

Nordic Endorsement of SIP-SDP Inter-IMS NNI Profile

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# Introduction

## Overview

This document describes an endorsement of GSMA PRD IR.95 (Inter-IMS SIP/SDP profile for the NNI between operators IMS networks) for the Nordic countries.

The document is structured as follows :-

* The summary of Selected Option Items at the NNI (see annex C of GSMA PRD IR.95)
* Endorsement Statement against the baseline text

# Summary of Selected Options at the NNI

This section 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 1 provides the list of option items selected by this document.

The format of table 1 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 | No | National LTE roaming not in scope |
| Non roaming II-NNI support | C.3.0.1 | Yes | Interconnect NNI is supported. |
| INFO Method | C.3.1.1 | No |  |
| MESSAGE Method | C.3.1.1 | Yes | Used for SMSoIP ~~and RCS Messaging services.~~ See IR.95 section 4~~.~~  ~~Applicable outside existing dialog.~~  ~~The content is application/vnd.3gpp.sms or message.cpim.~~ |
| REFER Method | C.3.1.1 | No | Not applicable to Interconnect NNI |
| 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 | Supported in IR.92 |
| SIP Session Timer | C.3.1.2.A | Yes | Possibly restrict range of timers, e.g. align with RFC 4028. |
| Replacing SIP Dialogs (replaces) | C.3.1.3 | Yes | Used for services such as CONF and adding a user to group chat |
| Session Participation (join) | C.3.1.4 | No | Not required for MMTel / RCS services |
| Conveying capabilities of UE | C.3.1.5 | Yes | Convey capabilities as per RFC 3840  Not applicable to the OPTIONS method in this profile. |
| Authorization of early media. | C.3.1.5.A | Yes | P-Early-Media header is supported across the interconnect NNI. See IR.95 section 10.4. |
| Managing the indication of the asserted service  (P-Asserted-Service header field) | C.3.1.6 | Yes | Identifies MMTel and RCS services across the NNI |
| Overlap signalling  (in-dialog / multiple INVITE) | C.3.1.7 | No | Overlap signalling is not in scope. See clause 6 of IR.95. |
| MIME Type | C.3.1.7.A | Yes | See clause 8 of IR.95. |
| Limitation of maximum length of a SIP message body | C.3.1.7.B | Yes | See clause 8 of IR.95. |
| UDP | C.3.1.8 | Yes | See clause 5 of IR.95. |
| TCP | C.3.1.8 | Yes | See clause 5 of IR.95. |
| SCTP | C.3.1.8 | ~~No~~ Yes | ~~Assumed not supported. See clause 5 of IR.95.~~ |
| 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 is supported – see section 10 of IR.95. |
| RTP/AVPF | C.3.1.9 | Yes | See clause 10 of IR.95. |
| 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 of IR.95. |
| DTMF Transport (SIP INFO mechanism) | C.3.1.10 | No |  |
| Subaddress  ("isub" parameter) | C.3.1.10A | No | Not a mandatory IR.92 service |
| IPv4 | C.3.1.11 | Yes | See clause 11 of IR.95. One or both of IP4/IP6 must be supported. This endorsement assumes IPv4 for both the user and control planes. |
| IPv6 | C.3.1.11 | No | See clause 11 of IR.95. |
| Malicious Communication Identification (MCID) | C.3.1.12 | No | Not a mandatory IR.92 service. |
| Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) | C.3.1.12 | Yes | Mandatory IR.92 service. Uses the P-Asserted-Id and Privacy headers. |
| Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) | C.3.1.12 | Yes | Mandatory IR.92 service. Uses the P-Asserted-Id and Privacy headers. |
| Anonymous Communication Rejection (ACR) | C.3.1.12 | Optional | Not a mandatory IR.92 service. Uses the P-Asserted-Id and Privacy headers. |
| Communication DIVersion (CDIV) excluding Communication Diversion Notification (CDIVN) | C.3.1.12 | Yes | Mandatory IR.92 service. Uses the History-Info header, cause codes in RFC 4458 and the 181 provisional response. If bi-laterally agreed, the DIVERSION header may be used as described in IETF RFC 5806 [91]) |
| Communication Waiting (CW) | C.3.1.12 | Yes | Mandatory IR.92 service. Uses the Alert-Info header in the 180 response. |
| Communication HOLD (HOLD) | C.3.1.12 | Yes | Mandatory IR.92 service |
| Message Waiting Indication (MWI) | C.3.1.12 | No | Mandatory IR.92 service but not relevant at the interconnect NNI |
| Incoming Communication Barring (ICB) | C.3.1.12 | Yes | Mandatory IR.92 service |
| Outgoing Communication Barring (OCB) | C.3.1.12 | ~~Yes~~No | Mandatory IR.92 service but not applicable to the I-NNI. |
| Completion of Communications to busy subscriber (CCBS) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Completion of Communications No Reply (CCNR) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Explicit Communication Transfer (ECT) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Customized Alerting Tone (CAT) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Customized Ringing Signal (CRS) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Closed User Group (CUG) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Personal Network Management (PNM) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Three Party (3PTY) | C.3.1.12 | No | 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 | No | 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 across the interconnect |
| Announcements (during established session) | C.3.1.12 | Yes | Media stream can be modified mid call in this profile. This uses the Call-Info header. |
| 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 | No | Not a mandatory IR.92 service |
| Completion of Communications Not Logged In (CCNL) | C.3.1.12 | No | Not a mandatory IR.92 service |
| Presence Service | C.3.1.12 | ~~Yes~~No | IR.90 service  Out of scope in this document |
| Messaging (Pager Mode) | C.3.1.12 | ~~Yes~~No | IR.90 service  Out of scope in this document |
| Messaging (Session Mode) | C.3.1.12 | ~~Yes~~No | IR.90 service  Out of scope in this document |
| Messaging (Session Mode Conferences) | C.3.1.12 | ~~Yes~~No | IR.90 service  Out of scope in this document |
| 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 IR.95 |
| Optimal Media Routing | C.3.1.13 | No | Related to roaming |
| Applying Forking | C.3.1.13 | ~~No~~Yes | See clause 10.1. |
| Transfer of IP multimedia service tariff information | C.3.1.13 | No |  |
| m=line | C.3.1.14 | Yes | See clause 10.4. of IR.95 |
| b=line | C.3.1.14 | Yes | See clause 10.4. of IR.95 |
| a=line | C.3.1.14 | Yes | See clause 10.4. of IR.95 |
| 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 | No | Applicable to roaming.  Not relevant to interconnect NNI. |
| SIP digest plus check of IP association | C.3.2.1 | No | Not relevant to interconnect NNI. |
| SIP digest plus Proxy Authentication | C.3.2.1 | No | Not relevant to interconnect NNI. |
| SIP digest with TLS | C.3.2.1 | No | Not relevant to interconnect NNI. |
| Inter-operator accounting | C.3.2.1A | No | No accouncting information sent across the NNI. |
| Inter-operator accounting for the transit scenario | C.3.2.1A | No | No accouncting information sent across the NNI. |
| The key of service profile for HSS query (P-Profile-Key header field) | C.3.2.2 | No |  |
| Dial string  ( "user=dialstring " parameter) | C.3.2.3 | No |  |
| Communication Diversion Notification (CDIVN) | C.3.2.4 | No | Not in IR.92 |
| Unstructured Supplemnentary Service Data (USSD) | C.3.2.4 | No | 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 | Not relevant to interconnect NNI. |
| Single Radio Video Call Continuity | C.3.2.5 | No | Not relevant to interconnect NNI. |
| Inter-UE Transfer (IUT) | C.3.2.5 | No | Not relevant to interconnect NNI. |
| CS to PS Single Radio Voice Call Continuity (SRVCC) | C.3.2.5 | No | Not relevant to interconnect NNI. |
| PS to CS Dual Radio Voice Call Continuity (DRVCC) | C.3.2.5 | No | Not relevant to interconnect NNI. |
| CS to PS Dual Radio Voice Call Continuity (DRVCC) | C.3.2.5 | No | Not relevant to interconnect NNI. |
| Registration of bulk number contacts | C.3.2.6 | No | Not relevant to interconnect NNI. |
| NOTIFY method | C.3.3.1 | ~~Yes~~ No | See clause 4 of IR.95.  Out of scope for interconnect NNI with no RCS services. |
| SUBSCRIBE method | C.3.3.1 | ~~Yes~~ No | See clause 4 of IR.95.  Out of scope for interconnect NNI with no RCS services. |
| PUBLISH method | C.3.3.1 | ~~Yes~~ No | See clause 4. IR.95. Used for RCS Social Presence .  Out of scope for this document (used for RCS Social Presence) |
| Inter-operator accounting | C.3.3.2 | No | No accouncting information sent across the NNI. |
| Inter-operator accounting for the transit sceenario | C.3.3.2 | No | No accouncting information sent across the NNI. |
| Globally Routable User Agent URIs (gruu) | C.3.3.3 | No |  |
| Media Feature Tags | C.3.3.4 | Yes | ~~See IR.95 clause 9~~. For this profile, the “audio” and “vdeo” feature tags are exchanged across the NNI. |
| 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 – see IR.95 clause 8.  Applicable to the legacy ISDN user-user service which is not in scope in this market. |
| Private network traffic  (P-Private-Network-Indication header field) | C.3.3.6 | ~~Optional~~  No | Not applicable to this document. |
| SIP URI | C.3.3.7 | Yes | See clause 7 of IR.95. |
| Tel URI | C.3.3.7 | Yes | See clause 7 of IR.95. |
| IM URI | C.3.3.7 | No | See clause 7 of IR.95. |
| PRES URI | C.3.3.7 | No | See clause 7 of IR.95. |
| Number Portability Routing Number  ("rn" and "npdi" parameter) | C.3.3.7 | Yes | See clause 7 of IR.95. |
| Calling Party’s Category  ("cpc" parameter) | C.3.3.7 | No |  |
| Originating Line Information  ("oli" parameter) | C.3.3.7 | No |  |
| Support of out-of-dialog OPTIONS method | C.3.3.8 | Yes | ~~Assumed used for RCS Capability Exchange~~  Shall be used for NNI heartbeat mechanism – see clause 4 of IR.95. TBC |

Table : Option Items Selection

# Endorsement Statement

The Endoresment Statement, based on Table I, uses the following schema :-

* Endorsed text is unchanged,
* Not-endorsed text is ~~struck through.~~ In addition, RCS releated functionality is also coloured in ~~red font as well as being struck out~~.
* Added text is underlined.

# GSMA PRD IR.95 Endorsement Text

# Introduction

## 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, Video Call and SMSoIP~~, and RCS 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.

## 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 NNI inter-operability. The requirements are derived from GSMA PRDs for the provision of IMS based voice and video ~~and RCS~~ services.

The VoLTE, Video Call and SMSoIP ~~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.84 – Video Share Phase 2 Interoperability Specification [7]~~
* ~~IR.79 – Image Share Interoperability Specification [8]~~
* ~~RCC.07 – RCS Advanced Communications Services & Client Specification [44]~~

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]~~

## Scope

This document specifies a SIP/SDP profile across the inter-IMS NNI in support of VoLTE, Video Call and SMSoIP~~, 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.

## Definition of Terms

| 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 |
| 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 |
| ITU-T | Telecoms Standardization Sector of ITU |
| 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 |
| VBD | Voice Band Data |
| XDM | XML Document Management |
| XML | Extended Mark-up Language |

## Document Cross-References

| Ref | Doc Number | Title |
| --- | --- | --- |
|  | 3GPP TS 29.165 R11 | Inter-IMS Network-Network Interface (NNI) |
|  | GSMA PRD IR.92 | IMS Profile for Voice and SMS |
|  | GSMA PRD IR.94 | IMS Profile for Conversational Video Service |
|  | GSMA PRD IR.65 | IMS Roaming NNI and Interworking Guidelines |
|  | GSMA PRD IR.90 | RCS Interworking Guidelines |
|  | GSMA PRD IR.88 | LTE Roaming NNI Guidelines |
|  | GSMA PRD IR.84 | Video Share Phase 2 Interoperability Specification |
|  | GSMA PRD IR.79 | Image Share Interoperability Specification |
|  | 3GPP TS 26.114 | IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction |
|  | ETSI TS 181 005 | Service & Capability Requirements |
|  | IETF RFC 3264 | An Offer/Answer Model with the Session Description Protocol (SDP) |
|  | IETF RFC 3261 | Session Initiation Protocol (SIP) |
|  | IETF RFC 4566 | Session Description Protocol (SDP) |
|  | IETF RFC 3262 | Reliability of Provisional Responses in the Session Initiation Protocol (SIP) |
|  | IETF RFC 3312 | Integration of Resource Management and Session Initiation Protocol (SIP) |
|  | IETF RFC 4028 | Session Timers in the Session Initiation Protocol (SIP) |
|  | 3GPP TS 29.238 | Interconnection Border Control Function (IBCF) – Transition Gateway (TrGW) interface, Ix Interface; Stage 3 |
|  | IETF RFC 3556 | SDP Bandwidth Modifiers for RTCP bandwidth |
|  | IETF RFC 3891 | The Session Initiation Protocol (SIP) “Replaces” Header |
|  | IETF RFC 3327 | Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts |
|  | IETF RFC 4488 | Suppression of Session Initiation Protocol (SIP) REFER Method Implicit Subscription |
|  | IETF RFC 4733 | RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals |
|  | GSMA PRD IR.67 | DNS/ENUM Guidelines for Service Providers and GRX/IPX Providers |
|  | ITU-T Rec. T.140 (1998) | Text Conversation Protocol for multimedia application, with amendment 1 (2000) |
|  | IETF RFC 4013 | RTP Payload for Text Conversation |
|  | 3GPP TS 33.210 | 3G Security; Network Domain Security (NDS); IP Network Layer Security |
|  | IETF RFC 4303 | IP Encapsulating Security Payload (ESP) |
|  | IETF RFC 5368 | Referring to Multiple Resources in the Session Initiation Protocol (SIP) |
|  | ITU-T Rec. H.263 (2005) | Video Coding for low bit rate communication |
|  | IETF RFC 3984 | RTP Payload format for ITU-T Rec. H.264 Video |
|  | 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” |
|  | IETF RFC 3551 | RTP Profile for Audio Video Conferences with Minimal Control |
|  | IETF RFC 4585 | Extended RTP Profile for Real Time Control Protocol (RTCP)-Based Feedback (RTP/AVPF) |
|  | OMA CPM | OMA CPM Conversation Functions (OMA-CPM-TS\_Conv\_Func-V1\_0-20120612-A) |
|  | IETF RFC 5939 | Session Description Protocol (SDP) Capability Negotiation |
|  | IETF RFC 4103 | RTP Payload for Text Conversation |
|  | 3GPP TS 29.079 | Optimal Media Routing within the IP Multimedia System (IMS); Stage 3 |
|  | IETF RFC 5621 | Message Body Handling in the Session Initiation Protocol (SIP) |
|  | IETF RFC 5547 | A Session Description Protocol (SDP) Offer/Answer Mechanism to enable File Transfer |
|  | IETF RFC 4483 | A Mechanism for Content Indirection in Session Initiation Protocol (SIP) Messages |
|  | IETF RFC 5438 | Instant Message Disposition Notification |
|  | IETF RFC 3842 | A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP) |
|  | IETF RFC 4575 | A Session Initiation Protocol (SIP) Event Package for Conference State |
|  | GSMA PRD RCC.07 | RCS Advanced Communications Services & Client Specification |
|  | 3GPP TS 29.163 | Interworking between IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched networks |
|  | IETF RFC 3966 | The Tel URI for telephone numbers |
|  | 3GPP TS 24.341 | Support of SMS over IP networks; Stage 3 |
|  | IETF RFC 4244 | An extension to the Session Initiation Protocol (SIP) for Request History Information |
|  | IETF RFC 4916 | Connected Identity in the Session Initiation Protocol (SIP) |
|  | IETF RFC 3680 | A Session Initiation Protocol (SIP) Event Package for registration |
|  | IETF RFC 3515 | The Session Initiation Protocol (SIP) Refer method |
|  | IETF RFC 3840 | Indicating User Agent Capabilities in the Session Initiation Protocol (SIP) |
|  | IETF RFC 3856 | A Presence Event Package for the Session Initiation Protocol (SIP) |
|  | IETF RFC 3857 | A Watcher-Information Event Template Package for the Session Initiation Protocol (SIP) |
|  | IETF RFC 3858 | An Extensible Mark-Up Language (XML) based format for Watcher Information |
|  | IETF RFC 3863 | Presence Information Data Format (PIDF) |
|  | IETF RFC 4479 | A data model for Presence |
|  | IETF RFC 4975 | The Message Session Relay Protocol (MSRP) |
|  | IETF RFC 5364 | Extensible Markup Language (XML) Format Extension for Representing Copy Control Attributes in Resource Lists |
|  | IETF RFC 5366 | Conference Establishment Using Request-Contained Lists in the Session Initiation Protocol (SIP) |
|  | IETF RFC 5368 | Referring to Multiple Resources in the Session Initiation Protocol (SIP) |
|  | 3GPP TS 29.658 | SIP Transfer of IP Multimedia Tariff Information; Protocol Specification. |
|  | 3GPP TS 24.337 | IP Multimedia (IM) Core Network (CN) Subsystem; IP Multimedia Subsystem (IMS) Inter UE Transfer; Stage 3 |
|  | 3GPP TS 24.237 | IP Multimedia (IM) Core Network (CN) Subsystem; IP Multimedia Subsystem (IMS) Service Continuity; Stage 3 |
|  | IETF RFC 5627 | Obtaining and using Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP) |
|  | IETF RFC 4412 | Communications Resource Priority in the Session Initiation Protocol (SIP). |
|  | 3GPP TS 24.628 | Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification |
|  | 3GPP TS 24.229 | IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP);Stage 3 |
|  | 3GPP TS 29.162 | Interworking between the IMS CN Subsystem and IP Networks |
|  | 3GPP TS 26.071 | Mandatory speech CODEC speech processing functions; AMR speech Codec; General description |
|  | 3GPP TS 26.090 | Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions |
|  | 3GPP TS 26.073 | ANSI C code for the Adaptive Multi Rate (AMR) speech codec |
|  | 3GPP TS 26.104 | ANSI-C code for the floating-point Adaptive Multi-Rate (AMR) speech codec |
|  | 3GPP TS 26.093 | Mandatory speech codec speech processing functions Adaptive Multi-Rate (AMR) speech codec; Source controlled rate operation |
|  | 3GPP TS 26.171 | Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; General description |
|  | 3GPP TS 26.173 | ANSI-C code for the Adaptive Multi-Rate – Wideband (AMR-WB) speech codec |
|  | 3GPP TS 26.190 | Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; Transcoding functions |
|  | 3GPP TS 26.193 | Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; Source controlled rate operation |
|  | 3GPP TS 26.204 | Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; ANSI-C code |
|  | 3GPP TS 23.228 | IP Multimedia Subsystem (IMS); Stage 2 |
|  | IETF RFC 7329 | A Session Identifier for the Session Initiation Protocol (SIP) |
|  | IETF RFC 6086 | Session Initiation Protocol (SIP) INFO Method and Package Framework |
|  | 3GPP TS 23.003 | Numbering, Addressing & Identification |
|  | IETF RFC 4694 | Number Portability Parameters for the "tel" URI |
|  | IETF RFC 3389 | Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN) |
|  | IETF RFC 4040 | RTP Payload Format for a 64 kbit/s Transparent Call |
|  | 3GPP TS 22.002 | Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN) |
|  | 3GPP TS 23.146 | Technical realization of facsimile group 3 (non-transparent) |
|  | ITU-T Rec. T.38 | Procedures for Group 3 Fax Communication over IP Networks |
|  | 8211-A356 | Address Formats for Swedish National SIP/SIP-I inteconnect |
|  | IETF RFC 5806 | Diversion Indication in SIP |

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

# Applicable Services

As stated previously, this profile supports VoLTE, Video Call and SMSoIP. ~~and RCS services. The RCS Services include all of the following:~~

* ~~Capability Exchange based on SIP OPTIONS (Note 1)~~
* ~~Capability Exchange via Presence (Note 1)~~
* ~~Social Presence Information based on OMA SIMPLE Presence and XML Document Management (XDM)~~
* ~~Chat based on OMA SIMPLE IM and Converged IP Messaging (CPM)~~
* ~~Standalone messaging based on OMA CPM~~
* ~~File Transfer based on OMA SIMPLE IM and CPM~~
* ~~Video Share based on GSMA PRD IR.84 [7]~~
* ~~Image Share based on GSMA PRD IR.79 [8]~~
* ~~Geo-location sharing based on OMA File Transfer and Location Application Programming Interface~~
* ~~Audio Messaging~~
* ~~Extension to Extension services~~

~~Note 1: If Capability Exchange via Presence is bilaterally agreed, then SIP OPTIONS need not to be supported for Capability Exchange, and vice versa. It is assumed that SIP OPTIONS is used in this market.~~

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

# IP Interconnection

IP interconnection is described in GSMA PRD IR.65 [4] and may be accomplished via IPX 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 profile across the NNI. The IMS 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 and TrGWs in the control and media planes respectively.



1. : Inter IMS NNI Reference Architecture

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

# 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.
* Not Applicable means that the method is not used at the NNI. If received, see section 4.1 for handling

| Method | Status | Additional Information |
| --- | --- | --- |
| INVITE | Mandatory | Includes both the initial INVITE and any subsequent re-INVITE |
| ACK | Mandatory |  |
| BYE | Mandatory |  |
| CANCEL | Mandatory |  |
| OPTIONS | Mandatory | ~~May be used~~ ~~Used for RCS Capability Exchange~~. ~~May~~ ~~also~~ Shall be used as a heartbeat mechanism on the NNI. - TBC |
| INFO | ~~Optional~~  Not Applicable | ~~May be used across the Roaming NNI for USSI (USSD over IMS).~~  ~~May also be used for MMTel features not included in GSMA PRD IR.92 [2] if bilaterally agreed (e.g. AOC, MCID 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~~  Optional | Used for SMS over IP~~.~~ (if mutually agreed between Operators)  ~~Also used for RCS messaging (pager mode and IMDN)~~ |
| NOTIFYh | ~~Mandatory~~  Not Applicable | ~~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 may be generated in response to any non-100 provisional response to an INVITE message specifying the ‘100rel’ option tag in a Require ~~Supported~~ header.  Note that this may only ~~be specified~~ occur 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~~  Not Applicable | ~~Used for RCS Social Presence and applicable at the roaming NNI only.~~ |
| REFER | ~~Mandatory~~  Not Applicable | ~~Used for MMTel services (e.g. CONF, ECT) and RCS Group Chat.~~ |
| REGISTER | ~~Mandatory~~  Not Applicable | ~~Applicable for roaming NNI only.~~ |
| SUBSCRIBE | ~~Mandatory~~  Not Applicable | ~~Applicable for roaming NNI for “regevent”~~  ~~Also used Used at the interconnect NNI for RCS Social Presence, MMTel services (e.g. CONF status), RCS Group Chat and Capability Exchange via Social Presence.~~ |
| UPDATE | Mandatory | Used for offer/answer exchange, session timer refresh etc. |

1. : Applicable SIP Methods

## 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.

## 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.

## SIP Header Handling

### 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].

### 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.

## 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] | | |

1. : Supported SIP Headers (Overall)

### 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 |
| --- | --- | --- | --- |
|  | P-Asserted-Identity | Trusted | Trust is mandatory at the roaming NNI |
|  | P-Access-Network-Info | Trusted | Should be trusted at the roaming NNI, even if “Not Trusted” at the non-roaming NNI |
|  | Resource-Priority | Not Trusted |  |
|  | History-Info | Trusted |  |
|  | P-Asserted-Service | Trusted |  |
|  | P-Charging-Vector | Trusted |  |
|  | P-Charging-Function-Addresses | Not Trusted |  |
|  | P-Profile-Key | Not Trusted | Shall always be “Not Trusted” at a non-Roaming NNI |
|  | P-Private-Network-Indication | Not Trusted |  |
|  | P-Served-User | Trusted  Not Trusted | Shall always be “Trusted” at the Roaming-NNI  Shall always be “Not Trusted” at a non-Roaming NNI |
|  | Reason | Trusted |  |
|  | P-Early-Media | Trusted |  |
|  | Feature-Caps | Trusted | Trust is mandatory at the roaming NNI |

1. : Guidelines for trust relationship for SIP headers at the II-NNI

## 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.

### 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.~~ There are no additional headers.

| 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~~ |
| ~~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]~~ | |

1. : 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.

### 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. |

1. : IBCF Header Manipulation

# SIP Message Transport

Both UDP and TCP 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 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].

# SIP Signalling Mode

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

# Numbering & Addressing

The routing of SIP signalling 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]. ~~Three scenarios are possible for outgoing SIP sessions:~~

The SIP-URI shall comprise a user part containing the E.164 number format (i.e. including the country code +45 or +46) and the user=phone URI parameter. In addition, the domain is recommended to be constructed based on the MNC and MCC of the target network as described in section 13.2 of 3GPP TS 23.003 [83]. Alternately, the domain names used at the IMS NNI may also be any valid domain name (based on normal FQDN rules) and based on mutual agreement between the interconnecting Operators as described in GSMA PRDs IR.65 [4] and IR.67 [23]. .Example valid SIP URIs would be :-

* <SIP:+45XXXXXXXXXX@ims.mncBBB.mccAAA.3gppnetwork.org;user=phone>
* <SIP:+46XXXXXXXXXX@ims.mncBBB.mccAAA.3gppnetwork.org;user=phone>
* <SIP:+47XXXXXXXXXX@ims.mncBBB.mccAAA.3gppnetwork.org;user=phone>
* <SIP:+45XXXXXXXXXX@OperatorName.dk;user=phone>
* <SIP:+45XXXXXXXXXX@OperatorName.com;user=phone>
* <SIP:+46XXXXXXXXXX@OperatorName.se;user=phone>
* <SIP:+46XXXXXXXXXX@OperatorName.com;user=phone>
* <SIP:+47XXXXXXXXXX@OperatorName.no;user=phone>
* <SIP:+47XXXXXXXXXX@OperatorName.com;user=phone>

where, XXXXXXXXXX represents the MSIDN digit string of the target subscriber/UE, BBB is the MNC Number of the target Operator and AAA the MCC Number of the target Operator.

The TEL URI shall comprise the E164 number. Example valid TEL URIs would be :-

* <TEL:+45XXXXXXXXXX>
* <TEL:+46XXXXXXXXXX>
* <TEL:+47XXXXXXXXXX>

where, XXXXXXXXXX represents the MSIDN digit string of the target subscriber/UE.

For Number Portability, it is assumed that the existing solution in CS shall be re-used whereby the originating Network does a data dip to its Operator specific database (i.e. no National Number Portability database in Denmark nor Sweden) and then forward the call to the ported-to Operator. The solutions differ between Denmark and Sweden.

Denmark

The SIP Request-URI shall be constructed in two steps as follows:-

1. Construct a TEL-URI as described in IETF RFC 4694 [84]
2. Optionally, convert the TEL-URI to a SIP-URI as described in section 19.1.6 of IETF RFC 3261 [12].

As an example, consider a request to a subscriber number +45XXXXXXXXXX which has been ported to another Operator (with MCC=AAA and MNC=CCC). Also, assume that the Number Portability Routing Number is +45YYYYYYYYYY. In this case, the TEL URI is firstly created as per IETF RCF 4696 [84]:-

* <TEL:+45XXXXXXXXXX;npdi;rn=+45YYYYYYYYYY>

This TEL URI is then converted to a SIP URI as described in IETF 3261 [12] to yield :-

* <SIP:+45XXXXXXXXXX;npdi;rn=+45YYYYYYYYYY@ims.mncCCC.mccAAA.3gppnetwork.org>; user=phone

Sweden

The SIP Request-URI shall be constructed in two steps as follows:-

1. Construct a TEL-URI as described in the Swedish national standard [90]
2. Optionally, convert the TEL-URI to a SIP-URI as described in section 19.1.6 of IETF RFC 3261 [12].

As an example, consider a request to a subscriber number +46XXXXXXXXXX which has been ported to another Operator (with MCC=AAA and MNC=CCC). Also, assume that the Number Portability Routing Number of the target Operator is YYY. In this case, the TEL URI is created as follows :-

* [TEL:+46394YYYXXXXXXXXXX](TEL:+46394YYYXXXXXXXXXX;npdi)

This TEL URI is then converted to a SIP URI as described in IETF 3261 [12] to yield :-

[SIP:+46394YYYXXXXXXXXXX @ims.mncCCC.mccAAA.3gppnetwork.org](SIP:+46394YYYXXXXXXXXXX%20@ims.mncCCC.mccAAA.3gppnetwork.org); user=phone

* ~~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.~~

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 to be 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).

# SIP Message Bodies

~~3GPP TS 29.165 [1] states that the MIME 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 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.~~  Used in this profile to carry multiple message bodies for SDP and PSTN XML. |
| ~~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. Thus Not Applicable.  ~~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.~~ |
| ~~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 legacy bearer/ISDN information (see 3GPP TS 29.163 [45] annex F). This is ~~conditionally~~ supported where IMS is used ~~as a transit network to connect CS-networks~~ to support legacy bearer services.  Use of this message body enables a SIP based interconnect to functionally match a SIP-I based interconnect. For this market, it is assumed that only a SIP based interconnect will be used. |
| ~~message/sipfrag~~ | ~~Used to convey SIP session progress. This is conditionally supported if MMTel Ad-Hoc Conference / RCS Group Chat service is used.~~ Applicable only to the Roaming NNI. |
| ~~application/x-session-info~~ | ~~Used to convey additional digits in a SIP INFO for overlap sending. Not applicable to this profile.~~ |
| ~~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.~~ Applicable only to the Roaming NNI. |
| ~~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. 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).~~ Applicable only to the Roaming NNI. |

1. : 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.

# 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 RFC 4028 [16]. |
| 100rel | Enables reliable provisional responses – see RFC 3262 [14]. An example would be conveying SDP in a 18X response.  In this profile, an originating UE includes this tag in the SIP Supported header in the INVITE message. If the terminating UE supports and wishes to use reliable provisional responses, then this tag is included in a 18X response in the SIP Require header. |
| precondition | Enables negotiation of resource reservation for segmented QOS – see RFC 3312 [15].  In this profile, an originating UE includes the tag in the SIP Supported header. If the terminating UE also supports the precondition mechanism, then the tag is included in a 18X message in a SIP Require header. |
| ~~path~~ | ~~Used for the roaming NNI scenario as part of IMS registration – see RFC 3327 [20].~~ Applicable to Roaming NNI only. |
| replaces | Applicable to MMTel services CONF and ECT – see RFC 3891 [19]. |
| histinfo | Used in MMTel Call Forwarding – see 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]~~ |
| ~~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]. |

1. : SIP Option Tags

# 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.~~

## 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.

Each Operator at the NNI shall independently decide if preconditions are applicable to its respective Network. Where there is a mismatch in Operator requirments, the receiving Network at the NNI shall take any necessary remedial action. In practice, this means that a Network that does not want to use Preconditions may receive an INVITE message with SDP indicating the use/support of Preconditions. In this case, the ingress IBCF must remove preconditions from the message by removing both the SIP Option tag and the related SDP media attribites.

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 offer/answer exchanges are typically required to establish the bearer (e.g., INVITE/18x plus UPDATE/200 OK(UPDATE)).

Media flows may be subsequently modified within an existing SIP session via a new offer carried within a SIP re-INVITE or SIP UPDATE message.

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 seven (7) 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].

## 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 (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].~~

## 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-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 in-hand codec(s) are exclusive of a bi-laterally agreed codec list. See clause 10.2.5.

### Audio Codecs

For this profile, the following audio codecs are mandatory, in line with GSMA PRD IR.92 [3]:

* Adaptive Multi-Rate (AMR) speech codec (described in 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] and 3GPP TS 26.204 [79]).

In addition, to support interoperability with non-3GPP access inter-connect, the following audio codecs ~~are also recommended~~ are also mandated for this profile:

* G.711-A law (see IETF RFC 3551 [32])
* Comfort Noise (see IETF RFC 3389 [85])
* Clearmode (see IETF RFC 4040 [86])

NOTE 1:-The Comfort Noice codec would be applicable to sessions using the G.711 codec

NOTE 2:-The Clearmode codec is required to support legacy BS2X (Asynchronous Bearer Service) and BS3X (Synchronous Bearer Service). See 3GPP TS 22.002 [87].

~~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.~~

### Video Codecs

As mandated in GSMA PRD IR.94 [3], the following video codec is mandatory:

* 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

~~As mandated in GSMA PRD IR.84 [7], the following video codecs are mandatory:~~

* ~~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.263-2000 [29] profile 0 level 45. 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.~~

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

### Audio/Video Codec Negotiation/Handling at the NNI

The network shall have the ability to perform transcoding between the media types and codecs it supports. The network may implement these capabilities in various network elements; however, the remainder of this section assumes they are provided by the IBCF / TrGW. For this market, transcoding shall be the responsibility of the terminating network of a session traversing the NNI.

To facilitate interoperability, the IBCF shall not preclude other codecs being offered across the NNI. To this end, the following behaviour is applicable to this profile:

* SDP Offers containing other than mandatory codecs shall be sent across the NNI if present with at least one mandatory codec. In this case, the receiving Network may or may not recognise the non-mandatory codec(s) but shall convey the Offer to the terminating UE which may recognise the non-mandatory codec(s).
* SDP Offers containing zero mandatory codecs may be sent across the NNI based on Operator Preference/Policy.
* If an SDP Offer containing zero mandatory codecs is not permitted to be sent/received at the NNI, then it shall be rejected with a SIP final response of 606 “Not Acceptable” with SDP message body containing the supported mandatory codecs.

### ~~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].~~

### DTMF

DTMF digits 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].

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.

### Voice Band Data (VBD)

VBD covers legacy services which transport data in the frequency band of the narrowband voice spectrum. Such services may be categorized into three major applications areas: data modem, facsimile and text telephony. For this market, group 3 facsimile services are in scope as defined in 3GPP TS 23.146 [88].

Facsimile shall be transported via ITU-T Rec. T.38 ([89]) and clarified via annex K of 3GPP TS 29.163 [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.

## SDP Contents

SDP is defined in IETF RFC 4566 [13].

SDP usage shall be compliant with the Offer/Answer rules in IETF RFC 3264 [11]

NAT 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].

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]. Note that the version must be incremented in each each offer/answer exchange as described in IETF RFC 3264 [11].  - <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  (t-line) | The time "t=" line consists of two fields  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 and to “udptl” for image media.  The <fmt> is set to one or more RTP payload numbers for RTP/AVP ~~and to “\*” for TCP/MSRP~~ and to “t38” for udptl. .  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”~~  “m=image 1234 udptl t38” |
| 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.  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 dynamic payload numbers)  a= rtpmap: <payload type> <encoding name>/<clock rate> [/<encoding parameters>]  Example: “a=rtpmap:100 AMR-WB/16000”  “a=rtpmap:97 CLEARMODE/8000”  (for providing payload specific parameter)  a= fmtp:<format> <format specific parameters>  Example:- “a=fmtp: 100 octet-align 1”  (for defining packetization rate)  a= ptime: <time>  Example: a=ptime: 20  (for segmented QOS indication as per 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)~~  ~~a=tcap:1~~  ~~a=pcfg:1 t=1~~  ~~(for RTCP Feedback as per IETF RFC 4585 [33])~~  ~~a=rtcp-fb:98 nack rpsi~~  (for T38 Facsimile media, as annex K of 3GPP TS 29.163 [4])  Example:-  a=T38FaxVersion:2  a=T38MaxBitRate:14400  a=T38FaxFillBitRemoval:FALSE  a=T38FaxTranscodingMMR:FALSE  a=T38FaxTranscodingJBIG:FALSE  a=T38FaxRateManagmwent:transferredTCF  a=T38FaxMaxBuffer:1800  a=T38FaxMaxDatagram:150  a=T38FaxUdpEC:t38UDPRedundancy |
| 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 RS / RR.  ~~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” |

1. : SDP Contents

## 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.

# 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.~~

For this market, IPv4 shall be used and IPv6 may be used if bilaterally agreed.

~~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].~~

# Document Management

## Document History

|  |  |  |  |  |
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| Version | Date | Brief Description of Change | Approval Authority | Editor / Company |
| 0.1 | 17/02/2016 | Endorsement Notice for IR.95 for the Danish Market | N/A | Wayne Cutler, GSMA |
| 0.2 | 03/05/2016 | Updated following comments from Operators and F2F meeting on 20th April. | N/A | Wayne Cutler, GSMA |
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| 1.2 | 20/01/2017 | Changes due to email comments from Swedish Operators. Scope of document changed to Nordic countries only. | N/A | Wayne Cutler, GSMA |
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