SIP-SDP Inter-IMS NNI Profile

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This is a Non-binding Permanent Reference Document of the GSMA

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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, SMSoIP 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, SMSoIP, 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]
- IR.74 – Video Share Interoperability Specification [7]
- 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.88 – LTE Roaming NNI Guidelines [6]
- 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, SMSoIP, 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

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

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

1.4.2 Terms

Term	Description
Confirming SDP Offer	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

Term	Description
	payload type.

1.5 Document Cross-References

Ref	Doc Number	Title
[1]	3GPP TS 29.165 R11	Inter-IMS Network-Network Interface (NNI)
[2]	GSMA PRD IR.92	IMS Profile for Voice and SMS
[3]	GSMA PRD IR.94	IMS Profile for Conversational Video Service
[4]	GSMA PRD IR.65	IMS Roaming NNI and Interworking Guidelines
[5]	GSMA PRD IR.90	RCS Interworking Guidelines
[6]	GSMA PRD IR.88	LTE Roaming NNI Guidelines
[7]	GSMA PRD IR.74	Video Share Interoperability Specification
[8]	Void	
[9]	3GPP TS 26.114	IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction
[10]	ETSI TS 181 005	Service & Capability Requirements
[11]	IETF RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
[12]	IETF RFC 3261	Session Initiation Protocol (SIP)
[13]	IETF RFC 4566	Session Description Protocol (SDP)
[14]	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
[15]	IETF RFC 3312	Integration of Resource Management and Session Initiation Protocol (SIP)
[16]	IETF RFC 4028	Session Timers in the Session Initiation Protocol (SIP)
[17]	3GPP TS 29.238	Interconnection Border Control Function (IBCF) – Transition Gateway (TrGW) interface, Ix Interface; Stage 3
[18]	IETF RFC 3556	SDP Bandwidth Modifiers for RTCP bandwidth
[19]	IETF RFC 3891	The Session Initiation Protocol (SIP) “Replaces” Header
[20]	IETF RFC 3327	Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts
[21]	IETF RFC 4488	Suppression of Session Initiation Protocol (SIP) REFER Method Implicit Subscription
[22]	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
[23]	GSMA PRD IR.67	DNS/ENUM Guidelines for Service Providers and GRX/IPX Providers
[24]	ITU-T Rec. T.140 (1998)	Text Conversation Protocol for multimedia application, with amendment 1 (2000)
[25]	IETF RFC 4013	RTP Payload for Text Conversation
[26]	3GPP TS 33.210	3G Security; Network Domain Security (NDS); IP Network

Ref	Doc Number	Title
		Layer Security
[27]	IETF RFC 4303	IP Encapsulating Security Payload (ESP)
[28]	IETF RFC 5368	Referring to Multiple Resources in the Session Initiation Protocol (SIP)
[29]	ITU-T Rec. H.263 (2005)	Video Coding for low bit rate communication
[30]	IETF RFC 3984	RTP Payload format for ITU-T Rec. H.264 Video
[31]	ITU-T Rec. H.264 (2005)	Advanced video coding for generic audiovisual services ISO/IEC 14496-10:2005: “Information technology – Coding of audio-visual objects – Part 10: Advanced Video Coding”
[32]	IETF RFC 3551	RTP Profile for Audio Video Conferences with Minimal Control
[33]	IETF RFC 4585	Extended RTP Profile for Real Time Control Protocol (RTCP)-Based Feedback (RTP/AVPF)
[34]	OMA CPM	OMA CPM Conversation Functions (OMA-CPM-TS_Conv_Func-V1_0-20120612-A)
[35]	IETF RFC 5939	Session Description Protocol (SDP) Capability Negotiation
[36]	IETF RFC 4103	RTP Payload for Text Conversation
[37]	3GPP TS 29.079	Optimal Media Routing within the IP Multimedia System (IMS); Stage 3
[38]	IETF RFC 5621	Message Body Handling in the Session Initiation Protocol (SIP)
[39]	IETF RFC 5547	A Session Description Protocol (SDP) Offer/Answer Mechanism to enable File Transfer
[40]	IETF RFC 4483	A Mechanism for Content Indirection in Session Initiation Protocol (SIP) Messages
[41]	IETF RFC 5438	Instant Message Disposition Notification
[42]	IETF RFC 3842	A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
[43]	IETF RFC 4575	A Session Initiation Protocol (SIP) Event Package for Conference State
[44]	GSMA PRD RCC.07	RCS 7.0 Advanced Communications Services & Client Specification
[45]	3GPP TS 29.163	Interworking between IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched networks
[46]	IETF RFC 3966	The Tel URI for telephone numbers
[47]	3GPP TS 24.341	Support of SMS over IP networks; Stage 3
[48]	IETF RFC 4244	An extension to the Session Initiation Protocol (SIP) for Request History Information
[49]	IETF RFC 4916	Connected Identity in the Session Initiation Protocol (SIP)
[50]	IETF RFC 3680	A Session Initiation Protocol (SIP) Event Package for registration
[51]	IETF RFC 3515	The Session Initiation Protocol (SIP) Refer method

Ref	Doc Number	Title
[52]	IETF RFC 3840	Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)
[53]	IETF RFC 3856	A Presence Event Package for the Session Initiation Protocol (SIP)
[54]	IETF RFC 3857	A Watcher-Information Event Template Package for the Session Initiation Protocol (SIP)
[55]	IETF RFC 3858	An Extensible Mark-Up Language (XML) based format for Watcher Information
[56]	IETF RFC 3863	Presence Information Data Format (PIDF)
[57]	IETF RFC 4479	A data model for Presence
[58]	IETF RFC 4975	The Message Session Relay Protocol (MSRP)
[59]	IETF RFC 5364	Extensible Markup Language (XML) Format Extension for Representing Copy Control Attributes in Resource Lists
[60]	IETF RFC 5366	Conference Establishment Using Request-Contained Lists in the Session Initiation Protocol (SIP)
[61]	IETF RFC 5368	Referring to Multiple Resources in the Session Initiation Protocol (SIP)
[62]	3GPP TS 29.658	SIP Transfer of IP Multimedia Tariff Information; Protocol Specification.
[63]	3GPP TS 24.337	IP Multimedia (IM) Core Network (CN) Subsystem; IP Multimedia Subsystem (IMS) Inter UE Transfer; Stage 3
[64]	3GPP TS 24.237	IP Multimedia (IM) Core Network (CN) Subsystem; IP Multimedia Subsystem (IMS) Service Continuity; Stage 3
[65]	IETF RFC 5627	Obtaining and using Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP)
[66]	IETF RFC 4412	Communications Resource Priority in the Session Initiation Protocol (SIP).
[67]	3GPP TS 24.628	Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification
[68]	3GPP TS 24.229	IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP);Stage 3
[69]	3GPP TS 29.162	Interworking between the IMS CN Subsystem and IP Networks
[70]	3GPP TS 26.071	Mandatory speech CODEC speech processing functions; AMR speech Codec; General description
[71]	3GPP TS 26.090	Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions
[72]	3GPP TS 26.073	ANSI C code for the Adaptive Multi Rate (AMR) speech codec
[73]	3GPP TS 26.104	ANSI-C code for the floating-point Adaptive Multi-Rate (AMR) speech codec

Ref	Doc Number	Title
[74]	3GPP TS 26.093	Mandatory speech codec speech processing functions Adaptive Multi-Rate (AMR) speech codec; Source controlled rate operation
[75]	3GPP TS 26.171	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; General description
[76]	3GPP TS 26.173	ANSI-C code for the Adaptive Multi-Rate – Wideband (AMR-WB) speech codec
[77]	3GPP TS 26.190	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; Transcoding functions
[78]	3GPP TS 26.193	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; Source controlled rate operation
[79]	3GPP TS 26.204	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; ANSI-C code
[80]	3GPP TS 23.228	IP Multimedia Subsystem (IMS); Stage 2
[81]	IETF RFC 7329	A Session Identifier for the Session Initiation Protocol (SIP)
[82]	IETF RFC 6086	Session Initiation Protocol (SIP) INFO Method and Package Framework
[83]	3GPP TS 26.441	Codec for Enhanced Voice Services (EVS); General overview
[84]	3GPP TS 26.442	Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point)
[85]	3GPP TS 26.443	Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)
[86]	3GPP TS 26.445	Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description
[87]	3GPP TS 26.447	Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets
[88]	3GPP TS 26.449	Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects
[89]	3GPP TS 26.450	Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)
[90]	3GPP TS 26.451	Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)
[91]	GSMA PRD IR.51	IMS Profile for Voice, Video and SMS over Wi-Fi
[92]	3GPP TS 23.334	IP Multimedia Subsystem (IMS) Application Level Gateway (IMS-ALG) - IMS Access Gateway (IMS-AGW) interface: Procedures descriptions

Ref	Doc Number	Title
[93]	IETF RFC 4687	RTP Payload Format and File Storage Format
[94]	IETF RFC 6184	RTP Payload Format for H.264 Video
[95]	IETF RFC 3389	RTP Payload for Comfort Noise
[96]	GSMA PRD RCC.71	RCS Universal Profile Service Definition Document
[97]	GSMA PRD NG.105	ENUM Guidelines for Service Providers and IPX Providers
[98]	3GPP TS 26.446	Codec for Enhanced Voice Sevices (EVS); AMR-WB Backwards Compatible Functions

Note: Unless otherwise stated, the latest version of the referenced specifications applies.

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.

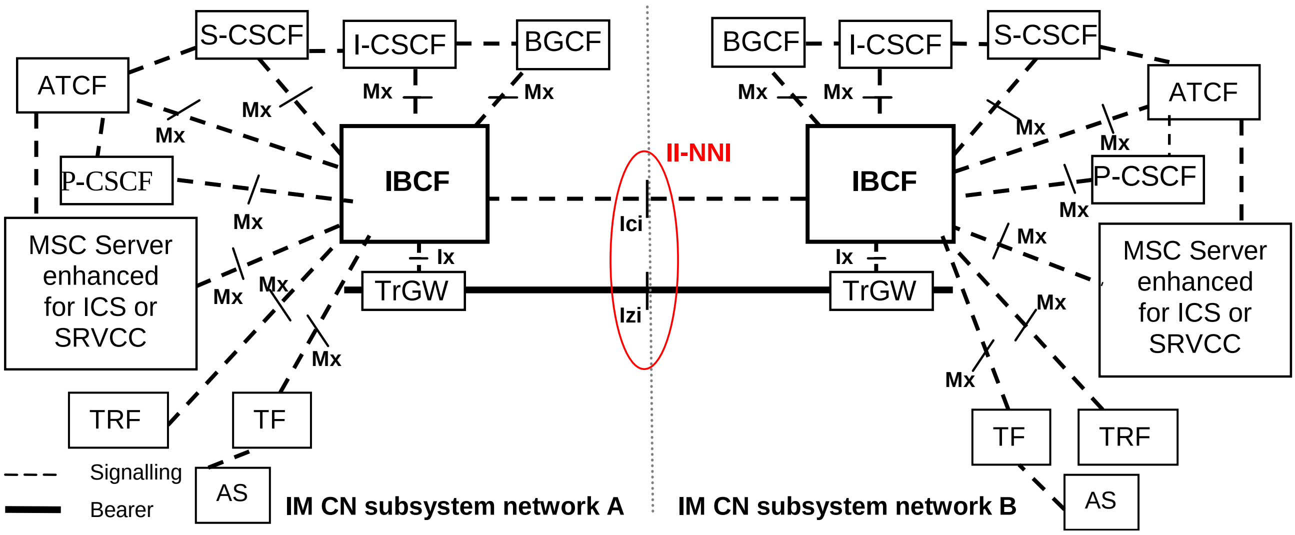


Figure 1 : Inter IMS NNI Reference Architecture

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.

Method	Status	Additional Information
INVITE	Mandatory	Includes both the initial INVITE and any subsequent re-INVITE
ACK	Mandatory	
BYE	Mandatory	
CANCEL	Mandatory	

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

Method	Status	Additional Information
UPDATE	Mandatory	Used for offer/answer exchange, session timer refresh etc.

Table 1: Applicable SIP Methods

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.

Header	Status	Additional Information
Contribution-ID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Conversation-ID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
InReplyTo-Contribution-ID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Message-Expires	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Message-UID	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
Session-Replaces	o	Defined in OMA CPM [34] – not included in 3GPP TS 29.165 [1]
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.

Item	Header Field	Suggested Trust Relationship	Comment
[1]	P-Asserted-Identity	Trusted	Trust is mandatory at the roaming NNI
[2]	P-Access-Network-Info	Trusted	Should be trusted at the roaming NNI, even if “Not Trusted” at the non-roaming NNI
[3]	Resource-Priority	Not Trusted	
[4]	History-Info	Trusted	
[5]	P-Asserted-Service	Trusted	Mandatory for terminating sessions delivered over Roaming NNI, when using LBO architecture [4] for all services that have an ISCI defined.
[6]	P-Charging-Vector	Trusted	
[7]	P-Charging-Function-Addresses	Not Trusted	
[8]	P-Profile-Key	Not Trusted	Shall always be “Not Trusted” at a non-Roaming NNI
[9]	P-Private-Network-Indication	Not Trusted	
[10]	P-Served-User	Trusted Not Trusted	Shall always be “Trusted” at the Roaming-NNI Shall always be “Not Trusted” at a non-Roaming NNI
[11]	Reason	Trusted	
[12]	P-Early-Media	Trusted	
[13]	Feature-Caps	Trusted	Trust is mandatory at the roaming NNI

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.

Header	Related Methods / Responses
Contribution-ID	INVITE request OR MESSAGE request Both with status do
Conversation-ID	INVITE request OR MESSAGE request – both with status do

Header	Related Methods / Responses
InReplyTo-Contribution-ID	INVITE request OR MESSAGE request – both with status do
Message-Expires	INVITE request – with status do
Message-UID	MESSAGE request OR 200 OK (MESSAGE) response OR BYE request – all with status do.
Session-Replaces	INVITE request – with status do
Note: Status meaning is as defined in Table B.2.1 of 3GPP TS 29.165 [1]	

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.

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

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 bi-lateral 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:

MIME Type	Additional Info
application/SDP	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.
multipart/mixed	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.
multipart/related	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]).
multipart/alternative	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.
message/external-body	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.

MIME Type	Additional Info
message/cpim	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.
message/imdn+xml	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.
application/vnd.etsi.mcid+xml	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.
application/vnd.3gpp.cw+xml	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.
application/vnd.3gpp.comm-div-info+xml	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.
application/vnd.etsi.aoc+xml	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.
application/vnd.etsi.cug+xml	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.
application/vnd.etsi.sci+xml	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.
application/vnd.etsi.pstn+xml	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.
message/sipfrag	Used to convey SIP session progress. This is conditionally supported if MMTel Ad-Hoc Conference / RCS Group Chat service is used.
application/x-session-info	Used to convey additional digits in a SIP INFO for overlap sending. Not applicable to this profile.

MIME Type	Additional Info
application/pidf+xml, application/pidf-diff+xml, application/watcherinfo+xml, application/xcap-diff+xml, application/vnd.oma.suppnot+xml, application/simple-filter+xml	Conditionally supported in RCS Social Presence is applicable at the NNI.
application/resource-lists+xml, application/rlmi+xml	Used to convey a list of target users for MMTel Ad-Hoc Conference & RCS Group Chat.
application/load-control+xml	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.
application/im-iscomposing+xml	Used to convey SIMPLE IM. Conditionally supported for RCS messaging services for interworking between SIMPLE IM and CPIM.
application/simple-message-summary+xml	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].
application/vnd.3gpp.sms	Conditionally supported at the NNI if the SMS over IP service is in scope – see 3GPP TS 24.341 [47].
application/vnd.3gpp.ussd	Used for MMI at the roaming NNI. Optional to this profile and may be supported if bilaterally agreed.
application/vnd.3gpp.iut+xml application/vnd.3gpp.replication+xml	Used for inter-UE transfer. Optional in this profile and may be supported if bilaterally agreed. See 3GPP TS 24.337 [63].
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	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].
application/3gpp-ims+xml	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.
application/reginfo+xml	Conditionally supported for the roaming NNI.
application/conference-info+xml	Conditionally supported if conference services are supported across the NNI (e.g. MMTel Ad-Hoc Conference, RCS Group Chat).

Table 6: SIP Message Bodies

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

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

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

Table 7: SIP Option Tags

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-aw-recv” is present or not in an EVS RTP payload type of the received SDP offer, the originating network can set the MIME parameter “ch-aw-recv” 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”:

    - 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.
    - 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]).
- Adaptive Multi-Rate Wideband (AMR-WB) codec (described in 3GPP TS 26.114 [9], 3GPP TS 26.171 [75], 3GPP TS 26.190 [77], 3GPP TS 26.173 [76], 3GPP TS 26.193 [78] and 3GPP TS 26.204 [79]).
- If super-wideband or fullband speech communications are supported over the II-NNI, then the EVS codec is also mandatory as described in 3GPP Release 12 TS 26.114 [9], 3GPP Release 12 TS 26.441 [883], 3GPP Release 12 TS 26.445 [86], 3GPP Release 12 TS 26.442 [84], 3GPP Release 12 TS 26.443 [85], 3GPP Release 12 TS 26.447 [87], 3GPP Release 12 TS 26.449 [88], 3GPP Release 12 TS 26.450 [489] and 3GPP Release 12 TS 26.451 [90].
- 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,
- ITU-T Recommendation H.264 Constrained Baseline Profile (CBP) Level 3.1 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]

- ITU-T Recommendation H.265 (HEVC) Main Profile, Main Tier, Level 3.1 as specified in 3GPP Release 12 TS 26.114 [9] section 5.2.2.

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)

If supported, Global Text Telephony (GTT) messages must use ITU-T Recommendation T.140 [24] real-time text according to the rules and procedures specified in 3GPP TS 26.114 [9] and with clarifications in Annex B.2 in GSMA PRD IR.92 [2].

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.

See 3GPP TS 26.114 [9] and clause 3.2 of GSMA PRD IR.92 [2] for further guidance on the encoding and transport of DTMF events.

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 perfomed, 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.

SDP Attribute	Profile Settings
Version (v-line)	The value must always be equal to zero: v=0.
Origin (o-line)	The origin line consists of six fields: (<username>, <sess-id>, <sess-version>, <nettype>, <addrtype> and <unicast-address>). - <user name> should contain an hyphen - <session ID> and <version> should contain one or mode 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=video 1234 RTP/AVP 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/16000” (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	Profile Settings
	a= ptime:<time> Example: a=ptime:20 (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=tcap:<capability number> <transport protocol list> a=pcfg:<config number> <potential config list> a=acfg:<config number> <selected config list> Example : a=tcap:1 RTP/AVPF a=pcfg:1 t=1 (for RTCP Feedback as per IETF RFC 4585 [33] and IETF RFC 5104 [95]) a=rtcp-fb:<payload type> <feedback type> <feedback parameters> Example : a=rtcp-fb:* nack rpsi a=rtcp-fb:* ccm fir a=rtcp-fb:* ccm tmmbr (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>:<bandwdith>. 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.

SDP Attribute	Profile Settings
	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.

Header	Reference(s)	Example Header
Accept	IETF RFC 3261	*Accept:application/sdp,message/cpim*
Accept-Contact	IETF RFC 3841	*Accept-Contact:*;mobility="mobile";methods="INVITE"*
Accept-Encoding	IETF RFC 3261	*Accept-Encoding:gzip*
Accept-Language	IETF RFC 3261	*Accept-Language: da, en-gb;q=0.8, en;q=0.7*
Accept-Resource-Priority	IETF RFC 4412	*Accept-Resource-Priority:ets.0*
Alert-Info	IETF RFC 3261	*Alert-Info:<urn:alert:service:call-waiting>*
Allow	IETF RFC 3261	*Allow:INVITE, ACK, UPDATE, PRACK, CANCEL, PUBLISH, MESSAGE, OPTIONS,SUBSCRIBE,NOTIFY*
Allow-Events	IETF RFC 6665	*Allow-Events: conference*
Answer-Mode	IETF RFC 5373	*Answer-Mode: Auto*
Authentication-Info	IETF RFC 3261 IETF RFC 3310	*Authentication-Info: nextnonce="47364c23432d2e131a5fb210812c"*
Authorization	IETF RFC 3261 IETF RFC 3310 3GPP TS 24.229	*Authorization: Digest, username="jon.dough@mobile.biz", realm="RoamingUsers@mobile.biz", nonce="CjPk9mRqNuT25eRkajM09uTl9nM09uTl9nMz5OX25PZz==", uri="sip:home.mobile.biz",qop=auth-int, nc=00000001,cnonce="0a4f113b", response="6629fae49393a05397450978507c4ef1", opaque="5ccc069c403ebaf9f0171e9517f40e41"*
Call-Id	IETF RFC 3261	*Call-Id:12345@mydomain.com*
Call-Info	IETF RFC 3261	*Call-Info: <http://wwww.example.com/alice/photo.jpg> ;purpose=icon*
Contact	IETF RFC 3261	*Contact:<sip:ibcf4@operator1.com:5060>*
Content-Disposition	IETF RFC 3261	*Content-Disposition:session*
Content-Encoding	IETF RFC 3261	*Content-Encoding:gzip*
Content-Language	IETF RFC 3261	*Content-Language:fr*
Content-Length	IETF RFC 3261	*Content-Length:146*
Content-Type	IETFRFC 3261	*Content-Type:application/sdp*
Contribution-ID	OMA CPM	*Contribution-ID: abcdef-1234-5678-90ab-cdef01234567*
Converstaion-ID	OMA CPM	*Conversation-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6*

Header	Reference(s)	Example Header
CSeq	IETF RFC 3261	*CSeq:2 CANCEL*
Date	IETF RFC 3261	*Date Sat,13 Nov 2010 23:29:00 GMT*
Error-Info	IETF RFC 3261	*Error-Info: <sip:announcement10@example.com>*
Event	IETF RFC 6665 IETF RFC 6446 IETF RFC 3680	*Event:reg*
Expires	IETF RFC 3261	*Expires:3600*
Feature-Caps	IETF RFC 6809 3GPP TS 24.229	*Feature-Caps:+3gpp.trf=sip:trf3.operator3.com*
Flow-Timer	IETF RFC 5626	*Flow-Timer: 30*
From	IETF RFC 3261	*From: <sip:+12125551212@operator1.com;user=phone>; tag=145688*
Geolocation	IETF RFC 6442	*Geolocation:<user1@operator1.com>*
Geolocation-Error	IETF RFC 6442	*Geolocation-Error:100;code="Cannot Process Location"*
Geolocation-Routing	IETF RFC 6442	*Geolocation-Routing:yes*
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?Reason=SIP%3Bcause%3D486%3Btext%3D"Busy Here">;index=1.2, <sip:User5@UA5.example.com>;index=1.3*
Info-Package	IETF RFC 6086	*Info-Package:foo*
InReplyTo-Contribution-ID	OMA CPM	*InReplyTo-Contribution-ID: 01234567-89ab-cdef-0123-456789abcdef*
Join	IETF RFC 3911	*Join: 1234@example.com;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-Acces-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:+14085264000@operator1.com;user=phone*
P-Asserted-	IETF RFC 6050	P-Asserted-Service: *urn:urn-7:3gpp-service.ims.icsi.mmtel*

Header	Reference(s)	Example Header
Service		
P-Associated-URI	IETF RFC 7315	P-Associated-URI:sip:user2@operator3.com
P-Called-Party-ID	IETF RFC 7315	P-Called-Party-ID: sip:user1-business@example.com
P-Charging-Vector	IETF RFC 7315	P-Charging-Vector: icid-value=1234bc9876e; icid-generated-at=192.0.6.8; orig-ioi=home1.net
P-Early-Media	IETF RFC 5009	P-Early-Media:supported
P-Preferred-Service	IETF RFC 6050	P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
P-Private-Network-Indication	IETF RFC 7316	P-Private-Network-Indication: example.com
P-Profile-Key	IETF RFC 5002	P-Profile-Key:sip:user3@operator4.com
P-Refused-URI-List	IETF RFC 5318	P-Refused-URI-List: sip:friends-list@example.net; members=<cid:an3bt8jf03@example.net>
P-Served-User	IETF RFC 5502	P-Served-User:sip:user1@operator2.com;sescase=orig
P-Visited-Network-ID	IETF RFC 7315	P-Visited-Network-ID:"Visited network number 1"
Path	IETF RFC 3327	Path:<sip:P2.example.com;lr>,<sip:P1.example.com;lr>
Permission-Missing	IETF RFC 5360	Permission-Missing: userC@example.com
Policy-Contact	IETF RFC 6794	Policy-Contact: sip:server5@example.com
Priority	IETF RFC 3261	Priority: emergency
Priv-Answer-Mode	IETF RFC 5373	Priv-Answer-Mode: Auto
Privacy	IETF RFC 3323 IETF RFC 3325	Privacy:user Privacy:id
Proxy-Authenticate	IETF RFC 3261	Proxy-Authenticate: Digest realm="atlanta.com", domain="sip:ss1.carrier.com", qop="auth", nonce="f84f1cec41e6cbe5aea9c8e88d359", opaque="", stale=FALSE, algorithm=MD5
Proxy-Authorization	IETF RFC 3261	Proxy-Authorization: Digest username="Alice", realm="atlanta.com", nonce="c60f3082ee1212b402a21831ae", response="245f23415f11432b3434341c022"
Proxy-Require	IETF RFC 3261	Proxy-Require: foo
Rack	IETF RFC 3262	Rack:10
Reason	IETF RFC 3326	Reason:Q.850;cause=16;text="Terminated"
Record-Route	IETF RFC 3261	Record-Route: <sip:server10.biloxi.com;lr>, <sip:bigbox3.site3.atlanta.com;lr>
Recv-Info	IETF RFC 6086	Recv-Info: bar

Header	Reference(s)	Example Header
Referred-By	IETF RFC 3892	*Referred-By: <sip:alice@phone1.example.org>*
Refer-Sub	IETF RFC 4488	*Refer-Sub=false*
Refer-To	IETF RFC 3515	*Refer-To: <sip:dave@bobster.example.org? Replaces=425928%40bobster.example.com.3%3B to-tag%3D7743%3Bfrom-tag%3D6472>*
Reject-Contact	IETF RFC 3841	*Reject-Contact: *;actor="msg-taker";video*
Replaces	IETF RFC 3891	*Replaces: 425928@bobster.example.org;to-tag=7743;from-tag=6472*
Reply-To	IETF RFC 3261	*Reply-To: Bob <sip:user4@example.com>*
Request-Disposition	IETF RFC 3841	*Request-Disposition: proxy*
Require	IETF RFC 3261	*Require:Path*
Resource-Priority	IETF RFC 4412	*Resource-Priority:ets.1*
Retry-After	IETF RFC 3261	*Retry-After: 3600*
Route	IETF RFC 3261	*Route: <sip:bigbox3.site3.atlanta.com;lr>, <sip:server10.biloxi.com;lr>*
RSeq	IETF RFC 3262	*RSeq:10*
Security-Client	IETF RFC 3329	*Security-Client: ipsec-ike*
Security-Server	IETF RFC 3329	*Security-Server: ipsec-ike*
Security-Verify	IETF RFC 3329	*Security-Verify:ipsec-ike; q=0.1*
Service-Route	IETF RFC 3608	*Service-Route:<sip:P1.example.com;lr>,<sip:P2.example.com;lr>*
Session-Expires	IETF RFC 4028	*Session-Expires:3600;refresher=uac*
Session-Replaces	OMA CPM	*Session-Replaces: abcdef-1234-5678-90ab-cdef01234567*
Session-ID	IETF RFC 7329	*Session-ID: 0123456789abcdef123456789abcdef0*
SIP-Etag	IETF RFC 3903	*SIP-Etag:12345*
SIP-If-Match	IETF RFC 3903	*SIP-If-Match:12345*
Subscription-State	IETF RFC 6665	*Subscription-State:active*
Supported	IETF RFC 3261	*Supported:100Rel,timer,precondition*
Suppress-If-Match	IETF RFC 5839	*Suppress-If-Match:34567*
Target-Dialog	IETF RFC 4538	*Target-Dialog:*
Timestamp	IETF RFC 3261	*Timestamp:64*
To	IETF RFC 3261	*To: <sip:+12126661212@operator2.com;user=phone>;tag=123456*
Trigger-Consent	IETF RFC 5360	*Trigger-Consent: sip:user1@example.com; target-uri="sip:user2@example.xcom"*
Unsupported	IETF RFC 3261	*Unsupported:100Rel*

Header	Reference(s)	Example Header
User-Agent	IETF RFC 3261	*User-Agent:Softphone Beta1.5*
Via	IETF RFC 3261	*Via: SIP/2.0/UDP operator1.com:5060;branch=z9hG4bK87asdks7*
Warning	IETF RFC 3261	*Warning: 307 isi.edu "Session parameter 'foo' not understood"*
WWW-Authenticate	IETF RFC 3261 IETF RFC 3310 3GPP TS 24.229	*WWW-Authenticate: Digest realm="atlanta.com", domain="sip:boxesbybob.com", qop="auth", nonce="f84f1cec41e6cbe5aea9c8e88d359", opaque="", stale=FALSE, algorithm=MD5*

Table 9: SIP Header Examples

Annex B SIP Message Examples (Informational)

Appendix 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.

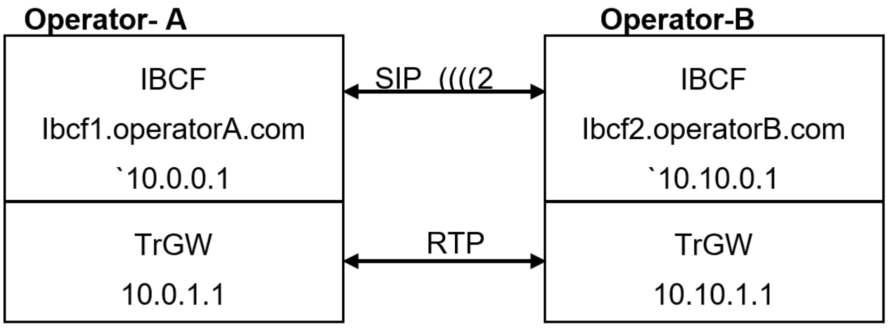


Figure 2 : NNI Configuration

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.

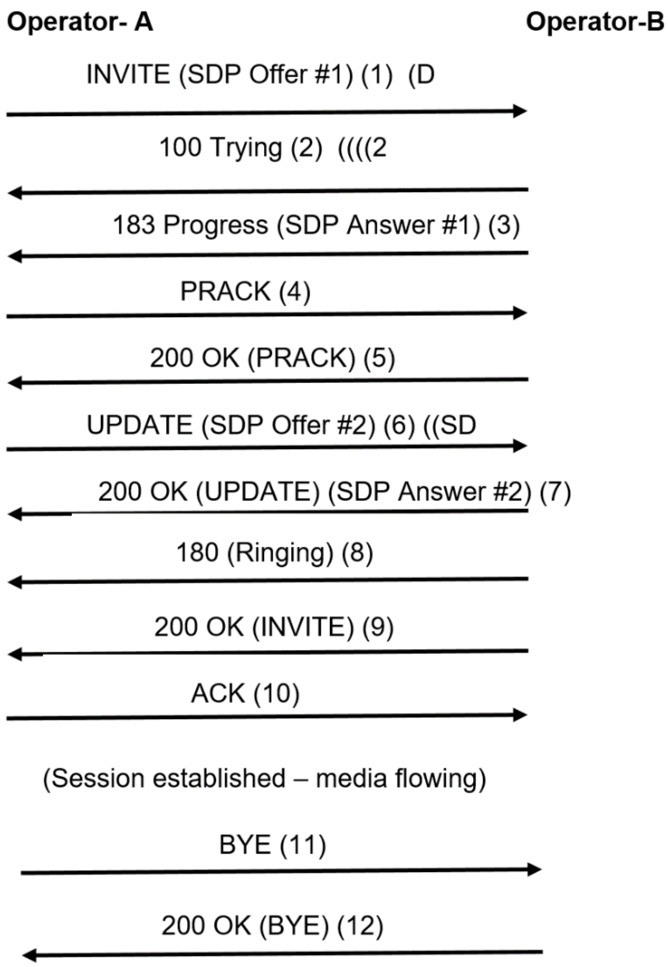


Figure 3: NNI Session Establishment & Teardown

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, NOTITY

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-ioi=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%3gpp-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

<u>Message 2 – 100 Trying</u>

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

<u>Message 3 – 183 Progress (SDP Answer #1)</u>

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%3gpp-service.ims.icsi.mmtel"

Content-Type: application/sdp

Content-Length: 450

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 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

<u>Message 7 – 200 OK (UPDATE) (SDP Answer #2)</u>

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=-

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

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;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

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%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

<u>Message 10 – ACK</u>

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:10.0.1.1:5060;lr

Contact: <sip:10.0.0.1:5060>

Content-Length:0

<u>Message 11 – BYE</u>

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 appendix covers the case of a voice and video multi-media session establishment and teardown. The message flow and numbering is identical to that of appendix 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-appendix with references made to messages in appendix B.1.

Message 1 – INVITE (SDP Offer #1)

As appendix 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%3gpp-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

<u>Message 2 – 100 Trying</u>

As appendix B.1.

<u>Message 3 – 183 Progress (SDP Answer #1)</u>

As appendix B.1 with 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:10.10.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel;**video**"

Content-Type: application/sdp

Content-Length: 878

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

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 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 4 – PRACK

As appendix B.1.

Message 5 – 200 OK (PRACK)

As appendix B.1.

Message 6 – UPDATE (SDP Offer #2)

As appendix 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

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 tmmbr

a=extmap:7 urn:3gpp:video-orientation

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)

As appendix 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 tmmbr

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

<u>Message 8 – 180 Ringing</u>

As appendix 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"

<u>Message 9 – 200 OK (INVITE)</u>

As appendix 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"

<u>Message 10 – ACK</u>

As appendix B.1.

<u>Message 11 – BYE</u>

As appendix B.1.

<u>Message 12 – 200 OK (BYE)</u>

As appendix B.1.

B.3 Use of session timer

The session establishment sequence in appendix 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.

.

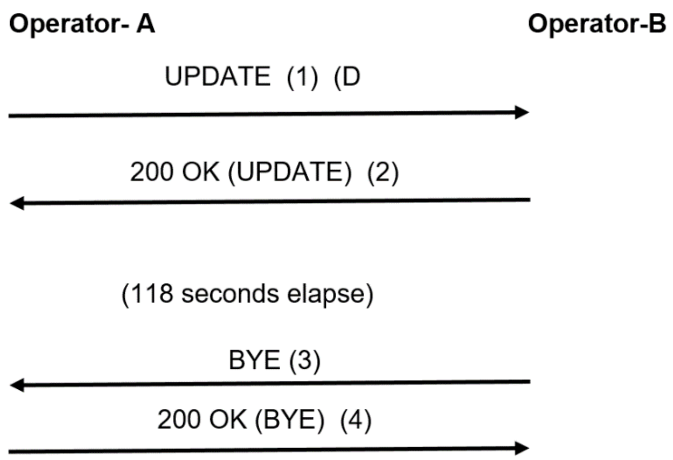


Figure 4 : Session Timer

<u>Message 1 – UPDATE</u>

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

<u>Message 2 – 200 OK (UPDATE)</u>

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

<u>Message 3 – BYE</u>

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

<u>Message 4 – 200 OK (BYE)</u>

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 appendix 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 appendix 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

<table>
<tr><th></th><th>Originating UE EVS +AMR-WB capable</th><th>Originating IMS EVS+ AMR-WB capable</th><th>Terminating IMS AMR-WB capable</th><th>Terminating UE AMR-WB capable</th></tr>
<tr><td>SIP Invite</td><td colspan="4">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</td></tr>
<tr><td rowspan="2">SIP 183 Progress</td><td></td><td colspan="3">←--- Confirming SDP answer
m=audio 3000 RTP/AVP 97
a=rtpmap:97 AMR-WB/16000
a=fmtp:97 mode-set=0,1,2</td></tr>
<tr><td colspan="2">←--------------------- Confirming SDP answer
(modification 1a – replace AMR-WB with EVS IO mode)
m=audio 3000 RTP/AVP 96
a=rtpmap:96 EVS/16000
a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1</td><td colspan="2"></td></tr>
<tr><td rowspan="2">SIP UPDATE</td><td colspan="2">Subsequent SDP offer ------------------------→
m=audio 3000 RTP/AVP 96
a=rtpmap:96 EVS/16000
a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1</td><td colspan="2"></td></tr>
<tr><td></td><td colspan="3">Subsequent SDP offer ---→
(modification 1b – replace EVS IO mode with AMR-WB)
m=audio 3000 RTP/AVP 97
a=rtpmap:97 AMR-WB/16000
a=fmtp:97 mode-set=0,1,2</td></tr>
<tr><td>SIP 200 OK</td><td></td><td colspan="3">←--- Subsequent SDP answer
m=audio 3000 RTP/AVP 97
a=rtpmap:97 AMR-WB/16000
a=fmtp:97 mode-set=0,1,2</td></tr>
</table>

	←---------------------- Subsequent SDP answer (modification 1a – replace AMR-WB with EVS IO mode) m=audio 3000 RTP/AVP 96 a=rtpmap:96 EVS/16000 a=fmtp:96 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1	

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>
<tr><th></th><th>Originating UE AMR-WB capable</th><th>Originating IMS AMR-WB capable</th><th>Terminating IMS EVS + AMR-WB capable</th><th>Terminating UE EVS + AMR-WB capable</th></tr>
<tr><td rowspan="2">SIP Invite</td><td colspan="3">Initial SDP offer --→
m=audio 3000 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
a=fmtp:96 mode-set=0,1,2</td><td></td></tr>
<tr><td colspan="2"></td><td colspan="2">Final SDP offer -------------------------------------→
(modification 2a – probe for EVS support)
m=audio 3000 RTP/AVP 97 96
a=rtpmap:97 EVS/16000
a=fmtp:97 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1
a=rtpmap:96 AMR-WB/16000
a=fmtp:96 mode-set=0,1,2</td></tr>
<tr><td rowspan="2">SIP 183 Progress</td><td colspan="2"></td><td colspan="2">←------------------------- Confirming SDP answer
m=audio 3000 RTP/AVP 97
a=rtpmap:97 EVS/16000
a=fmtp:97 br=5.9-24.4;bw=nb-swb; mode-set=0,1,2;evs-mode-switch=1</td></tr>
<tr><td colspan="3">←-- Confirming SDP answer
(modification 2b – replace EVS IO mode with AMR-WB)
m=audio 3000 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
a=fmtp:96 mode-set=0,1,2</td><td></td></tr>
<tr><td rowspan="2">SIP UPDATE</td><td colspan="3">Subsequent SDP offer ---→
m=audio 3000 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
a=fmtp:96 mode-set=0,1,2</td><td></td></tr>
<tr><td colspan="2"></td><td colspan="2">Subsequent SDP offer ----------------------------→
(modification 2c – replace AMR-WB with EVS IO mode)
m=audio 3000 RTP/AVP 97
a=rtpmap:97 EVS/16000
a=fmtp:97 br=5.9-24.4; bw=nb-swb; mode-set=0,1,2; evs-mode-switch=1</td></tr>
</table>

<table>
<tr><td rowspan="2">SIP 200 OK</td><td></td><td>←------------------------ Subsequent SDP answer
m=audio 3000 RTP/AVP 97
a=rtpmap:97 EVS/16000
a=fmtp:97 br=5.9-24.4.2;bw=nb-swb, mode-set=0,1,2;evs-mode-switch=1</td></tr>
<tr><td>Subsequent SDP answer ←--
(modification 2b – replace EVS IO mode with AMR-WB)
m=audio 3000 RTP/AVP 96
a=rtpmap:96 AMR-WB/16000
a=fmtp:96 mode-set=0,1,2</td><td></td></tr>
</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.

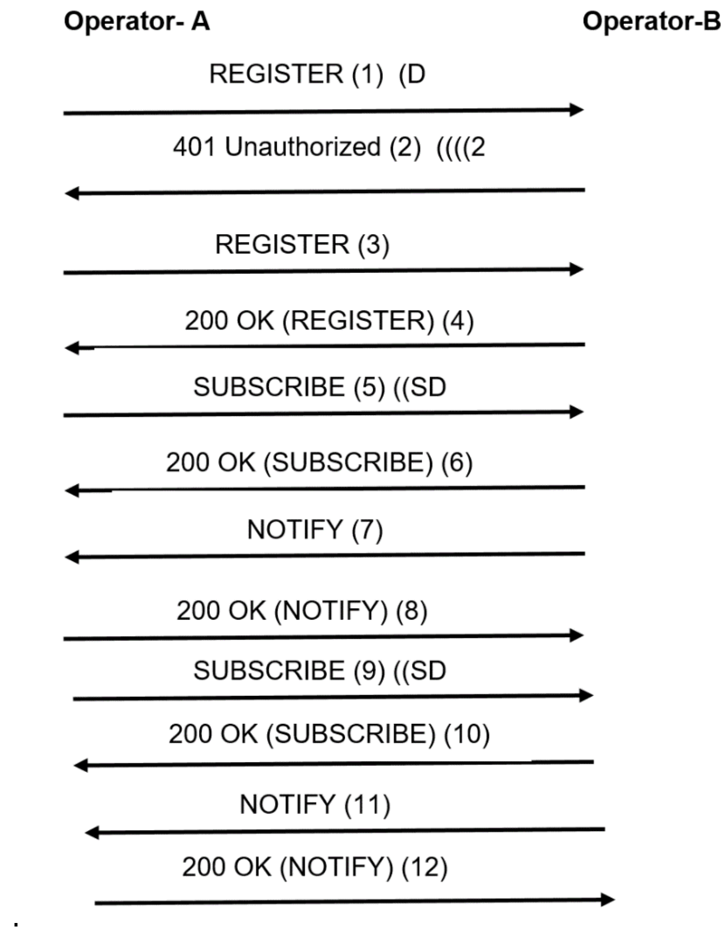


Figure 5 : IMS Registration

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

<u>Message 1 – REGISTER</u>
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-ioi=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.iari-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.rcs.ftthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7%3A3gpp-application.ims.iari.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=”12ab34bc4567b45RTFGH12323asdfghGHDe34500oOlP+”, algorithm=AKAv1-MD5, ik=2345, ck=5678

Content-Length: 0

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-ioi=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.iari-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.rcs.ftthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7%3A3gpp-application.ims.iari.gsma-vs"

Content-Length: 0

<u>Message 4 – 200 OK (REGISTER)</u>

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>

Content-Length: 0

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>

Content-Type: application/reginfo+xml

Content-Length: xx

[Registration Info] - see IETF RFC 3680 [50]

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 appendix 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

Addditionally, 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:

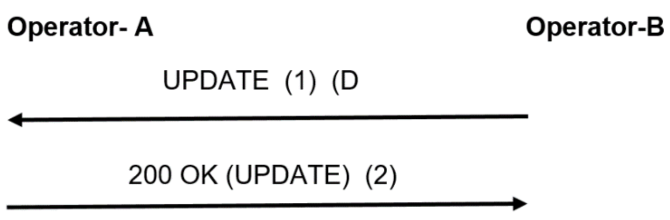


Figure 6: TIP (from Change)

The message contents are as follows (assuming session establishment as per appendix 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

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:10.10.0.1:5060>

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 Pivacy 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.

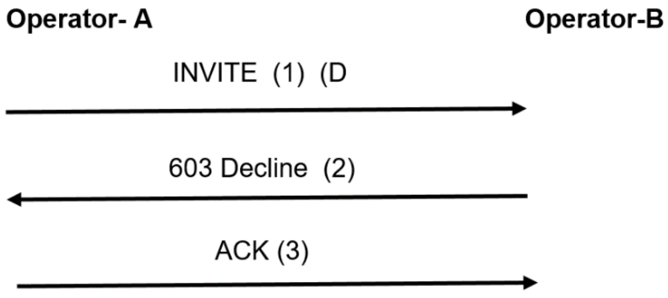


Figure 7 : OCB

Message 1 – INVITE
As Appendix 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 Appendix 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.

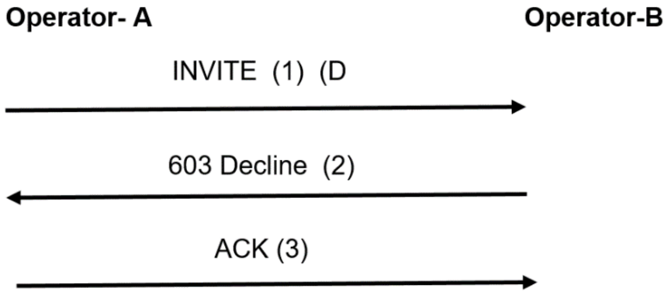


Figure 8: ICB

Message 1 – INVITE

As Appendix 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 Appendix B.1.

Communication Hold (CH)

This service is realised via an additional offer/answer exchange where the bi-driectional media (as established in appendix 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.

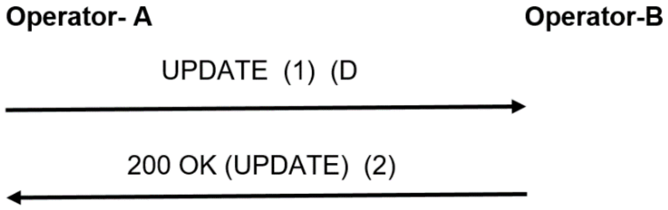


Figure 9 : Communication Hold

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 (CFNRc), Communication Deflection (CD) and No Reply (CFNR). Note that CD is not in scope in 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 :

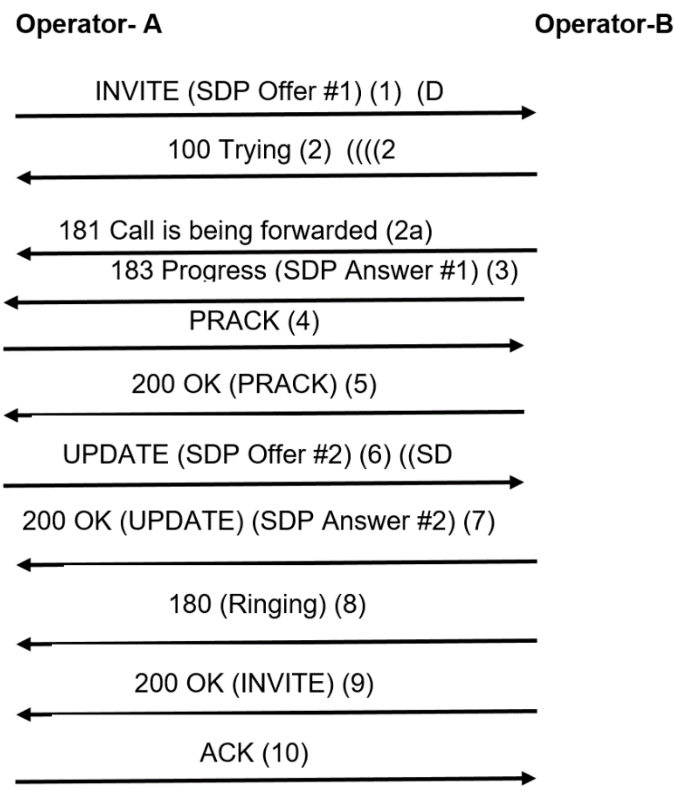


Figure 10: Call Forwarding

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 appendix B.1 – and therefore only differences from appendix B.1 will be shown here.

<u>Message 1 – INVITE (SDP Offer #1)</u>
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

<u>Message 2 – 100 Trying</u>

As appendix B.1.

<u>Message 2a – 181 Call is being forwarded</u>

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%3gpp-service.ims.icsi.mmtel"

Content-Length: 0

Message 3 – 183 Progress (SDP Answer #1)

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 4 – PRACK

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

Message 5 – 200 OK (PRACK)

As appendix 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 appendix 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.

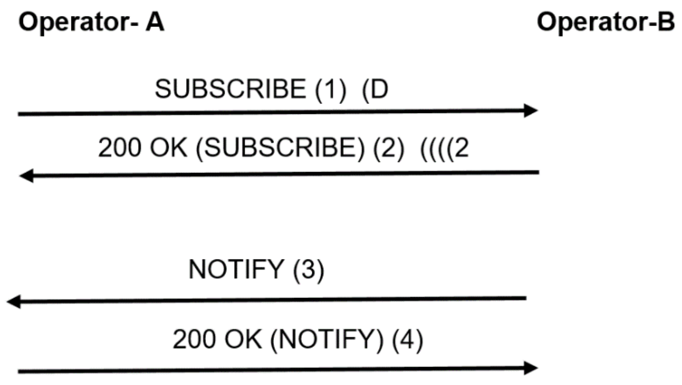


Figure 11 : Message Waiting

The message contents are as shown below.

<u>Message 1 – SUBSCRIBE</u>
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

<u>Message 2 – 200 OK (SUBSCRIBE)</u>

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:10.10.0.1:5060>

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=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 10.10.0.1:5060;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

Contact: <sip:10.0.0.1:5060>

Content-Length: 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 appendix 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.

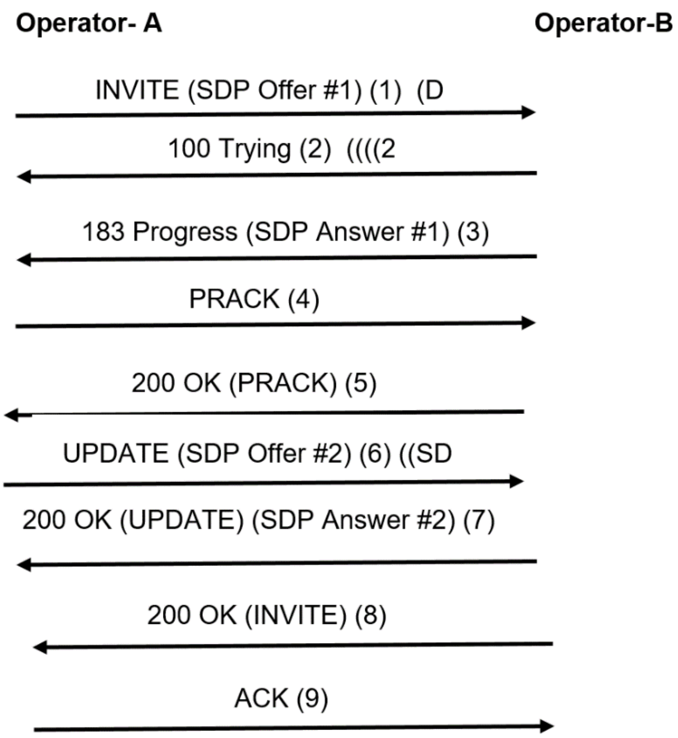


Figure 12: Ad-Hoc Conference (Invoking the Conference-Factory)

The above message sequence is very similar to that in appendix 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 appendix B.1 are shown below.

<u>Message 1 – INVITE (SDP Offer #1)</u>

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

From: <sip:+447960306800@operatorB.com;user=phone>;tag=123456

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

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

<u>Message 2 – 100 Trying</u>

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.

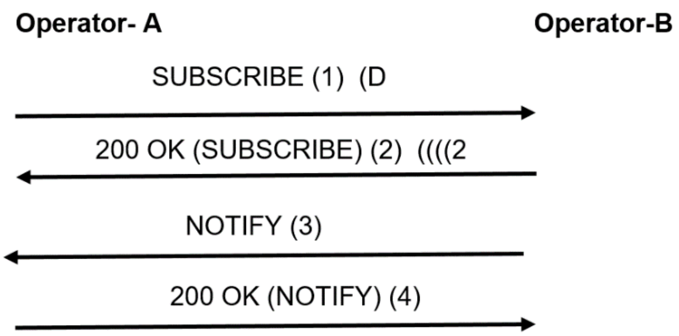


Figure 13 : Conference Package Subscription

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=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: 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

<u>Message 2 – 200 OK (SUBSCRIBE)</u>

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-ioi=operatorA.com; term-ioi=operatorB.com

Call-ID: abc1234588@operatorB.com

CSeq: 1 SUBSCRIBE

Event:conference

Expires: 7200

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

<u>Message 3 – NOTIFY</u>

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]

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=3467894a

Call-ID: abc1234588@operatorB.com

CSeq: 12 NOTIFY

Contact: <sip:10.0.0.1:5060>

Content-Length: 0

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.

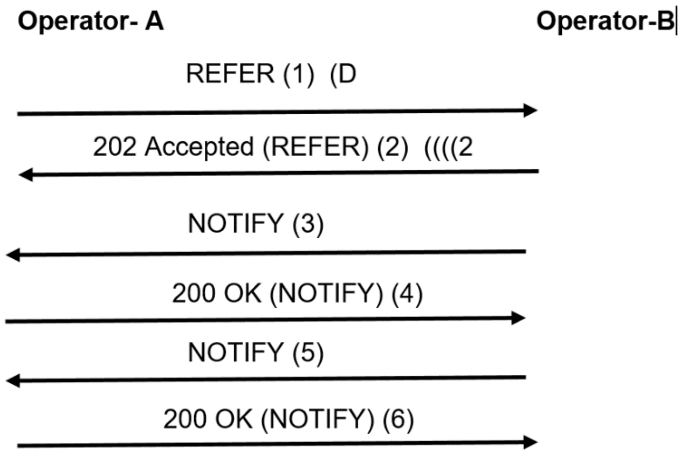


Figure 14 : Referring a User Into the Conference

The message contents are as shown below.

<u>Message 1a – REFER (to an existing party)</u>

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: norefersub

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: <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 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: +497789123456@operatorC.com;user=phone;method=INVITE>

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

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

Call-ID: 56789102@operatorB.com

CSeq: 1 REFER

Refer-Sub: false

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=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

Contact: <sip:10.0.0.1:5060>

Content-Length: 0

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:10.10.0.1:5060>

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.

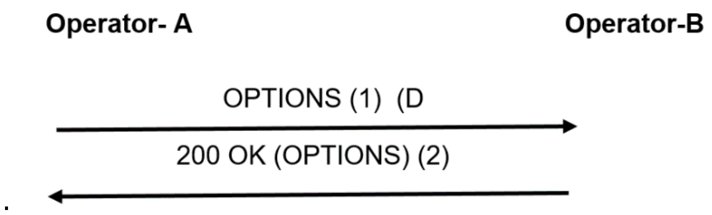


Figure 15 : RCS Capability Exchange

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+xm, application/resource-lists+xml, application/rlmi+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-service.ims.icsi.oma.cpm.session"; +g.3gpp.iari-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.rcs.ftthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7:3gpp-application.ims.iari.gsma-vs, urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopullft, urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopush", +g.3gpp.smsip

Route: sip:10.10.0.1:5060;lr

Contact: <sip:10.0.0.1:5060>; +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-service.ims.icsi.oma.cpm.filetransfer; urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.session"; +g.3gpp.iari-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.rcs.ftthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7:3gpp-application.ims.iari.gsma-vs, urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopullft, urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopush, urn%3Aurn-7%3A3gpp-application.ims.iari.rcse.sp ",+g.3gpp.smsip

Content-Length: 0

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: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

Accept-Contact: +g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel, urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.msg; urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"; +g.3gpp.iari-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.rcs.ftthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7:3gpp-application.ims.iari.gsma-vs",+g.3gpp.smsip

Contact: <sip:10.10.0.1:5060>; +g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel, urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.msg; urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg"; +g.3gpp.iari-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.rcs.ftthumb, urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-is, urn:urn-7:3gpp-application.ims.iari.gsma-vs", +g.3gpp.smsip

Content-Length: 0

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.

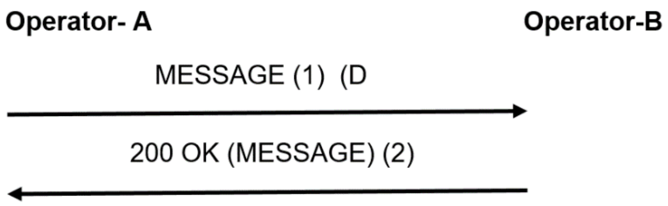


Figure 16 : RCS Messaging (Pager Mode)

The message contents are shown below:

<u>Message 1 – MESSAGE</u>

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;lr

Contact: <sip:10.0.0.1:5060>;+g.3gpp.icsi – ref ="urn%3Aurn-7%3gpp-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.

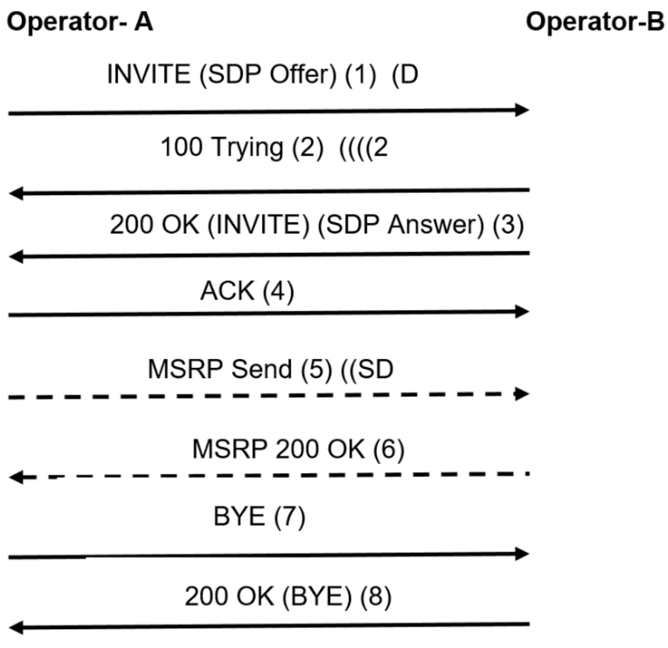


Figure 17 : CPM Message (Large Message Mode)

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer)</u>

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, NOTITY

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

Conversation-ID: 1234-5678-9abcd

Contribution-ID: 0012-3456-123

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

Contact: <sip:10.0.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.icsi.oma.cpm.largemsg"

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

<u>Message 3 – 200 OK (INVITE) (SDP Answer)</u>

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;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%3gpp-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. Blah blah blah ………blah.

</body>

-------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

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 appendix B.9. 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-appendix with references made to messages in appendix B.9.

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/jshA7weztas;tcp

a=sendonly

a=setup:active

a=file-selector:name:"My cool picture.jpg" type:image/jpeg size:32349

a=file-transfer-id:vBnG916bdberum2fF

a=file-disposition:render

Message 2 – 100 Trying

As appendix B.9.

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

<u>Message 4 – ACK</u>

As appendix B.9.

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

<u>Message 5 – MSRP SEND</u>

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: 1-2048/32349

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://10.10.1.1:6000/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 appendix B.9.

Message 8 – 200 OK (BYE)

As appendix B.9.

B.12 Appendix B.11 – 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 appendix 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 sub-appendix with references made to messages in appendix B.2.

Message 1 – INVITE (SDP Offer #1)

As appendix 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:10.0.0.1:5060>; +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;QCIF=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

<u>Message 2 – 100 Trying</u>

As appendix B.2.

<u>Message 3 – 183 Prpgress (SDP Answer #1)</u>

As appendix 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

a=conf qos remote sendrecv

b=AS: 768

<u>Message 4 – PRACK</u>

As appendix B.2.

<u>Message 5 – 200 OK (PRACK)</u>

As appendix B.2.

<u>Message 6 – UPDATE (SDP Offer #2)</u>

As appendix 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=fmtp: 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

<u>Message 7 – 200 OK (UPDATE) (SDP Answer #2)</u>

As appendix 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-parameter-sets=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 appendix 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 appendix 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 10 – ACK

As appendix B.2.

Message 11 – BYE

As appendix B.2.

Message 12 – 200 OK (BYE)

As appendix 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.iari.rcse.ft" and urn:urn-7:3gpp-application.ims.iari.rcs.ftthumb" indicate that either CPM or SIMPLE based FT is supported. This appendix 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 appendix B.9. 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-appendix with references made to messages in appendix B.9.

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer)</u>

………

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%3gpp-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

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

a=file-transfer-id:vBnG916bdberum2fF

a=file-disposition:render

a:file-icon:cid:icon12345

--boundary123

Content-Type: image/jpeg

Content-Transfer-Encoding: binary

Content-ID: <icon1234>

Content-Length: 200

Content-Disposition: icon

[..small preview icon...]

--boundary123--

<u>Message 2 – 100 Trying</u>

As appendix B.9

<u>Message 3 – 200 OK (INVITE) (SDP Answer)</u>

……..

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:10.10.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-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 *

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:name:"sunrise.jpg" type:image/jpeg size:60000

a=file-transfer-id:vBnG916bdberum2fF

Message 4 – ACK

As appendix B.9

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 appendix B.9.

Message 8 – 200 OK (BYE)

As appendix B.9.

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 appendix 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 appendix B.9. 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-appendix with references made to messages in appendix B.9.

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/jshA7weztas;tcp

a=sendonly

a=setup:active

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

a=file-transfer-id:vBnG916bdberum2fF

a=file-disposition:render

a:file-icon:cid:icon12345

--boundary123

Content-Type: image/jpeg

Content-Transfer-Encoding: binary

Content-ID: <icon1234>

Content-Length: 200

Content-Disposition: icon

[..small preview icon...]

--boundary123--

Message 2 – 100 Trying

As appendix B.9

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=-

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: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 appendix B.9

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://10.10.1.1:6000/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://10.10.1.1:6000/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 appendix B.9.

Message 8 – 200 OK (BYE)

As appendix B.9.

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.iari.rcse.im” indicates that either CPM or SIMPLE based IM/Chat is supported. This appendix covers the CPM case. The message flow and numbering for the RCS CPM based Chat is identical to appendix B.12. 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-appendix with references made to messages in appendix B.12.

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer)</u>

………

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

<u>Message 2 – 100 Trying</u>

As appendix B.12

<u>Message 3 – 200 OK (INVITE) (SDP Answer)</u>

……..

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%3gpp-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

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=sendrecv

a=setup:passive

Message 4 – ACK

As appendix B.12

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 appendix B.12.

Message 8 – 200 OK (BYE)

As appendix B.12.

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 appendix covers the SIMPLE case. The message flow and numbering for the RCS CPM based Chat is identical to appendix 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-appendix with references made to messages in appendix B.13.

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer)</u>

………

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

<u>Message 2 – 100 Trying</u>

As appendix B.13

<u>Message 3 – 200 OK (INVITE) (SDP Answer)</u>

……..

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 appendix B.13

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/jshA7weztas;tcp

Message-ID: 12339sdqwer

Byte-Range: 1-43

Content-Type: text/plain; charset=utf8

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 appendix B.13

Message 8 – 200 OK (BYE)

As appendix B.13.

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 appendix B.12), This appendix covers the CPM case. The message flow and numbering for the RCS CPM based Geolocation Push is identical to appendix B.12. 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-appendix with references made to messages in appendix B.12. The corresponding SIMPLE based Geolocation push flow may be derived by applying similar changes to the message contents in appendix B.13.

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer)</u>

………

Accept-Contact: +g.3gpp.iari-ref=”urn%3Aurn-7%3A3gpp-service.ims.iari.rcs.geopush”

Contact: <sip:10.0.0.1:5060>;+g.3gpp.iari - ref="urn%3Aurn-7%3gpp-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

<u>Message 2 – 100 Trying</u>

As appendix B.12

<u>Message 3 – 200 OK (INVITE) (SDP Answer)</u>

……..

Accept-Contact: +g.3gpp.iari-ref=”urn%3Aurn-7%3A3gpp-service.ims.iari.rcs.geopush”

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

Content-Type: application/sdp

Content-Length: 324

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
a=file-selector: type:application/rcspushlocation+xml size:500
a=file-transfer-id:vBnG916bdberum2fF

<u>Message 4 – ACK</u>

As appendix B.12

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

<u>Message 5 – MSRP SEND</u>

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: 2000-12-13T13:40:00-08:00

Subject: My Location Information

Content-Type: application/rcspushlocation+xml

Content-ID: <56789>

(location XML block – 500 octets)

-------d93kswow+

<u>Message 6 – MSRP 200 OK</u>

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 appendix B.12.

Message 8 – 200 OK (BYE)

As appendix B.12.

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 appendix B.12), This appendix covers the CPM case. The message flow and numbering for the RCS CPM based Geolocation Pull is identical to appendix B.12. 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-appendix with references made to messages in appendix B.12. The corresponding SIMPLE based Geolocation pull flow may be derived by applying similar changes to the message contents in appendix B.13.

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer)</u>

………

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%3gpp-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/rcspushlocation+xml

<u>Message 2 – 100 Trying</u>

As appendix B.12

<u>Message 3 – 200 OK (INVITE) (SDP Answer)</u>

……..

Accept-Contact: +g.3gpp.iari-ref=”urn%3Aurn-7%3A3gpp-service.ims.iari.rcs.geopullft”

Contact: <sip:10.10.0.1:5060>;+g.3gpp.icsi - ref="urn%3Aurn-7%3gpp-service.ims.iari.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://10.10.1.1:6000/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 appendix B.12

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 appendix B.12.

Message 8 – 200 OK (BYE)

As appendix B.12.

B.19 Social Presence (Overview)

The RCS Social Presence service consists of a number of discrete parts. To aid understanding, this appendix 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 appendix B.7),
- The publication of a Presence Information Document onto the Presence Server by the Presentity (see appendix B.19),
- The subscription of a Watcher to be notified of the Presence Information of the Presentity (see appendix B.20).
- The subscription by the Presentity to be notified of the list of Watchers associated with the Presentity's Presence Document (see appendix B.21) and to be able to verify that new watchers may be added to the list.

B.20 Publication of Social Presence Information.

This message interchange 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.

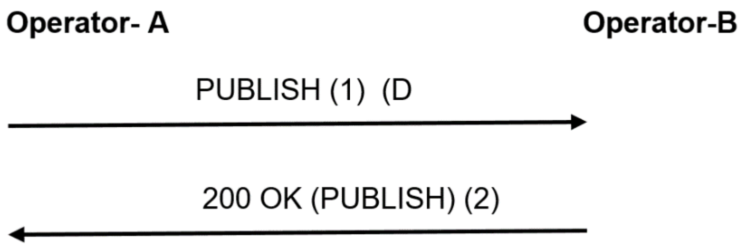


Figure 18 : Publication of Presence Information

The message contents are shown below:

<u>Message 1 – PUBLISH</u>

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=”abch+236788a”; orig-ioi=operatorA.com; term-ioi=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].

<u>Message 2 – 200 OK (PUBLISH)</u>

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 appendix B.21), a second notification is sent.

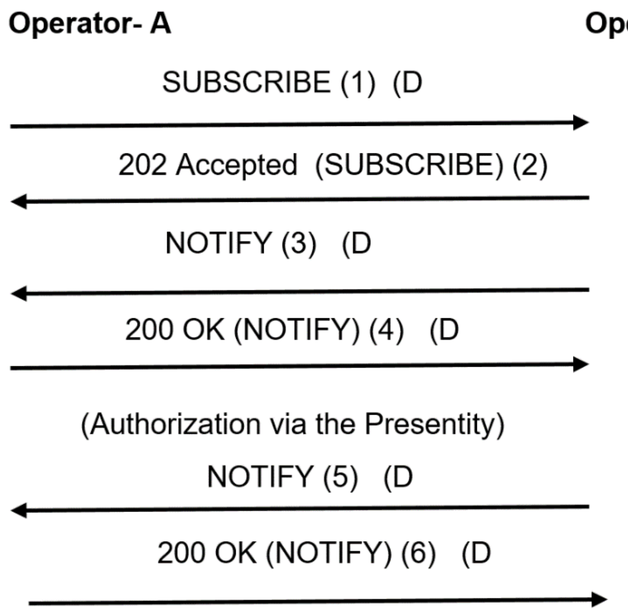


Figure 19 : Subscription to Presence Information

The message contents are shown below:

<u>Message 1 – SUBSCRIBE</u>

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

Via: SIP/2.0/UDP 10.0.0.1:5060;branch=12345678778

From: <sip:+397850316900@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-ioi=operatorA.com; term-ioi=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 @operatorA.com;user=phone>;tag=56ad7111

To: <sip:+447960306800@operatorB.com;user=phone>; tag=3467894a

P-Charging-Vector: icid-value="abch+236788a"; orig-ioi=operatorA.com; term-ioi=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 @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

Document Management

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>

Content-Type: application/pidf+xml

Content-Length: xx

[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

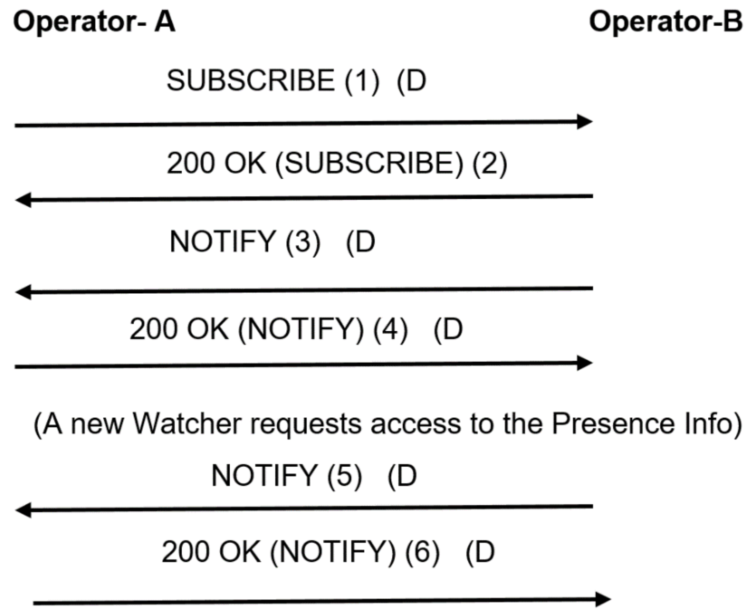
CSeq: 13 NOTIFY

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 appendix B.19 (and is not shown below).



.

Figure 20 : Subscription to Presence Watcher Information

The message contents are shown below:

<u>Message 1 – SUBSCRIBE</u>

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

Event: presence.winfo

Accept: application/watcherinfo+xml

Expires: 3600

Max-Forwards:70

P-Charging-Vector: icid-value=”abch+236788a”; orig-ioi=operatorA.com; term-ioi=operatorB.com
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=12345678778

From: <sip: <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:10.10.0.1:5060>

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: <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

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 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>;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

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=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

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 appendix B.7) 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]).

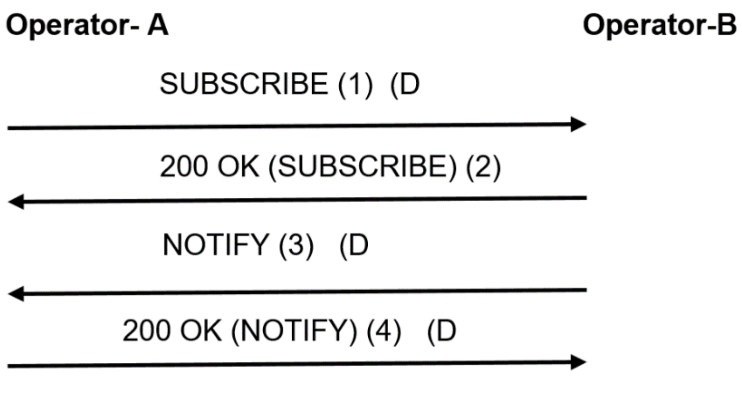


Figure 21 : Capability Discovery via Presence

The message contents are shown below:

<u>Message 1 – SUBSCRIBE</u>

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-ioi=operatorA.com; term-ioi=operatorB.com
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=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-ioi=operatorA.com; term-ioi=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 appendix 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 appendix 14). 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.

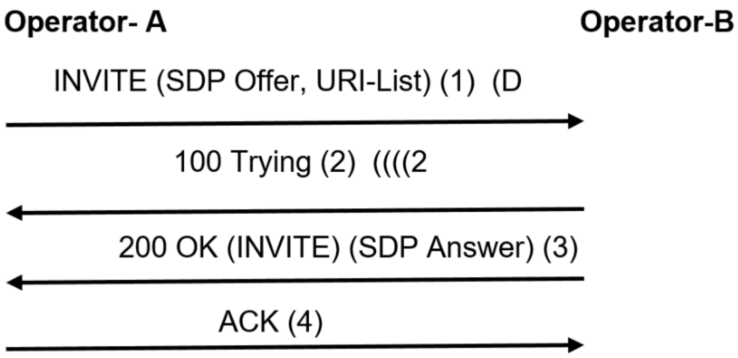


Figure 22 : Group Chat Establishment

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer, URI-List)</u>

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, NOTITY

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

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:msrp://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

   <?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>
   --boundary1—
```

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.

<u>Message 2 – 100 Trying</u>

SIP/2.0 100 Trying

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

Content-Length: 0

<u>Message 3 – 200 OK (INVITE)</u>

SIP/2.0 200 OK

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

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

<u>Message 4 – ACK</u>

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:10.0.1.1:5060;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 appendix 15) 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.

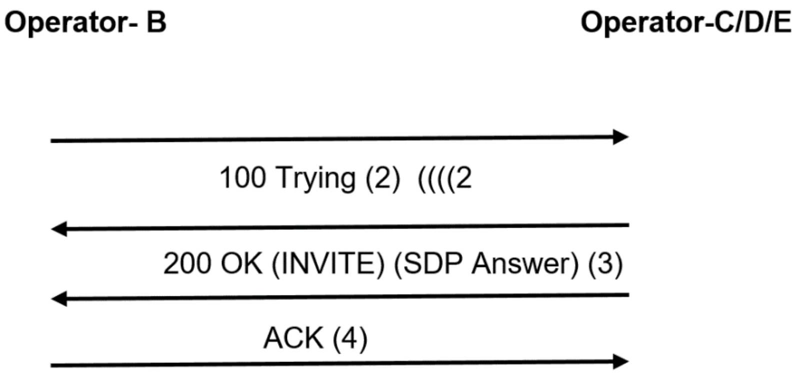


Figure 23 : Inviting a Target User into a Group Chat

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

<u>Message 1 – INVITE (SDP Offer, URI-List)</u>

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-ioi=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/jshA7weztas;tcp
a=sendrecv
a=setup:active
b=AS: 45
--boundary1
Content-Type: application/resource-lists+xml
Content-Disposition: recipient-list-history; handling=optional
  <?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" cp:copyControl="to" />
```

<entry uri="sip:anonymous@anonymous.invalid" cp:copyControl="to"
cp:count="1"/>

<entry uri="sip:+391234567890@operatorE.com" cp:copyControl="to"/>

</list>

</resource-lists>

--boundary1—

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@opertaorB.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;refresher=uac

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/abcA7wept654;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 appendix 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 userr (see appendix 14). 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.

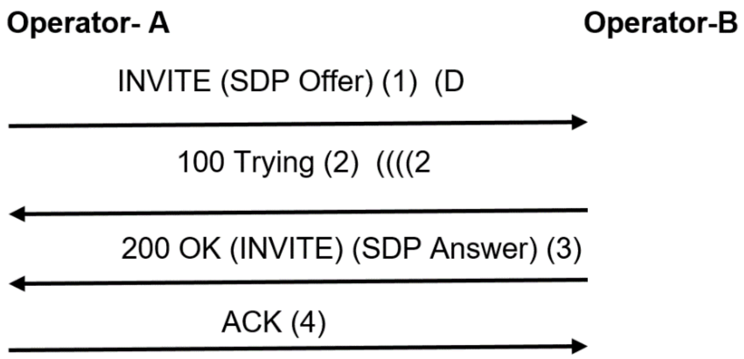


Figure 24 : Group Chat Establishment

The message contents are shown below:

<u>Message 1 – INVITE (SDP Offer, URI-List)</u>

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, NOTITY

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

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

a=sendrecv

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@opertaorB.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 10.0.0.1:5060;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

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

From: <sip:+447960306800@operatorB.com;user=phone>;tag=56784321

CSeq: 1 INVITE

Call-ID: dgh1234567@operatorB.com

Max-Forwards:70

Route: sip:10.0.1.1:5060;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 appendix 15) 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.

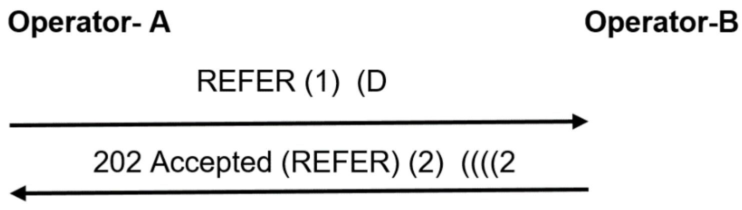


Figure 25 : Referring to a User In Conference

The message contents are as shown below, and cover both cases (moving the existing user and adding new users).

<u>Message 1a – REFER (to an existing party)</u>

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

<u>Message 2a – 202 ACCEPTED</u>

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).

<u>Message 1b – REFER (to new users)</u>

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

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>
```


<u>Message 2b – 202 ACCEPTED</u>

```
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 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 The List of Selected Option Items for the NNI (Informative)

This annex is provides a cross reference to annex C of 3GPP TS 29.165[1] which provides a list of items that are recommended to be selected by inter-operator agreements for the interconnection between IMS operators using the II-NNI. The applicability of option items are selected based on main body of this document. Table 34 provides the list of option items selected by this document.

The format table 34 is as follows :

- The first column is the option item as listed in annex C of 3GPP TS 29.165 [1],
- The second column provides a reference to the corresponding table in annex C of 3GPP TS 29.165 [1].
- The third column describes the applicability of the option item in this profile in terms of "yes" (applicable), "no" (not applicable) and "optional" (items that are outside the scope of this profile but may be supported by bilateral agreement),
- The fourth column provides further details / additional information including references to clauses in this document.

Option Item	Reference	Applicability	Details / Further info
Roaming II-NNI support	C.3.0.1	Yes	Roaming is supported in this profile. Local break out is assumed. RAVEL (Open Mobile Alliance) is applicable (subject to bilateral agreement between the HPLMN & VPLMN operators).
Non roaming II-NNI support	C.3.0.1	Yes	Interconnect NNI is supported.
INFO Method	C.3.1.1	Optional	See clause 4 of this profile.
MESSAGE Method	C.3.1.1	Yes	See clause 4 of this profile. Used for MMTEL and RCS (Rich Communications Services) services.
REFER Method	C.3.1.1	Yes	See clause 4 of this profile. Used for MMTEL and RCS services.
SIP Overload Control	C.3.1.A	No	
Feedback Control	C.3.1.A	No	
Event Control	C.3.1.A	No	
Negotiation of resource reservation (precondition)	C.3.1.2	Yes	See clause 10.1 of this profile.
SIP Session Timer	C.3.1.2.A	Yes	
Replacing SIP Dialogs (replaces)	C.3.1.3	Yes	Used for MMTel & RCS services.

Option Item	Reference	Applicability	Details / Further info
Session Participation (join)	C.3.1.4	No	Not required for MMTel / RCS services
Conveying capabilities of UE	C.3.1.5	Yes	
Authorization of early media.	C.3.1.5.A	Yes	P-Early-Media header is supported in this profile.
Managing the indication of the asserted service (P-Asserted-Service header field)	C.3.1.6	Yes	P-Asserted-Service header is supported in this profile – identifying MMTel and RCS services.
Overlap signalling (in-dialog / multiple INVITE)	C.3.1.7	No	Overlap signalling is not in scope. See clause 6.
MIME Type	C.3.1.7.A	Yes	See clause 8.
Limitation of maximum length of a SIP message body	C.3.1.7.B	Yes	See clause 8.
UDP	C.3.1.8	Yes	See clause 5.
TCP	C.3.1.8	Yes	See clause 5.
SCTP	C.3.1.8	Optional	See clause 5.
Speech media (m=audio)	C.3.1.9	Yes	Speech is supported (IR.92)
Video media (m=video)	C.3.1.9	Yes	Video is supported (IR.94, IR.90)
Other Media	C.3.1.9	Yes	Message & Text (T.140) are supported – see section 10.
RTP/AVPF	C.3.1.9	Yes	See clause 10. Used for video calls (IR.94)
Transmission Control Protocol	C.3.1.9	Yes	Used for MSRP support for RCS services.
Other user plane protocols	C.3.1.9	No	
DTMF Transport (telephone-event)	C.3.1.10	Yes	See clause 10.2.4.
DTMF Transport (SIP INFO mechanism)	C.3.1.10	No	
Subaddress ("isub" parameter)	C.3.1.10A	Optional	Not a mandatory IR.92 service
IPv4	C.3.1.11	Yes	See clause 11
IPv6	C.3.1.11	Yes	See clause 11
Malicious	C.3.1.12	Optional	Not a mandatory IR.92 service

Option Item	Reference	Applicability	Details / Further info
Communication Identification (MCID)			
Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR)	C.3.1.12	Yes	Mandatory IR.92 service
Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR)	C.3.1.12	Yes	Mandatory IR.92 service
Anonymous Communication Rejection (ACR)	C.3.1.12	Optional	Not a mandatory IR.92 service
Communication DIVersion (CDIV) excluding Communication Diversion Notification (CDIVN)	C.3.1.12	Yes	Mandatory IR.92 service
Communication Waiting (CW)	C.3.1.12	Yes	Mandatory IR.92 service
Communication HOLD (HOLD)	C.3.1.12	Yes	Mandatory IR.92 service
Message Waiting Indication (MWI)	C.3.1.12	Yes	Mandatory IR.92 service
Incoming Communication Barring (ICB)	C.3.1.12	Yes	Mandatory IR.92 service
Outgoing Communication Barring (OCB)	C.3.1.12	Yes	Mandatory IR.92 service
Completion of Communications to busy subscriber (CCBS)	C.3.1.12	Optional	Not a mandatory IR.92 service
Completion of Communications No Reply (CCNR)	C.3.1.12	Optional	Not a mandatory IR.92 service
Explicit Communication Transfer (ECT)	C.3.1.12	Optional	Not a mandatory IR.92 service
Customized Alerting	C.3.1.12	Optional	Not a mandatory IR.92 service

Option Item	Reference	Applicability	Details / Further info
Tone (CAT)			
Customized Ringing Signal (CRS)	C.3.1.12	Optional	Not a mandatory IR.92 service
Closed User Group (CUG)	C.3.1.12	Optional	Not a mandatory IR.92 service
Personal Network Management (PNM)	C.3.1.12	Optional	Not a mandatory IR.92 service
Three Party (3PTY)	C.3.1.12	Optional	Not a mandatory IR.92 service
Conference (CONF)	C.3.1.12	Yes	Mandatory IR.92 service (Ad-Hoc Conference).
Flexible Alerting (FA)	C.3.1.12	Optional	Not a mandatory IR.92 service
Announcements (during session establishment)	C.3.1.12	Yes	Both P-Early-Media and Alert-Info headers are supported in this profile.
Announcements (during established session)	C.3.1.12	Yes	Media stream can be modified mid call in this profile. Also, Call-Info header is supported.
Announcements (when communication request rejected)	C.3.1.12	Yes	All 3 cited options in 3GPP TS 29.165 are supported in this profile.
Advice of Charge (AOC)	C.3.1.12	Optional	Not a mandatory IR.92 service
Completion of Communications Not Logged In (CCNL)	C.3.1.12	Optional	Not a mandatory IR.92 service
Presence Service	C.3.1.12	Yes	IR.90 service
Messaging (Pager Mode)	C.3.1.12	Yes	IR.90 service
Messaging (Session Mode)	C.3.1.12	Yes	IR.90 service
Messaging (Session Mode Conferences)	C.3.1.12	Yes	IR.90 service
Delivery of original destination identity	C.3.1.12	No	
Other additional service using other SIP extensions	C.3.1.12	No	None identified in this profile
Optimal Media Routing	C.3.1.13	Yes	OMR is applicable to this profile for RAVEL
Applying Forking	C.3.1.13	Optional	See clause 10.1.
Transfer of IP multimedia service tariff information	C.3.1.13	Optional	

Option Item	Reference	Applicability	Details / Further info
m=line	C.3.1.14	Yes	See clause 10.4.
b=line	C.3.1.14	Yes	See clause 10.4.
a=line	C.3.1.14	Yes	See clause 10.4.
Public Safety Answering Point (PSAP) Callback	C.3.1.15	No	PSAP callback is not in IR.92
IMS AKA plus IPsec ESP	C.3.2.1	Yes	Aligns with IR.92
SIP digest plus check of IP association	C.3.2.1	Optional	
SIP digest plus Proxy Authentication	C.3.2.1	Optional	
SIP digest with TLS	C.3.2.1	Optional	
Inter-operator accounting	C.3.2.1A	Yes	
Inter-operator accounting for the transit scenario	C.3.2.1A	Yes	
The key of service profile for HSS query (P-Profile-Key header field)	C.3.2.2	Optional	Optional header – see clause 4.5.
Dial string ("user=dialstring " parameter)	C.3.2.3	No	
Communication Diversion Notification (CDIVN)	C.3.2.4	Optional	Not in IR.92
Unstructured Supplemnentary Service Data (USSD)	C.3.2.4	Optional	Not in IR.92
IMS Centralized Services (ICSI)	C.3.2.5	Yes	
PS to CS Single Radio Voice Call Continuity (SRVCC)	C.3.2.5	No	
Single Radio Video Call Continuity	C.3.2.5	No	
Inter-UE Transfer (IUT)	C.3.2.5	No	
CS to PS Single Radio Voice Call	C.3.2.5	No	

Option Item	Reference	Applicability	Details / Further info
Continuity (SRVCC)			
PS to CS Dual Radio Voice Call Continuity (DRVCC)	C.3.2.5	No	
CS to PCS Dual Radio Voice Call Continuity (DRVCC)	C.3.2.5	No	
Registration of bulk number contacts	C.3.2.6	No	
NOTIFY method	C.3.3.1	Yes	See clause 4. Used in IMS registration, MMTEL and RCS services
SUBSCRIBE method	C.3.3.1	Yes	See clause 4. Used in IMS registration, MMTEL and RCS services
PUBLISH method	C.3.3.1	Yes	See clause 4. Used for RCS.
Inter-operator accounting	C.3.3.2	Yes	
Inter-operator accounting for the transit sceenario	C.3.3.2	Yes	
Globally Routable User Agent URIs (gruu)	C.3.3.3	No	
Media Feature Tags	C.3.3.4	Yes	See Appendix B.
User to User Call Control Information in SIP for ISDN interworking (uui)	C.3.3.5	No	Applicable if transit traffic between fixed line IMS cores – which is not in scope of this profile.
Private network traffic (P-Private-Network-Indication header field)	C.3.3.6	Optional	
SIP URI	C.3.3.7	Yes	See clause 7.
Tel URI	C.3.3.7	No	See clause 7.
IM URI	C.3.3.7	No	See clause 7.
PRES URI	C.3.3.7	No	See clause 7.
Number Portability Routing Number ("rn" and "npdi" parameter)	C.3.3.7	Optional	
Calling Party's Category ("cpc" parameter)	C.3.3.7	Optional	
Originating Line	C.3.3.7	Optional	

Option Item	Reference	Applicability	Details / Further info
Information ("oli" parameter)			
Support of out-of-dialog OPTIONS method	C.3.3.8	Yes.	See clause 4. Applicable to RCS Capability Exchange

Table 1: Option Items Selection

Annex D Document Management

D.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
.1	7 November 2014	New DRAFT PRD submitted for IREG WG approval	IREG67	Mark McGinley, AT&T
1.0	16 February 2015	PSMC Approval (post PSMC #130)	PSMC	Mark McGinley, AT&T
2.0	22 February 2017	Implemented approved change requests: • IR.95 CR 1002 • IR.95 CR 1003 • IR.95 CR 1004 • IR.95 CR 1005 • IR.95 CR 1006 • IR.95 CR 1007	PSMC	Mark McGinley, AT&T
3.0	29 August 2017	Implemented approved change requests: • IR.95 CR 1008 • IR.95 CR 1009 • IR.95 CR 1010 • IR.95 CR 1011	TG (formerly, PSMC)	Mark McGinley, AT&T
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