel number shall be "/1" or omitted.

200 OK for UPDATE (Step 7)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A1, A10 and A22, and with the following exceptions:

Header/param	Value/remark
Require option-tag	*precondition*
Content-Type media-type	*application/sdp*
Content-Length value	length of message-body
Message-body	SDP body of the 200 response copied from the received UPDATE and modified as follows: - IP address on "c=" lines and transport port on "m=" lines changed to indicate to which IP address and port the UE should start sending the media; - "o=" line identical to previous SDP sent by SS except that sess-version is incremented; - Attributes for preconditions: *a=curr:qos remote sendrecv*

180 Ringing (Step 8)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A3.

PRACK (Step 9)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK for PRACK (Step 10)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying condition A10.

200 OK for INVITE (Step 11)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A1, A10, and A19.

ACK (Step 12)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A3.

A.15.2 MTSI MO Video Call / without preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	→		INVITE	UE sends INVITE with the first SDP offer.
2	←		100 Trying	SS sends a 100 Trying provisional response.
3	←		183 Session Progress	SS sends an SDP answer.
4	→		PRACK	UE acknowledges reception of 183 Session Progress.
5	←		200 OK	SS responds to PRACK.
6	←		180 Ringing	SS sends 180 Ringing reliably.
7	→		PRACK	UE acknowledges reception of 180 Ringing.
8	←		200 OK	SS responds to PRACK.
9	←		200 OK	SS responds to INVITE.
10	→		ACK	UE acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MO Call Setup" in Annex A.2.1 of TS 34.229-1 [2] applying conditions A1, A3, A4, A28, A29, A30, and A31, and with the following exceptions:

Header/param	Value/Remark
Supported	
option-tag	*precondition*

Message-body	The following SDP types and values. **Session description:** *v=0* *o*=(username) (sess-id) (sess-version) IN (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS*: (bandwidth-value) **Time description:** *t*= (start-time) (stop-time) **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) [Note 2] *b=RR:* (bandwidth-value) [Note 2] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; max-red*= (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=5.9-24.4; bw=nb-swb; max-red*= (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=13.2; bw=swb; max-red*= (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-13.2; bw=swb; max-red*= (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *EVS/16000* [Note 3, 9, 10] *a=fmtp:* (format) *br=9.6-24.4; bw=swb; max-red*= (att-field) [Note 4, 5, 10] *a=rtpmap:* (payload type) *AMR-WB/16000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red*= (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/16000* *a=fmtp:* (format) *a=rtpmap:* (payload type) *AMR/8000* [Note 3, 9] *a=fmtp:* (format) *mode-change-capability=2; max-red*= (att-field) [Note 4, 6] *a=rtpmap:* (payload type) *telephone-event/8000* *a=fmtp:* (format) *a=ecn-capable-rtp: leap ect=0* [Note 7] *a=rtcp-fb:* nack ecn* [Note 7] *a=rtcp-xr:ecn-sum* [Note 7] *a=rtcp-rsize* [Note 7] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 8] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:WVNfX19zZW1jdGwgKCkgewkyMjA7fQp9 CnVubGVz|2^20|* *1:4FEC_ORDER=FEC_SRTP"* [Note 8] **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) or RTP/AVP (fmt) [Note 11] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=tcap:1 RTP/AVPF* [Note 11] *a=pcfg:1 t=1* [Note 11] *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1;level-id=(att-field) *a=tcap:1 RTP/AVPF* [Note 11] *a=pcfg:1 t=1* [Note 11] *a=rtpmap:* (payload type) *H264/90000* *a=fmtp:* (format) *profile-level-id*= (att-field)

	Note 1: At least one "c=" field shall be present. Note 2: The RR value shall be greater than 0. The RS value can be any value. Note 3: The channel number shall be "/1" or omitted. Note 4: The max-red values from 0 to 220 are allowed. Note 5: The parameters dtx, dtx-recv and evs-mode-switch shall not be present. Note 6: The parameters mode-set, mode-change-period, mode-change-neighbor, crc, robust-sorting and interleaving shall not be included. Note 7: Attributes for ECN Capability may be present if the UE supports Explicit Congestion Notification. Note 8: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 9: The ordering of payload types shall be as listed, i.e., EVS before AMR-WB before AMR according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3]. Note 10: The EVS payload type shall carry at least one of the five EVS configurations according to NG.114 [31] and corresponding capability A.22/4 of TS 34.229-2 [3]. Note 11: The tcap/pcfg attributes are present if RTP/AVP is present on the m line.

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A2.

183 Session Progress (Step 3)

Use the default message "183 Session Progress for INVITE" in Annex A.2.3 of TS 34.229-1 [2] applying condition A2, and with the following exceptions:

Header/param	Value/Remark
Require option-tag	 *precondition*
Message-body	The following SDP types and values. **Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:65* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 1, 2] *b=AS:65* *b=RS:* (bandwidth-value) [Note 3] *b=RR:* (bandwidth-value) [Note 3] **Attributes for media:** *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 8] *a=fmtp:* (format) *br=13.2; bw=swb; mode-set=0,1,2; max-red=220* [Note 8] *a=rtpmap:* (payload type) *EVS/16000/1* [Note 1, 9] *a=fmtp:* (format) *br=5.9-13.2; bw=nb-swb; mode-set=0,1,2, max-red=220* [Note 9] *a=ecn-capable-rtp: leap ect=0* [Note 6] *a=rtcp-fb:* nack ecn* [Note 6] *a=rtcp-xr:ecn-sum* [Note 6] *a=ptime:20* *a=maxptime:240* **Attributes for media security mechanism:** *a=3ge2ae: requested* [Note 7] *a=crypto:1 AES_CM_128_HMAC_SHA1_80inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4* [Note 7] **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) [Note 1] *b=AS:* (bandwidth-value) [Note 1] *b=RS:* (bandwidth-value) [Note 1] *b=RR:* (bandwidth-value) [Note 1] **Attributes for media:** *a=acfg:1 t=1* [Note 10] *a=rtpmap:* (payload type) *H265/90000* [Note 1] *a=fmtp:* (format) (format specific parameters) [Note 1] Note 1: The values for fmt, bandwidth, payload type, format and format specific parameters are copied from step 1. Note 2: Transport port is the port number of the SS (see RFC 3264 clause 6). Note 3: The bandwidth-value is copied from step 1. Note 4: All present br, br-send and br-recv parameter=value pairs are copied from step 1. Note 5: bw, bw-send and bw-recv parameter are copied from bw at step 1. Note 6: Attributes for ECN Capability are present if the UE supports Explicit Congestion Notification. Note 7: Attributes for media plane security are present if the use of end-to-access-edge security is supported by UE. Note 8: This EVS configuration is sent if UE sent it as the first of its EVS configurations in INVITE. Note 9: This EVS configuration is sent if UE did not send "br=13.2; bw=swb" as the first of its EVS configurations in INVITE. Note 10: Present if tcap/pcfg attributes were included in step 1

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying conditions A1 and A7.

200 OK for PRACK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A10 and A22.

180 Ringing (Step 6)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A14.

PRACK (Step 7)

Editor's note: probably to be fixed.

s

200 OK for PRACK (Step 8)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying condition A10.

200 OK for INVITE (Step 9)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A1, A10, and A19.

ACK (Step 10)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A1 and A3.

A.16 MTSI MT Video Call / 5GS

A.16.1 MTSI MT Video Call / with preconditions / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	←		INVITE	SS sends INVITE with the first SDP offer.
2	→		100 Trying	Optional step: UE may send a 100 Trying provisional response.
3	→		183 Session Progress	UE sends 183 Session Progress response reliably, including an SDP answer.
4	←		PRACK	SS acknowledges reception of 183 Session Progress.
5	→		200 OK	UE responds to PRACK.
6	←		UPDATE	SS sends a second SDP offer
7	→		200 OK	UE responds to UPDATE, including an SDP answer.
8	→		180 Ringing	UE sends 180 Ringing.
9	←		PRACK	Conditional step: if UE sent 180 Ringing reliably, SS acknowledges reception of 180 Ringing
10	→		200 OK	Conditional step: if UE sent 180 Ringing reliably, UE responds to PRACK.
10A				Make UE accept the voice call.
11	→		200 OK	UE responds to INVITE.
12	←		ACK	SS acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MT Call" in Annex A.2.9 of TS 34.229-1 [2] applying conditions A1, A3, A4, A7, and A8, and with the following exceptions:

Header/param	Value/remark
Supported	
option-tag	*precondition*
Content-Type	
media-type	*application/sdp*
Content-Length	
value	length of message-body

Message-body	**Session description:** *v=0* *o=- 1111111111 1111111111 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:540* **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP 96 97 98 99 100* *b=AS:65* *b=RS:0* *b=RR:2000* **Attributes for media:** *a=rtpmap: 96 EVS/16000/1* *a=fmtp: 96 br=13.2; bw=swb; max-red=220* *a=rtpmap:97 AMR-WB/16000/1* *a=fmtp:97 mode-change-capability=2; max-red=220* *a=rtpmap: 98 telephone-event/16000* *a=fmtp: 98 0-15* *a=rtpmap:99 AMR/8000/1* *a=fmtp:99 mode-change-capability=2; max-red=220* *a=rtpmap: 100 telephone-event/8000* *a=fmtp: 100 0-15* *a=ptime:20* *a=maxptime:240* **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv* **Media description:** *m=video (transport port) RTP/AVPF 101* *a=tcap:1 RTP/AVPF* *a=pcfg:1 t=1* *b=AS: 540* *b=RS: 0* *b=RR: 5000* **Attributes for media:** *a=rtpmap: 101 H265/90000* *a=fmtp: 101 profile-id=1; level-id=93; * *sprop-vps=QAEMAf//AWAAAAMAgAAAAwAAAwBaLAUg; * *sprop-sps=QgEBAWAAAAMAgAAAAwAAAwBaoAaiAeFlLktIvQB3CAQQ; * *sprop-pps=RAHAcYDZIA==* *a=imageattr:101 send [x=848,y=480] recv [x=848,y=480]* *a=rtcp-fb:* trr-int 5000* *a=rtcp-fb:* nack* *a=rtcp-fb:* nack pli* *a=rtcp-fb:* ccm fir* *a=rtcp-fb:* ccm tmmbr* *a=extmap:4 urn:3gpp:video-orientation* **Attributes for preconditions:** *a=curr:qos local none* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos optional remote sendrecv*

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A2.

183 Session Progress (Step 3)

Use the default message "183 Session Progress" in Annex A.2.3 of TS 34.229-1 [2] applying condition A2 and A6, and with the following exceptions:

Header/param	Value/remark
Require option-tag	 *precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body length of message-body
Message-body	**Session description:** *v=0* *o*=(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) *s*=(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) *br=13.2; bw=swb; max-red*=(att-field) **Attributes for preconditions:** *a=curr:qos local none* or *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* **Media description:** *m=video (transport port) RTP/AVPF* (fmt) *a=acfg:1 t=1* *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1; level-id=93; \ **Attributes for preconditions:** *a=curr:qos local none* or *a=curr:qos local sendrecv* *a=curr:qos remote none* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A8, A11, A18, and A22.

UPDATE (step 6)

Use the default message "UPDATE" in Annex A.2.5 of TS 34.229-1 [2] applying condition A3, and with the following exceptions:

Header/param	Value/remark
Require option-tag	*precondition*
Content-Type	
media-type	*application/sdp*
Content-Length	
value	length of message-body
Message-body	**Session description:** *v=0* *o=- 1111111111 1111111112 IN* (addrtype) (unicast-address for SS) *s=-* *c=IN* (addrtype) (connection-address for SS) *b=AS:540* Time description: *t=0 0* Media description: *m=audio* (transport port) *RTP/AVP 96* *b=AS:65* *b=RS:0* *b=RR:2000* Attributes for media: *a=rtpmap:96 EVS/16000/1* *a=fmtp:96 br=(*att-field*); bw=(*att-field*); max-red=220* [Note 2] *a=ptime:20* *a=maxptime:240* Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote none* or *curr:qos remote sendrecv* [Note 1] *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Media description: *m=video (transport port) RTP/AVPF 101* *a=tcap:1 RTP/AVPF* *a=pcfg:1 t=1* *b=AS: 540* *b=RS: 0* *b=RR: 5000* Attributes for media: *a=rtpmap: 101 H265/90000* *a=fmtp: 101 profile-id=1; level-id=93; * *sprop-vps=QAEMAf//AWAAAAMAgAAAAwAAAwBaLAUg; * *sprop-sps=QgEBAWAAAAMAgAAAAwAAAwBaoAaiAeFlLktIvQB3CAQQ; * *sprop-pps=RAHAcYDZIA==* *a=imageattr:101 send [x=848,y=480] recv [x=848,y=480]* *a=rtcp-fb:* trr-int 5000* *a=rtcp-fb:* nack* *a=rtcp-fb:* nack pli* *a=rtcp-fb:* ccm fir* *a=rtcp-fb:* ccm tmmbr* *a=extmap:4 urn:3gpp:video-orientation* Attributes for preconditions: *a=curr:qos local sendrecv* *a=curr:qos remote none* or *curr:qos remote sendrecv* [Note 1] *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: Use the value (none/sendrecv) received from 183 Session Progress and attribute a=curr:qos local. Note 2: The br and bw values are taken from step 3.

200 OK (step 7)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A2, A11, A18, and A22, and with the following exceptions:

Header/param	Value/remark
Require option-tag	*precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body length of message-body
Message-body	**Session description:** *v=0* *o=*(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) [Note 4] *s=*(session *name)* *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) [Note 2, 3] **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote sendrecv* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* **Media description:** *m=video* (transport port) *RTP/AVPF* (fmt) *a=acfg:1 t=1* *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1; level-id=93; \ **Attributes for preconditions:** *a=curr:qos local sendrecv* *a=curr:qos remote sendrecv* *a=des:qos mandatory local sendrecv* *a=des:qos mandatory remote sendrecv* Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked Note 3: Parameters for the AMR codec are not checked Note 4: "o=" line identical to previous SDP sent by UE except that sess-version is incremented by one.

180 Ringing (Step 8)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A14, and with the following exceptions:

Header/param	Value/remark
Content-Type media-type	Header not present
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body 0
Message-body	Not present

PRACK (Step 9)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK (Step 10)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A8, A11, and A22.

200 OK (Step 11)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A8, A11, and A22.

ACK (Step 12)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A3.

A.16.2 MTSI MT Video Call / without preconditions / 5GS

Expected sequence

Step	Direction UE SS	Message	Comment
1	←	INVITE	SS sends INVITE with the first SDP offer.
2	→	100 Trying	Optional step: UE may send a 100 Trying provisional response.
3	→	183 Session Progress	UE sends 183 Session Progress response reliably, including an SDP answer.
4	←	PRACK	SS acknowledges reception of 183 Session Progress.
5	→	200 OK	UE responds to PRACK.
6	→	180 Ringing	UE sends 180 Ringing.
7	←	PRACK	Conditional step: if UE sent 180 Ringing reliably, SS acknowledges reception of 180 Ringing
8	→	200 OK	Conditional step: if UE sent 180 Ringing reliably, UE responds to PRACK.
8A			Make UE accept the voice call.
9	→	200 OK	UE responds to INVITE.
10	←	ACK	SS acknowledges.

Specific Message Contents

INVITE (Step 1)

Use the default message "INVITE for MT Call" in Annex A.2.9 of TS 34.229-1 [2] applying conditions A1, A3, A4, A7, and A8, and with the following exceptions:

<table>
<tr><th>Header/param</th><th>Value/remark</th></tr>
<tr><td>Supported
option-tag</td><td>
precondition</td></tr>
<tr><td>Content-Type
media-type</td><td>
application/sdp</td></tr>
<tr><td>Content-Length
value</td><td>
length of message-body</td></tr>
<tr><td>Message-body</td><td>Session description:
v=0
o=- 1111111111 1111111111 IN (addrtype) (unicast-address for SS)
s=-
c=IN (addrtype) (connection-address for SS)
b=AS:540

Time description:
t=0 0

Media description:
m=audio (transport port) RTP/AVP 96 97 98 99 100
b=AS:65
b=RS:0
b=RR:2000

Attributes for media:
a=rtpmap: 96 EVS/16000/1
a=fmtp: 96 br=13.2; bw=swb; max-red=220
a=rtpmap:97 AMR-WB/16000/1
a=fmtp:97 mode-change-capability=2; max-red=220
a=rtpmap: 98 telephone-event/16000
a=fmtp: 98 0-15
a=rtpmap:99 AMR/8000/1
a=fmtp:99 mode-change-capability=2; max-red=220
a=rtpmap: 100 telephone-event/8000
a=fmtp: 100 0-15
a=ptime:20
a=maxptime:240

Media description:
m=video (transport port) RTP/AVPF 101
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS: 540
b=RS: 0
b=RR: 5000

Attributes for media:
a=rtpmap: 101 H265/90000
a=fmtp: 101 profile-id=1; level-id=93; \
  sprop-vps=QAEMAf//AWAAAAMAgAAAAwAAAwBaLAUg; \
  sprop-sps=QgEBAWAAAAMAgAAAAwAAAwBaoAaiAeFlLktIvQB3CAQQ; \
  sprop-pps=RAHAcYDZIA==
a=imageattr:101 send [x=848,y=480] recv [x=848,y=480]
a=rtcp-fb:* trr-int 5000
a=rtcp-fb:* nack
a=rtcp-fb:* nack pli
a=rtcp-fb:* ccm fir
a=rtcp-fb:* ccm tmmbr
a=extmap:4 urn:3gpp:video-orientation</td></tr>
</table>

100 Trying (Step 2)

Use the default message "100 Trying for INVITE" in Annex A.2.2 of TS 34.229-1 [2] applying condition A2.

183 Session Progress (Step 3)

Use the default message "183 Session Progress" in Annex A.2.3 of TS 34.229-1 [2] applying condition A2 and A6, and with the following exceptions:

Header/param	Value/remark
Require option-tag	 *precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body length of message-body
Message-body	**Session description:** *v=0* *o=*(user-name) (sess-id) (sess-version) *IN* (addrtype) (unicast-address for UE) *s=*(session name) *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) **Time description:** *t=0 0* **Media description:** *m=audio* (transport port) *RTP/AVP* (fmt) [Note 2] *c=IN* (addrtype) (connection-address for UE) [Note 1] *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:*(payload type) *EVS/16000* [Note 2] *a=fmtp:*(format) *br=13.2; bw=swb; max-red=*(att-field) **Media description:** *m=video (transport port) RTP/AVPF* (fmt) *a=acfg:1 t=1* *b=AS:* (bandwidth-value) *b=RS:* (bandwidth-value) *b=RR:* (bandwidth-value) **Attributes for media:** *a=rtpmap:* (payload type) *H265/90000* *a=fmtp:* (format) profile-id=1; level-id=93; \ Note 1: At least one "c=" field shall be present. Note 2: The value for fmt, payload type and format is not checked

PRACK (Step 4)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK for PRACK (Step 5)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A8, A11, A18, and A22.

180 Ringing (Step 6)

Use the default message "180 Ringing for INVITE" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A14, and with the following exceptions:

Header/param	Value/remark
Content-Type media-type	Header not present
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body 0
Message-body	Not present

PRACK (Step 7)

Use the default message "PRACK" in Annex A.2.4 of TS 34.229-1 [2] applying condition A3.

200 OK for PRACK (Step 8)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A8, A11, and A22.

200 OK for INVITE (Step 9)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2] applying conditions A8, A11, and A22.

ACK (Step 10)

Use the default message "ACK" in Annex A.2.6 of TS 34.229-1 [2] applying conditions A2 and A3.

A.17 Generic test procedure for putting a MTSI speech call to hold or to resume the call from the UE / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	->		INVITE or UPDATE	UE sends INVITE or UPDATE with a SDP offer to hold or resume the call
2	<-		100 Trying	The SS responds to the INVITE with a 100 Trying provisional response
3	<-		200 OK	The SS responds to INVITE or UPDATE with 200 OK to indicate that the remote UE is no more sending any media (call hold) or resumes sending media (call resume)
4	->		ACK	Conditional: If the UE sent INVITE in step 1 then UE acknowledges the receipt of 200 OK for INVITE

Specific Message Contents

INVITE or UPDATE (Step 1)

Use the default message “INVITE for MO call setup” in Annex A.2.1 of TS 34.229-1 [2] applying conditions A1, A3, A5 and A28 or “UPDATE” in Annex A.2.5 applying conditions A1 and A6 of TS 34.229-1 [2]. In case of an INVITE the UE shall use also the same URI in the request line as the SS has sent in the Contact header of an earlier message within the same dialog (in case of an UPDATE ref. to A.2.5 of TS 34.229-1 [2]).

The UE shall include support for precondition in the Supported header field.

The UE shall include an SDP body as described in A.4.1, Step 4, (respectively Step 4 of A.15.1, for holding a video call), but with the following exceptions and clarifications:

- the sess-version number of the SDP shall be incremented by one; and
- the direction-tag for the current-status remote segment shall be "sendrecv"; and
- the UE shall either add a session level direction attribute (and remove the direction attributes of all the media lines) or modify the direction attributes of all the media lines as follows:
  - in case of Call Hold
    - If the directionality of the media lines were originally as "recvonly" then the directionality attributes within the INVITE in step 1 shall be "inactive"
    - If the directionality of the media lines were originally as "sendrecv" then the directionality attributes within the INVITE in step 1 shall be "sendonly"
  - in case of Call Resume
    - the UE shall restore the value of the directionality attributes within the SDP body their original values (the UE may use either a single session level attribute or separate attributes for each media line).

100 Trying for INVITE (Step 2) optional step used when UE sent INVITE in step 1

Use the default message “100 Trying for INVITE” in Annex A.2.2 of TS 34.229-1 [2], applying condition A1.

200 OK for INVITE or UPDATE (Step 3)

Use the default message “200 OK for other requests than REGISTER or SUBSCRIBE” in Annex A.3.1 of TS 34.229-1 [2], applying conditions A1, A10 and A19, with the following exceptions:

Header/param	Value/remark
Require option-tag	 *precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	 length of message-body
Message-body	SDP body of the 200 OK response copied from the received INVITE or UPDATE but modified as follows: - "o=" line identical to previous SDP sent by SS except that sess-version is incremented by one - IP address on "c=" line and transport port on "m=" lines changed to indicate to which IP address and port the UE should send the media; and In case of Call Hold: - "sendonly" direction attribute inverted to "recvonly". Note that this applies to “a=sendonly” direction attributes only, not to the direction tags found in preconditions.

ACK (Step 4) conditional step used when UE sent INVITE in step 1

Use the default message “ACK” in Annex A.2.7 of 34.229-1 [2], applying conditions A1, A3 and A5.

A.18 Generic test procedure for putting a MTSI speech call to hold or to resume the call from the SS / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	<-		INVITE	SS sends INVITE with a SDP offer to hold or resume the call
2	->		100 Trying	Optional: The UE responds with a 100 Trying provisional response
3	->		200 OK	The UE responds to INVITE with 200 OK to indicate that the UE is no more sending any media (call hold) or resumes sending media (call resume)
4	<-		ACK	The SS acknowledges the receipt of 200 OK for INVITE

Specific Message Contents

INVITE (Step 1)

Use the default message “INVITE for MT call setup” in Annex A.2.9 of TS 34.229-1 [2], applying conditions A1 and A5, with the below exceptions. The SS uses the same URI in the request line as the UE has sent in the Contact header of the original INVITE request creating this dialog.

The SS shall include support for precondition in the Supported header field.

In case of Call Hold, the SS shall include the same lines in the SDP body as finally accepted for the MTSI call, i.e., the last SDP sent by the SS, with the following exceptions:

- version number of the SDP shall be incremented; and
- each media line shall carry direction attribute “a=sendonly”.

In case of Call Resume, the SS shall include the same lines in the SDP body as sent in the message for Call Hold with the following exceptions:

- version number of the SDP shall be incremented; and
- each media line shall carry direction attribute “a=sendrecv”.

100 Trying for INVITE (Step 2)

Use the default message “100 Trying for INVITE” in Annex A.2.2 of TS 34.229-1 [2], applying condition A2.

200 OK for INVITE (Step 3)

Use the default message “200 OK for other requests than REGISTER or SUBSCRIBE” in Annex A.3.1 of TS 34.229-1 [2], applying conditions A2, A5, A11, A20 and A22 with the following exceptions:

Header/param	Value/remark
Require option-tag	 *precondition*
Content-Type media-type	 *application/sdp*
Content-Length value	header shall be present if UE uses TCP to send this message and if there is a message body length of message-body
Message-body	SDP answer to the SDP offer contained in the INVITE including: - All mandatory SDP lines as specified in RFC 4566 [38]. - The same number of media lines (“m=”) as in the INVITE. - All the media lines having directionality as "recvonly" In case of Call Hold: All the media lines having direction attribute “a=recvonly”. In case of Call Resume: All the media lines having direction attribute “a=sendrecv”.

ACK (Step 4)

Use the default message “ACK” in Annex A.2.7 of TS 34.229-1 [2], applying conditions A2, A3 and A5.

A.19 MTSI conference creation / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1-7			Steps 1-7 of Annex A.4.1	The same messages as in steps 1-7 of Annex A.4.1
8	<-		200 OK	The SS responds INVITE with 200 OK and gives the final conference URI within the response
9	->		ACK	The UE acknowledges the receipt of 200 OK for INVITE
			EXCEPTION: steps 10–13 describe optional behaviour depending on UE configuration. The SS shall wait up to 3s for the SUBSCRIBE of step 10	
10	->		SUBSCRIBE	UE subscribes the conference event
11	<-		200 OK	SS responds to the subscription
12	<-		NOTIFY	SS sends the initial state of the conference event to the UE
13	->		200 OK	UE responds to the NOTIFY

Specific Message Contents

The specific message contents for steps 1–7 is otherwise identical to what have been specified in Annex A.4.1, but with the exceptions as below:

INVITE (Step 2)

Header/param	Value/remark
Request-Line	
Request-URI	*sip:mmtel@conf-factory* appended with px_IMS_HomeDomainName
To	
addr-spec	*sip:mmtel@conf-factory* appended with px_IMS_HomeDomainName

183 Session in Progress for INVITE (Step 4)

Header/param	Value/remark
Contact	
addr-spec	*sip:temporary@conf-factory*. appended with px_IMS_HomeDomainName
feature-param	*isfocus*
Record-Route	
rec-route	<*sip:orig@scscf.3gpp.org;lr*>, <sip:SS P-CSCF address: protected server port of SS;*lr*>

200 OK for INVITE (Step 8)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2], applying conditions A1 and A5 with the following exceptions:

Header/param	Value/remark
Record-Route	
rec-route	Same value as in the 183 response
Contact	
addr-spec	*sip:final@conf-factory*. appended with px_IMS_HomeDomainName
feature-param	*Isfocus*

ACK (Step 9)

Use the default message "ACK" in Annex A.2.7 of TS 34.229-1 [2], applying conditions A1 and A3, with the following exceptions:

Header/param	Value/remark
Request-Line	
Request-URI	*sip:final@conf-factory*. appended with px_IMS_HomeDomainName

SUBSCRIBE (Step 10)

Use the default message "SUBSCRIBE for conference event package" in Annex A.5.1 of TS 34.229-1 [2], applying conditions A1 and A7.

200 OK (Step 11)

Use the default message "200 OK for SUBSCRIBE" in Annex A.5.2 of TS 34.229-1 [2], applying condition A1.

NOTIFY (Step 12)

Use the default message "MT NOTIFY for conference event package" in Annex A.5.3 of TS 34.229-1 [2], applying condition A3.

200 OK (Step 13)

Use the default message "200 OK for other requests than REGISTER or SUBSCRIBE" in Annex A.3.1 of TS 34.229-1 [2], applying conditions A5 and A22.

A.20 Generic test procedure for Inviting user to conference by sending a REFER request to the conference focus / 5GS

Expected sequence

Step	Direction		Message	Comment
	UE	SS		
1	->		REFER	UE sends REFER to SS referring to the conference
2	<-		202 Accepted	The SS responds with a 202 final response
3	<-		NOTIFY	The SS sends initial NOTIFY for the implicit subscription created by the REFER request
4	->		200 OK	The UE responds the NOTIFY with 200 OK
5	<-		NOTIFY	The SS sends a NOTIFY related to REFER request to confirm that the invited user was able to join the conference
6	->		200 OK	The UE responds the NOTIFY with 200 OK
7	<-		NOTIFY	Conditional: If the UE has subscribed the conference event package, the SS sends a NOTIFY for conference event package to inform that the invited user was able to join the conference
8	->		200 OK	Conditional: The UE responds the NOTIFY with 200 OK

Specific Message Contents

REFER (Step 1)

Use the default message "MO REFER" in Annex A.2.10 of TS 34.229-1[2], applying conditions A1 and A5, with the following exceptions:

Header/param	Value/remark
Request-URI	*sip:final@conf-factory.* appended with px_IMS_HomeDomainName
Refer-To addr-spec	 SIP URI or tel URI of the user invited to the conference. If an active session exists, the Replaces header in the header portion of the SIP URI shall be included (mandatory inclusion is stated in NG.114 [31]) and set to the dialog ID of the active session according to RFC 3891 [40]. In this case, if the user has been invited with a tel URI, the UE shall convert the tel URI to a SIP URI according to RFC 3261 [6] clause 19.1.6. (NOTE: the dialog ID is percent encoded according to RFC 3986 [41]).
To addr-spec tag **Route** route-param	 remote SIP URI as used in To header in step 2 of A.16 remote tag of the dialog with the conference focus created in step 2 of A.16 URIs of the Record-Route header of 183 response sent in step 4 of A.16 in reverse order

NOTIFY (Step 3)

Use the default message "MT NOTIFY for refer package" in Annex A.2.11 of TS 34.229-1 [2], applying condition A1, with the following exceptions:

Header/param	Value/remark
Message-body	*SIP/2.0 100 Trying*

NOTIFY (Step 5)

Use the default message "MT NOTIFY for refer package" in Annex A.2.11 of TS 34.229-1 [2], applying condition A1, with the following exceptions:

Header/param	Value/remark
Subscription-State	
substate-value	*terminated*
expires	omitted from the request
reason	*noresource*
Message-body	*SIP/2.0 200 OK*

NOTIFY (Step 7)

Use the default message "NOTIFY for conference event package" in Annex A.5.3 of TS 34.229-1 [2], applying conditions A1 and A3, with the following exceptions:

Header/param	Value/remark
Message-body	<?xml version="1.0" encoding="UTF-8"?> <conference-info xmlns="urn:ietf:params:xml:ns:conference-info" entity="sip:final@conf-factory. appended with px_IMS_HomeDomainName" state="partial" version=" value as in previous notification for conference event package but incremented by one" <users> <user entity=" SIP URI or tel URI of the invited user"> <endpoint entity=" Contact URI of the invited user"> <status>connected</status> <joining-method>dialed-in</joining-method> <media id="1"> <type>audio</type> <label> unique identifier for the media stream between the focus and the endpoint of the invited user (e.g. 11223)</label> <src-id>random SSRC value</src-id> <status>sendrecv</status> </media> </endpoint> </users> </conference-info>

Annex B (informative): Change history

Change history							
Date	Meeting	TDoc	CR	Rev	Cat	Subject/Comment	New version
2019-10	RAN5#85	R5-197746	-	-	-	First draft version V0.1.0 made available	0.1.0
2019-11	RAN5#85	R5-198832	-	-	-	Second draft version V0.2.0 made available, implementing pCRs R5-197934, R5-198899, R5-198239, R5-198240, and R5-198241	0.2.0
2020-06	RAN5#87-e	R5-201458	-	-	-	Third draft version V0.3.0 made available, implementing pCRs R5-202693, R5-202686, R5-202687, R5-202688, R5-202689, R5-202678, R5-202679, R5-202680, R5-202681, R5-202682, R5-202690, R5-202683, R5-202684, R5-202685, R5-202691, R5-202692	0.3.0
2020-06	RAN5#87-e	-	-	-	-	Raised to v15.0.0	15.0.0
2020-09	RAN5#88-e	R5-203437	0004	-	F	Corrections to A.2 on IMS Registration	15.1.0
2020-09	RAN5#88-e	R5-203439	0006	-	F	New generic procedure for MT Call Release	15.1.0
2020-09	RAN5#88-e	R5-203440	0007	-	F	Adding references as needed	15.1.0
2020-09	RAN5#88-e	R5-203441	0008	-	F	Corrections to test cases 7.6 and 7.7	15.1.0
2020-09	RAN5#88-e	R5-203442	0009	-	F	Corrections to test case 6.1	15.1.0
2020-09	RAN5#88-e	R5-203443	0010	-	F	Corrections to test case 6.2	15.1.0
2020-09	RAN5#88-e	R5-203445	0012	-	F	Corrections to test case 6.4	15.1.0
2020-09	RAN5#88-e	R5-203447	0013	-	F	Corrections to test case 6.5	15.1.0
2020-09	RAN5#88-e	R5-203448	0014	-	F	Corrections to test case 6.6	15.1.0
2020-09	RAN5#88-e	R5-203452	0015	-	F	Corrections to test case 6.7	15.1.0
2020-09	RAN5#88-e	R5-203453	0016	-	F	Corrections to test case 6.8	15.1.0
2020-09	RAN5#88-e	R5-203454	0017	-	F	Corrections to test case 6.9	15.1.0
2020-09	RAN5#88-e	R5-203461	0018	-	F	Corrections to MTSI MT Voice Call TC 7.6	15.1.0
2020-09	RAN5#88-e	R5-203462	0019	-	F	Corrections to Annex A.5.1	15.1.0
2020-09	RAN5#88-e	R5-204477	0001	1	F	Addition of IMS NR TC 9.4-MT Concatenated SMS	15.1.0
2020-09	RAN5#88-e	R5-204478	0002	1	F	Addition of IMS NR TC 9.5-MO SMS RP-ERROR	15.1.0
2020-09	RAN5#88-e	R5-204479	0003	1	F	Addition of IMS NR TC 10.1-emergency call with registration and Location	15.1.0
2020-09	RAN5#88-e	R5-204480	0005	1	F	Adding details for A.3 for IMS Emergency Registration	15.1.0
2020-09	RAN5#88-e	R5-204481	0011	1	F	Corrections to test case 6.3	15.1.0
2020-09	RAN5#88-e	R5-204482	0020	1	F	Addition of new IMS 5GS test case 9.1	15.1.0
2020-09	RAN5#88-e	R5-204483	0021	1	F	Addition of new IMS 5GS test case 9.2	15.1.0
2020-09	RAN5#88-e	R5-204484	0022	1	F	New generic IMS procedures for use in EPS fallback	15.1.0
2020-09	RAN5#88-e	R5-204485	0023	1	F	Addition of NR TC 8.18 Barring of All Incoming Calls / except for a specific user / 5GS	15.1.0
2020-09	RAN5#88-e	R5-204486	0024	1	F	Addition of NR TC 9.3 Mobile Originating Concatenated SMS / 5GS	15.1.0
2020-09	RAN5#88-e	R5-204487	0026	1	F	Addition of new IMS test case 8.1	15.1.0
2020-12	RAN5#89-e	R5-205113	0029		F	Corrections to A.9 on EPS Fallback	15.2.0
2020-12	RAN5#89-e	R5-205115	0030		F	Corrections to A.2 and addition of A.10	15.2.0
2020-12	RAN5#89-e	R5-205157	0033		F	Correction to 5GS IMS test case 9.3	15.2.0
2020-12	RAN5#89-e	R5-205158	0034		F	Correction to 5GS IMS test case 9.4	15.2.0
2020-12	RAN5#89-e	R5-205159	0035		F	Correction to 5GS IMS test case 9.5	15.2.0
2020-12	RAN5#89-e	R5-205185	0037		F	New generic procedure for Re-Registration	15.2.0
2020-12	RAN5#89-e	R5-205217	0043		F	Corrections to test case 6.6	15.2.0
2020-12	RAN5#89-e	R5-205219	0044		F	Corrections to A.7	15.2.0
2020-12	RAN5#89-e	R5-205220	0045		F	New References	15.2.0
2020-12	RAN5#89-e	R5-205311	0046		F	Corrections to generic procedure A.4 on MO Voice Call	15.2.0
2020-12	RAN5#89-e	R5-205312	0047		F	Corrections to generic procedure A.5 on MT Voice Call	15.2.0
2020-12	RAN5#89-e	R5-205518	0057		F	Corrections to A.6	15.2.0
2020-12	RAN5#89-e	R5-205585	0063		F	Correction to Clause A.2	15.2.0
2020-12	RAN5#89-e	R5-206287	0031	1	F	Correction to 5GS IMS test case 9.1	15.2.0
2020-12	RAN5#89-e	R5-206372	0032	1	F	Correction to 5GS IMS test case 9.2	15.2.0
2020-12	RAN5#89-e	R5-206373	0036	1	F	New generic procedure for Mobile Initiated De-Registration	15.2.0
2020-12	RAN5#89-e	R5-206374	0038	1	F	Introduction of generic procedures for IMS MO and MT SMS	15.2.0
2020-12	RAN5#89-e	R5-206375	0039	1	F	Addition of MTSI MT Voice Call Test Case 7.8	15.2.0
2020-12	RAN5#89-e	R5-206376	0040	1	F	Addition of MTSI MT Voice Call Test Case 7.9	15.2.0
2020-12	RAN5#89-e	R5-206377	0041	1	F	Addition of MTSI MT Voice Call Test Case 7.11	15.2.0
2020-12	RAN5#89-e	R5-206378	0048	1	F	Editorial correction to add the title of section 10	15.2.0
2020-12	RAN5#89-e	R5-206379	0049	1	F	Addition of IMS NR TC 7.3-MO Voice 421 Extension Required	15.2.0
2020-12	RAN5#89-e	R5-206468	0050	2	F	Addition of IMS NR TC 7.12-MO Voice MO-MT UE with-without preconditions	15.2.0
2020-12	RAN5#89-e	R5-206381	0051	1	F	Addition of IMS NR TC 7.10-MT Voice without preconditions and SDP offer	15.2.0
2020-12	RAN5#89-e	R5-206382	0058	1	F	Update test case 7.4, 7.5, 7.6 and 7.7	15.2.0
2020-12	RAN5#89-e	R5-206383	0060	1	F	Correction to 5GS IMS TC 6.1	15.2.0
2020-12	RAN5#89-e	R5-206384	0062	1	F	Addition of New IMS over 5GS TC 7.2 MTSI MO Voice Call / 504 Server Time-out / 5GS	15.2.0
2020-12	RAN5#89-e	R5-206385	0068	1	F	Addition of new IMS over 5GS TC 7.1 MTSI MO Voice Call / 503 Service Unavailable / 5GS	15.2.0
2020-12	RAN5#89-e	R5-206386	0069	1	F	Correction to 5GS IMS TC 6.7	15.2.0

2021-01	RAN5#89-e	-	-	-	-	History correction for R5-206468. Corrected parts of implementations of R5-206287 and R5-206386	15.2.1
2021-03	RAN5#90-e	R5-210056	0070	-	F	Corrections to A.5 on MT Voice Call	15.3.0
2021-03	RAN5#90-e	R5-210057	0071	-	F	Corrections to test case 6.3	15.3.0
2021-03	RAN5#90-e	R5-210095	0073	-	F	Corrections to test case 6.4	15.3.0
2021-03	RAN5#90-e	R5-210196	0084	-	F	Corrections to SMS test case 9.5	15.3.0
2021-03	RAN5#90-e	R5-210259	0085	-	F	Corrections to A.11	15.3.0
2021-03	RAN5#90-e	R5-210343	0090	-	F	Corrections and extensions to test case 7.4	15.3.0
2021-03	RAN5#90-e	R5-210348	0091	-	F	Adding NG.114 dependencies to Annex A.4	15.3.0
2021-03	RAN5#90-e	R5-210659	0104	-	F	Withdrawing NR IMS TC 7.3-MO voice-UE preconditions enabled but not included in INVITE	15.3.0
2021-03	RAN5#90-e	R5-210882	0114	-	F	Correction to NR IMS TC 7.1-Shorter Retry-after period	15.3.0
2021-03	RAN5#90-e	R5-211334	0072	1	F	Addition of IMS over 5GS test case 7.25	15.3.0
2021-03	RAN5#90-e	R5-211421	0074	1	F	Addition of IMS over 5GS test case 7.27	15.3.0
2021-03	RAN5#90-e	R5-211422	0075	1	F	Addition of IMS over 5GS test case 7.28	15.3.0
2021-03	RAN5#90-e	R5-211423	0076	1	F	Addition of IMS over 5GS test case 7.29	15.3.0
2021-03	RAN5#90-e	R5-211424	0077	1	F	Addition of IMS over 5GS test case 7.30	15.3.0
2021-03	RAN5#90-e	R5-211425	0078	1	F	Addition of IMS over 5GS test case 7.31	15.3.0
2021-03	RAN5#90-e	R5-211426	0079	1	F	Addition of IMS over 5GS test case 7.32	15.3.0
2021-03	RAN5#90-e	R5-211427	0080	1	F	Addition of IMS over 5GS test case 7.33	15.3.0
2021-03	RAN5#90-e	R5-211428	0081	1	F	Addition of IMS over 5GS test case 7.34	15.3.0
2021-03	RAN5#90-e	R5-211429	0083	1	F	Update test case 7.4, 7.5, 7.6 and 7.7	15.3.0
2021-03	RAN5#90-e	R5-211430	0092	1	F	Editorial corrections to TS 34.229-5	15.3.0
2021-03	RAN5#90-e	R5-211431	0094	1	F	Addition of IMS over 5GS TC 7.14	15.3.0
2021-03	RAN5#90-e	R5-211432	0095	1	F	Adding references	15.3.0
2021-03	RAN5#90-e	R5-211433	0096	1	F	Addition of A.15.1 MTSI MO Video Call / with preconditions / 5GS	15.3.0
2021-03	RAN5#90-e	R5-211434	0099	1	F	Correction to IMS over 5GS TC 7.2	15.3.0
2021-03	RAN5#90-e	R5-211435	0100	1	F	Correction to IMS over 5GS TC 7.11	15.3.0
2021-03	RAN5#90-e	R5-211436	0102	1	F	Correction to NR IMS TC 7.10-Content Type not present	15.3.0
2021-03	RAN5#90-e	R5-211437	0103	1	F	Correction to NR IMS A.9.2-Optional UPDATE after EPS fallback	15.3.0
2021-03	RAN5#90-e	R5-211438	0105	1	F	Correction to NR IMS TC 10.1-Conformance requirement update	15.3.0
2021-03	RAN5#90-e	R5-211439	0106	1	F	Addition of NR IMS TC 7.26-MO CAT forking model	15.3.0
2021-03	RAN5#90-e	R5-211440	0107	1	F	Addition of NR IMS TC 8.26-MO hold without announcement	15.3.0
2021-03	RAN5#90-e	R5-211441	0108	1	F	Addition of NR IMS TC 8.28-MT hold without announcement	15.3.0
2021-03	RAN5#90-e	R5-211442	0109	1	F	Addition of NR IMS TC 8.30-Subscription to MWI event	15.3.0
2021-03	RAN5#90-e	R5-211443	0110	1	F	Addition of NR IMS TC 8.31-Creating and leaving conference	15.3.0
2021-03	RAN5#90-e	R5-211444	0111	1	F	Addition of NR IMS TC 8.32-Inviting user to conference by REFER	15.3.0
2021-03	RAN5#90-e	R5-211445	0112	1	F	Addition of NR IMS TC 8.34-Three way session	15.3.0
2021-03	RAN5#90-e	R5-211446	0113	1	F	Addition of NR IMS TC 8.36-MO explicit communication transfer	15.3.0
2021-03	RAN5#90-e	R5-211447	0115	1	F	Addition of new IMS over 5GS TC 8.38 Communication Waiting and cancelling the call / 5GS	15.3.0
2021-06	RAN5#91-e	R5-212047	0116	-	F	Corrections to test case 7.4 regarding NG.114 Profile Requirements	15.4.0
2021-06	RAN5#91-e	R5-212048	0117	-	F	Corrections to generic procedure A.4 on MO Voice Call	15.4.0
2021-06	RAN5#91-e	R5-212060	0118	-	F	New References	15.4.0
2021-06	RAN5#91-e	R5-212076	0119	-	F	Corrections to TC 7.26	15.4.0
2021-06	RAN5#91-e	R5-212104	0121	-	F	Addition of IMS over 5GS test case 7.19	15.4.0
2021-06	RAN5#91-e	R5-212127	0123	-	F	Corrections to TC 6.4	15.4.0
2021-06	RAN5#91-e	R5-212160	0125	-	F	Corrections to TC 7.25	15.4.0
2021-06	RAN5#91-e	R5-212208	0126	-	F	Corrections to TC 7.8	15.4.0
2021-06	RAN5#91-e	R5-212209	0127	-	F	Corrections to TC 7.7	15.4.0
2021-06	RAN5#91-e	R5-212387	0133	-	F	Corrections to TC 10.1	15.4.0
2021-06	RAN5#91-e	R5-212388	0134	-	F	Corrections to A.6	15.4.0
2021-06	RAN5#91-e	R5-213367	0161	-	F	Addition of MTSI MO video call without precondition	15.4.0
2021-06	RAN5#91-e	R5-213520	0120	1	F	Addition of IMS over 5GS test case 7.18	15.4.0
2021-06	RAN5#91-e	R5-213521	0122	1	F	Corrections to TC 7.1	15.4.0
2021-06	RAN5#91-e	R5-213522	0128	1	F	Corrections to TC 8.36	15.4.0
2021-06	RAN5#91-e	R5-213523	0130	1	F	Corrections regarding a=sendrecv and a=inactive	15.4.0
2021-06	RAN5#91-e	R5-213524	0131	1	F	Addition of generic procedure for MT Video Call	15.4.0
2021-06	RAN5#91-e	R5-213525	0132	1	F	Corrections to TC 7.11	15.4.0
2021-06	RAN5#91-e	R5-213526	0135	1	F	Correction to NR IMS TC 8.26-MO hold without announcement	15.4.0
2021-06	RAN5#91-e	R5-213527	0136	1	F	Correction to NR IMS TC 8.28-MT hold without announcement	15.4.0
2021-06	RAN5#91-e	R5-213528	0137	1	F	Correction to NR IMS TC 8.34-Three way session	15.4.0
2021-06	RAN5#91-e	R5-213529	0139	1	F	Addition of NR IMS Generic Procedure A.17-MTSI speech hold or resume	15.4.0
2021-06	RAN5#91-e	R5-213530	0140	1	F	Addition of NR IMS Generic Procedure A.18-MTSI speech hold or resume from SS	15.4.0
2021-06	RAN5#91-e	R5-213531	0141	1	F	Addition of NR IMS Generic Procedure A.19-MTSI conference creation	15.4.0
2021-06	RAN5#91-e	R5-213532	0142	1	F	Addition of NR IMS Generic Procedure A.20-REFER inviting user to conference	15.4.0
2021-06	RAN5#91-e	R5-213533	0162	1	F	Corrections to IMS over 5GS Test Case 7.31 and 7.32	15.4.0
2021-06	RAN5#91-e	R5-213534	0146	1	F	Corrections to IMS over 5GS Test Case 6.2	15.4.0

2021-06	RAN5#91-e	R5-213535	0148	1	F	Clarifications on usage of preconditions	15.4.0
2021-06	RAN5#91-e	R5-213536	0149	1	F	Addition of NR IMS 7.13-MT Voice with RTCP disabled	15.4.0
2021-06	RAN5#91-e	R5-213537	0150	1	F	Addition of NR IMS 7.15-MO Video without preconditions	15.4.0
2021-06	RAN5#91-e	R5-213538	0151	1	F	Addition of NR IMS 7.16-MT Video with preconditions	15.4.0
2021-06	RAN5#91-e	R5-213539	0152	1	F	Addition of NR IMS 7.17-MT Video without preconditions	15.4.0
2021-06	RAN5#91-e	R5-213540	0153	1	F	Addition of NR IMS 7.20-MO Voice-add and remove video-with preconditions	15.4.0
2021-06	RAN5#91-e	R5-213541	0154	1	F	Addition of NR IMS 7.21-MO Voice-add and remove video-without preconditions	15.4.0
2021-06	RAN5#91-e	R5-213542	0155	1	F	Addition of NR IMS 7.23-MT Voice-add and remove video-without preconditions	15.4.0
2021-06	RAN5#91-e	R5-213543	0156	1	F	Addition of NR IMS 7.24-Forking-two responses one cancel	15.4.0
2021-06	RAN5#91-e	R5-213544	0157	1	F	Addition of new IMS over 5GS TC 7.22 MTSI MT Voice Call / add video and remove video / with preconditions at both originating UE and terminating UE / 5GS	15.4.0
2021-06	RAN5#91-e	R5-213545	0160	1	F	Correction to test cases 7.33	15.4.0
2021-06	RAN5#91-e	R5-213674	0147	1	F	Corrections to IMS over 5GS Test Case 7.14	15.4.0

History

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