SIP-SDP Inter-IMS NNI Profile
Version 6.0
09 June 2019

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1 Introduction

1.1 Overview

This document describes a SIP/SDP profile for interconnection and roaming NNI between operators’ IMS networks for the purposes of exchanging traffic originating from and terminating to the respective operators’ customers. This document profiles SIP/SDP for the GSMA defined IMS based services (VoLTE Voice over LTE, Video Call, SMS over IP, and RCS Rich Communications Services services) as described in the relevant GSMA PRDs cited in clause 2.

This profile is intended to be a generic NNI profile that may be applied to any such inter-operator interconnect, including IPX and direct bilateral interconnect on regional and international basis, thereby promoting commonality and facilitating interoperability. Where options are supported based on bilateral agreement, such agreement is between the respective Operators or between Operator and IPX Provider dependent on the type of interconnect.

It is also acknowledged that some organizations may still wish to define national specific interconnect profiles. Whilst it is hoped that such national specific variants can be avoided, this document may also be used as a basis for such national specific variants and thereby minimising duplication of effort.

1.2 Relationship to Existing Standards

This document is a profile of 3GPP Release-11 TS 29.165 [1] and provides clarifications and recommendations to that technical specification to facilitate inter-IMS (IP Multimedia System) NNI (Network-Network Interface) inter-operability. The requirements are derived from GSMA PRDs for the provision of IMS-based voice, video and RCS services.

The VoLTE, Video Call, SMS over IP, VoWiFi, and RCS services are described in the following GSMA PRDs:

- IR.92 – IMS Profile for Voice and SMS [2]
- IR.94 – IMS Profile for Conversational Video Service [3]
- RCC.07 – RCS Advanced Communications Services & Client Specification [44]
- IR.51 – IMS Profile for Voice, Video, and SMS over WiFi [91]

The NNI aspects are described in the following GSMA PRDs:

- IR.65 – IMS Roaming NNI and Interworking Guidelines [4]
- IR.90 – RCS Interworking Guidelines [5]

1.3 Scope

This document specifies a SIP/SDP profile across the inter-IMS NNI in support of VoLTE, Video Call, SMS over IP, VoWiFi, and RCS services. The VoLTE and Video Call Services are based on 3GPP MMTel, specifically the sub-set of MMTel services as described in GSMA
PRDs IR.92 [2] and IR.94 [3]. In the remainder of this document, the Voice and Video Call supplementary services shall be referred to as MMTel services.

The exact set of services to be supported is determined by mutual agreement between operators. Where a reduced set of services is agreed, an appropriate subset of this profile is applicable.

There are two aspects for the NNI profiled within this document, namely the Interconnect NNI and the Roaming NNI which are defined as below:

- **Interconnect-NNI** – This term applies when the NNI is used to exchange traffic between the serving network (home or visited network) of the originating device, and the home network of the called party.

- **Roaming NNI** – This term applies when the NNI is used to exchange traffic between the home and visited networks of a roaming device, i.e. when using Local Break-Out (LBO) where the P-CSCF is in the Visited Network. It is also inclusive of a scenario whereby the IMS core network is under different administrative control to the home Operator’s access network (i.e. hosted solution).

Note: The use of the specification in support of fixed line access is not precluded.

### 1.4 Definition of Acronyms and Terms

#### 1.4.1 Acronyms

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<th>Term</th>
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<tr>
<td>3GPP</td>
<td>3rd Generation Partnership Project</td>
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<td>AMR</td>
<td>Adaptive Multi-Rate</td>
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<td>AMR-WB</td>
<td>AMR Wide Band</td>
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<td>CONF</td>
<td>Conference (MMTel Service)</td>
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<td>CPM</td>
<td>Converged IP Messaging</td>
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<tr>
<td>DTMF</td>
<td>Dual Tone Multi Frequency</td>
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<td>ECT</td>
<td>Explicit Communication Transfer (MMTel Service)</td>
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<tr>
<td>EVRC</td>
<td>Enhanced Variable Rate Codec</td>
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<tr>
<td>EVS</td>
<td>Enhanced Voice Services</td>
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<td>GSMA</td>
<td>GSM Association</td>
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<tr>
<td>IBCF</td>
<td>Interconnection Border Control Function</td>
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<td>IETF</td>
<td>Internet Engineering Task Force</td>
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<td>II-NNI</td>
<td>Inter-IMS Network to Network Interface</td>
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<td>IMDN</td>
<td>Instant Message Delivery Notification</td>
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<tr>
<td>IMS</td>
<td>IP Multimedia Subsystem</td>
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<tr>
<td>IP</td>
<td>Internet Protocol</td>
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<td>IPX</td>
<td>IP Exchange</td>
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<td>ISDN</td>
<td>Integrated Services Digital Network</td>
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<td>ITU</td>
<td>International Telecommunications Union</td>
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### Terms

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<td>ITU-T</td>
<td>Telecommunications Standardization Sector of ITU</td>
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<td>LTE</td>
<td>Long Term Evolution</td>
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<td>MaaP</td>
<td>Messaging as a Platform</td>
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<td>MCID</td>
<td>Malicious Call Identification (MMTel Service)</td>
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<td>MIME</td>
<td>Multipurpose Internet Mail Extensions</td>
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<td>MMTel</td>
<td>Multimedia Telephony</td>
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<td>MSRP</td>
<td>Message Sending Relay Protocol</td>
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<td>NNI</td>
<td>Network-Network Interface</td>
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<td>OMA</td>
<td>Open Mobile Alliance</td>
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<td>PRD</td>
<td>Permanent Reference Document</td>
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<td>RAVEL</td>
<td>Roaming Architecture for Voice over IMS with Local Breakout</td>
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<td>RCS</td>
<td>Rich Communications Services</td>
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<td>RFC</td>
<td>Request For Comments</td>
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<td>RTCP</td>
<td>Real Time Control Protocol</td>
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<td>RTP</td>
<td>Real Time Protocol</td>
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<td>SCTP</td>
<td>Stream Control Transmission Protocol</td>
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<td>SDP</td>
<td>Session Description Protocol</td>
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<td>SIP</td>
<td>Session Initiation Protocol</td>
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<td>SMS</td>
<td>Short Messaging Service</td>
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<td>SIMPLE</td>
<td>SIP for Instant Messaging and Presence Leveraging Extensions</td>
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<td>SRVCC</td>
<td>Single Radio Voice Call Continuity</td>
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<td>TCP</td>
<td>Transmission Control Protocol</td>
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<td>TrGW</td>
<td>Transition Gateway</td>
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<td>TS</td>
<td>Technical Specification</td>
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<td>UP</td>
<td>Universal Profile</td>
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<td>VBD</td>
<td>Voice Band Data</td>
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<td>VoLTE</td>
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<td>VoWiFi</td>
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<td>XDM</td>
<td>XML Document Management</td>
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<td>XML</td>
<td>Extended Mark-up Language</td>
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<td>WiFi</td>
<td>Wireless Fidelity</td>
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#### 1.4.2 Terms

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<td>Confirming SDP Offer</td>
<td>The second SDP offer of the two offer/answer exchanges that is possibly required if preconditions are used. This second SDP offer is started by the originating or terminating UE, e.g., with the SDP of the SIP Update request, is used for precondition status update (see GSMA PRD IR.92 [2]) and confirms the selected RTP protocol.</td>
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## 1.5 Document Cross-References

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<td>Inter-IMS Network-Network Interface (NNI)</td>
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<td>IMS Profile for Voice and SMS</td>
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<td>[3]</td>
<td>GSMA PRD IR.94</td>
<td>IMS Profile for Conversational Video Service</td>
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<td>[8]</td>
<td>Void</td>
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<td>[9]</td>
<td>3GPP TS 26.114</td>
<td>IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction</td>
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<td>IETF RFC 3261</td>
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<tr>
<td>[85]</td>
<td>3GPP TS 26.443</td>
<td>Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)</td>
</tr>
<tr>
<td>[86]</td>
<td>3GPP TS 26.445</td>
<td>Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description</td>
</tr>
<tr>
<td>[87]</td>
<td>3GPP TS 26.447</td>
<td>Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets</td>
</tr>
<tr>
<td>[88]</td>
<td>3GPP TS 26.449</td>
<td>Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects</td>
</tr>
<tr>
<td>[89]</td>
<td>3GPP TS 26.450</td>
<td>Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)</td>
</tr>
<tr>
<td>[90]</td>
<td>3GPP TS 26.451</td>
<td>Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)</td>
</tr>
<tr>
<td>[91]</td>
<td>GSMA PRD IR.51</td>
<td>IMS Profile for Voice, Video and SMS over Wi-Fi</td>
</tr>
<tr>
<td>[92]</td>
<td>3GPP TS 23.334</td>
<td>IP Multimedia Subsystem (IMS) Application Level Gateway (IMS-ALG) - IMS Access Gateway (IMS-AGW) interface: Procedures descriptions</td>
</tr>
</tbody>
</table>
2 Applicable Services

As stated previously, this profile supports VoLTE, Video Call, SMSoIP, VoWiFi, and RCS services. The RCS Services, as defined in GSMA PRD RCC.71 [95] include all of the following:

- Capability Exchange based on SIP OPTIONS (Note 1)
- Capability Exchange via Presence (Note 1)
- 1:1 Chat/Group Chat
- Standalone messaging
- File Transfer based on HTTP (Hypertext Transfer Protocol)
- Video Share
- Enriched Calling (Call Composer, Call Unanswered, Shared Map, Shared Sketch)
- Image Share based on GSMA PRD IR.79 [8]
- Geo-location Push
- Chatbot Sessions for MaaP
- Extension to Extension services

The relevant feature tags that are applicable to the above service set are as listed in Section 5 of GSMA PRD IR.92 [2].

Note 1: If Capability Exchange via Presence is bilaterally agreed, then (Session Initiation Protocol) SIP OPTIONS need not to be supported for Capability Exchange, and vice versa.

Note 2: RCS also cites Voice call and Video call. However, these are based on GSMA PRDs IR.92 [2], IR.51 [91] and IR.94 [3] respectively.

3 IP Interconnection

IP interconnection is described in GSMA PRD IR.65 [4] and may be accomplished via IPX (IP Exchange) or else via direct point-point connectivity. Whichever option is chosen is determined by mutual agreement of the operators and has no impact on the SIP/SDP (Session Initiation Protocol /Session Description Protocol) profile across the NNI (Network-
Network Interface). The IMS (IP Multimedia Subsystem) inter-NNI reference architecture (from 3GPP TS 29.165 [1]) is as shown in Figure 1 below. It is seen that the NNI interface consists of the Ici and Izi reference points between the peer IBCFs (Interconnection Border Control Function) and TrGWs (Transition Gateway) in the control and media planes respectively.

![Inter IMS NNI Reference Architecture](image)

Figure 1: Inter IMS NNI Reference Architecture

As stated in 3GPP TS 29.165 [4], IMS roaming may be performed across the II-NNI (Inter-IMS Network to Network Interface) subject to agreements between the operators. Such use of the NNI (the roaming-NNI) is in scope for this profile.

### 4 SIP Methods & Headers

Table 1 describes the SIP methods that are applicable in this profile. Each method is tagged as Mandatory / Optional where:

- Mandatory means that the SIP method must be supported at each end,
- Optional means that the SIP method must be supported dependent on a specific service or capability being applicable at the NNI. If no such service or capability is applicable, then the method is Not Applicable. Example services/capabilities are given. The list of example services/capabilities does not preclude others being applied via bilateral agreement.

<table>
<thead>
<tr>
<th>Method</th>
<th>Status</th>
<th>Additional Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE</td>
<td>Mandatory</td>
<td>Includes both the initial INVITE and any subsequent re-INVITE</td>
</tr>
<tr>
<td>ACK</td>
<td>Mandatory</td>
<td></td>
</tr>
<tr>
<td>BYE</td>
<td>Mandatory</td>
<td></td>
</tr>
<tr>
<td>CANCEL</td>
<td>Mandatory</td>
<td></td>
</tr>
<tr>
<td>Method</td>
<td>Status</td>
<td>Additional Information</td>
</tr>
<tr>
<td>----------</td>
<td>------------</td>
<td>---------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>OPTIONS</td>
<td>Mandatory</td>
<td>May be used for RCS Capability Exchange. May also be used as a heartbeat mechanism on the NNI.</td>
</tr>
<tr>
<td>INFO</td>
<td>Optional</td>
<td>May be used across the Roaming NNI for USSI (USSD over IMS).</td>
</tr>
<tr>
<td></td>
<td></td>
<td>May also be used for MMTel (Multimedia Telephony) features not included in GSMA PRD IR.92 [2] if</td>
</tr>
<tr>
<td></td>
<td></td>
<td>bilaterally agreed (e.g. AOC, MCID (Malicious Call Identification (MMTel Service) etc. - see section 12</td>
</tr>
<tr>
<td></td>
<td></td>
<td>of 3GPP TS 29.165 [1])</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Older implementations (pre-IETF RFC 4028 [16]) may use INFO as a session heartbeat via bilateral</td>
</tr>
<tr>
<td></td>
<td></td>
<td>agreement. In this case, a 200 OK response must be sent in reply to an INFO request that is</td>
</tr>
<tr>
<td></td>
<td></td>
<td>syntactically correct and well structured, as defined in IETF RFC 6086 [82].</td>
</tr>
<tr>
<td>MESSAGE</td>
<td>Mandatory</td>
<td>Used for SMS over IP.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Also used for RCS messaging (pager mode and IMDN, Instant Message Delivery Notification)</td>
</tr>
<tr>
<td>NOTIFY</td>
<td>Mandatory</td>
<td>Used for roaming NNI for “regevent” notification.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Used at the interconnect NNI for MMTel services (e.g. CONF status), RCS Social Presence, Group Chat and</td>
</tr>
<tr>
<td>PRACK</td>
<td>Mandatory</td>
<td>PRACK shall be generated in response to any non-100 provisional responses to an INVITE message</td>
</tr>
<tr>
<td></td>
<td></td>
<td>specifying the ‘100rel’ option tag in a Require header.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PRACK may be generated in response to any non-100 provisional responses to an INVITE message</td>
</tr>
<tr>
<td></td>
<td></td>
<td>specifying the ‘100rel’ option tag in a Supported header.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Note that this may only be specified in the provisional response if the corresponding INVITE</td>
</tr>
<tr>
<td></td>
<td></td>
<td>message indicated support of reliable provisional responses via inclusion of a ‘100rel’ option tag in</td>
</tr>
<tr>
<td></td>
<td></td>
<td>either a Supported or Require header.</td>
</tr>
<tr>
<td>PUBLISH</td>
<td>Mandatory</td>
<td>Used for RCS Social Presence and applicable at the roaming NNI only.</td>
</tr>
<tr>
<td>REFER</td>
<td>Mandatory</td>
<td>Used for MMTel services (e.g. CONF, ECT) and RCS Group Chat.</td>
</tr>
<tr>
<td>REGISTER</td>
<td>Mandatory</td>
<td>Applicable for roaming NNI only.</td>
</tr>
<tr>
<td>SUBSCRIBE</td>
<td>Mandatory</td>
<td>Applicable for roaming NNI for “regevent”.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Also used at the interconnect NNI for RCS Social Presence, MMTel services (e.g. CONF status), RCS</td>
</tr>
</tbody>
</table>
4.1 SIP Method Handling

If a SIP method is received and recognized but not supported, it shall be rejected as defined in IETF RFC 3261 [12] with a SIP 405 "Method not allowed" response which shall include an ALLOW header field containing a list of supported methods.

If a SIP method is received and is not recognized (i.e. not implemented), it shall be rejected as defined in IETF RFC 3261 [12] by a 501 "Not Implemented" response.

Note: In order to prevent a given request being repeatedly re-sent, an IBCF may change a response code prior to forwarding the message across the NNI to a 403 "Forbidden" response.

4.2 SIP Status Code Handling

SIP responses are handled according to IETF RFC 3261 [12].

As stated in IETF RFC 3261 [12], if a non-recognized final response is received in a SIP message then it shall be treated as being equivalent to the x00 response code of that class.

As stated in IETF RFC 3261 [12], if a non-recognized 18x provisional response (i.e. not referenced in the section 4.3.4.3, Table 3), is received in a SIP message, then it shall be treated as being equivalent to a 183 "Session Progress" response.

4.3 SIP Header Handling

4.3.1 SIP Requests

If a mandatory header is absent or malformed in the request, the request shall be rejected as defined in IETF RFC 3261 [12] with a SIP 400 "Bad Request" response.

If an unrecognized option tag is present in the Require header, the Request shall be rejected with a SIP 420 "Bad Extension" response. Other non-supported headers and parameters shall be ignored.

The headers or parameters that are not mentioned in the tables from Section 4.4 are considered as not applicable headers or parameters and shall be ignored as defined in IETF RFC 3261 [12].

4.3.2 SIP Responses

If a header necessary for processing the response is absent or malformed in a final 2XX response to an INVITE request, the response shall be acknowledged by sending an ACK and then the dialog shall be terminated with a SIP BYE.

If a header necessary for processing the response is absent or malformed in a provisional response, the response shall be discarded.
If a header necessary for processing the response is absent or malformed in other final responses (i.e. except a 2XX response), the response shall be treated as the 500 "Server Internal Failure" response.

If a non-supported SIP header or parameter is received in a SIP response, it shall be ignored. Headers not listed in the subsequent tables in clause 4.4 and 4.5 are considered to be non-supported, unless there is a related bilateral agreement.

### 4.4 SIP Header Support (Summary)

This clause summarises the SIP headers that are supported across the NNI across all SIP methods and responses. The supported headers are as documented in Table A.1 of 3GPP TS 29.165 [1] with additions as shown in Table 2 below. All the cited additional headers in Table 2 below are applicable only to RCS services using OMA CPM.

<table>
<thead>
<tr>
<th>Header</th>
<th>Status</th>
<th>Additional Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Contribution-ID</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
<tr>
<td>Conversation-ID</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
<tr>
<td>InReplyTo-</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
<tr>
<td>Contribution-ID</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
<tr>
<td>Message-Expires</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
<tr>
<td>Message-UID</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
<tr>
<td>Session-Replaces</td>
<td>o</td>
<td>Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]</td>
</tr>
</tbody>
</table>

Note: Status meaning is as defined in Table A.2 of 3GPP TS 29.165 [1]

#### Table 2: Supported SIP Headers (Overall)

### 4.4.1 Trust Relationships

Section 6.1.1.3.1 of 3GPP TS 29.165 [1] identifies the SIP headers that are subject to trust relationships.

The basic assumption for an MNO interconnecting to another MNO or IPX service provider using this service profile should be that there is a trust relationship. Nevertheless, some services are mainly applied within a single network, and SIP header related to such service should therefore not be passed over the II-NNI.

Table 3 below provides guidelines for trust relationships over the II-NNI to be used as complement to Table 6.2 of 3GPP TS 29.165 [1].

Note: The guidelines provided has taken an international II-NNI as the basis, when applied within a country, national regulatory requirements may also need to be considered.
<table>
<thead>
<tr>
<th>Item</th>
<th>Header Field</th>
<th>Suggested Trust Relationship</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>[1]</td>
<td>P-Asserted-Identity</td>
<td>Trusted</td>
<td>Trust is mandatory at the roaming NNI</td>
</tr>
<tr>
<td>[2]</td>
<td>P-Access-Network-Info</td>
<td>Trusted</td>
<td>Should be trusted at the roaming NNI, even if “Not Trusted” at the non-roaming NNI</td>
</tr>
<tr>
<td>[5]</td>
<td>P-Asserted-Service</td>
<td>Trusted</td>
<td>Mandatory for terminating sessions delivered over Roaming NNI, when using LBO architecture [4] for all services that have an ISCI defined.</td>
</tr>
<tr>
<td>[6]</td>
<td>P-Charging-Vector</td>
<td>Trusted</td>
<td></td>
</tr>
<tr>
<td>[7]</td>
<td>P-Charging-Function-Addresses</td>
<td>Not Trusted</td>
<td></td>
</tr>
<tr>
<td>[8]</td>
<td>P-Profile-Key</td>
<td>Not Trusted</td>
<td>Shall always be “Not Trusted” at a non-Roaming NNI</td>
</tr>
<tr>
<td>[9]</td>
<td>P-Private-Network-Indication</td>
<td>Not Trusted</td>
<td></td>
</tr>
<tr>
<td>[10]</td>
<td>P-Served-User</td>
<td>Trusted</td>
<td>Shall always be “Trusted” at the Roaming-NNI</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Not Trusted</td>
<td>Shall always be “Not Trusted” at a non-Roaming NNI</td>
</tr>
<tr>
<td>[12]</td>
<td>P-Early-Media</td>
<td>Trusted</td>
<td></td>
</tr>
<tr>
<td>[13]</td>
<td>Feature-Caps</td>
<td>Trusted</td>
<td>Trust is mandatory at the roaming NNI</td>
</tr>
</tbody>
</table>

Table 3: Guidelines for trust relationship for SIP headers at the II-NNI

4.5 SIP Header Support (Per Method / Response)

This clause provides header details on a per SIP Method / Response basis. The header details per method/response are as tables B.3.1 through B.16.2 in 3GPP TS 29.165 [1] with any modifications/clarifications as described in this section.

4.5.1 Additional Headers

There are a number of headers that are applicable only to RCS services using CPM that are defined in OMA CPM [34] and are not present in 3GPP TS 29.165 [1]. Table 4 below lists these headers and the methods/responses in which they may occur.

<table>
<thead>
<tr>
<th>Header</th>
<th>Related Methods / Responses</th>
</tr>
</thead>
<tbody>
<tr>
<td>Contribution-ID</td>
<td>INVITE request OR MESSAGE request</td>
</tr>
<tr>
<td></td>
<td>Both with status do</td>
</tr>
<tr>
<td>Conversation-ID</td>
<td>INVITE request OR MESSAGE request – both with status do</td>
</tr>
<tr>
<td>Header</td>
<td>Related Methods / Responses</td>
</tr>
<tr>
<td>--------------------</td>
<td>----------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>InReplyTo-</td>
<td>INVITE request OR MESSAGE request – both with status do</td>
</tr>
<tr>
<td>Contribution-ID</td>
<td></td>
</tr>
<tr>
<td>Message-Expires</td>
<td>INVITE request – with status do</td>
</tr>
<tr>
<td>Message-UID</td>
<td>MESSAGE request OR 200 OK (MESSAGE) response OR BYE request – all with status do.</td>
</tr>
<tr>
<td>Session-Replaces</td>
<td>INVITE request – with status do</td>
</tr>
</tbody>
</table>

**Note:** Status meaning is as defined in Table B.2.1 of 3GPP TS 29.165 [1]

<table>
<thead>
<tr>
<th>Header</th>
<th>Header Modification / Manipulation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call-ID</td>
<td>May be overwritten by the IBCF for topology hiding at the Interconnect-NNI but must not be</td>
</tr>
<tr>
<td></td>
<td>overwritten at the Roaming NNI (see section 5.10.1 of 3GPP TS 24.229 [68]).</td>
</tr>
<tr>
<td>Content-Length</td>
<td>IBCF shall recalculate this number when message bodies are altered (e.g. SDP).</td>
</tr>
<tr>
<td>From</td>
<td>Can be anonymised by the IBCF based on the Privacy header.</td>
</tr>
<tr>
<td>Max-Forwards</td>
<td>May be overwritten by the IBCF.</td>
</tr>
<tr>
<td>Record-Route</td>
<td>This header may be overwritten for topology hiding.</td>
</tr>
<tr>
<td>Route</td>
<td>IBCF may include the identity of its peer in this header. For Roaming NNI, this header may also</td>
</tr>
<tr>
<td></td>
<td>include the identity of HPLMN S-CSCF or VPLMN P-CSCF.</td>
</tr>
<tr>
<td>Via</td>
<td>May be overwritten by the IBCF for topology hiding.</td>
</tr>
</tbody>
</table>

**Table 4: Additional Headers**

Other headers that are not cited may be transited or removed at the NNI based on local operator policy and/or bi-lateral agreement.

### 4.5.2 Header Manipulation

This section provides options/mandated actions on the manipulation of certain headers by the IBCF at the NNI. Note that the listed headers are not an exhaustive list and that header manipulation by the IBCF is dependent on operator policy.

<table>
<thead>
<tr>
<th>Header</th>
<th>Header Modification / Manipulation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call-ID</td>
<td>May be overwritten by the IBCF for topology hiding at the Interconnect-NNI but must not be</td>
</tr>
<tr>
<td></td>
<td>overwritten at the Roaming NNI (see section 5.10.1 of 3GPP TS 24.229 [68]).</td>
</tr>
<tr>
<td>Content-Length</td>
<td>IBCF shall recalculate this number when message bodies are altered (e.g. SDP).</td>
</tr>
<tr>
<td>From</td>
<td>Can be anonymised by the IBCF based on the Privacy header.</td>
</tr>
<tr>
<td>Max-Forwards</td>
<td>May be overwritten by the IBCF.</td>
</tr>
<tr>
<td>Record-Route</td>
<td>This header may be overwritten for topology hiding.</td>
</tr>
<tr>
<td>Route</td>
<td>IBCF may include the identity of its peer in this header. For Roaming NNI, this header may also</td>
</tr>
<tr>
<td></td>
<td>include the identity of HPLMN S-CSCF or VPLMN P-CSCF.</td>
</tr>
<tr>
<td>Via</td>
<td>May be overwritten by the IBCF for topology hiding.</td>
</tr>
</tbody>
</table>

**Table 5: IBCF Header Manipulation**

### 5 SIP Message Transport

Both UDP (User Datagram Protocol) and TCP (Transmission Control Protocol) transport are mandatory. If UDP is the transport of choice, then TCP should be used for large messages to avoid fragmentation as described in clause 18.1.1 of IETF RFC 3261 [12]. As stated in clause 18.1.1 of IETF RFC 3261 [12], it is recommended that an existing TCP connection be reused if a request is destined to an IP address, port, and transport to which an existing connection is already open.

SCTP (Stream Control Transmission Protocol) is optional and may be used if bilaterally agreed between the operators. As stated in 3GPP TS 29.165 [1], this option is favourable if the operators would like to improve reliability over the ICI interface.
Dependent on the nature of the connection, security may be applied at the NNI based on bilateral agreement. As stated in 3GPP TS 29.165 [1], the security mechanisms are as defined in 3GPP TS 33.210 [26]. It is recommended to use Network Domain Security (NDS) for IMS Protocols as defined in Annex C of 3GPP TS 33.210 [26] which mandates the use of IPSEC ESP [27].

6 SIP Signalling Mode

En-bloc signalling shall apply to this profile. The format of the address is described in clause 7.

7 Numbering & Addressing

The routing of SIP (Session Initiated Protocol) signaling over the IMS NNI requires use of SIP URIs or tel URIs routable at the NNI per clause 6.4 of GSMA PRD IR.65 [7]. Five scenarios are possible for outgoing SIP sessions:

- An identifier used for routing at the NNI may be formatted as a SIP URI whose user part is based on an E.164 format Public User Identity, and whose domain part is routable at the NNI. The “user=phone” parameter must be appended to such a URI.
- An identifier used for routing at the NNI may be formatted as a SIP URI whose user part is alphanumeric and whose domain part is routable at the NNI.
- An identifier used for routing at the NNI may by bilateral agreement be formatted as a SIP URI whose user part is based on a local telephone number (as defined in IETF RFC 3966 [46]), and whose domain part is routable at the NNI. In this case, the local number must be qualified via the phone-context parameter as defined in IETF RFC 3966 [46] which may be specified in terms of its global number (e.g. “+44”) or via a domain name. The format of the permitted phone-context is also subject to bilateral agreement. The “user=phone” URI parameter must also be appended in this case.
- An identifier used for routing at the NNI may be formatted as a tel URI containing E.164 format Public User Identity.
- An identifier used for routing at the NNI may, if agreed bilaterally, be defined as a tel URI containing a local telephone number (as defined in IETF RFC 3966 [46]). In this case, the local number must be qualified via the phone-context parameter as defined in IETF RFC 3966 [46], which may be specified in terms of its global number (e.g., “+44”) or via a domain name. The format of the permitted phone-context is also subject to bilateral agreement.

It is recommended that Number Portability is handled as described in GSMA PRD IR.105 [96].

In all cases, if a SIP URI is entered by the user, its domain part may have to be converted by the originating network in order to be routable at the NNI.

The originating Service Provider discovers the SIP point of contact (e.g. IMS IBCF) specified by the terminating Service Provider as described in clause 4.5.2 of GSMA PRD IR.67 [23].

As specified in IETF RFC 3261, the application layer address to which a SIP message is delivered is identified in the Request-URI. To reach that address the message may traverse a sequence of SIP-aware network elements belonging to one or more network.
To constrain its path, a “stack” of URIs may be encoded in one or more Route headers and appended to the SIP message. At each network element that performs SIP routing, a SIP-aware network element toward which the message is to be forwarded is identified by the URI at the top of that stack. A network element that owns the resource identified by the topmost URI removes that URI from the stack; and removes the associated Route header if it contains no additional URIs. If a message contains no Route headers, it is forwarded based on the URI in the Request line (i.e., the Request-URI).

8 SIP Message Bodies

3GPP TS 29.165 [1] states that the MIME (Multipurpose Internet Mail Extensions) type “application/sdp” and multipart message bodies (multipart/mixed, multipart/related and multipart/alternative) shall be supported according to IETF RFC 5621 [38] over the NNI.

3GPP TS 29.165 [1] also lists a number of other MIME types may be supported over the NNI based on agreement between operators. A number of these additional MIME types are related to MMTel or RCS services.

Table 6 below lists the MIME types that are recommended to be supported in this profile, based on the related services in scope across the NNI:

<table>
<thead>
<tr>
<th>MIME Type</th>
<th>Additional Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>application/SDP</td>
<td>Mandatory. Used to carry SDP bodies to describe MMTel audio/video sessions and RCS TCP/MSRP (Message Sending Relay Protocol) sessions. The IBCF (in conjunction with information received from the TrGW) manipulates SDP message bodies.</td>
</tr>
<tr>
<td>multipart/mixed</td>
<td>Mandatory to align with 3GPP TS 29.165 [1]. Used in RCS messaging where multiple message bodies are included to send an initial message as well as negotiate a TCP/MSRP session. The IBCF manipulates the SDP to reflect the TCP/MSRP session traversing the TrGW.</td>
</tr>
<tr>
<td>multipart/related</td>
<td>Mandatory to align with 3GPP TS 29.165 [1]. Used in RCS FT to enable multiple message bodies to be included to both negotiate a TCP/MSRP session and include a thumbnail file preview (see IETF RFC 5547 [39]).</td>
</tr>
<tr>
<td>multipart/alternative</td>
<td>Despite being mandatory in 3GPP TS 29.165 [1], not specifically used for MMTel or RCS at the current time. Need not be manipulated by the IBCF. May be transited or removed by the IBCF based on operator preference.</td>
</tr>
<tr>
<td>message/external-body</td>
<td>Used in RCS messaging/FT to pass a reference to stored content, identified via a URI (see IETF RFC 4483 [40]). Conditionally supported in this profile of RCS messaging/FT is in scope across the NNI. Conveyed unchanged by the IBCF.</td>
</tr>
<tr>
<td>MIME Type</td>
<td>Additional Info</td>
</tr>
<tr>
<td>---------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>message/cpim</td>
<td>Used in RCS standalone (pager mode) messaging. Conditionally supported in this profile if RCS messaging is in scope across the NNI. Conveyed unchanged by the IBCF.</td>
</tr>
<tr>
<td>message/imdn+xml</td>
<td>Used in RCS messaging to inform the sender of message delivery/read (see IETF RFC 5438 [41]). Conditionally supported in this profile if RCS messaging is in scope across the NNI. Conveyed unchanged by the IBCF.</td>
</tr>
<tr>
<td>application/vnd.etsi.mcid+xml</td>
<td>Used in the MMTel MCID service (see 3GPP TS 24.616). This service is not mandated in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.3gpp.cw+xml</td>
<td>Used in n/w based Communication Waiting, which is not required in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.3gpp.comm-div-info+xml</td>
<td>Used in diversion notification, which is not required in GSMA PRD IR.92 [2] and this message body thus optional in this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.etsi.aoc+xml</td>
<td>Used for MMTel Advice of Charge Service, which is not required in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.etsi.cug+xml</td>
<td>Used for MMTel CUG Service, which is not required in GSMA PRD IR.92 [2] and this message body is thus optional in this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.etsi.sci+xml</td>
<td>Used for the transfer of real time charge information between the Charge Determination Point and Charge Recording Point (see 3GPP TS 29.658 [62]). Optional and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.etsi.pstn+xml</td>
<td>Used to convey ISDN information (see 3GPP TS 29.163 [45]). This is conditionally supported where IMS is used as a transit network to connect CS-networks.</td>
</tr>
<tr>
<td>message/sipfrag</td>
<td>Used to convey SIP session progress. This is conditionally supported if MMTel Ad-Hoc Conference / RCS Group Chat service is used.</td>
</tr>
<tr>
<td>application/x-session-info</td>
<td>Used to convey additional digits in a SIP INFO for overlap sending. Not applicable to this profile.</td>
</tr>
</tbody>
</table>
### Table 6: SIP Message Bodies

The IBCF is primarily a transit point and may manipulate the application/SDP message body.

<table>
<thead>
<tr>
<th>MIME Type</th>
<th>Additional Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>application/pidf+xml, application/pidf-diff+xml, application/watcherinfo+xml, application/xcap-diff+xml, application/vnd.oma.suppnnot+xml, application/simple-filter+xml</td>
<td>Conditionally supported in RCS Social Presence is applicable at the NNI.</td>
</tr>
<tr>
<td>application/resource-lists+xml, application/rimi+xml</td>
<td>Used to convey a list of target users for MMTel Ad-Hoc Conference &amp; RCS Group Chat.</td>
</tr>
<tr>
<td>application/load-control+xml</td>
<td>Used to exchange overload control information. The related internet draft is not yet agreed. Therefore, this is optional for this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/im-iscomposing+xml</td>
<td>Used to convey SIMPLE IM. Conditionally supported for RCS messaging services for interworking between SIMPLE IM and CPIM.</td>
</tr>
<tr>
<td>application/simple-message-summary+xml</td>
<td>Conditionally supported at the roaming NNI if the MMTel Message Waiting service is used. This service is included in GSMA PRD PRD IR.92 [2].</td>
</tr>
<tr>
<td>application/vnd.3gpp.sms</td>
<td>Conditionally supported at the NNI if the SMS over IP service is in scope – see 3GPP TS 24.341 [47].</td>
</tr>
<tr>
<td>application/vnd.3gpp.ussd</td>
<td>Used for MMI at the roaming NNI. Optional to this profile and may be supported if bilaterally agreed.</td>
</tr>
<tr>
<td>application/vnd.3gpp.iut+xml, application/vnd.3gpp.replication+xml</td>
<td>Used for inter-UE transfer. Optional in this profile and may be supported if bilaterally agreed. See 3GPP TS 24.337 [63].</td>
</tr>
<tr>
<td>application/vnd.3gpp.access-transfer-events+xml, application/vnd.3gpp.mid-call+xml, application/vnd.3gpp.srvcc-ext+xml, application/vnd.3gpp.srvcc-info+xml, application/vnd.3gpp.state-and-event-info+xml</td>
<td>Applicable to the roaming NNI and used for SRVCC (Single Radio Voice Call Continuity). Optional in this profile and may be supported if bilaterally agreed. See 3GPP TS 24.237 [64].</td>
</tr>
<tr>
<td>application/3gpp-ims+xml</td>
<td>Generic 3GPP XML body. This is optional in this profile and may be passed unaltered by the IBCF subject to bilateral agreement at the NNI.</td>
</tr>
<tr>
<td>application/reginfo+xml</td>
<td>Conditionally supported for the roaming NNI.</td>
</tr>
<tr>
<td>application/conference-info+xml</td>
<td>Conditionally supported if conference services are supported across the NNI (e.g. MMTel Ad-Hoc Conference, RCS Group Chat).</td>
</tr>
</tbody>
</table>
(e.g., due to its interaction with the TrGW to enable media flows to traverse the TrGW, and optionally to facilitate the media transcoding as described in 3GPP TS 23.228, Annex I, Section I.3.3 [80]).

If media transcoding is provided, then it shall be done as described in Section 10.3 of this GSMA PRD.

All other permitted message bodies are transited unchanged. The permitted MIME type of message bodies are selected based on local operator policy and/or bi-lateral agreement at the NNI dependent on the services supported at the NNI.

The IBCF may limit the size of SIP message bodies and take remedial action should that size be exceeded. The limit is agreed on a bilateral basis. The remedial action as specified in clause 5.10.6.3 of 3GPP TS 24.229 [68].

Other (unrecognised) message bodies may be removed or transited based on local operator policy and/or bi-lateral agreement at the NNI. The default action in this profile would be to remove such a message body.

9 SIP Options Tags

SIP Option tags are not removed when transited across the NNI.

Table 6.1.3.1 in 3GPP TS 29.165 [1] provides a list of capabilities supported across the NNI, a number of which are related to the use of SIP Option tags. All of the cited option tags are included in Table 6.1.3.1 of 3GPP TS 29.165 [1].

Table 7 below provides a list of the SIP Option Tags applicable to the scope/services of this profile based on GSMA PRD IR.65 [4] and GSMA PRD IR.90 [5] and carried in the SIP Supported or Require headers.

This list below in Table 7 does not preclude other tags being transmitted across the NNI nor a given tag being used in relation to other services.

<table>
<thead>
<tr>
<th>Tag</th>
<th>Additional Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>timer</td>
<td>Enables SIP session keep-alive – see IETF RFC 4028 [16].</td>
</tr>
<tr>
<td>100rel</td>
<td>Enables reliable provisional responses – see IETF RFC 3262 [14]. An example would be conveying SDP in a 18X response.</td>
</tr>
<tr>
<td>precondition</td>
<td>Enables negotiation of resource reservation for segmented QOS – see IETF RFC 3312 [15]</td>
</tr>
<tr>
<td>path</td>
<td>Used for the roaming NNI scenario as part of IMS registration – see IETF RFC 3327 [20].</td>
</tr>
<tr>
<td>replaces</td>
<td>Applicable to MMTel services CONF and ECT – see IETF RFC 3891 [19].</td>
</tr>
<tr>
<td>histinfo</td>
<td>Used in MMTel Call Forwarding – see IETF RFC 4244 [48].</td>
</tr>
<tr>
<td>multiple-refer</td>
<td>Used for the roaming NNI when referring to multiple parties to be added to an existing conference – see IETF RFC 5368 [28]</td>
</tr>
</tbody>
</table>
Table 7: SIP Option Tags

<table>
<thead>
<tr>
<th>Tag</th>
<th>Additional Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>norefersub</td>
<td>Used for the roaming NNI and enables implicit subscription to be notified of the progress of the associated REFER – see IETF RFC 4488 [21]</td>
</tr>
<tr>
<td>from-change</td>
<td>Related to the TIP service – see IETF RFC 4916 [49].</td>
</tr>
<tr>
<td>gruu</td>
<td>Used for the roaming NNI as part of IMS registration – see IETF RFC 5627 [65].</td>
</tr>
<tr>
<td>recipient-list-invite</td>
<td>Used for the roaming NNI when creating a conference via a list of URIs – see IETF RFC 5366 [60].</td>
</tr>
<tr>
<td>resource-priority</td>
<td>Used to denote priority for a SIP session – see IETF RFC 4412 [66].</td>
</tr>
</tbody>
</table>

10 Media Control

Media control shall follow the SIP SDP Offer/Answer model as documented in IETF RFC 3264 [11] to control the underlying user plane during a SIP session.

MMTel services shall negotiate voice, video or voice/video media flows. A single SIP session shall be able to negotiate a voice media flow, a video media flow or both a voice and video media flow. Video may be added to voice during a SIP session or subtracted from a multimedia (voice and video) session during a SIP session.

If the RCS Video Share service is used, video flows shall be negotiated. If any other RCS service is used (with the exception of pager-mode messaging, which has no user plane), TCP/MSRP sessions shall be negotiated to support the required media plane flows.

10.1 SIP SDP Offer / Answer

SIP SDP information shall be supported in the body of INVITE, ACK, PRACK, UPDATE, 200 OK (INVITE, PRACK, UPDATE) and 18x (INVITE) messages.

In line with GSMA PRD IR.92 [2], SIP preconditions must be supported by the UE but may be disabled by the network operator. It is out of scope of this profile whether the IBCF or some other network element is responsible for removing the related SDP media attributes to disable preconditions.

Irrespective of whether the network operator has disabled preconditions, the IBCF must support receiving SDP both with and without media attributes relating to preconditions.

If SIP preconditions are permitted, then the IBCF shall transit the related SDP media attributes to enable preconditions to be negotiated end-to-end, and thus two SDP offer/answer exchanges may be required to establish the bearer (e.g., INVITE/18x plus UPDATE/200 OK(UPDATE)).

The confirming SDP offer contains exactly the RTP payload type of the Selected Codec, as determined by the first SDP answer, e.g., in SIP 18x, see GSMA PRD IR.92 [2].

Note: Typically, also the RTP payload type of the telephone-event codec is included, but this is not used for encoding of media flows and not mentioned here any further.
Both the originating and the terminating IMS network must forward this RTP payload type of the Selected Code within the confirming SDP offer unmodified, unless transcoding or repacking is inserted.

Media flows may be subsequently modified within an existing SIP session via a new offer SDP carried within a SIP re-INVITE or SIP UPDATE message. Such a new SDP offer may include more RTP payload types for a potential modification of the media flows.

As stated in GSMA PRD IR.92 [2], SIP forking is recommended to be supported for inter-operability and forward-compatibility reasons, and the IBCF should be able to maintain at least forty (40) parallel early dialogues until receiving the final response on one of them. The IBCF/TrGW must support receiving media on one of these early dialogues as described in 3GPP TS 24.628 [67].

It is recommended that SDP offers and SDP answers are transited unchanged across the II-NNI to facilitate inter-operability and avoid transcoding. For voice and video, the content of the SDP offer and the content of the SDP answer is as described in section 3.2.1 and 2.4.3 of GSMA PRD IR.92 [2] and section 3.3.1 of GSMA PRD IR.94 [3] respectively, unless the SDP offer is modified by the originating network before reaching the II-NNI as specified below.

Some modifications of RTP (Real Time Protocol) payload types in the SDP offer by the originating network are permissible (except in a confirming SDP offer):

- The originating network can add additional RTP payload types describing codecs or codec configurations available via transcoding as less preferred alternatives (by placing those RTP payload types as last RTP payload types in the related SDP m-line), e.g. to increase the likelihood of interoperability.
- The originating network can remove some offered RTP payload types. However, the originating network must retain at least one AMR (Adaptive Multi-Rate) and one AMR-WB (AMR Wide Band) RTP payload type in the SDP offer. If the originating network and the subsequent II-NNI support super-wideband or fullband calls and such a call is offered, the originating network must also retain EVS RTP payload types as listed in section 2.4.3.3 of GSMA IR.92 [2].
- The originating network can restrict the configuration associated with an offered RTP payload type by modifying MIME parameters as allowed according to the SDP offer-answer rules specified for those MIME parameters (for AMR and AMR-WB see IETF RFC 4867 [93], for EVS see 3GPP Release 12 TS 26.445 [86], for H.264 see IETF RFC 6184 [94]). However, if a "mode-set" parameter is added for the AMR codec type, only value "mode-set=0,2,4,7" is recommended. If a "mode-set" parameter is added for the AMR-WB codec type, only value "mode-set=0,1,2" is recommended.
- Regardless, whether the MIME parameter "ch-awrecv" is present or not in an EVS RTP payload type of the received SDP offer, the originating network can set the MIME parameter "ch-awrecv" to any value allowed by TS 26.445 [86] in the forwarded SDP offer in an EVS RTP payload type.
- In the scenario of re-packing between AMR-WB and EVS IO mode at the II-NNI (see section 10.3.3.1), an EVS RTP payload type containing the MIME parameter value "evs-mode-switch=1" is here referred as an "EVO IO payload type."
• The originating network can consistently replace an AMR-WB payload type with an EVS IO payload type in all SDP answers, as long as it also replaces that EVS IO payload type with the corresponding AMR-WB payload type in subsequent SDP offers. The originating network must not make such replacement in an initial SDP offer.

Similar modifications of RTP payload types in the received SDP offer are permissible for the terminating IMS network, before the final SDP offer is sent to the terminating client.

• If present in an EVS RTP payload type of the received SDP offer, the terminating network can keep the MIME parameter "ch-aw-recv" or increase its value, but it cannot decrease its value.

• If not present in an EVS RTP payload type of the received SDP offer, the terminating network can set the MIME parameter "ch-aw-recv" to any value allowed by 3GPP TS 26.445 [86], before the final SDP offer is sent to the terminating network.

• In the scenario of re-packaging between AMR-WB and EVS IO mode at II-NNI (see section 10.3.3.1), where an EVS RTP payload type contains the MIME parameter value "evs-mode-switch=1" is here referred as an "EVS IO payload type":
  o The terminating network can, when the received initial SDP offer contains AMR-WB but no EVS payload types, add an EVS IO payload type with highest priority to the final SDP offer sent to the terminating client.
  o If such added EVS IO payload type in the final SDP offer was accepted by the terminating UE, the terminating network must consistently replace the corresponding AMR-WB payload type with the previously added EVS IO payload type and vice versa in any succeeding SDP offer/answer, as long as the received SDP contains only those corresponding speech payload types. Any subsequent received SDP offer containing more speech payload types should be handled as an initial SDP offer (see above).

10.2 RTP Profile

As stated in GSMA PRD IR.92 [2] and GSMA PRD IR.88 [7], the Real Time Protocol (RTP) profile and Audio Video Profile (RTP/AVP) (IETF RFC 3551 [32]) shall be used for voice sessions and Video Share sessions respectively.

As stated in GSMA PRD IR.94 [3], the Extended RTP Profile for Real Time Control Protocol (RTCP)-based Feedback (RTP/AVPF) (IETF RFC 4585 [33]) must be used for video telephony sessions. If the initial offer using RTP/AVPF is rejected, then a new offer shall be sent by the UE using RTP/AVP for the video telephony session.

If any RCS service is used, except for pager-mode messaging (which has no user plane), and the Video Share service, TCP/MSRP sessions shall be negotiated to support the required media plane flows as defined by IETF RFC 4975 [58] and described by GSMA PRD RCC.07 [44].

10.3 Codecs

The codecs in this profile are based on those mandated in the cited GSMA PRDs.
In general, whilst codec negotiation takes place end-end, the codecs supported at the NNI are subject to bi-lateral agreement. This profile defines a number of mandatory codecs at the NNI.

Further, this profile does not preclude other codecs being bilaterally agreed and nor should the IBCF unnecessarily remove codecs from a list received in an offer/answer as codec negotiation is an end-to-end negotiation.

Where possible, transcoding is to be avoided at the NNI. However, it is acknowledged that the IBCF/TrGW may need to perform transcoding at a given NNI where the received codec(s) are exclusive of a bi-laterally agreed codec list.

### 10.3.1 Audio Codecs

For this profile, the following audio codecs are mandatory, to be supported over the II-NNI, in line with GSMA PRD IR.92 [2]:

- Adaptive Multi-Rate (AMR) codec (described in 3GPP TS 26.114 [9], 3GPP TS 26.071 [70], 3GPP TS 26.090 [71], 3GPP TS 26.073 [72], and 3GPP TS 26.104 [73]).
- The telephone-event codec according to IETF RFC 4733 [22], further refined in 3GPP TS 26.114 [9], with all relevant RTP clock rates.

An SDP offer over the II-NNI must contain at least the following:

- An AMR RTP payload type with no mode set specified or with mode-set=0,2,4,7, and/or an AMR-WB RTP payload type with no mode set specified or with mode-set=0,1,2.
- The telephone-event/8000 and/or the telephone-event/16000 RTP payload types, chosen such that a separate, matching payload type is included for every RTP clock rate used by speech codecs that are included in the same SDP offer.
- If a super-wideband of fullband call is offered and supported over the II-NNI based on bilateral agreement, EVS RTP payload types as listed in section 2.4.3.3 of IR.92 [2].

In addition, to support interoperability with non-3GPP access inter-connect, the following audio codecs are also recommended to be supported for this profile over the II-NNI:

- G.711 (see IETF RFC 3551 [32]), using payload type 8 (A-law) and/or 0 (Mu-Law) dependent on market considerations.
- Comfort Noise codec as specified in IETF RFC 3389 [95], for use with audio codecs lacking built-in comfort noise support, such as e.g. G.711.

Other audio codecs may be supported (e.g., G.729A and G.722 (see IETF RFC 3551 [32])) based on bilateral agreement and are out of scope of this profile.
10.3.2 Video Codecs

Video codecs may be supported over the II-NNI based on agreement between operators. If video codecs are supported, the requirements in the present sub clause apply.

For this profile, the following video codecs and related profiles are mandatory to be supported over the II-NNI, in line with GSMA PRD IR.94 [3):

- ITU-T Recommendation H.264 Constrained High Profile (CHP) Level 3.1 as specified in 3GPP release 13 TS 26.114 [9] section 5.2.2,
- ITU-T Recommendation H.264 Constrained Baseline Profile (CBP) Level 1.2 as specified in 3GPP Release 10 TS 26.114 [9] section 5.2.2,

For this profile, the following video codec and related profile is recommended to be supported over the II-NNI, in line with GSMA PRD IR.94 [3]


Video in line with GSMA PRD IR.74 [7] and RCC.07 [44] may be supported over the II-NNI based on agreement between operators. If video in line with GSMA PRD IR.74 [7] and RCC.07 [44] is supported, in addition to the codecs and profiles described above, the following video codecs and related profiles are mandatory to be supported over the II-NNI, in line with GSMA PRD IR.74 [7] and RCC.07 [44]:

- ITU-T Recommendation H.264 Constrained Baseline Profile (CBP) level 1.3 is mandatory. RTP payload format for H.264 is used as specified in IETF RFC 3984 [30].
- ITU-T Recommendation H.263 [29] profile 0 level 45 is mandatory. Note that the indication of H.263 profile 0 level 45 in SDP implies support of H.263 profile 0 level 10. When using the H.263 video codec, only QCIF resolution video must be supported for Video Share. The recommended frame rate is 15fps. The ‘framerate’ and ‘framesize’ media-level SDP attributes are used as specified in IETF RFC 4566 [13], to indicate the same.

**Note:** The configuration parameters of the H.264 codec profiles used over the II-NNI (e.g. resolution, frame rate, maximum bitrate…) can be subject to bilateral agreement.

Other video codecs may be supported based on bilateral agreement and are out of scope of this profile.

10.3.3 Codec Negotiation/Handling at the NNI

The network shall have the ability to perform transcoding between the codecs it supports within each media type. The network may implement these capabilities in various network elements; however, the remainder of this section assumes they are provided by the IBCF / TrGW.

To facilitate interoperability, the IBCF shall not preclude other codecs being offered across the NNI.
10.3.3.1 Re-packing between AMR-WB and EVS IO modes

In the following:

- AMR-WB IO mode is used as a synonym for AMR-WB interoperable mode (as an alternative implementation) using IETF RFC4867 (AMR-WB RTP) [93] payload format; and
- EVS IO mode is used as a synonym for AMR-WB interoperable mode using TS 26.445, Annex A (EVS RTP) [86] payload format.

The network may enforce usage of the EVS IO mode as replacement for AMR-WB by providing re-packaging the RTP payload at II-NNI. This relies on:

- SDP modification in SIP network nodes is feasible for the case when an EVS capable network is connected to a network without EVS capability but supporting AMR-WB.
- The SDP-modifying SIP network nodes are stateful, in the way they are capable to follow a SIP dialog and compare an SDP answer with the corresponding SDP offer.

NOTE: A UE as described in [2] supporting EVS and offering super-wideband or full-band speech communications is capable of EVS IO mode [9].

There are two ways to enforce the use of EVS in the EVS-supporting call leg:

- Using EVS primary mode in the EVS call leg and performing transcoding to AMR-WB in the network. The EVS call leg then benefits from improved error robustness, both in uplink and downlink. The improved EVS error robustness especially in channel aware mode allows to relax BLER target in that call leg which corresponds to an improved SNR and indoor/outdoor cell coverage.
- Using EVS IO mode and perform re-packaging in the network. This re-packaging procedure may be executed already during SRVCC if during an EVS VoLTE to VoLTE call, one UE switches to the CS network and AMR-WB codec. This also enables the use case where EVS IO mode is directly negotiated during call setup. It is likely advantageous with regard to error robustness and audio quality to use EVS IO mode instead of legacy AMR-WB.

NOTE: In case the AMR-WB codec is negotiated, the UEs may use the AMR-WB IO mode as the alternative implementation of AMR-WB for the sending and/or receiving direction. It is in the UE vendor’s sole discretion to implement AMR-WB using AMR-WB IO mode. Hence, the operator/IMS network can only enforce the use of the EVS IO mode but not of AMR-WB IO.

Active Control and User Plane network nodes are assumed in the following, with IMS used as a synonym for all network nodes in the EPC and IMS involved in the call setup:

Originating UE <=> Originating IMS <=> Terminating IMS <=> Terminating UE

It is assumed that a UE supporting EVS can only be capable of using EVS if the network it is attached to also supports EVS. In all other cases, the UE is considered to be a non-EVS-capable UE.
The following SDP modifications in call signalling reflect the alternative connections enforcing the use of EVS IO mode:

1. An EVS-capable originating IMS (oIMS) that receives an SDP offer from the originating UE (oUE) containing one or more EVS payload types (oUE is EVS capable), and further receives the corresponding, confirming SDP answer from the terminating UE (tUE) not containing any EVS payload types (tUE is not EVS capable), replaces:

a) "AMR-WB" with "EVS IO mode" in this confirming answer and subsequent SDP answers sent towards the oUE (modification 1a), and
b) "EVS IO mode" with "AMR-WB" in subsequent SDP offers sent towards the tUE (modification 1b).

2. An EVS-capable terminating IMS (tIMS) that receives an initial SDP offer from the oIMS not containing any EVS payload types (oIMS is not EVS capable):

a) Adds an "EVS IO mode" payload type with highest priority to the final SDP offer forwarded to the tUE (modification 2a).
b) If tIMS receives a corresponding, confirming SDP answer from the tUE containing "EVS IO mode" (tUE is EVS capable), tIMS replaces:

i. "EVS IO mode" with "AMR-WB" in this confirming SDP answer and subsequent SDP answers sent towards oIMS (modification 2b), and
ii. "AMR-WB" with "EVS IO mode" in any subsequent SDP offers sent towards tUE that contain only those speech payload types, e.g. in SIP UPDATE (modification 2c). Subsequent SDP offers containing more speech payload types should be handled as initial SDP offers (see above).

All modifications are assumed to be implemented in the IMS network instances already managing SDP messages, e.g. they may be implemented in P-CSCF or other IMS network instances such as the ATCF (Access Transfer Control Function) already present for SRVCC. The re-packaging may be implemented e.g. in Media Resource Function Processor (MRFP) or Access Transfer Gateways (ATGW).

SDP examples are provided in Annex B.5.

10.3.4 Global Text Telephony (GTT)


10.3.5 DTMF

DTMF (Dual Tone Multi Frequency) events shall be conveyed across the II-NNI via the "Named Telephone Event" payload format defined in IETF RFC 4733 [22], as specified in Annex G of 3GPP TS 26.114 [9], using the relevant telephone-event codec.

10.4 Early Media Detection

Early media shall be supported in this profile.

As stated in GSMA PRD IR.92 [2], early media may be indicated by the presence of a P-Early-Media header in a re-INVITE/UPDATE/PRACK request or a 18x (INVITE) response. A P-Early-Media shall be transited across the NNI and all values of the P-Early-Media header shall be supported at the NNI. In addition, the IBCF shall configure its TrGW in a manner consistent with the P-Early-Media header to ensure that the related media traverses the TrGW.

10.5 SDP Contents

SDP is defined in IETF RFC 4566 [13].

SDP usage shall be compliant with the Offer/Answer rules in IETF RFC 4566 [13].

NAPT may be performed at the NNI in the user plane via the TrGW under the control of the IBCF via the Ix reference point as defined in 3GPP TS 29.238 [17]. If NA(P)T is applied at the NNI, then the IP address and port in the c=line and m=line respectively must be modified.

For handling MSRP media, Border Elements shall follow GSMA PRD IR.90 [5]. In the case of SIMPLE IM, if NA(P)T is performed, then the IP address and port must also be modified in the a-path attribute, whilst leaving the session identity unchanged. For a summary of the different procedures for MSRP handling, see also section 5.19 of 3GPP TS 23.334 [92].

The SDP contents applicable to the MMTel and RCS services in this document are summarized in Table 8 below.

<table>
<thead>
<tr>
<th>SDP Attribute</th>
<th>Profile Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Version (v-line)</td>
<td>The value must always be equal to zero: v=0.</td>
</tr>
</tbody>
</table>
| Origin (o-line) | The origin line consists of six fields: 
- `<username>`, `<sess-id>`, `<sess-version>`, `<nettype>`, `<addrtype>` and `<unicast-address>`.
- `<username>` should contain anhyphen
- `<session ID>` and `<version>` should contain one or more digits as described in IETF RFC 4566 [8]
- `<network type>` shall be set to IN
- `<address type>` shall be set to IP4 or IP6
The Address Type shall be set to “IP4” or “IP6” depending on the addressing scheme.
- `<address>` should contain the fully qualified domain name or IP address of the media end point (typically the TrGW).
Example: 
"o= 0 0 IN IP4 10.1.2.3"
| Session (s-line) | The session name "s=" line contains a single field
`s= <session name>
Example: 
"s=" |
| Time | The time "t=" line consists of two fields |
### SDP Attribute | Profile Settings
---|---
(t-line) | `t=<start time> and <stop time>`.  
Example:  
```
t=0 0
```

Connection (c-line) | The connection "c=" line consists of three fields,  
`c=<nettype> <addr type> <connection-address>`  
The `<nettype>` shall be "IN"  
The `<addr type>` shall be "IP4" or "IP6"  
The `<connection-address>` shall be an IPv4 or IPv6 address as signified by the `<addr type>`.
Example:  
```
c=IN IP4 10.10.1.1
```

Media (m-line) | The media "m=" line consists of four fields `<media>`, `<port>`, `<proto>` and `<fmt>`  
The `<media>` field shall be set to "audio" or "video" or "message" or "text" or "image".  
The `<port>` is set to the port number that shall send/receive the media. The port number must be even for audio/video/text/image.  
The `<proto>` is set to "RTP/AVPF" for video and to "RTP/AVP" for audio/video/text media and to "TCP/MSRP" for message media.  
The `<fmt>` is set to one or more RTP payload type numbers for RTP/AVP and RTP/AVPF and to "*" for TCP/MSRP.
Examples:  
```
m=audio 1234 RTP/AVP 100 8 0
```
```
m=video 1234 RTP/AVPF 100 102
```
```
m=message 1239 TCP/MSRP *
```
```
m=text 1234 RTP/AVP 99
```

Media Attributes (a-lines) | The attribute "a=" line consists of one or two fields `<attribute>` or `<attribute>:<value>`. Many attribute lines are defined, of which most are related to a specific payload type.
One or more media attribute lines may be included dependent on the payload type. In general, attribute lines should be transited at the NNI to facilitate media interworking. Some common media attribute lines are shown below:

(For stream activity)  
a=inactive/recvonly/sendonly/sendrecv  
Example: "a=inactive"

(For identifying RTP payload type numbers and mapping them to codec types)  
a= rtpmap: `<payload type number> <encoding name>/<clock rate> [</encoding parameters>]`  
Example: "a=rtpmap:100 AMR-WB/8000"

(For providing payload type specific parameters)  
a= fmtp: `<format> <format specific parameters>`  
Example: "a=fmtp:100 mode-change-capability=2"  
NOTE: When m= line profile is (S)RTP/AVP(F), "format" is always an RTP payload type number

(For defining packetization time in ms)
### SDP Attribute

<table>
<thead>
<tr>
<th>SDP Attribute</th>
<th>Profile Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>a=ptime:&lt;time&gt;</td>
<td>Example: a=ptime:20</td>
</tr>
</tbody>
</table>
  *(for segmented QoS indication as per IETF RFC 3312 [15])* |
| a=curr:<precondition type> <status-type> <direction> |
| a=des:<precondition type> <strength> <status-type> <direction> |
| a=conf:<precondition type> <status-type> <direction> |
  *Example:*
| a=curr:qos local none |
| a=des:qos mandatory local sendrecv |
| a=curr:qos remote none |
| a=des:qos optional remote sendrecv |
| a=conf:qos remote sendrecv |
  *(for OMR as per 3GPP TS 29.079 [37])* |
| a=visited-realm:<instance> <realm> <addrtype> <addr> <port> |
| a=omr-m-cksum:<hexNumber> |
| a=omr-s-cksum:<hexNumber> |
| a=omr-codecs:<instance> <proto> 1*:<codec> |
| a=omr-m-att:<instance>:1*:<attribute> |
| a=omr-s-att:<instance>:1*:<attribute> |
| a=omr-m-bw:<instance>:<bandwidth> |
| a=omr-s-bw:<instance>:<bandwidth> |
  *Example:*
| a=visited-realm:1 Xa.operatorx.net IN IP4 192.0.2.1 49170 |
| a=visited-realm:2 XY.operator.netx IN IP4 13.24.1.1 66000 |
| a=omr-m-cksum:89 |
| a=omr-s-cksum:0 |
  *(for RTP Profile Negotiation as per IETF RFC 5939 [35])* |
| a=ticap:<capability number> <transport protocol list> |
| a=pcfg:<config number> <potential config list> |
| a=acfg:<config number> <selected config list> |
  *Example:*
| a=ticap:1 RTP/AVPF |
| a=pcfg:1 t=1 |
  *(for RTCP Feedback as per IETF RFC 4585 [33] and IETF RFC 5104 [95])* |
| a=rtp-fb:<payload type> <feedback type> <feedback parameters> |
  *Example:*
| a=rtp-fb:* nack rpsi |
| a=rtp-fb:* ccm fir |
| a=rtp-fb:* ccm tmbr |
  *(for RTP Header Extensions as per IETF RFC 5285 [96])* |
| a=extmap:<ext id> <extension attributes> |
  *Example:*
| a=extmap:7 urn:3gpp:video-orientation |
  *(for TCP Connection establishment for MSRP as per IETF RFC 4975 [83])* |
| a=path:msrp://10.10.1.1:1239/jshA7weztas:tcp |

**Bandwidth (b-line)**

The bandwidth "b=" line consists two fields `<bwtype>:<bandwidth>`.

For voice and video call sessions, as mandated in GSMA PRDs IR.92 [2] and IR.94 [3] the `<bwtype>` is set to AS for the RTP media part, and to RS / RR for RTCP.
For GTT sessions, as mandated in GSMA PRD IR.92 [2], the <bwtype> is set to AS, with RS and RR set to zero.

For video share and RCS sessions, the <bwtype> is set to AS.

The <bandwidth> defines the peak bandwidth in units of kbits/sec (for AS) or bits/sec (for RS/RR) and includes up to and including the IP layer as defined in IETF RFC 4566 [13] and IETF RFC 3566 [18].

Examples:
“b=RR:100”, “b=AS:100”

### Table 8: SDP Contents

#### 10.6 RTP/RTCP Packet Source

In a SIP session, the same IP address and port number shall be used to send and receive RTP packets (symmetric IP address and port number). Further, the port number for sending/receiving RTCP packets shall be equal to "the port number negotiated for RTP" + 1.

SDP Bandwidth Modifiers for RTCP (see IETF RFC 3556 [18]) are supported on the NNI.

#### 11 IP Version

As stated in 3GPP TS 29.165 [1], the supported IP version at the NNI may be IPv4 only, IPv6 only or both IPv4 and IPv6.

In case IPv4 only and IPv6 only networks are interconnected, the involved IBCF and TrGWs shall apply the IP version interworking procedures as indicated in 3GPP TS 29.162 [69].

#### 12 Inter-Operator Accounting

Inter-operator accounting shall be performed as described in section 11.2 of 3GPP TS 29.165 [1].
## Annex A  SIP Header Examples (Informational)

Table 9 below provides examples and references to the relevant RFCs etc. for the SIP headers in scope in this profile.

<table>
<thead>
<tr>
<th>Header</th>
<th>Reference(s)</th>
<th>Example Header</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accept</td>
<td>IETF RFC 3261</td>
<td>Accept:application/sdp.message/cpim</td>
</tr>
<tr>
<td>Accept-Contact</td>
<td>IETF RFC 3261</td>
<td>Accept-Contact:*;mobility=&quot;mobile&quot;;methods=&quot;INVITE&quot;</td>
</tr>
<tr>
<td>Accept-Encoding</td>
<td>IETF RFC 3261</td>
<td>Accept-Encoding:gzip</td>
</tr>
<tr>
<td>Accept-Language</td>
<td>IETF RFC 3261</td>
<td>Accept-Language: da, en-gb;q=0.8, en,q=0.7</td>
</tr>
<tr>
<td>Accept-Resource-Priority</td>
<td>IETF RFC 4412</td>
<td>Accept-Resource-Priority:ets.0</td>
</tr>
<tr>
<td>Alert-Info</td>
<td>IETF RFC 3261</td>
<td>Alert-Info:<a href="">urn:alert:service:call-waiting</a></td>
</tr>
<tr>
<td>Allow</td>
<td>IETF RFC 3261</td>
<td>Allow:INVITE, ACK, UPDATE, PRACK, CANCEL, PUBLISH, MESSAGE, OPTIONS, SUBSCRIBE, NOTIFY</td>
</tr>
<tr>
<td>Allow-Events</td>
<td>IETF RFC 6665</td>
<td>Allow-Events: conference</td>
</tr>
<tr>
<td>Answer-Mode</td>
<td>IETF RFC 5373</td>
<td>Answer-Mode: Auto</td>
</tr>
<tr>
<td>Authentication-Info</td>
<td>IETF RFC 3261 IETF RFC 3310 3GPP TS 24.229</td>
<td>Authentication-Info: nextnonce=&quot;47364c23432d2e131a5fb210812c&quot;</td>
</tr>
<tr>
<td>Authorization</td>
<td>IETF RFC 3261 IETF RFC 3310 3GPP TS 24.229</td>
<td>Authorization: Digest, username=&quot;<a href="mailto:jon.dough@mobile.biz">jon.dough@mobile.biz</a>&quot;, realm=&quot;<a href="mailto:RoamingUsers@mobile.biz">RoamingUsers@mobile.biz</a>&quot;, nonce=&quot;CjPk9mRQnTu25eRkajM09uT19nM09uT19nMz5OX25PZ2=&quot;&quot;, url=&quot;sip:home.mobile.biz&quot;,&quot;qop=auth-int, nc=00000001, cnonce=&quot;0a4f113b&quot;, response=&quot;6629fae49393a0539745978507c4ef1&quot;, opaque=&quot;5ccc069c403ebaf9f0171e951740e41&quot;</td>
</tr>
<tr>
<td>Call-Id</td>
<td>IETF RFC 3261</td>
<td>Call-Id:<a href="mailto:12345@mydomain.com">12345@mydomain.com</a></td>
</tr>
<tr>
<td>Call-Info</td>
<td>IETF RFC 3261</td>
<td>Call-Info:<a href="http://www.example.com/alice/photo.jpg">http://www.example.com/alice/photo.jpg</a>, purpose=icon</td>
</tr>
<tr>
<td>Contact</td>
<td>IETF RFC 3261</td>
<td>Contact:<a href="">sip:ibcf4@operator1.com:5060</a></td>
</tr>
<tr>
<td>Content-Disposition</td>
<td>IETF RFC 3261</td>
<td>Content-Disposition:session</td>
</tr>
<tr>
<td>Content-Encoding</td>
<td>IETF RFC 3261</td>
<td>Content-Encoding:gzip</td>
</tr>
<tr>
<td>Content-Language</td>
<td>IETF RFC 3261</td>
<td>Content-Language:fr</td>
</tr>
<tr>
<td>Content-Length</td>
<td>IETF RFC 3261</td>
<td>Content-Length:146</td>
</tr>
<tr>
<td>Content-Type</td>
<td>IETF RFC 3261</td>
<td>Content-Type:application/sdp</td>
</tr>
<tr>
<td>Contribution-ID</td>
<td>OMA CPM</td>
<td>Contribution-ID: abedef-1234-5678-90ab-cdef01234567</td>
</tr>
<tr>
<td>Conversation-ID</td>
<td>OMA CPM</td>
<td>Conversation-ID: 1f1d4f9e-7dec-11d0-a765-00a0c91e6bf6</td>
</tr>
<tr>
<td>Header</td>
<td>Reference(s)</td>
<td>Example Header</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------------------</td>
<td>-----------------------------------------------------------------------</td>
</tr>
<tr>
<td>CSeq</td>
<td>IETF RFC 3261</td>
<td>CSeq:2 CANCEL</td>
</tr>
<tr>
<td>Date</td>
<td>IETF RFC 3261</td>
<td>Date Sat, 13 Nov 2010 23:29:00 GMT</td>
</tr>
<tr>
<td>Error-Info</td>
<td>IETF RFC 3261</td>
<td>Error-Info: <a href="">sip:announcement10@example.com</a></td>
</tr>
<tr>
<td>Event</td>
<td>IETF RFC 6685</td>
<td>Event: reg</td>
</tr>
<tr>
<td>Expires</td>
<td>IETF RFC 3261</td>
<td>Expires: 3600</td>
</tr>
<tr>
<td>Feature-Caps</td>
<td>IETF RFC 6809</td>
<td>Feature-Caps+:3gpp.trf=sip:trf3.operator3.com</td>
</tr>
<tr>
<td>Flow-Timer</td>
<td>IETF RFC 5626</td>
<td>Flow-Timer: 30</td>
</tr>
<tr>
<td>From</td>
<td>IETF RFC 3261</td>
<td>From: <a href="">sip:+12125551212@operator1.com;user=phone</a>; tag=145688</td>
</tr>
<tr>
<td>Geolocation</td>
<td>IETF RFC 6442</td>
<td>Geolocation:<a href="mailto:user1@operator1.com">user1@operator1.com</a></td>
</tr>
<tr>
<td>Geolocation-Error</td>
<td>IETF RFC 6442</td>
<td>Geolocation-Error:100;code=&quot;Cannot Process Location&quot;</td>
</tr>
<tr>
<td>Geolocation-Routing</td>
<td>IETF RFC 6442</td>
<td>Geolocation-Routing:yes</td>
</tr>
</tbody>
</table>
| History-Info      | IETF RFC 4244       | History-Info: <sip:Bob@P1.example.com>; index=1, <sip:Bob@P2.example.com>; index=1.1, <sip:User3@UA3.example.com>?
<p>|                  |                     | Reason=SIP%3Bcause%3D486%3Btext%3D&quot;Busy Here&quot;; indexes=1.2, <a href="">sip:User5@UA5.example.com</a>; index=1.3 |
| Info-Package      | IETF RFC 6086       | Info-Package:foo                                                     |
| InReplyTo-ID      | OMA CPM             | InReplyTo- Contribution-ID: 01234567-89ab-cdef-0123-456789abedef    |
| Join              | IETF RFC 3911       | Join: <a href="mailto:1234@example.com">1234@example.com</a>; to-tag=3456; from-tag=6789                   |
| Max-Breadth       | IETF RFC 5393       | Max-Breadth: 20                                                      |
| Max-Forwards      | IETF RFC 3261       | Max-Forwards: 70                                                     |
| Message-Expires   | OMA CPM             | Message-Expires: 259200                                               |
| Message-UID       | OMA CPM             | Message-UID: 4392                                                    |
| MIME-Version      | IETF RFC 3261       | MIME-Version: 1.0                                                    |
| Min-Expires       | IETF RFC 3261       | Min-Expires: 40                                                      |
| Min-SE            | IETF RFC 4028       | Min-SE: 60                                                           |
| Organization      | IETF RFC 3261       | Organization: My Company                                             |
| P-Access-Network-Info | IETF RFC 7315     | P-Access-Network-Info: 3GPP-E-UTRAN-FDD; e-utran-cell-id: 3gpp=1234 |
| P-Answer-State    | IETF RFC 4964       | P-Answer-State: Unconfirmed                                          |
| P-Asserted-Identity| IETF RFC 3325       | P-Asserted-Identity: sip:<a href="mailto:+14085264000@operator1.com">+14085264000@operator1.com</a>; user=phone     |
| P-Asserted-Service | IETF RFC 6050       | P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel             |</p>
<table>
<thead>
<tr>
<th>Header</th>
<th>Reference(s)</th>
<th>Example Header</th>
</tr>
</thead>
<tbody>
<tr>
<td>Service</td>
<td>IETF RFC 7315</td>
<td>P-Associated-URI: sip:<a href="mailto:user2@operator3.com">user2@operator3.com</a></td>
</tr>
<tr>
<td>P-Associated-URI</td>
<td>IETF RFC 7315</td>
<td>P-Associated-URI: sip:<a href="mailto:user2@operator3.com">user2@operator3.com</a></td>
</tr>
<tr>
<td>P-Called-Party-ID</td>
<td>IETF RFC 7315</td>
<td>P-Called-Party-ID: sip:<a href="mailto:user1-business@example.com">user1-business@example.com</a></td>
</tr>
<tr>
<td>P-Charging-Vector</td>
<td>IETF RFC 7315</td>
<td>P-Charging-Vector: icid-value=1234bc9876e; icid-generated-at=192.0.6.8; orig-iod=home1.net</td>
</tr>
<tr>
<td>P-Early-Media</td>
<td>IETF RFC 5009</td>
<td>P-Early-Media: supported</td>
</tr>
<tr>
<td>P-Preferred-Service</td>
<td>IETF RFC 6050</td>
<td>P-Preferred-Service: urn:urn-7.3gpp-service.ims.icsi.mmtel</td>
</tr>
<tr>
<td>P-Private-Network-Indication</td>
<td>IETF RFC 7316</td>
<td>P-Private-Network-Indication: example.com</td>
</tr>
<tr>
<td>P-Profile-Key</td>
<td>IETF RFC 5002</td>
<td>P-Profile-Key: sip:<a href="mailto:user3@operator4.com">user3@operator4.com</a></td>
</tr>
<tr>
<td>P-Refused-URI-List</td>
<td>IETF RFC 5318</td>
<td>P-Refused-URI-List: sip:<a href="mailto:friends-list@example.net">friends-list@example.net</a>; members=<a href="">cid:an3bi8f03@example.net</a></td>
</tr>
<tr>
<td>P-Served-User</td>
<td>IETF RFC 5502</td>
<td>P-Served-User: sip:<a href="mailto:user1@operator2.com">user1@operator2.com</a>; sescase=orig</td>
</tr>
<tr>
<td>P-Visited-Network-ID</td>
<td>IETF RFC 7315</td>
<td>P-Visited-Network-ID: &quot;Visited network number 1&quot;</td>
</tr>
<tr>
<td>Path</td>
<td>IETF RFC 3327</td>
<td>Path:<a href="">sip:P2.example.com;lr</a>,<a href="">sip:P1.example.com;lr</a></td>
</tr>
<tr>
<td>Permission-Missing</td>
<td>IETF RFC 5360</td>
<td>Permission-Missing: <a href="mailto:userC@example.com">userC@example.com</a></td>
</tr>
<tr>
<td>Policy-Contact</td>
<td>IETF RFC 6794</td>
<td>Policy-Contact: sip:<a href="mailto:server5@example.com">server5@example.com</a></td>
</tr>
<tr>
<td>Priority</td>
<td>IETF RFC 3261</td>
<td>Priority: emergency</td>
</tr>
<tr>
<td>Priv-Answer-Mode</td>
<td>IETF RFC 5373</td>
<td>Priv-Answer-Mode: Auto</td>
</tr>
<tr>
<td>Privacy</td>
<td>IETF RFC 3323</td>
<td>Privacy:user</td>
</tr>
<tr>
<td></td>
<td>IETF RFC 3325</td>
<td>Privacy:id</td>
</tr>
<tr>
<td>Proxy-Authenticate</td>
<td>IETF RFC 3261</td>
<td>Proxy-Authenticate: Digest realm=&quot;atlanta.com&quot;, domain=&quot;sip:ss1.carrier.com&quot;, qop=&quot;auth&quot;, nonce=&quot;f841ce4c41e66cbe5ae9c8e88d359&quot;, opaque=&quot;&quot;, stale=FALSE, algorithm=MD5</td>
</tr>
<tr>
<td>Proxy-Authorization</td>
<td>IETF RFC 3261</td>
<td>Proxy-Authorization: Digest username=&quot;Alice&quot;, realm=&quot;atlanta.com&quot;, nonce=&quot;c60f3082ee1212b402a21831ae&quot;, response=&quot;245f23415f11432b3434341c022&quot;</td>
</tr>
<tr>
<td>Proxy-Require</td>
<td>IETF RFC 3261</td>
<td>Proxy-Require: foo</td>
</tr>
<tr>
<td>Rack</td>
<td>IETF RFC 3262</td>
<td>Rack:10</td>
</tr>
<tr>
<td>Reason</td>
<td>IETF RFC 3326</td>
<td>Reason: Q.850; cause=16; text=&quot;Terminated&quot;</td>
</tr>
<tr>
<td>Record-Route</td>
<td>IETF RFC 3261</td>
<td>Record-Route: <a href="">sip:server10.biloxi.com;lr</a>, <a href="">sip:bigbox3.site3.atlanta.com;lr</a></td>
</tr>
<tr>
<td>Recv-Info</td>
<td>IETF RFC 6086</td>
<td>Recv-Info: bar</td>
</tr>
<tr>
<td>Header</td>
<td>Reference(s)</td>
<td>Example Header</td>
</tr>
<tr>
<td>------------------------</td>
<td>------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Referred-By</td>
<td>IETF RFC 3892</td>
<td>Referred-By: <a href="mailto:sip.alice@phone1.example.org">sip.alice@phone1.example.org</a></td>
</tr>
<tr>
<td>Refer-Sub</td>
<td>IETF RFC 4488</td>
<td>Refer-Sub=false</td>
</tr>
<tr>
<td>Refer-To</td>
<td>IETF RFC 3515</td>
<td>Refer-To: <a href="">sip:dave@bobster.example.org</a>? Replaces=<a href="mailto:425928@bobster.example.org">425928@bobster.example.org</a>;to-tag=7743;from-tag=6472</td>
</tr>
<tr>
<td>Reject-Contact</td>
<td>IETF RFC 3841</td>
<td>Reject-Contact: &quot;actor=&quot;msg-taker&quot;,&quot;video</td>
</tr>
<tr>
<td>Replaces</td>
<td>IETF RFC 3891</td>
<td>Replaces: <a href="mailto:425928@bobster.example.org">425928@bobster.example.org</a>;to-tag=7743;from-tag=6472</td>
</tr>
<tr>
<td>Reply-To</td>
<td>IETF RFC 3261</td>
<td>Reply-To: Bob <a href="">sip:user4@example.com</a></td>
</tr>
<tr>
<td>Request-Disposition</td>
<td>IETF RFC 3841</td>
<td>Request-Disposition: proxy</td>
</tr>
<tr>
<td>Require</td>
<td>IETF RFC 3261</td>
<td>Require:Path</td>
</tr>
<tr>
<td>Resource-Priority</td>
<td>IETF RFC 4412</td>
<td>Resource-Priority:ets.1</td>
</tr>
<tr>
<td>Retry-After</td>
<td>IETF RFC 3261</td>
<td>Retry-After: 3600</td>
</tr>
<tr>
<td>Route</td>
<td>IETF RFC 3261</td>
<td>Route: <a href="">sip:bigbox3.site3.atlanta.com;lr</a>, <a href="">sip:server10.biloxi.com;lr</a></td>
</tr>
<tr>
<td>RSeq</td>
<td>IETF RFC 3262</td>
<td>RSeq:10</td>
</tr>
<tr>
<td>Security-Client</td>
<td>IETF RFC 3329</td>
<td>Security-Client: ipsec-ike</td>
</tr>
<tr>
<td>Security-Verify</td>
<td>IETF RFC 3329</td>
<td>Security-Verify:ipsec-ike; q=0.1</td>
</tr>
<tr>
<td>Server</td>
<td>IETF RFC 3261 &amp;</td>
<td>Server: PRD-IR92/11 term-Vendor1/Model1-XXXX device-type/feature-phone mno-custom/none</td>
</tr>
<tr>
<td>GSMA PRD IR.92</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Service-Route</td>
<td>IETF RFC 3608</td>
<td>Service-Route: <a href="">sip:P1.example.com;lr</a>, <a href="">sip:P2.example.com;lr</a></td>
</tr>
<tr>
<td>Session-Expires</td>
<td>IETF RFC 4028</td>
<td>Session-Expires:3600;refresher=uac</td>
</tr>
<tr>
<td>Session-Replaces</td>
<td>OMA CPM</td>
<td>Session-Replaces: abcdedf-1234-5678-90ab-cdef01234567</td>
</tr>
<tr>
<td>Session-ID</td>
<td>IETF RFC 7329</td>
<td>Session-ID: 0123456789abcdedf123456789abcdedf0</td>
</tr>
<tr>
<td>SIP-Etag</td>
<td>IETF RFC 3903</td>
<td>SIP-Etag:12345</td>
</tr>
<tr>
<td>SIP-If-Match</td>
<td>IETF RFC 3903</td>
<td>SIP-If-Match:12345</td>
</tr>
<tr>
<td>Subscription-State</td>
<td>IETF RFC 6665</td>
<td>Subscription-State:active</td>
</tr>
<tr>
<td>Supported</td>
<td>IETF RFC 3261</td>
<td>Supported:100Rel,timer,precondition</td>
</tr>
<tr>
<td>Suppress-If-Match</td>
<td>IETF RFC 5839</td>
<td>Suppress-If-Match:34567</td>
</tr>
<tr>
<td>Target-Dialog</td>
<td>IETF RFC 4538</td>
<td>Target-Dialog:</td>
</tr>
<tr>
<td>Timestamp</td>
<td>IETF RFC 3261</td>
<td>Timestamp:64</td>
</tr>
<tr>
<td>To</td>
<td>IETF RFC 3261</td>
<td>To: <a href="">sip:+12126661212@operator2.com;user=phone</a>;tag=1234567</td>
</tr>
</tbody>
</table>
| Trigger                | IETF RFC 5360    | Trigger-Consent: sip:user1@example.com; target-
<table>
<thead>
<tr>
<th>Header</th>
<th>Reference(s)</th>
<th>Example Header</th>
</tr>
</thead>
<tbody>
<tr>
<td>Consent</td>
<td>IETF RFC 3261</td>
<td><code>un=&quot;sip:user2@example.xcom&quot;</code></td>
</tr>
<tr>
<td>Unsupported</td>
<td>IETF RFC 3261</td>
<td><code>Unsupported:100Rel</code></td>
</tr>
<tr>
<td>User-Agent</td>
<td>IETF RFC 3261 &amp; GSMA PRD IR.92</td>
<td><code>User-Agent: PRD-IR92/11 term-Vendor1/Model1-XXXX device-type/feature-phone mno-custom/none</code></td>
</tr>
<tr>
<td>Via</td>
<td>IETF RFC 3261</td>
<td><code>Via: SIP/2.0/UDP operator1.com:5060;branch=z9hG4bK87asdks7</code></td>
</tr>
<tr>
<td>Warning</td>
<td>IETF RFC 3261</td>
<td><code>Warning: 307 isi.edu &quot;Session parameter 'foo' not understood&quot;</code></td>
</tr>
</tbody>
</table>

**Table 9: SIP Header Examples**
Annex B  SIP Message Examples (Informational)

Annex B provides example message flows and message contents across the NNI.

It is assumed that each operator has FQDNs of ibcf1.operatorA.com at the NNI and ibcf2.operatorB.com and IP addresses of 10.0.0.1/10.0.1.1 and 10.10.0.1/10.10.1.1 for control and media respectively as shown in figure 2. In each case, the IBCF configures media pin holes on its respective TrGW.

Figure 2: NNI Configuration
B.1 Voice Session Establishment & Teardown

Figure 3 illustrates the assumed message sequence for the establishment and teardown of a SIP session across the NNI.

```
Operator-A  Operator-B
INVITE (SDP Offer #1)
100 Trying (2)
183 Progress (SDP Answer #1) (3)
PRACK (4)
200 OK (PRACK) (5)
UPDATE (SDP Offer #2) (6)
200 OK (UPDATE) (SDP Answer #2) (7)
180 (Ringing) (8)
200 OK (INVITE) (9)
ACK (10)
(Session established – media flowing)
BYE (11)
200 OK (BYE) (12)
```
Figure 3: NNI Session Establishment & Teardown

Message 1 – INVITE (SDP Offer #1)

INVITE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Allow: INVITE,PRACK,ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY
P-Asserted-Identity: Tel:+397850316900
P-Access-Network-Info:3GPP-E-UTRAN-FDD:e-utran-cell-id-3gpp=1234
Privacy: Id
P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com
Max-Forwards:70
P-Early-Media: supported
Supported: 100rel, timer, precondition, histinfo, from-change
Session-Expires: 180;refresher=uac
Min-SE:90
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060;>;+g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Content-Type: application/sdp
Content-Length: 559

v=0
s=-
o=- 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=audio 52000 RTP/AVP 98 99 100 101
b=AS:41
b=RS:0
b=RR:500
a=inactive
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;max-red=0
a=rtpmap:99 AMR/8000
a=fmtp:99 mode-change-capability=2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv

Message 2 – 100 Trying
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Content-Length: 0

Message 3 – 183 Progress (SDP Answer #1)
SIP/2.0 183 Progress
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Allow: INVITE,PRACK,ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY
P-Asserted-Identity: Tel:+447960306800
Privacy: none
P-Access-Network-Info:3GPP-E-UTRAN-FDD:e-utran-cell-id:3gpp=5678910
P-Early-Media: supported
Require:100rel
RSeq:1234
Supported: 100rel, timer, precondition, histinfo, from-change
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=OperatorB.com
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Contact: <sip:10.10.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Content-Type: application/sdp

v=0
s=-
o=- 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=audio 53000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=inactive
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv

Message 4 – PRACK

PRACK sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 2 PRACK
Call-ID: dgh1234567@operatorA.com
RAck:1234
Max-Forwards:70
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

Message 5 – 200 OK (PRACK)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 2 PRACK
Call-ID: dgh1234567@operatorA.com
Content-Type: application/sdp
Content-Length: 0

Message 6 – UPDATE (SDP Offer #2)
UPDATE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 3 UPDATE
Call-ID: dgh1234567@operatorA.com
Max-Forwards:70
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Type: application/sdp
Content-Length: 452

v=0
s=-
o=- 0 0 IN4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=audio 52000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=sendrecv
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv

Message 7 – 200 OK (UPDATE) (SDP Answer #2)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345

CSeq: 3 UPDATE

Call-ID: dgh1234567@operatorA.com

Contact: <sip:10.10.0.1:5060>

Content-Type: application/sdp

Content-Length: 419

v=0
s=-
a=0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=audio 53000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=sendrecv
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional remote sendrecv
Message 8 – 180 Ringing

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34

To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345

CSeq: 1 INVITE

Call-ID: dgh1234567@operatorA.com

Allow: INVITE, PRACK, ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY

P-Asserted-Identity: Tel:+447960306800

Privacy: none

P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id=3gpp=5678910

P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=OperatorB.com

Supported: 100rel, timer, precondition, histinfo, from-change

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel

Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"

Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"

Content-Length: 0

Message 9 – 200 OK (INVITE)

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34

To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345

CSeq: 1 INVITE

Call-ID: dgh1234567@operatorA.com
Allow: INVITE, PRACK, ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY

P-Asserted-Identity: Tel:+447960306800

Privacy: none

P-Access-Network-Info: 3GPP-E-UTRAN-FDD; a-utran-cell-id-3gpp=5678910

P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com; term-oi=OperatorB.com

Supported: 100rel, timer, precondition, histinfo, from-change

Session-Expires: 150; refresher=uac

Require: timer

P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel

Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn:7:3gpp-service.ims.icsi.mmtel"

Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi-ref="urn%3Aurn:7:3gpp-service.ims.icsi.mmtel"

Content-Length: 0

Message 10 – ACK

ACK sip:+447960306800@operatorB.com; user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060; branch=1234567abcd
From: <sip:+397850316900@operatorA.com; user=phone>; tag=5678ab34
To: <sip:+447960306800@operatorB.com; user=phone>; tag=ade2345
CSeq: 4 ACK
Call-ID: dgh1234567@operatorA.com
Max-Forwards: 70
Route: sip:10.0.0.1:5060;l
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

Message 11 – BYE

BYE sip:+447960306800@operatorB.com; user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 5 BYE
Call-ID: dgh1234567@operatorA.com
Max-Forwards: 70
Route: sip:10.10.0.1:5060;lr
Content-Length: 0

Message 12 – 200 OK (BYE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 5 BYE
Call-ID: dgh1234567@operatorA.com
Content-Length: 0
B.2 Multi-Media Session Establishment & Teardown

This annex covers the case of a voice and video multi-media session establishment and teardown. The message flow and numbering is identical to that of annex B.1. The main difference is in the SDP exchange. In addition, the MMTel ICSI will reflect that video telephony is also possible. These differences will be highlighted in this sub-annex with references made to messages in annex B.1.

Message 1 – INVITE (SDP Offer #1)

As annex B.1 with modified P-Asserted-Service, Contact & Accept-Contact headers plus multi-media SDP in the message body.

............

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel;video
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel;video"
Contact: <sip:10.0.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel;video"

Content-Type: application/sdp
Content-Length: 1059

v=0
s=
O=- 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=audio 52000 RTP/AVP 98 99 100 101
b=AS:41
b=RS:0
b=RR:500
a=inactive
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;max-red=0
a=rtpmap:99 AMR/8000
a=fmtp:99 mode-change-capability=2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
m=video 54000 RTP/AVPF 101 102
b=AS:768
b=RS:0
b=RR:2500
a=inactive
a=rtpmap:101 H264/90000
a=fmtp:101 profile-level-id=42C016; packetization-mode=1; sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g==
a=rtpmap:102 H263-2000/90000
a=fmtp:102 profile=0; level=10; QCIF=2
a=rtcp-fb:* ccm fir
a=rtcp-fb:* ccm tmmbr
a=extmap:7 urn:3gpp:video-orientation
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
Message 2 – 100 Trying
As annex B.1.

Message 3 – 183 Progress (SDP Answer #1)
As annex B.1 with multi-media SDP in the message body.

P-Asserted-Service: urn:urn:3gpp-service.ims.icsi.mmtel:video
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel:video"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel:video"
Content-Type: application/sdp
Content-Length: 878

v=0
s=-
a= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=audio 53000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=inactive
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240

V6.0
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
m=video 56000 RTP/AVPF 101
b=AS:768
b=RS:0
b=RR:2500
a=inactive
a=rtpmap:101 H264/90000
a=fmtp:101 profile-level-id=42C016;packetization-mode=1;sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g==
a=rtcp-fb:* ccm fir
a=rtcp-fb:* ccm tmbr
a=extmap:7 urn:3gpp:video-orientation
a=curr:qos local none
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv

Message 4 – PRACK
As annex B.1.

Message 5 – 200 OK (PRACK)
As annex B.1.
Message 6 – UPDATE (SDP Offer #2)

As annex B.1 with multi-media SDP in the message body.

Content-Type: application/sdp
Content-Length: 884

v=0
s=
o= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=audio 52000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=sendrecv
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
m=video 54000 RTP/AVPF 101
b=AS:768
Message 7 – 200 OK (UPDATE) (SDP Answer #2)
As annex B.1 with multi-media SDP in the message body.

Content-Type: application/sdp
Content-Length: 836

v=0
s=-
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=audio 53000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=sendrecv
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional remote sendrecv
m=video 56000 RTP/AVPF 101
b=AS:768
b=RS:0
b=RR:2500
a=sendrecv
a=rtpmap:101 H264/90000
a=fmtp:101 profile-level-id=42C016;packetization-mode=1;sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g==
a=rtcp-fb:* ccm fir
a=rtcp-fb:* ccm tmbr
a=extmap:7 urn:3gpp:video-orientation
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional remote sendrecv

Message 8 – 180 Ringing
As annex B.1. with modified P-Asserted-Service, Contact & Accept-Contact headers.
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel;video
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel;video"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel;video"

Message 9 – 200 OK (INVITE)
As annex B.1. with modified P-Asserted-Service, Contact & Accept-Contact headers.

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel;video
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel;video"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel;video"

Message 10 – ACK
As annex B.1.

Message 11 – BYE
As annex B.1.

Message 12 – 200 OK (BYE)
As annex B.1.
B.3 Use of session timer

The session establishment sequence in annex B.1 indicated that session timer (IETF RFC 4028 [16]) would be used with the UAC generating the refresh messages. The negotiated session time was 150 seconds. Therefore, the UAC shall generate a refresh message every 75 seconds. In the event of the refresh not being sent, then the UAS would terminate the session after 118 seconds.

Message 1 – UPDATE

UPDATE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 11 UPDATE
Call-ID: dgh1234567@operatorA.com
Supported: timer
Session-Expires:150;refresher=uac
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length:0
Message 2 – 200 OK (UPDATE)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 11 UPDATE
Call-ID: dgh1234567@operatorA.com
Require: timer
Session-Expires:150;refresher=uac
Contact: <sip:10.10.0.1:5060>
Content-Length:0

Message 3 – BYE

BYE sip:+397850316900@operatorA.com;user=phone   SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=5678910abc
To: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
From: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 BYE
Call-ID: dgh1234567@operatorA.com
Route: sip:10.0.0.1:5060;lr
Content-Length:0

Message 4 – 200 OK (BYE)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=5678910abc
To: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
From: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 BYE
Call-ID: dgh1234567@operatorA.com
Content-Length: 0
B.4 Use of Early Media

This annex covers the case of early media being sent from the terminating side (e.g. the call break out into the CS network and ring tone is sent from the terminating side. The message flow and numbering is identical to that of annex B.1. The only difference is in the 180 Ringing message, which is shown below:

Message 8 – 180 Ringing

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Allow: INVITE,PRACK,ACK,CANCEL,UPDATE,PUBLISH,OPTIONS,MESSAGE,BYE,REFER,SUBSCRIBE,NOTIFY
P-Asserted-Identity: Tel:+447960306800@operatorB.com
Privacy: none
P-Early-Media: sendonly
Supported: 100rel,timer,precondition,histinfo,from-change
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Contact: <sip:10.10.0.1:5060>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Content-Length: 0

The presence of the P-Early-Media header causes the IBCF to ensure that RTP packets are conveyed in the backward direction via the TrGW.
B.5 Re-packing between AMR-WB and EVS IO modes

The SDP modifications during the offer-answer along with the relevant parts of SIP commands are described below.

**Note:**

The message sequence is reduced to the codec negotiation relevant messages. Also note that SDP messages are shortened to improve readability.

B.5.1 Example 1: oUE + oIMS are EVS and AMR-WB capable, tUE + tIMS are only AMR-WB capable
### SIP Invite

<table>
<thead>
<tr>
<th>Originating UE</th>
<th>Originating IMS</th>
<th>Terminating IMS</th>
<th>Terminating UE AMR-WB capable</th>
</tr>
</thead>
<tbody>
<tr>
<td>EVS + AMR-WB capable</td>
<td>EVS+ AMR-WB capable</td>
<td>IMS</td>
<td>IMS</td>
</tr>
</tbody>
</table>

**Initial SDP offer**

- `m=audio 3000 RTP/AVP 96 97`
- `a=rtpmap:96 EVS/16000`
- `a=fmtp:96 br=5.9-24.4;bw=nb-swb`
- `a=rtpmap:97 AMR-WB/16000`
- `a=fmtp:97 mode-set=0,1,2`

**SIP 183 Progress**

<table>
<thead>
<tr>
<th></th>
<th>Confirming SDP answer</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><code>m=audio 3000 RTP/AVP 97</code></td>
</tr>
<tr>
<td></td>
<td><code>a=rtpmap:97 AMR-WB/16000</code></td>
</tr>
<tr>
<td></td>
<td><code>a=fmtp:97 mode-set=0,1,2</code></td>
</tr>
</tbody>
</table>

**SIP UPDATE**

<table>
<thead>
<tr>
<th></th>
<th>Subsequent SDP offer</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><code>m=audio 3000 RTP/AVP 96</code></td>
</tr>
<tr>
<td></td>
<td><code>a=rtpmap:96 EVS/16000</code></td>
</tr>
<tr>
<td></td>
<td><code>a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1</code></td>
</tr>
</tbody>
</table>

**SIP 200 OK**

<table>
<thead>
<tr>
<th></th>
<th>Subsequent SDP answer</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><code>m=audio 3000 RTP/AVP 97</code></td>
</tr>
<tr>
<td></td>
<td><code>a=rtpmap:97 AMR-WB/16000</code></td>
</tr>
<tr>
<td></td>
<td><code>a=fmtp:97 mode-set=0,1,2</code></td>
</tr>
</tbody>
</table>

**Applied modifications:**

- modification 1a: originating IMS network modifies received confirming and subsequent SDP answer by replacing AMR-WB with EVS IO mode (in the sent SDP answer)
- modification 1b: originating IMS network replaces EVS IO mode in received subsequent SDP offer with AMR-WB (in sent subsequent SDP offer)
With those SDP modifications, EVS IO mode is used on originating UE, while AMR-WB is used on terminating UE. The originating IMS network needs to perform RTP re-packaging to connect the two UEs, but transcoding is not necessary. In the example above, RTP re-packaging would include translating RTP payload type 96 in the RTP header received from the originating UE to RTP payload type 97 in the RTP header sent towards the terminating IMS network, and vice versa.
### B.5.2 Example 2: oUE + oIMS are only AMR-WB capable, tUE + tIMS are EVS and AMR-WB capable

<table>
<thead>
<tr>
<th></th>
<th>Initial SDP offer</th>
<th>Final SDP offer</th>
<th>Confirming SDP answer</th>
<th>Subsequent SDP offer</th>
<th>Subsequent SDP answer</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>SIP Invite</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Originating UE AMR-WB</td>
<td>m=audio 3000 RTP/AVP 96</td>
<td>Final SDP offer (modification 2a – probe for EVS support)</td>
<td>m=audio 3000 RTP/AVP 97</td>
<td>m=audio 3000 RTP/AVP 96</td>
<td></td>
</tr>
<tr>
<td>capability</td>
<td>a=rtpmap:96 AMR-WB/16000</td>
<td>a=rtppmap:97 EVS/16000</td>
<td>a=rtppmap:97 EVS/16000</td>
<td>a=rtppmap:96 AMR-WB/16000</td>
<td>a=rtppmap:96 AMR-WB/16000</td>
</tr>
<tr>
<td></td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td>a=fmtp:97 br=5.9-24.4;bw=nb-sw; mode-set=0,1,2;evs-mode-switch=1</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
</tr>
<tr>
<td><strong>SIP 183 Progress</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Confusing SDP answer</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>m=audio 3000 RTP/AVP 97</td>
<td></td>
<td>m=audio 3000 RTP/AVP 97</td>
<td>m=audio 3000 RTP/AVP 97</td>
<td></td>
</tr>
<tr>
<td></td>
<td>a=rtpmap:96 AMR-WB/16000</td>
<td></td>
<td>a=rtpmap:96 AMR-WB/16000</td>
<td>a=rtpmap:96 AMR-WB/16000</td>
<td>a=rtpmap:96 AMR-WB/16000</td>
</tr>
<tr>
<td></td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td></td>
<td>a=fmtp:97 br=5.9-24.4; bw=nb-sw; mode-set=0,1,2; evs-mode-switch=1</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
</tr>
<tr>
<td><strong>SIP UPDATE</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>m=audio 3000 RTP/AVP 96</td>
<td></td>
<td>m=audio 3000 RTP/AVP 97</td>
<td>m=audio 3000 RTP/AVP 97</td>
<td></td>
</tr>
<tr>
<td></td>
<td>a=rtpmap:96 AMR-WB/16000</td>
<td></td>
<td>a=rtpmap:97 EVS/16000</td>
<td>a=rtpmap:97 EVS/16000</td>
<td>a=rtpmap:96 AMR-WB/16000</td>
</tr>
<tr>
<td></td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td></td>
<td>a=fmtp:97 br=5.9-24.4; bw=nb-sw; mode-set=0,1,2; evs-mode-switch=1</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
</tr>
<tr>
<td><strong>SIP 200 OK</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Confusing SDP answer</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>m=audio 3000 RTP/AVP 97</td>
<td></td>
<td>m=audio 3000 RTP/AVP 97</td>
<td>m=audio 3000 RTP/AVP 97</td>
<td></td>
</tr>
<tr>
<td></td>
<td>a=rtpmap:96 AMR-WB/16000</td>
<td></td>
<td>a=rtpmap:97 EVS/16000</td>
<td>a=rtpmap:97 EVS/16000</td>
<td>a=rtpmap:96 AMR-WB/16000</td>
</tr>
<tr>
<td></td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td></td>
<td>a=fmtp:97 br=5.9-24.4; bw=nb-sw; mode-set=0,1,2; evs-mode-switch=1</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
<td>a=fmtp:96 mode-set=0,1,2</td>
</tr>
</tbody>
</table>
Applied modifications:

- modification 2a: terminating IMS network adds EVS IO mode with highest priority to received SDP initial offer to probe terminating UE for EVS (in final SDP offer)
- modification 2b: terminating IMS network replaces EVS IO mode in received confirming or subsequent SDP answer with AMR-WB (in the sent SDP answer)
- modification 2c: terminating IMS network replaces AMR-WB with EVS IO mode (in subsequent SDP offer)

With those SDP modifications, EVS IO mode is used on terminating UE, while AMR-WB is used on originating UE. The terminating IMS network needs to perform RTP re-packaging to connect the two UEs, but transcoding is not necessary. In the example above, RTP re-packaging would include translating RTP payload type 96 in the RTP header received from the originating IMS network to RTP payload type 97 in the RTP header sent towards the terminating UE, and vice versa.
B.6 IMS Registration

Figure 5 illustrates the assumed message sequence for IMS registration. This flow is applicable only to the roaming NNI (i.e. UE roaming in a visited network and performing IMS registration to its home network). The P-CSCF shall be in the visited network and the S-CSCF in the home network with the Mw reference point crossing the NNI. The SUBSCRIBE message is sent from both the P-CSCF and UE in the visited network to the S-CSCF in the home network. In this example, the UE is assumed to support MMTel services as well as a number of RCS services.

Figure 5: IMS Registration

In the above flow, it is assumed that user +447960306800@operatorB.com is roaming in the network of operatorA. The message contents are below:

Message 1 – REGISTER

REGISTER sip:operatorB.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=7895ad34
To: <sip:+447960306800@operatorB.com;user=phone>

CSeq: 1 REGISTER

Call-ID: dgh1234567@operatorB.com

P-Access-Network-Info: 3GPP-E-UTRAN-FDD; e-utran-cell-id-3gpp=1234

P-Visited-Network-Id: "operatorA.com"

Max-Forwards: 70

Authorization: Digest username=user1234567@operatorB.com, realm="operator.com", uri="sip:operator.com", nonce="", response=""

P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com

Require: path

Supported: gruu

Path: <sip:10.0.0.1:5060;lr>

Route: sip:10.10.0.1:5060;lr

Contact: <sip:10.0.0.1:5060>; +sip.instance="<urn:gsma:imei:90420156-025763-0>"; +g.3gpp.smssip, +g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel:video, urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm:msg; urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"; +g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-application.ims.ari.rcse.im, urn%3Aurn-7%3A3gpp-application.ims.ari.rcse.ft, urn%3Aurn-7%3A3gpp-application.ims.ari.gsm-is, urn:urn-7%3A3gpp-application.ims.ari.gsma-vs"

Content-Length: 0

Message 2 – 401 Unauthorised

SIP/2.0 401 Unauthorised

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+447960306800@operatorB.com;user=phone>;tag=7895ad34

To: <sip:+447960306800@operatorB.com;user=phone>;tag=34589012

CSeq: 1 REGISTER

Call-ID: dgh1234567@operatorB.com

WWW-Authenticate: Digest realm="scscf12.operator.com", nonce="12ab34bc4567b45RTFGH12323asdfghGHDe34500oIP+", algorithm=AKAv1-MD5, ik=2345, ck=5678
Message 3 – REGISTER

REGISTER sip:operatorB.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 2 REGISTER
Call-ID: dgh1234567@operatorB.com
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
P-Visited-Network-Id: "operatorA.com"
Max-Forwards:70
Authorization: Digest username=user1234567@operatorB.com, realm="scscf12.operator.com", uri="sip:operator.com
nonce="12ab34bc4567b45RTFGH12323asdfghGHDe34500oOlP+", response="2345bcdef567bf54ef"
P-Charging-Vector: icid-value="abch+236788a"; orig-oi=operatorA.com
Require: path
Supported: gruu
Path: <sip:10.0.0.1:5060;lr>
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>;+sip.instance="<urn:gsma:imei:90420156-025763-0>";
+g.3gpp.smsip, +g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel;video,
urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.msg; urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"; +g.3gpp.icii-ref="urn%3Aurn-7%3A3gpp-
application.ims.iari.rcse.im, urn%3Aurn-7%3A3gpp-application.ims.iari.rcse.ft, urn%3Aurn-
7%3A3gpp-application.ims.iari.rcse.fthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7%3A3gpp-application.ims.iari.gsma-vs"

Content-Length: 0

Message 4 – 200 OK (REGISTER)

SIP/2.0 200 OK
 Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>;tag=34578231
CSeq: 2 REGISTER
Call-ID: dgh1234567@operatorB.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id:3gpp=1234
P-Charging-Vector: icid-value="abch+236788a"; orig-ioi=operatorA.com; term-ioi=operatorB.com
P-Associated-URI:sip:+447960306800@operatorB.com;user=phone
Service-Route: <sip:10.10.0.1:5060;lr>
Contact: <sip:10.10.0.1:5060>;expires=3600
Content-Length: 0

Message 5 – SUBSCRIBE
SUBSCRIBE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=587903ab
From: <sip:ibcf4@operatorA.com>;tag=123456
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 10 SUBSCRIBE
Call-ID: 5634567@operatorA.com
Max-Forwards: 70
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Event: reg
P-Asserted-Id: <sip:10.0.0.1:5060;lr>
Expires:4000
Accept: application/reginfo+xml
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Message 6 – 200 OK (SUBSCRIBE)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=587903ab
From: <sip: ibcf4@operatorA.com>;tag=123456
To: <sip: +447960306800@operatorB.com;user=phone>;tag=7612345
CSeq: 10 SUBSCRIBE
Call-ID: 5634567@operatorA.com
Event: reg
Expires: 3600
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Contact: <sip:10.10.0.1:5060>
Content-Length: 0

Message 7 – NOTIFY

NOTIFY sip:ibcf4@operatorA.com SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=abc45678
To: <sip: ibcf4@operatorA.com>;tag=123456
From: <sip: +447960306800@operatorB.com;user=phone>;tag=7612345
Call-ID: 5634567@operatorA.com
CSeq: 20 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=3600
Event: reg
Accept: application/reginfo+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Message 8 – 200 OK (NOTIFY)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=abc45678
To: <sip: ibcf4@operatorA.com>;tag=123456
From: <sip: +447960306800@operatorB.com;user=phone>;tag=7612345
Call-ID: 5634567@operatorA.com
CSeq: 20 NOTIFY
Contact: <sip:10.10.0.1:5060>
Content-Length: 0

Messages 9-12
As messages 5-8 apart from the CSeq and Call-Id headers and From/To tags changing accordingly.
B.7 MMTel Services

This annex covers the impacts on the NNI services due to the mandatory set of MMTel services as listed in GSMA PRD IR.92 [2]. Dependent on the impact of the service, references/changes to basic call messages or additional call flows will be listed.

Originating Identification Presentation (OIP)

There is no further impact on basic call signalling due to OIP. The OIP service uses the P-Asserted-Identity and associated Privacy and From headers that are sent in the INVITE message.

Originating Identification Restriction (OIR)

There is no further impact on basic call signalling due to OIR. The OIR service causes the Privacy header in the INVITE message to be set to:

Privacy:Id

Additionally, the value ‘user’ can be added to the Privacy header and/or the From header can be anonymized.

Terminating Identification Presentation (TIP)

The TIP service uses the P-Asserted-Identity and associated Privacy headers that are sent in the 18X / 200 OK (INVITE) messages. There is one small change to the basic call signalling, namely that an additional option tag (“from-change” - see IETF RFC 4916 [49]) may be exchanged across the NNI. This option indicates that it is possible to pass a connected party identity subsequent to the session being established (i.e. post 200 OK (INVITE)/ACK). The modified identity would be passed in a FROM header in the URI portion and re-using the previous FROM tag used in the initial dialog establishment. The updated FROM header may be sent in a SIP UPDATE or re-INVITE message. An example of the UPDATE exchange is shown in figure 6:

Figure 6: TIP (from Change)

The message contents are as follows (assuming session establishment as per annex B.1):

Message 1 – UPDATE

UPDATE sip:+397850316900@operatorA.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=7654321

V6.0
To: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
From: <sip:+447960306812@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 UPDATE
Call-ID: dgh1234567@operatorA.com
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length:0

Message 2 – 200 OK (UPDATE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=7654321
To: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
From: <sip:+447960306812@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 UPDATE
Call-ID: dgh1234567@operatorA.com
Contact: <sip:10.0.0.1:5060>
Content-Length:0

Terminating Identification Restriction (TIR)
There is no further impact on basic call signalling due to TIR. The TIR service causes the Privacy headers in the 18X / 200 OK (INVITE) messages to be set as follows:

Privacy:Id

Communication Waiting (CW)
As stated in GSMA PRD IR.92 [2], it is assumed that the terminal based flavour of the service is used. The sole change is that the alert-info header is conveyed in the 180 Ringing message to inform the originating UE that this is a waiting communication, e.g.

Alert-Info:<urn:alert:service:call-waiting>

In addition, the AS may also (on receiving a 180 Ringing message with the Alert-Info header present) connect an indication which would be reflected as a modified Offer/Answer exchange between the AS/MRFC and UE.

Outgoing Communication Barring (OCB)
This includes a number of flavours, namely Barring of All Outgoing Calls, Barring of Outgoing International Calls & Barring of Outgoing Calls (ex Home Country). These features
are only applicable at the NNI in the roaming NNI scenario. The message flow is shown in figure 7. Note that the MMTel AS may optionally connect an indication prior to generating the 603 Decline response, which could result in related Offer/Answer exchanges between the AS/MRFC and UE.

![Message Flow Diagram]

**Figure 7: OCB**

**Message 1 – INVITE**
As Annex B.1.

**Message 2 – 603 Decline**
SIP/2.0 603 Decline
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Content-Length: 0

**Message 3 – ACK**
As Annex B.1.

**Incoming Communication Barring (ICB)**

This includes two flavours, namely Barring of All Incoming Calls & Barring of Incoming Calls when roaming. The message flow is shown in figure 8. Note that the MMTel AS may optionally connect an indication prior to generating the 603 Decline response, which could result in related Offer/Answer exchanges between the AS/MRFC and UE.
Figure 8: ICB

Message 1 – INVITE
As Annex B.1.

Message 2 – 603 Decline
SIP/2.0 603 Decline
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Content-Length: 0

Message 3 – ACK
As Annex B.1.

Communication Hold (CH)
This service is realised via an additional offer/answer exchange where the bi-directional media (as established in annex B.1) is set to sendonly. The message flow for CH is as shown in figure 9. Answer exchanges between the AS/MRFC and UE.
Figure 9: Communication Hold

Message 1 – UPDATE (SDP Offer #2)

UPDATE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 11 UPDATE
Call-ID: dgh1234567@operatorA.com
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Type: application/sdp
Content-Length: 452

v=0
s=-
o=- 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=audio 52000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=sendonly
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote none
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv

Message 2 – 200 OK (UPDATE) (SDP Answer #2)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 11 UPDATE
Call-ID: dgh1234567@operatorA.com
Contact: <sip:10.10.0.1:5060>
Content-Type: application/sdp
Content-Length: 429
v=0
s=
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=audio 53000 RTP/AVP 98 100
b=AS:30
b=RS:0
b=RR:500
a=recvonly
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 mode-change-capability=2;mode-set=0,1,2;max-red=0
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=ptime:20
a=maxptime:240
a=curr:qos local sendrecv
a=des:qos mandatory local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional remote sendrecv

Note that the MMTel AS may insert an announcement, in which case the new offer/answer would reflect the presence of the MRFP.

The session may be resumed to duplex speech with a further offer answer exchange with the media attribute line restored to “sendrecv”.
**Communication Forwarding (CFx)**

This service has a number of flavours, namely Unconditional (CFU), Not Logged In (CFNL), Busy (CFB), Not Reachable (CFNR), Communication Deflection (CD) and No Reply (CFNR). Note that CD is not in the scope of GSMA PRD IR.92 [5]. The service impacts both forward and backward SIP signalling across the NNI. In the example here, it is assumed that CFU is used. Further, it is assumed that an initial call forwarding occurs in the originating network with the call being offered across the NNI to the terminating network where a second call forwarding occurs. The message flow is as shown in figure 10.

![Message Flow Diagram](image)

**Figure 10: Call Forwarding**

The message contents are as shown below. Note that it is assumed that the originating party (+397850316900@operatorA) initially calls user +397850516900@operatorA and the call is forwarded (CFU) to +447960306800@operatorB. Subsequently, CFU is encountered in the terminating network and the call is re-targeted to its final destination of +447960306900@operatorB.

The message contents are very close to the corresponding messages in annex B.1 – and therefore only differences from annex B.1 will be shown here.
Message 1 – INVITE (SDP Offer #1)

INVITE sip:+447960306800@operatorB.com;user=phone;cause=302 SIP/2.0

History-Info:<sip:+397850516900@operatorA;user=phone?privacy=history>;index=1,
<sip:+447960306800@operatorB;user=phone;cause=302>;index=1.1

Message 2 – 100 Trying

As annex B.1.

Message 2a – 181 Call is being forwarded

SIP/2.0 181 Call is being forwarded
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345

CSeq: 1 INVITE

Call-ID: dgh1234567@operatorA.com

Allow: INVITE,PRACK,ACK,CANCEL,UPDATE,PUBLISH,OPTIONS,MESSAGE,BYE,REFER,
SUBSCRIBE,NOTIFY

P-Asserted-Identity: Tel:+447960306900@operatorB.com

Privacy: none

History-Info:<sip:+397850516900@operatorA;user=phone?privacy=history>;index=1,
<sip:+447960306800@operatorB;user=phone;cause=302>;index=1.1,
<sip:+447960306900@operatorB;user=phone;cause=302>;index=1.1.1

P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=9108910

P-Early-Media: supported

Supported: 100rel,timer,precondition,histinfo,from-change

P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel

Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"

Contact: <sip:10.10.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"

Content-Length: 0

Message 3 – 183 Progress (SDP Answer #1)
Message 4 – PRACK
PRACK sip:+447960306800@operatorB.com;user=phone;cause=302 SIP/2.0

Message 5 – 200 OK (PRACK)
As annex B.1.

Message 6 – UPDATE (SDP Offer #2)
UPDATE sip:+447960306800@operatorB.com;user=phone;cause=302 SIP/2.0

Message 7 – 200 OK (UPDATE) (SDP Answer #2)
As annex B.1.

Message 8 – 180 Ringing
P-Asserted-Identity: Tel:+447960306900@operatorB.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=9108910
History-Info:<sip:+397850516900@operatorA;user=phone?privacy=history>;index=1,
<sip:+447960306800@operatorB;user=phone;cause=302>;index=1.1,
<sip:+447960306900@operatorB;user=phone;cause=302>;index=1.1.1

Message 9 – 200 OK (INVITE)
P-Asserted-Identity: Tel:+447960306900@operatorB.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=9108910
History-Info:<sip:+397850516900@operatorA;user=phone?privacy=history>;index=1,
<sip:+447960306800@operatorB;user=phone;cause=302>;index=1.1,
<sip:+447960306900@operatorB;user=phone;cause=302>;index=1.1.1
Message 10 – ACK

ACK sip:+447960306800@operatorB.com;user=phone;cause=302 SIP/2.0

Message Waiting Indication (MWI)

As stated in GSMA PRD IR.92 [2], it is assumed that this service uses the Message Waiting Indication (see IETF RFC 3842 [42]).

This service crosses the NNI only in the roaming NNI case (i.e. UE is visited network and MMTel AS in the home network. The message sequence is shown in figure 11.

![Message Waiting Diagram](image)

**Figure 11 : Message Waiting**

The message contents are as shown below.

**Message 1 – SUBSCRIBE**

SUBSCRIBE sip:+447960306800@operatorB.com;user=phone SIP/2.0

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip: +447960306800@operatorB.com;user=phone>;tag=5678ab34

To: <sip: +447960306800@operatorB.com;user=phone>

CSeq: 1 SUBSCRIBE

Call-ID: abc1234567@operatorB.com

P-Asserted-Identity: Tel:+447960306800

P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234

Privacy: Id

Max-Forwards: 70

P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Expires: 7200
Event: message-summary
Accept: application/simple-message-summary
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

Message 2 – 200 OK (SUBSCRIBE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip: +447960306800@operatorB.com;user=phone>;tag=5678ab34
To: <sip: +447960306800@operatorB.com;user=phone>;tag=123456
CSeq: 1 SUBSCRIBE
Call-ID: abc1234567@operatorB.com
Event: message-summary
Expires: 7200
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

Message 3 – NOTIFY
NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip: +447960306800@operatorB.com;user=phone>;tag=5678ab34
From: <sip: +447960306800@operatorB.com;user=phone>;tag=123456
Call-ID: abc1234567@operatorB.com
CSeq: 12 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=7200
Event: Message-Summary
Accept: application/simple-message-summary
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Type: application/simple-message-summary
Content-Length: 90
Message-Waiting: yes
Message-Account: sip:+447960306800@operatorB.com;user=phone
Voice-Message: 4/1 (2/1)
Video-Message: 0/0 (0/0)
To: <+447960306800@operatorB.com;user=phone>
From: <+442476452864@operatorC.com;user=phone>
Subject: call me back!
Date: 19 Apr 20013 21:45:31 -0700
Priority: urgent
Message-ID: 27775334485@operatorB.com
Message-Context: voice-message
To: <+447960306800@operatorB.com;user=phone>
From: <+442476633123@operatorD.com;user=phone>
Subject: Where are you that late???
Date: 19 Apr 2013 23:45:31 -0700
Priority: urgent
Message-ID: 27775334485@operatorB.com
Message-Context: voice-message

Message 4 – 200 OK (NOTIFY)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=567891234a
To: <sip: +447960306800@operatorB.com;user=phone>;tag=5678ab34
From: <sip: +447960306800@operatorB.com;user=phone>;tag=123456
Call-ID: abc1234567@operatorB.com

V6.0
Ad-Hoc Conference

As stated in GSMA PRD IR.92 [2], this service is realised via the UE initiating a session with a conference factory and then adding further users (including those which currently have an active session with the initiating UE) via REFER requests to the conference factory. Assume that there is a session established as previously shown in annex B.1. The originating party decides to initiate a conference. Separate flows are shown (see figures 12 to 14). i) to invoke the conference factory, ii) subscribe to the conference package and iii) to move/add other participants to the conference. In terms of the NNI, these actions are applicable only to the roaming NNI (UE in the visited network and the AS/conference-factory in the home network). It should also be noted that the signalling to the other participants from the AS/conference-factory can cross the I-NNI.

Figure 12: Ad-Hoc Conference (Invoking the Conference-Factory)

The above message sequence is very similar to that in annex B.1 (essentially, the target is the conference factory rather than another UE and there is no 180 Ringing response). Therefore, for brevity, only differences from annex B.1 are shown below.

Message 1 – INVITE (SDP Offer #1)

INVITE sip: conference-factory1@operatorB.com SIP/2.0
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>

V6.0
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Call-ID: abc1234567@operatorB.com
P-Asserted-Identity: Tel: +4479606306800

Message 2 – 100 Trying
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>
Call-ID: abc1234567@operatorB.com

Message 3 – 183 Progress (SDP Answer #1)
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Call-ID: abc1234567@operatorB.com
Contact: <sip:10.10.0.1:5060>;isfocus

Message 4 – PRACK
PRACK sip: conference-factory1@operatorB.com SIP/2.0
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
Call-ID: abc1234567@operatorB.com

Message 5 – 200 OK (PRACK)
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
Call-ID: abc1234567@operatorB.com
Message 6 – UPDATE (SDP Offer #2)

UPDATE sip: conference-factory1@operatorB.com SIP/2.0
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
Call-ID: abc1234567@operatorB.com

Message 7 – 200 OK (UPDATE) (SDP Answer #2)

From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
Call-ID: abc1234567@operatorB.com
Contact: <sip:10.10.0.1:5060>;isfocus

Message 9 – 200 OK (INVITE)

From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
Call-ID: abc1234567@operatorB.com
Contact: <sip:10.10.0.1:5060>;isfocus

Message 9 – ACK

UPDATE sip: conference-factory1@operatorB.com SIP/2.0
From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456
To: <sip:conference-factory1@operatorB.com>;tag=8910123
Call-ID: abc1234567@operatorB.com

There is now a session set up between the user +447960306800@operatorB.com and the conference factory. The user now subscribes to the conference event package as shown in figure 13.
The message contents are as shown below.

**Message 1 – SUBSCRIBE**

SUBSCRIBE sip: conference-factory1@operatorB.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abed
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
To: <sip:conference-factory1@operatorB.com>
P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com; term-oi=operatorB.com
Call-ID: abc1234588@operatorB.com
CSeq: 1 SUBSCRIBE
P-Asserted-Identity: Tel:+447960306800
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
Privacy: Id
Max-Forwards: 70
Expires:7200
Event: conference
Accept: application/conference-info+xml
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

**Message 2 – 200 OK (SUBSCRIBE)**
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
To: <sip:conference-factory1@operatorB.com>;tag=3467894a
P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com; term-oi=operatorB.com
Call-ID: abc1234588@operatorB.com
CSeq: 1 SUBSCRIBE
Event: conference
Expires: 7200
P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com; term-oi=operatorB.com
Contact: <sip:10.10.0.1:5060>
Content-Length: 0

Message 3 – NOTIFY
NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=3467894a
Call-ID: abc1234588@operatorB.com
CSeq: 12 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=7200
Event: conference
Accept: application/conference-info+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Type: application/conference-info+xml
Content-Length: xx

[Conference Info] - see IETF RFC 4575 [43]
The controlling party will now add/move participants to the conference. This is done via a REFER to the AS/Conference-Factory as shown in figure 13. There are 2 distinct cases, namely the moving of a party in an existing 2-way session and the adding of a new party. In the latter case, parties may be added piece-meal or en-bloc. Message contents are shown for the moving of an existing party and addition of a single new party. Note that multiple parties may also be added in a single step via an XML body containing a URI-List (see IETF RFCs 5364 [59] & 5368 [61]).

The controlling party may be informed of the progress of the REFER request via SIP NOTIFY messages (there is an implicit subscription via the REFER request – see IETF RFC 3515 [51]). However, in this case, as the UE has already subscribed to the conference event package, then it is assumed that the REFER contains the Refer-Sub header set to false which over-rides the implicit subscription. The NOTIFY messages thus carry information relating to the conference progress.

**Figure 14 : Referring a User Info the Conference**
The message contents are as shown below.

**Message 1a – REFER (to an existing party)**

REFER sip: conference-factory1@operatorB.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
To: <sip:conference-factory1@operatorB.com>
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Call-ID: 56789102@operatorB.com
CSeq: 1 REFER
P-Asserted-Identity: Tel:+447960306800
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
Privacy: Id
Refer-Sub: false
Accept: application/sdp, message/sipfrag
Require: nofrefsu
Max-Forwards: 70
Refer-To:<sip: +397850316900@operatorA.com;user=phone;method=INVITE?Replaces=dgh123456 %40operatorA.com%3Bto-tag%3Dade2345%3Bfrom-tag%3D5678ab34?Require=replaces>
Referred-By: <sip:+447960306800@operatorB.com;user=phone>
Route: sip:10.10.0.1:5060;lr
Contact: <conference-factory1@operatorB.com>;isfocus
Content-Length: 0

The above message causes the conference factory to initiate a session to the cited (Refer-To) target which will contain the following headers (copied/derived from the REFER message).

Require: replaces
Referred-By: <sip:+447960306800@operatorB.com;user=phone>
Replaces: dgh1234565@operatorA.com;to-tag=ade2345;from-tag=5678ab34
Contact: <conference-factory1@operatorB.com>;isfocus

The target will accept this session and redirect the existing media endpoint to the conference factory, prior to sending a BYE to terminate the previous 2-way session.
Message 1b – REFER (to a new single party)

REFER sip: conference-factory1@operatorB.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
To: <sip:conference-factory1@operatorB.com>
P-Charging-Vector: icid-value="abch+23456y"; orig(ioi=operatorA.com; term(ioi=operatorB.com
Call-ID: 56789102@operatorB.com
CSeq: 1 REFER
P-Asserted-Identity: Tel:+447960306800
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
Privacy: Id
Max-Forwards: 70
Refer-To:<sip: +477969123456@operatorC.com;user=phone;method=INVITE>
Referred-By: sip:+447960306800@operatorB.com;user=phone
Refer-Sub: false
Accept: application/sdp, message/sipfrag
Require: nofibersub
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

The above message causes the conference factory to initiate a session to the cited (Refer-To) target which will add that party to the conference. The CONTACT header will contain the “isfocus” parameter (IETF RFC 3840 [52]).

Message 2 – 202 ACCEPTED

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
To: <sip:conference-factory1@operatorB.com>;tag=7824567
P-Charging-Vector: icid-value="abch+23456y"; orig(ioi=operatorA.com; term(ioi=operatorB.com
Message 3 – NOTIFY

NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=7824567
Call-ID: 56789102@operatorB.com
CSeq: 13 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=7200
Event: conference
Accept: application/conference-info+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Type: application/conference-info+xml
Content-Length: xx

[Conference Info] - see IETF RFC 4575 [43]

Message 4 – 200 OK (NOTIFY)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=7824567
Call-ID: 56789102@operatorB.com
CSeq: 13 NOTIFY
Messages 5 & 6 – As for 3 & 4 (with modified CSeq and updated XML body)

Note that if the implicit subscription to the progress of the REFER was not inhibited, then additional
notifications would also be sent. For completeness, examples are shown below:

```
NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=7824567
Call-ID: 56789102@operatorB.com
CSeq: 13 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=7200
Event: refer
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Type: message/sip-frag;version=2.0
Content-Length: 20
SIP/2.0 100 Trying
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=7824567
Call-ID: 56789102@operatorB.com
CSeq: 13 NOTIFY
Contact: <sip:10.0.0.1:5060>
Content-Length: 0
```
NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=7824567
Call-ID: 56789102@operatorB.com
CSeq: 14 NOTIFY
Max-Forwards: 70
Subscription-State: terminated;reason=noresource
Event: refer
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Type: message/sip-frag;version=2.0
Content-Length: 16

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
From: <sip:conference-factory1@operatorB.com>;tag=7824567
Call-ID: 56789102@operatorB.com
CSeq: 14 NOTIFY
Contact: <sip:10.0.0.1:5060>
Content-Length: 0
B.8 RCS Capability Exchange

Figure 15 illustrates a RCS Capability Exchange via SIP OPTIONS. This interchange may be done as a stand-alone transaction or else as part of an active SIP session (e.g. a voice call). The example below is assumed to be a stand-alone transaction.

![RCS Capability Exchange Diagram]

**Message 1 – OPTIONS (SDP Offer #1)**

OPTIONS sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 11 OPTIONS
Call-ID: dgh1234567@operatorA.com
P-Asserted-Identity: Tel:+397850316900
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id=3gpp=1234
Privacy: Id
Accept: application/sdp, multipart/mixed, multipart/related, message/external-body, message/cpim, message/imdn+xml, application/pidf+xml, application/pidf-diff+xml, application/watcherinfo+xml, application/xcap-diff+xml, application/vnd.oma.suppnot+xml, application/simple-filter+xml, application/resource-lists+xml, application/rmi+xml, application/im-iscomposing+xml, application/vnd.3gpp.sms
Max-Forwards:70
P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel;video, urn:7:3gpp-service.ims.icsi.oma.cpm.msg, urn:urn:7:3gpp-service.ims.icsi.oma.cpm.largemsg
Accept-Contact: +g.3gpp.icsi - ref="urn%3Aurn%7;3gpp-service.ims.icsi.mmtel;video, urn%3Aurn%7%3A3gpp-service.ims.icsi.oma.cpm.msg; urn%3Aurn%7%3A3gpp-service.ims.icsi.oma.cpm.largemsg; urn%3Aurn%7%3A3gpp-

Figure 15: RCS Capability Exchange
Message 2 – 200 OK (OPTIONS)

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34

To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345

CSeq: 11 OPTIONS

Call-ID: dgh1234567@operatorA.com

P-Asserted-Identity: Tel:+447960306800

Privacy: none

P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=5678910

P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel, urn:7:3gpp-service.ims.icsi.oma.cpm.msg, urn:7:3gpp-service.ims.icsi.oma.cpm.largemsg

Accept: application/sdp, multipart/mixed, multipart/related, message/external-body, message/cpim, message/imdn+xml, application/im-iscomposing+xml, application/vnd.3gpp.sms
The above exchange indicates that:

The sending party supports the following:

Multimedia Telephony, CPM Standalone Messaging (Pager & Large Message mode), CPM Session, IM, FT, FT Thumbnail, Image Share, Video Share, Geo-location Push, Geo-Location Pull-FT, Social Presence and SMS.

The receiving party supports the following offered features:

Voice Telephony, CPM Standalone Messaging (Pager & Large Message mode), IM, FT, FT Thumbnail, Image Share, Video Share and SMS.

Subsequent sessions may be established for the common set of supported services between the two users.

RCS sessions may be run standalone or in parallel with MMTel sessions. In the latter case, the RCS sessions may involve common endpoints (“in call services”) or different endpoints (“multi-tasking”). In all cases, separate SIP sessions are set up to enable the RCS services.
B.9 RCS CPM Messaging (Pager Mode)

It is assumed that a Capability Exchange has occurred. The message flow for the RCS CPM Pager Mode Messaging is shown in figure 16. Pager mode is recommended to be used if the message size is <1300 bytes.

![Figure 16: RCS Messaging (Pager Mode)](image)

The message contents are shown below:

**Message 1 – MESSAGE**

```
INVITE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 23 MESSAGE
Call-ID: dgh1234567@operatorA.com
P-Asserted-Identity: Tel:+397850316900
Privacy: Id
Max-Forwards: 70
Conversation-ID: 1234-5678-9abcd
Contribution-ID: 0012-3456-123
P-Early-Media: supported
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.msg
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-
                      service.ims.icsi.oma.cpm.message"
Route: sip:10.10.0.1:5060;
Contact: <sip:10.0.0.1:5060>;+g.3gpp.icsi – ref ="urn%3Aurn-7%3A3gpp-
                      service.ims.icsi.oma.com.message"
```
Content-Type: message/CPIM

Content-Length: 338

From: Fred <sip:+397850316900@operatorA.com;user=phone>
To: Bob <sip:+447960306800@operatorB.com;user=phone>
DateTime: 2000-12-13T13:40:00-08:00
Subject: What are you doing tonight?
Content-Type: text/xml; charset=utf8
Content-ID:<1234567789>

<body>
Fancy going to the pub tonight? I think there may be a quiz on.
</body>

Message 2 – 200 OK (MESSAGE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 23 MESSAGE
Call-ID: dgh1234567@operatorA.com
P-Asserted-Identity: Tel:+447960306800
Privacy: none
Content-Length: 0
B.10 RCS CPM Messaging (Large Message Mode)

It is assumed that a Capability Exchange has occurred. The message flow and numbering for the RCS CPM Messaging for Large Message Mode is shown in figure 17. A SIP session is set up with the SDP Offer/Answer enabling a TCP connection to be established to carry MSRP.

![Figure 17: CPM Message (Large Message Mode)](image)

The message contents are shown below:

**Message 1 – INVITE (SDP Offer)**

```
INVITE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=z9hG4bK1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Allow: INVITE,PRACK,ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY
P-Asserted-Identity: Tel:+397850316900
P-Access-Network-Info:3GPP-E-UTRAN-FDD:e-utran-cell-id-3gpp=1234
Privacy: Id
Max-Forwards:70
P-Early-Media: supported
```

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Conversation-ID: 1234-5678-9abcd

Supported: 100rel, timer, precondition, histinfo, from-change

Session-Expires: 180; refresher=uac

Min-SE: 90

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.largemsg

Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"

Route: sip:10.10.0.1:5060;lr

Content-Type: application/sdp

Content-Length: 244

v=0

s=""

o=0 0 IN IP4 10.0.1.1

t=0 0

c=IN IP4 10.0.1.1

m=message 4000 TCP/MSRP *

a=accept-types:message/cpim

a=accept-wrapped-types:*

a=path:msrp://10.0.1.1:4000/jshA7weztas;tcp

a=sendonly

a=file-selector:size:1500

a=setup:active

Message 2 – 100 Trying

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34

To: <sip:+447960306800@operatorB.com;user=phone>

CSeq: 1 INVITE

Call-ID: dgh1234567@operatorA.com

Content-Length: 0

V6.0
Message 3 – 200 OK (INVITE) (SDP Answer)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=z9hG4bK1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorA.com
Allow: INVITE, PRACK, ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY
P-Asserted-Identity: Tel:+447960306800
Privacy: none
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id=3gpp=5678910
Supported: 100rel, timer, precondition, histinfo, from-change
Session-Expires: 150;refresher=uac
Require: timer
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.largemsg
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"
Contact: <sip:10.10.0.1:5060>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"
Content-Type: application/sdp
Content-Length: 244
v=0
s=
o-- 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp
a=recvonly
a=setup:passive
Message 4 – ACK

ACK sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 2 ACK
Call-ID: dgh1234567@operatorA.com
Max-Forwards:70
Route: sip:10.0.1.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length:0

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]).

Message 5 – MSRP SEND

MSRP d93kswow SEND
To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-1766/*byte range includes the CPIM message contents*/
Content-Type: message/cpim

From: Fred <sip+397850316900@operatorA.com;user=phone>
To: Bob <sip:+447960306800@operatorB.com;user=phone>
DateTime: 2000-12-13T13:40:00-08:00
Subject: What are you doing tonight?
Content-Type:text/xml; charset=utf8
Content-ID:<1234567789>
<body>
The message goes in here.
Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp

Message 7 – BYE
BYE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 3 BYE
Call-ID: dgh1234567@operatorA.com
Max-Forwards:70
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length:0

Message 8 – 200 OK (BYE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+397850316900@operatorA.com;user=phone>;tag=5678ab34
To: <sip:+447960306800@operatorB.com;user=phone>;tag=ade2345
CSeq: 3 BYE
Call-ID: dgh1234567@operatorA.com
Content-Length: 0
B.11 RCS Image Share (IS)

It is assumed that a Capability Exchange has occurred. The message flow and numbering for the RCS Image Share (IS) is identical to that of annex B.10. The main difference is in the feature tags that are exchanged and the contents of the SDP exchange and MSRP flow. These differences will be highlighted in this annex with references made to messages in annex B.10.

Message 1 – INVITE (SDP Offer)

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.iari-ref="urn:urn-7:3gpp-application.ims.iari.gsma-is"
Contact: <sip:10.0.0.1:5060>; +g.3gpp.iari-ref="urn:urn-7:3gpp-application.ims.iari.gsma-is"
Route: sip:10.10.0.1:5060;lr
Content-Type: application/sdp
Content-Length: 316

v=0
s=
O= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types: *
a=path:msrp://10.0.1.1:4000/jshA7wetras;tcp
a=sendonly
a=setup:active
a=file-selector:name:"My cool picture.jpg" type:image/jpeg size:32349
a=file-transfer-id:vBnG916bdberum2IF
a=file-disposition:render

Message 2 – 100 Trying

As annex B.10.

Message 3 – 200 OK (INVITE) (SDP Answer)
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.iari-ref="urn:urn-7:3gpp-application.ims.iari.gsma-is"
Contact: <sip:10.10.0.1:5060>;+g.3gpp.iari-ref="urn:urn-7:3gpp-application.ims.iari.gsma-is" Content-Type: application/sdp

Content-Length: 244

v=0
s=
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:*
a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp
a=recvonly
a=setup:passive
a=file-selector:name:"My cool picture.jpg" type:image/jpeg size:32349
a=file-transfer-id:vBnG916bdberum2fF
a=file-disposition:render

Message 4 – ACK
As annex B.10.

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58])

Message 5 – MSRP SEND
MSRP d93kswow SEND
To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7wezetas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-2048/32349
Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
------d93kswow+

In this case, there will be multiple MSRP chunks (assumed 16 – each with the same Message-ID) until the complete image is transferred.

Message 7 – BYE
As annex B.10.

Message 8 – 200 OK (BYE)
As annex B.10.
B.12 RCS Video Share (VS)

It is assumed that a Capability Exchange has occurred. In this example, it is assumed that the users do not support multi-media telephony but do support RCS VS (note that if the users both supported multi-media telephony, then an MMTel uni-directional session would have been established).

The message flow and numbering is identical to that of annex B.2. The main difference is in the SDP exchange where a video codec is negotiated together with a uni-directional media path. In addition, the will reflect that video telephony is also possible. These differences will be highlighted in this -annex with references made to messages in annex B.2.

**Message 1 – INVITE (SDP Offer #1)**

As annex B.2 with modified P-Asserted-Service, Contact & Accept-Contact headers plus a single media line (for video) in the SDP message body.

```
...........
P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.iari-ref="urn:urn:7:3gpp-application.ims.iari.gsma-vs"
Contact: <sip:10.0.1.0:5060>; +g.3gpp.iari-ref="urn:urn:7:3gpp-application.ims.iari.gsma-vs"
Content-Type: application/sdp
Content-Length: 473
v=0
s=-
o-- 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=video 4000 RTP/AVP 101 102
a=inactive
a=rtpmap: 101 H264/90000
a=fmtp: 101 profile-level-id=42C016; packetization-mode=1; sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g== octet-align=1
a=rtpmap: 102 H263-2000/90000
a=fmtp 102 profile=0;level=10;QCIIF=2
a=curr qos local none
a=des qos mandatory local sendrecv
```
a=curr qos remote none
a=des qos optional remote sendrecv
a=conf qos remote sendrecv
b=AS: 768

Message 2 – 100 Trying
As annex B.2.

Message 3 – 183 Progress (SDP Answer #1)
As annex B.2 with modified headers as in the INVITE plus video SDP in the message body.

----------
P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.mmtel
Accept-Contact: : +g.3gpp.iari-ref="urn:urn:7:3gpp-application.ims.iari.gsma-vs"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.iari-ref="urn:urn:7:3gpp-application.ims.iari.gsma-vs"
Content-Type: application/sdp
Content-Length: 403
v=0
s=-
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=video 6000 RTP/AVP 101
a=inactive
a=rtpmap: 101 H264/90000
a=fmtp: 101 profile-level-id=42C016; packetization-mode=1;sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g== octet-align=1
a=curr qos local none
a=des qos mandatory local sendrecv
a=curr qos remote none
a=des qos optional remote sendrecv
Message 4 – PRACK
As annex B.2.

Message 5 – 200 OK (PRACK)
As annex B.2.

Message 6 – UPDATE (SDP Offer #2)
As annex B.2 with only video media SDP in the message body and the media being uni-directional.

Content-Type: application/sdp
Content-Length: 405

v=0
s=
o= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=video 4000 RTP/AVP 101
a=sendonly
a=rtpmap: 101 H264/90000
a=fmt: 101 profile-level-id=42C016; packetization-mode=1; sprop-parameter-sets=Z0LAFukDwKMg,aM4G4g== octet-align=1
a=curr qos local sendrecv
a=des qos mandatory local sendrecv
a=curr qos remote none
a=des qos optional remote sendrecv
a=conf qos remote sendrecv
b=AS: 768
Message 7 – 200 OK (UPDATE) (SDP Answer #2)

As annex B.2 with only single media video SDP in the message body. Also, the SDP answer reflects
the unidirectional media flow.

```
Content-Type: application/sdp
Content-Length: 382

v=0
s=-
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=video 6000 RTP/AVP 101
a=recvonly
a=rtpmap: 101 H264/90000
a=fmtp: 101 profile-level-id=42C016; packetization-mode=1;sprop-parameters=Z0LAFukDwKMg,aM4G4g== octet-align=1
a=curr qos local sendrecv
a=des qos mandatory local sendrecv
a=curr qos remote sendrecv
a=des qos optional remote sendrecv
b=AS: 768
```

Message 8 – 180 Ringing

As annex B.2. with modified P-Asserted-Service, Contact & Accept-Contact headers.

```
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.iari-ref="urn:urn-7:3gpp-application.ims.iari.gsma-vs"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.iari-ref="urn:urn-7:3gpp-application.ims.iari.gsma-vs"
```

Message 9 – 200 OK (INVITE)
As annex B.2. with modified P-Asserted-Service, Contact & Accept-Contact headers.

P-Asserted-Service: urn:7:3gpp-service.ims.icsi.mmtel
Accept-Contact: +g.3gpp.iari-ref="urn:7:3gpp-application.ims.iari.gsma-vs"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.iari-ref="urn:7:3gpp-application.ims.iari.gsma-vs"

Message 10 – ACK
As annex B.2.

Message 11 – BYE
As annex B.2.

Message 12 – 200 OK (BYE)
As annex B.2.
B.13 RCS FT (CPM Based)

It is assumed that a Capability Exchange has occurred. The tags "urn:urn-7:3gpp-application.ims.iani.rcse.ft" and "urn:urn-7:3gpp-application.ims.iani.rcs.ftthumb" indicate that either CPM or SIMPLE based FT is supported. This annex covers the CPM case. This example also shows a thumbnail being transferred ahead of the FT itself and the use of the multipart/related message body. The message flow and numbering for the RCS CPM based FT is identical to annex B.10. The main difference is in the feature tags that are exchanged and the contents of the SDP exchange and MSRP flow. These differences will be highlighted in this annex with references made to messages in annex B.10.

The message contents are shown below:

Message 1 – INVITE (SDP Offer)

```
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.filetransfer
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.filetransfer"
Contact: <sip:10.0.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.filetransfer"
Content-Type: multipart/related; type="application/sdp"; boundary=boundary123
Content-Length: 792
---boundary123
Content-Type: application/sdp
Content-Length: 374
v=0
s=-
o= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.0.1.1:4000/jshA7weztas;tcp
a=sendonly
a=setup:active
```

V6.0
Message 2 – 100 Trying
As annex B.10

Message 3 – 200 OK (INVITE) (SDP Answer)

P-Asserted-Service: urn:urn:7:3gpp-service.ims.icsi.oma.cpm.filetransfer
Accept-Contact: <g.3gpp.icsi> "urn%3Aurn%7%3A3gpp-service.ims.icsi.oma.cpm.filetransfer"
Contact: <sip:10.10.0.1:5060>; <g.3gpp.icsi> "urn%3Aurn%7%3A3gpp-service.ims.icsi.oma.cpm.filetransfer"
Content-Type: application/sdp
Content-Length: 322

v=0
s--
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
Message 4 – ACK

As annex B.10

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]). Note that it is assumed that the maximum chunk size is 500K bytes (as recommended in GSMA PRD IR.90 [5]). In this case, multiple MSRP messages are exchanged until the file transfer is complete. The CPIM header shall appear only in the first chunk that is sent.

Message 5 – MSRP SEND

MSRP d93kswow SEND

To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-50000/60220
Content-Type: message/cpim
From: Fred <sip:+397850316900@operatorA.com;user=phone>
To: Bob <sip:+447960306800@operatorB.com;user=phone>
DateTime: 2000-12-13T13:40:00-08:00
Subject: Sunrise Picture

Content-Type: image/jpeg
Content-ID: <12345>
(jpeg contents – first block)

--------d93kswow--
Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
--------d93kswow+
One more chunk would now be sent with the same MSRP Message-ID and including the rest of the JPEG file (and no CPIM headers). However, the MSRP Content-Type would still be set to message/cpim.

Message 7 – BYE
As annex B.10.

Message 8 – 200 OK (BYE)
As annex B.10.
B.14 RCS FT (SIMPLE Based)

It is assumed that a Capability Exchange has occurred. The tags "urn:urn-7:3gpp
application.ims.iari.rcse.ft" and urn:urn-7:3gpp-application.ims.iari.rcs.ftthumb" indicate that
either CPM or SIMPLE based FT is supported. This annex covers the SIMPLE case. This
example also shows a thumbnail being transferred ahead of the FT itself and the use of
the multipart/related message body. The message flow and numbering for the RCS SIMPLE
based FT is identical to annex B.10. The main difference is in the feature tags that are
exchanged and the contents of the SDP exchange and MSRP flow. These differences will be
highlighted in this annex with references made to messages in annex B.10.

The message contents are shown below:

Message 1 – INVITE (SDP Offer)

The following headers are absent: Contribution-ID, Conversation-ID & P-Asserted-Service.

........

Accept-Contact: +g.oma.sip-im

Contact: <sip:10.0.0.1:5060>;+g.oma.sip-im

Content-Type: multipart/related; type="application/sdp";boundary=boundary123

Content-Length: 766

--boundary123

Content-Type: application/sdp

Content-Length: 343

v=0

s=--
o= 0 0 IN IP4 10.0.1.1

t=0 0

c=IN IP4 10.0.1.1

m=message 4000 TCP/MSRP *

a=accept-types:image/jpeg

a=path:msrp://10.0.1.1:4000/jshA7wezas;tcp

a=sendonly

a=setup:active

a=file-selector:name:"sunrise.jpg" type:image/jpeg size:60000

a=file-transfer-id:vBnG916bdberum2fF

V6.0
Message 2 – 100 Trying

As annex B.10

Message 3 – 200 OK (INVITE) (SDP Answer)

The P-Asserted-Service header is absent.

Accept-Contact: +g.oma.sip-im
Contact: <sip:10.10.0.1:5060>;+g.oma.sip-im
Content-Type: application/sdp
Content-Length: 298

v=0
s=-
as=0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:image/jpeg
a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp
a=recvonly
a=setup:passive
a=file-selector:name:"sunrise.jpg" type:image/jpeg size:60000
a=file-transfer-id:vBnG916bdberum2fF

Message 4 – ACK

As annex B.10.

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]) and multiple MSRP messages are exchanged the file transfer is complete (i.e. assuming the maximum chunk size of 50K bytes).

Message 5 – MSRP SEND

MSRP d93kswow SEND
To-Path: msrp://10.10.1.1:60000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-50000/60000
Content-Type: image/jpeg
(jpeg contents)
-------d93kswow+

Message 6 – MSRP 200 OK

MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:60000/abcA7wept654;tcp
-------d93kswow+

One more chunk would be sent with the same MSRP Message-ID and including the rest of the JPEG file.

Message 7 – BYE

As annex B.10.
Message 8 – 200 OK (BYE)

As annex B.10.
B.15 RCS 1-To-1 Chat (CPM Based)

It is assumed that a Capability Exchange has occurred. The tag "urn:urn-7:3gpp-application.ims.iasi.rcse.im" indicates that either CPM or SIMPLE based IM/Chat is supported. This annex covers the CPM case. The message flow and numbering for the RCS CPM based Chat is identical to annex B.13. The main difference is in the feature tags that are exchanged and the contents of the SDP exchange and MSRP flow. These differences will be highlighted in this sub-annex with references made to messages in annex B.13.

The message contents are shown below:

Message 1 – INVITE (SDP Offer)

```
........
P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.session
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.session"
Contact: <sip:10.0.0.1:5060>; +g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.oma.cpm.session"
Content-Type: application/sdp
Content-Length: 217
v=0
s=
O= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.0.1.1:4000/jshA7weztas;tcp
a=sendrecv
a=setup:active
```

Message 2 – 100 Trying

As annex B.13.
Message 3 – 200 OK (INVITE) (SDP Answer)

.......... 

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.session
Accept-Contact: +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.session"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.session"
Content-Type: application/sdp
Content-Length: 244

v=0
s=
o= 0 0 IN IP4 10.10.1.1
i=0 0

m=message 6000 TCP/MSRP *

a=accept-types:message/cpim
a=accept-wrapped-types: *

a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp
a=sendrecv
a=setup:passive

Message 4 – ACK

As annex B.13.

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]). Since this is IM/Chat, each chunk shall contain a CPIM header and contents.

Message 5 – MSRP SEND

MSRP d93kswow SEND

To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-276
Content-Type: message/cpim
From: Fred <sip:+397850316900@operatorA.com;user=phone>
To: Bob <sip:+447960306800@operatorB.com;user=phone>
DateTime: 2000-12-13T13:40:00-08:00
Subject: How are you?

Content-Type: text/plain; charset=utf8
Content-ID: <56789>
How are you? Haven’t seen you for a while.
-------d93kswow+

Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
-------d93kswow+

Message 7 – BYE
As annex B.13.

Message 8 – 200 OK (BYE)
As annex B.13.
B.16 RCS 1-To-1 Chat (SIMPLE Based)

It is assumed that a Capability Exchange has occurred. The tag "urn:urn-7:3gpp-application.ims.iari.rcse.im" indicates that either CPM or SIMPLE based IM/Chat is supported. This annex covers the SIMPLE case. The message flow and numbering for the RCS CPM based Chat is identical to annex B.14. The main difference is in the feature tags that are exchanged and the contents of the SDP exchange and MSRP flow. These differences will be highlighted in this sub-annex with references made to messages in annex B.14.

The message contents are shown below:

**Message 1 – INVITE (SDP Offer)**

```
........
Accept-Contact: +g.oma.sip-im
Contact: <sip:10.0.0.1:5060>;+g.oma.sip-im
Content-Type: application/sdp
Content-Length: 179
v=0
s=
o= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types:* 
a=path:msrp://10.0.1.1:4000/jshA7weztas:tcp
a=sendrecv
a=setup:active
```

**Message 2 – 100 Trying**

As annex B.14.

**Message 3 – 200 OK (INVITE) (SDP Answer)**

V6.0
Accept-Contact: +g.oma.sip-im
Contact: <sip:10.10.0.1:5060>;+g.oma.sip-im
Content-Type: application/sdp
Content-Length: 184
v=0
s=
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:*
a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp
a=sendrecv
a=setup:passive

Message 4 – ACK
As annex B.14.

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]) and MSRP messages are exchanged to carry the chat session contents.

Message 5 – MSRP SEND
MSRP d93kswow SEND
To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7wez tas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-43
Content-Type: text/plain; charset=utf8
How are you? Haven’t seen you for a while.
Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
-------d93kswow+

Message 7 – BYE
As annex B.14.

Message 8 – 200 OK (BYE)
As annex B.14.
B.17 RCS Geolocation Push (CPM Based)

It is assumed that a Capability Exchange has occurred. The tag "urn:urn-7:3gpp-application.ims.iari.rcse.geopush" indicates that Geolocation information may be transferred via FT which can be CPM or SIMPLE based. In addition, the feature tag for FT must also have been exchanged (see annex B.13). This annex covers the CPM case. The message flow and numbering for the RCS CPM based Geolocation Push is identical to annex B.13. The main difference is in the feature tags that are exchanged and the contents of the SDP exchange and MSRP flow. These differences will be highlighted in this sub-annex with references made to messages in annex B.13. The corresponding SIMPLE based Geolocation push flow may be derived by applying similar changes to the message contents in annex B.14.

The message contents are shown below:

Message 1 – INVITE (SDP Offer)

```
Accept-Contact: +g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-service.ims.iari.rcs.geopush"
Content-Type: application/sdp
Content-Length: 347
v=0
s=
 o= 0 0 IN IP4 10.0.1.1
 t=0 0
 c=IN IP4 10.0.1.1
 m=message 4000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.0.1.1:4000/jshA7weztas;tcp
a=sendonly
a=setup:active
a=file-selector: type:application/rcspushlocation+xml size:500
a=file-transfer-id:vBnG916bdberum2fF
a=file-disposition:render
```
Message 2 – 100 Trying
As annex B.13.

Message 3 – 200 OK (INVITE) (SDP Answer)

Accept-Contact: +g.3gpp.1ari-ref="urn%3Aurn-7%3Agpp-service.ims.1ari.rcs.geopush"
Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi - ref="urn%3Aurn-7%3Agpp-service.ims.1ari.rcs.geopush "
Content-Type: application/sdp
Content-Length: 324

v=0
s=-
ao= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp
a=recvonly
a=setup:passive
a=file-selector: type:application/rcspushlocation+xml size:500
a=file-transfer-id: vBnG916bdberum2fF

Message 4 – ACK
As annex B.13.

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]).
Message 5 – MSRP SEND
MSRP d93kswow SEND
To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-749
Content-Type: message/cpim
From: Fred <sip:+397850316900@operatorA.com;user=phone>
To: Bob <sip:+447960306800@operatorB.com;user=phone>
DateTime: 2000-12-13T13:40:00
Subject: My Location Information
Content-Type: application/rcspushlocation+xml
Content-ID: <56789>
(location XML block – 500 octets)

Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp

Message 7 – BYE
As annex B.13.

Message 8 – 200 OK (BYE)
As annex B.13.
B.18 RCS Geolocation Pull (CPM Based)

It is assumed that a Capability Exchange has occurred. The tag "urn:urn-7:3gpp-application.ims.iari.rcse.geopullft" indicates that Geolocation information may be transferred via FT which can be CPM or SIMPLE based. In addition, the feature tag for FT must also have been exchanged (see annex B.13). This annex covers the CPM case. The message flow and numbering for the RCS CPM based Geolocation Pull is identical to annex B.13. The main difference is in the feature tags that are exchanged and the contents of the SDP exchange and MSRP flow. These differences will be highlighted in this sub-annex with references made to messages in annex B.13. The corresponding SIMPLE based Geolocation pull flow may be derived by applying similar changes to the message contents in annex B.14.

The message contents are shown below:

Message 1 – INVITE (SDP Offer)

.......... 
Accept-Contact: +g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-service.ims.iari.rcs.geopullft"
Contact: <sip:10.0.0.1:5060>;+g.3gpp.iari - ref="urn%3Aurn-7%3A3gpp-service.ims.iari.rcs.geopullft"
Content-Type: application/sdp
Content-Length: 342
v=0
s=-
o= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.0.1.1:4000/jshA7weztas:tcp
a=recvonly
a=setup:active
a=file-selector: type:application/rcpushlocation+xml

Message 2 – 100 Trying

As annex B.13.
Message 3 – 200 OK (INVITE) (SDP Answer)

        Accept-Contact: +g.3gpp.ari-ref="urn%3Aurn-7%3Agpp-service.ims.ari.rcs.geopullft"
Contact: <sip:10.10.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3Agpp-service.ims.ari.rcs.geopullft "
Content-Type: application/sdp
Content-Length: 325

v=0
s=
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:message/cpim
a=accept-wrapped-types: *
a=path:msrp://10.10.1:6000/abcA7wept654;tcp
a=sendonly
a=setup:passive
a=file-selector: type:application/rcspushlocation+xml size: 520
a=file-transfer-id:vBnG916bdberum2fF
a=file-disposition:render

Message 4 – ACK
As annex B.13.

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]).

Message 5 – MSRP SEND
MSRP d93kswow SEND
To-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
From-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
Message-ID: 12339sdqwer
Byte-Range: 1-779
Content-Type: message/cpim
From: Bob <sip:+447960306800@operatorB.com;user=phone>
To: Fred <sip:+397850316900@operatorA.com;user=phone>
DateTime: 2000-12-13T13:40:00-08:00
Subject: My Requested Location Information
Content-Type: application/rcspushlocation+xml
Content-ID: <56789>
(location XML block – 520 octets)
-------d93kswow+

Message 6 – MSRP 200 OK
MSRP d93kswow 200 OK
To-Path: msrp://10.10.1.1:6000/abcA7wept654;tcp
From-Path: msrp://10.0.1.1:4000/jshA7weztas;tcp
-------d93kswow+

Message 7 – BYE
As annex B.13.

Message 8 – 200 OK (BYE)
As annex B.13.
B.19 Social Presence (Overview)

The RCS Social Presence service consists of a number of discrete parts. To aid understanding, this annex provides a brief overview of those constituent parts prior to each of the parts being described subsequently.

The service consists of the following:

- A Capability Exchange between users to indicate mutual support of Social Presence (see annex B.8),
- The publication of a Presence Information Document onto the Presence Server by the Presentity (see annex B.20),
- The subscription of a Watcher to be notified of the Presence Information of the Presentity (see annex B.21),
- The subscription by the Presentity to be notified of the list of Watchers associated with the Presentity’s Presence Document (see annex B.22) and to be able to verify that new watchers may be added to the list.
B.20 Publication of Social Presence Information.

This message exchange occurs across the roaming NNI only. It is assumed that the user is roaming and has previously registered and is an RCS user subscribing to the Social Presence service. The user uploads his Presence Information to the Presence Server (located in the home network). The message flow is shown in figure 18.

![Figure 18: Publication of Presence Information](image)

The message contents are shown below:

**Message 1 – PUBLISH**

PUBLISH sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 PUBLISH
Call-ID: dgh1234567@operatorB.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
P-Asserted-Identity: sip:+447960306800@operatorB.com;user=phone
Event: presence
Expires: 3600
Max-Forwards:70
P-Charging-Vector: icid-value="abc+236788a"; orig-iei=operatorA.com; term-iei=operatorB.com
Route: sip:10.10.0.1:5060;lr
Content-Type: application/pidf+xml
Content-Length: xx

[Presence XML Document] – see IETF RFC 3863 [56].
Message 2 – 200 OK (PUBLISH)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>;tag=34578231
CSeq: 1 PUBLISH
Call-ID: dgh1234567@operatorB.com
SIP-Etag: dx400345w
Expires: 3000
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
P-Asserted-Identity: sip:+447960306800@operatorB.com;user=phone
P-Charging-Vector: icid-value="abch+236788a"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Content-Length: 0
B.21 Subscription to Social Presence Information.

This interchange allows a user (the Watcher) to subscribe to and (post authorization) receive the Presence Information of the target Presentity. The message flow is shown in figure 19. In this flow, the new Watcher firstly receives a 202 Accepted response followed by a dummy notification. Following authorization via the Presentity (not shown below – see annex B.22), a second notification is sent.

![Message Flow Diagram]

**Figure 19 : Subscription to Presence Information**

The message contents are shown below:

**Message 1 – SUBSCRIBE**

```
SUBSCRIBE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+3977850316900@operatorA@operatorA.com;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 SUBSCRIBE
Call-ID: dgh1234567@operatorA.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
P-Asserted-Identity: sip:+3977850316900@operatorA.com;user=phone
Event: presence
Accept: application/pidf+xml
```
.expires: 3600
max-forwards: 70
p-charging-vector: icid-value="abch+236788a"; orig-oi=operatorA.com; term-oi=operatorB.com
route: sip:10.10.0.1:5060;lr
contact: <sip:10.0.0.1:5060>
content-length: 0

message 2 – 202 accepted (subscribe)
sip/2.0 202 accepted
via: sip/2.0/udp 10.0.0.1:5060;branch=12345678778
from: <sip:+397850316900@operatorA.com;user=phone>;tag=56ad7111
to: <sip:+447960306800@operatorB.com;user=phone>;tag=3467894a
p-charging-vector: icid-value="abch+236788a"; orig-oi=operatorA.com; term-oi=operatorB.com
call-id: dgh1234567@operatorA.com
cseq: 1 subscribe
event: presence
expires: 3600
contact: <sip:10.10.0.1:5060>
content-length: 0

message 3 – notify
notify sip:+397850316900@operatorA.com;user=phone sip/2.0
via: sip/2.0/udp 10.10.0.1:5060;branch=567891234a
to: <sip:+397850316900@operatorA.com;user=phone>;tag=56ad7111
from: <sip:+447960306800@operatorB.com;user=phone>;tag=3467894a
call-id: abc1234588@operatorB.com
cseq: 12 notify
max-forwards: 70
subscription-state: pending; expires=3600
event: presence
Accept: application/pidf+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Length: 0

**Message 4 – 200 OK (NOTIFY)**

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+397850316900@operatorA @operatorA.com;user=phone>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=3467894a
Call-ID: abc1234588@operatorB.com
CSeq: 12 NOTIFY
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

**Message 5 – NOTIFY**

NOTIFY sip:+397850316900@operatorA.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+397850316900@operatorA @operatorA.com;user=phone>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=3467894a
Call-ID: abc1234588@operatorB.com
CSeq: 13 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=3600
Event: presence
Accept: application/pidf+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
[Presence Information document] – see IETF RFC 3863 [56].

Message 6 – 200 OK (NOTIFY)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+397850316900@operatorA @operatorA.com;user=phone>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=3467894a
Call-ID: abc1234588@operatorB.com
CSeq: 13 NOTIFY
Contact: <sip:10.0.0.1:5060>
Content-Length: 0
B.22 Subscription to Social Presence Watcher Information

This interchange allows the Presentity to subscribe to and receive the information relating to the Watchers of the Presentity’s Presence Information. This enables the Presentity to be informed of newly arrived Watchers and authorize them to be permitted to receive the Presence Information. The message flow is shown in figure 20. This message interchange occurs across the roaming NNI only. It is assumed that the Presentity is roaming and has previously registered, is an RCS user subscribing to the Social Presence service and has also uploaded his Presence Information. In this flow, the Presentity Watcher firstly receives a 200 OK response followed by a notification of existing watchers. Following the arrival of a new Watcher (not shown below), a second notification is sent requesting authorization of the new Watcher to receive the Presence Information. The authorization (or otherwise) is enabled by the uploading of a new Presence Document as in annex B.20 (and is not shown below).

![Figure 20: Subscription to Presence Watcher Information](image)

The message contents are shown below:

**Message 1 – SUBSCRIBE**

```
SUBSCRIBE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+447960306800@operatorB.com;user=phone>;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 SUBSCRIBE
Call-ID: dgh1234567@operatorB.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
P-Asserted-Identity: sip:+447960306800@operatorB.com;user=phone
```

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Message 2 – 200 OK (SUBSCRIBE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: <sip:+447960306800@operatorB.com;user=phone>;user=phone>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>; tag=123456
CSeq: 1 SUBSCRIBE
Call-ID: dgh1234567@operatorB.com
P-Charging-Vector: icid-value="abch+236788a"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Event:presence.winfo
Expires: 3600
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

Message 3 – NOTIFY
NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: <sip:+447960306800@operatorB.com;user=phone>;user=phone>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=123456
Call-ID: abc1234588@operatorB.com
CSeq: 12 NOTIFY
A new Watcher now subscribes to the Presence Information of the Presentity and a further notification is sent to the Presentity containing the identity of the new Watcher.

**Message 5 – NOTIFY**

NOTIFY sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=56789123479
To: <sip:+447960306800@operatorB.com;user=phone>;user=phone>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=123456
Call-ID: abc1234588@operatorB.com
CSeq: 12 NOTIFY
Contact: <sip:10.0.0.1:5060>
Content-Length: 0
CSeq: 13 NOTIFY
Max-Forwards: 70
Subscription-State: Active;Expires=3600
Event: presence.winfo
Accept: application/watcherinfo+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Type: application/watcherinfo+xml
Content-Length: xx

[Watcher Info] – see IETF RFC 3858 [55].

Message 6 – 200 OK (NOTIFY)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234789
To: <sip:+447960306800@operatorB.com;user=phone>;user=phone>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=123456
Call-ID: abc1234588@operatorB.com
CSeq: 13 NOTIFY
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

On being informed of the new Watcher, the Presentity may authorize or refuse the new Watcher access to the Presence Information. This is done via a new PUBLISH message (not shown). In addition, if the new Watcher is authorised, then a further NOTIFY message is sent to reflect that fact that the status of the new Watcher has changed from “pending” to “active” (see IETF RFC 3857 [54]). This further NOTIFY is also not shown.
B.23 Capability Discovery by Presence

It is also possible to perform Capability Discovery as a bi-laterally agreed option via Social Presence. This mechanism firstly requires a Capability Exchange via OPTIONS (see annex B.8) and the exchange of the following tag:

- +g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-application.ims.iari.rcse.dp"

Thereafter, the set of RCS Services are exchanged via Social Presence using an anonymous SUBSCRIBE message. The message flow is shown in figure 21. The service tags are included in the message body of the NOTIFY message. The correspondence between OPTIONS service tags and Presence service tags is documented in clause 2.6.1.3.1 of the GSMA RCS v.5.1 Client & Services specification ([44]).

The message contents are shown below:

**Message 1 – SUBSCRIBE**

SUBSCRIBE sip:+447960306800@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: Anonymous <sip: Anonymous@anonymous.invalid>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>
CSeq: 1 SUBSCRIBE
Call-ID: dgh1234567@operatorA.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD:e-utran-cell-id-3gpp=1234
Event: presence
Accept: application/pidf+xml
Expires: 0
Max-Forwards: 70
P-Charging-Vector: icid-value="abch+236788a"; orig-ori=operatorA.com; term-ori=operatorB.com
Route: sip:10.10.0.1:5060;
Contact: <sip:10.0.0.1:5060>
Content-Length: 0

Message 2 – 200 OK (SUBSCRIBE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778
From: Anonymous <sip: Anonymous@anonymous.invalid>;tag=56ad7111
To: <sip:+447960306800@operatorB.com;user=phone>; tag=4567hj8
CSeq: 1 SUBSCRIBE
Call-ID: dgh1234567@operatorA.com
P-Charging-Vector: icid-value="abch+236788a"; orig-ori=operatorA.com; term-ori=operatorB.com
Event: presence
Expires: 0
Contact: <sip:10.10.0.1:5060>
Content-Length: 0

Message 3 – NOTIFY
NOTIFY sip: Anonymous@anonymous.invalid SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: Anonymous <sip: Anonymous@anonymous.invalid>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=4567hj8
CSeq: 12 NOTIFY
Call-ID: abc1234588@operatorB.com
Max-Forwards: 70
Subscription-State: Active;Expires=0
Event: presence
Accept: application/pidf+xml
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Type: application/pidf+xml
Content-Length: xx

[Presence Service Info] – see IETF RFCs 3856 [53] & 4479 [57].

Message 4 – 200 OK (NOTIFY)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=567891234a
To: Anonymous <sip: Anonymous@anonymous.invalid>;tag=56ad7111
From: <sip:+447960306800@operatorB.com;user=phone>; tag=4567hj8
CSeq: 12 NOTIFY
Call-ID: abc1234588@operatorB.comFromntact: <sip:10.0.0.1:5060>
Content-Length: 0
B.24 RCS Group Chat (from start of session)

This annex shows the Group Chat service when it is created at the start of a session. A Group Chat (as 1:1 Chat) may be CPM based or SIMPLE based (see appendices 13 & 14). In this example, the latter is shown. Group Chat (from session start) is applicable only to the roaming NNI (i.e. UE in visited network and Chat Conference Factory in the home network). It is assumed that the initiating UE is roaming and registered and has performed a Capability Exchange with the other users in the group chat session (see annex B.15). In this example, the UE initiating the Group Chat session sends an INVITE to the Conference Factory which contains a multi-part body containing SDP and a URI-List of the other users in the group chat session (see IETF RFCs 5364 [59] & 5366 [60]). The message flow is shown in figure 22.

![Figure 22: Group Chat Establishment](image)

The message contents are shown below:

Message 1 – INVITE (SDP Offer, URI-List)

```plaintext
INVITE sip:Chat-Factory2@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@opertaorB.com>
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorB.com
Allow: INVITE,PRACK,ACK,CANCEL,UPDATE,PUBLISH,OPTIONS,MESSAGE,BYE,REFER,SUBSCRIBE,NOTIFY
P-Asserted-Identity: Tel:+447960306800
Privacy: none
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
```

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Max-Forwards: 70

P-Early-Media: supported

Supported: 100rel, timer, precondition, histinfo, from-change

Session-Expires: 180; refresher=uac

Min-SE: 90

Accept-Contact: +g.oma.sip-im

Contact: <sip:10.0.0.1:5060>; +g.oma.sip-im

Route: sip:10.10.0.1:5060;lr

Allow-Events: dialog

Accept: application/sdp, message/sipfrag

Require: recipient-list-invite

Content-Type: multipart/mixed; boundary="boundary1"

Content-Length: 890

--boundary1

Content-Type: application/sdp

v=0

s=-

o=- 0 0 IN IP4 10.0.1.1

t=0 0

c=IN IP4 10.0.1.1

m=message 4000 TCP/MSRP *

a=accept-types: *

a=path: mср://10.0.1.1:4000/jshA7weztas;tcp

a=sendrecv

a=setup:active

b=AS: 45

--boundary1

Content-Type: application/resource-lists+xml

Content-Disposition: recipient-list
The XML URI-List identifies 3 other parties for the group chat session, one of whom is indicated as being anonymized. This will impact on the XML blocks sent to the other users as will be seen subsequently when each of those users are invited into the group chat session by the Conference Server.

Message 2 – 100 Trying

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@operatorB.com>;tag=123456789
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorB.com
Content-Length: 0

Message 3 – 200 OK (INVITE)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@operatorB.com>;tag=123456789
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321
CSeq: 1 INVITE

Call-ID: dgh1234567@operatorB.com

Allow: INVITE, PRACK, ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY

P-Asserted-Identity: Tel:+447960306800

Privacy: none

P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=5678910

Supported: 100rel, timer, precondition, histinfo, from-change

Session-Expires: 150; refresher=uac

Require: timer

Accept-Contact: +g.oma.sip-im

Contact: <sip:10.10.0.1:5060>; +g.oma.sip-im

Content-Type: application/sdp

Content-Length: 184

v=0

s=

o= 0 0 IN IP4 10.10.1.1

t=0 0

c=IN IP4 10.10.1.1

m=message 6000 TCP/MSRP *

a=accept-types:*

a=path:msrp://10.10.1.1:6000/abcA7wept654;tcp

a=sendrecv

a=setup:passive

Message 4 – ACK

ACK sip:Chat-Factory2@operatorB.com;user=phone SIP/2.0

Via: SIP/2.0/UDP 10.0.0.1:5060; branch=1234567abcd

To: <sip:Chat-Factory2@opertaorB.com>; tag=123456789
There is now a session established between the initiating UE and Chat-Factory, and a TCP connection can be established to carry the MSRP (as shown in annex B.16) from the UE to the conference resource.

The initiating UE will also subscribe to the conference event package to be notified of conference progress/changes. This is identical to the flow shown in figure 13.

The next step is for the Conference-Factory to initiate sessions to the other 3 users into the group chat session. The message flow is shown in figure 23.

Figure 23: Inviting a Target User into a Group Chat

The message contents for the user in Operator C’s network is shown below:

Message 1 – INVITE (SDP Offer, URI-List)

INVITE sip:+331234567890@operatorC.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=5678910defg
From: <sip:Chat-Factory2@opertaorB.com>; tag=89101112
To: <sip:+331234567890@operatorC.com;user=phone>
CSeq: 10 INVITE
Call-ID: abc1234567@operatorB.com
Allow: INVITE, PRACK, ACK, CANCEL, UPDATE, PUBLISH, OPTIONS, MESSAGE, BYE, REFER, SUBSCRIBE, NOTIFY
P-Asserted-Identity: sip:Chat-Factory2@operatorB.com
P-Charging-Vector: icid-value="1adbg23456y"; orig-oi=operatorB.com
Max-Forwards: 70
P-Early-Media: supported
Supported: 100rel, timer, precondition, histinfo, from-change
Session-Expires: 180; refresher=uac
Min-SE: 90
Accept-Contact: +g.oma.sip-im
Contact: <sip:10.10.0.1:5060>; +g.oma.sip-im
Route: sip:10.20.0.1:5060;lr
Allow-Events: dialog, conference
Accept: application/sdp, message/sipfrag
Content-Type: multipart/mixed; boundary="boundary1"
Content-Length: 888
--boundary1
Content-Type: application/sdp
v=0
s=
O= 0 0 IN IP4 10.0.1.1
T= 0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types:*
a=path:msrp://10.10.1.1:4000/jshA7wez tas;tcp
a=sendrecv
a=setup:active
b=AS: 45
The XML URI-List has been modified to reflect that the content is a recipient list history plus the anonymity of the second user.

Message 2 – 100 Trying

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=5678910defg
From: <sip:Chat-Factory2@operatorB.com>; tag=89101112
To: <sip:+331234567890@operatorC.com;user=phone>;tag=45678912
CSeq: 10 INVITE
Call-ID: abc1234567@operatorB.com
Content-Length: 0

Message 3 – 200 OK (INVITE)
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=5678910defg
From: <sip:Chat-Factory2@opertaorB.com>; tag=89101112
To: <sip:+331234567890@operatorC.com;user=phone>;tag=45678912
CSeq: 10 INVITE
Call-ID: abc1234567@operatorB.com
Content-Length: 0
Allow: INVITE,PRACK,ACK,CANCEL,UPDATE,MESSAGE,BYE,REFER,SUBSCRIBE,NOTIFY
P-Asserted-Identity: Tel:+331234567890
Privacy: none
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234568909
P-Charging-Vector: icid-value="1adbg23456y"; orig-ioi=operatorB.com; term-ioi=operatorC.com
Supported: 100rel,timer,precondition,histinfo,from-change
Session-Expires: 150;refresh=0
Require:timer
Accept-Contact: +g.oma.sip-im
Contact: <sip:10.20.0.1:5060>;+g.oma.sip-im
Content-Type: application/sdp
Content-Length: 184

v=0
s=
o= 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:*
a=path:msrp://10.20.1.1:6000/abца7вепт654;tcp
a=sendrecv
a=setup:passive
Message 4 – ACK

ACK sip:+331234567890@operatorC.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.10.0.1:5060;branch=5678910defg
From: <sip:Chat-Factory2@opertaorB.com>; tag=89101112
To: <sip:+331234567890@operatorC.com;user=phone>;tag=45678912
CSeq: 10 INVITE
Call-ID: abc1234567@operatorB.com
Max-Forwards:70
Route: sip:10.20.1.1:5060;lr
Contact: <sip:10.10.0.1:5060>
Content-Length:0

Having exchanged the SDP, a TCP connection is created over which the MSRP messages will run (see IETF RFC 4975 [58]) and MSRP messages are exchanged to carry the chat session contents from the Conference Server to the target user.

At this point, the Group Chat session is established and MSRP messages are sent into the Conference Server where they are replicated and forwarded to all other users in the Group Chat session.
B.25 RCS Group Chat (from initial 1:1 Chat)

This annex shows the Group Chat service when it is created from an initial 1:1 Chat. As stated previously, the Chat may be CPM based or SIMPLE based (see appendices 13 & 14). In this example, the latter is shown. Group Chat (from initial 1:1 Chat) is applicable only to the roaming NNI (i.e. UE in visited network and Chat Conference Factory in the home network). It is assumed that the initiating UE is roaming and registered and has already set up a 1:1 with another user (see annex B.15). The UE, initiating the Group Chat session, initially sends an INVITE to the Conference Factory. Thereafter, the existing user will be moved by a REFER message onto the Group Chat and new users will be added to the Group Chat via another REFER message which contains a multi-part body containing SDP and a URI-List of the users to be added to the group chat session (see IETF RFCs 5364 [59] & 5368 [61]). The message flow to create the group chat session is shown in figure 24.

![Figure 24: Group Chat Establishment](image)

The message contents are shown below:

Message 1 – INVITE (SDP Offer, URI-List)

INVITE sip:Chat-Factory2@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@operatorB.com>
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorB.com
Allow: INVITE,PRACK,ACK,CANCEL,UPDATE,PUBLISH,OPTIONS,MESSAGE,BYE,REFER, SUBSCRIBE,NOTIFY
P-Asserted-Identity: Tel:+447960306800
Privacy: none
P-Charging-Vector: icid-value="abch+23456y"; orig-oi=operatorA.com; term-oi=operatorB.com
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
Max-Forwards: 70
P-Early-Media: supported
Supported: 100rel, timer, precondition, histinfo, from-change
Session-Expires: 180:refresher=uac
Min-SE: 90
Accept-Contact: +g.oma.sip-im
Contact: <sip:10.0.0.1:5060>; +g.oma.sip-im
Route: sip:10.10.0.1:5060;lr
Accept: application/sdp, message/sipfrag
Content-Type: application/sdp
Content-Length:

v=0
s=
O= 0 0 IN IP4 10.0.1.1
t=0 0
c=IN IP4 10.0.1.1
m=message 4000 TCP/MSRP *
a=accept-types:*
a=setup:active
b=AS: 45

Message 2 – 100 Trying
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@operatorB.com>;tag=123456789
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321

V6.0
Message 3 – 200 OK (INVITE)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@operatorB.com>;tag=123456789
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorB.com
Allow: INVITE,PRACK,ACK,CANCEL,UPDATE,PUBLISH,OPTIONS,MESSAGE,BYE,REFER,SUBSCRIBE,NOTIFY
P-Asserted-Identity: Tel:+447960306800
Privacy: none
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=5678910
Supported: 100rel,timer,precondition,histinfo,from-change
Session-Expires: 150;refresher=uac
Require:timer
Accept-Contact: +g.oma.sip-im
Contact: <sip:10.10.0.1:5060>;+g.oma.sip-im
Content-Type: application/sdp
Content-Length: 184
v=0
s=-
a=audio 0 0 IN IP4 10.10.1.1
t=0 0
c=IN IP4 10.10.1.1
m=message 6000 TCP/MSRP *
a=accept-types:*
Message 4 – ACK

ACK sip:Chat-Factory2@operatorB.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
To: <sip:Chat-Factory2@opertaorB.com>;tag=123456789
From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321
CSeq: 1 INVITE
Call-ID: dgh1234567@operatorB.com
Max-Forwards:70
Route: sip:10.0.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length:0

There is now a session established between the initiating UE and Chat-Factory, and a TCP connection can be established to carry the MSRP (as shown in annex B.16) from the UE to the conference resource.

The initiating UE will also subscribe to the conference event package to be notified of conference progress/changes. This is identical to the flow shown in figure 13.

The next step is for the controlling UE to move the existing other user (on the 1:1 chat) to the group chat followed by adding new users to the group chat. Both are accomplished via a REFER message, as shown in figure 25.
The message contents are as shown below, and cover both cases (moving the existing user and adding new users).

**Message 1a – REFER (to an existing party)**

```
REFER sip: Chat-factory2@operatorB.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
To: <sip:Chat-factory2@operatorB.com>
P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com
Call-ID: 56789102@operatorB.com
CSeq: 1 REFER
P-Asserted-Identity: Tel:+447960306800
P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234
Privacy: Id
Max-Forwards: 70
Refer-To:<sip:+397850316900@operatorA.com;user=phone;method=INVITE?Replaces=dgh123456 %40operatorA.com%3Bto-tag%3Dade2345%3Bfrom-tag%3D5678ab34?Require=replaces>
Referred-By: sip:+447960306800@operatorB.com;user=phone
Refer-Sub: false
Accept: application/sdp, message/sipfrag
Require: norefersub
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Length: 0
```

**Message 2a – 202 ACCEPTED**
SIP/2.0 202 Accepted

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321

To: <sip:conference-factory1@operatorB.com>;tag=7824567

P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com

Refer-Sub: false

Call-ID: 56789102@operatorB.com

CSeq: 1 REFER

Contact: <sip:10.10.0.1:5060>

Content-Length: 0

The above REFER message causes the chat factory to initiate a session to the cited (Refer-To) target which will contain the following headers (copied/derived from the REFER message).

Require: replaces

Referred-By: <sip:+447960306800@operatorB.com;user=phone>

Replaces: dgh1234565@operatorA.com;to-tag=ade2345;from-tag=5678ab34

Contact: <conference-factory1@operatorB.com>;isfocus

The target will accept this session and redirect the existing media endpoint to the chat factory, prior to sending a BYE to terminate the previous 2-way session.

In addition, the controlling UE will be notified of progress (having subscribed to the conference package – see figure 13).

Message 1b – REFER (to new users)

REFER sip: conference-factory1@operatorB.com SIP/2.0

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd

From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321

To: <sip:conference-factory1@operatorB.com>

P-Charging-Vector: icid-value="abch+23456y"; orig-ioi=operatorA.com; term-ioi=operatorB.com

Call-ID: 56789102@operatorB.com

CSeq: 2 REFER

P-Asserted-Identity: Tel:+447960306800
GSM Association
Official Document IR.95 - SIP-SDP Inter-IMS NNI Profile

P-Access-Network-Info:3GPP-E-UTRAN-FDD;e-utran-cell-id-3gpp=1234

Privacy: Id
Max-Forwards: 70
Refer-To: cid:cn35t8jf02@operatorB.com
Refer-Sub: false
Accept: application/sdp, message/sipfrag
Require: multiple-refer, norefersub
Referred-By: <sip:+447960306800@operatorB.com;user=phone>
Route: sip:10.10.0.1:5060;lr
Contact: <sip:10.0.0.1:5060>
Content-Type: application/resource-lists+xml
Content-Disposition: recipient-list
Content-ID: <cn35t8jf02@operatorB.com>
Content-Length: 539

<?xml version="1.0" encoding="UTF-8"?>
<resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"
    xmlns:cp="urn:ietf:params:xml:ns:copyControl">
  <list>
    <entry uri="sip:+331234567890@operatorC.com;user=phone" cp:copyControl="to" />
    <entry uri="sip:+491234567890@operatorD.com;user=phone" cp:copyControl="to"
        cp:anonymize="true"/>
    <entry uri="sip:+391234567890@operatorE.com;user=phone" cp:copyControl="to"/>
  </list>
</resource-lists>

Message 2b – 202 ACCEPTED

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 10.0.0.1:5060;branch=1234567abcd
From: <sip:+447960306800@operatorB.com;user=phone>;tag=564321
The above message causes the conference factory to initiate a session to the cited (Refer-To) targets which will add that party to the chat conference. The CONTACT header will contain the "isfocus" parameter (see IETF RFC 3840 [52]). The addition of these parties is as shown in figure 23. In addition, the controlling user will be notified of conference progress via SIP NOTIFY messages (via subscription to conference event package).
Annex C  Document Management

C.1  Document History

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