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IMS interconnect specification Telia Sweden

Description
Specification of technical details for IMS based SIP interconnect to be applied between Telia and other operators in Sweden

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1 Introduction

Telia and other operators are gradually more widely using IMS for communication service execution. As part of this evolution, also SIP interconnect will be based on IMS interfaces and standards.

This document specifies the technical details of this IMS based SIP interconnect to be applied between Telia and other operators in Sweden.

Intention is also to put the specification in the context of existing other networks than IMS, e.g. PSTN and PLMN and by that hopefully serve as input for dialogue as part of migration that goes beyond the specific details of the Inter IMS NNI.

1.1 Purpose

Purpose of this document is to specify the requirements for SIP and SDP interconnect to be used between Telia and other operators in Sweden for session-based communication.

1.2 Scope

This document covers the SIP Interconnect between Telia and other operators in Sweden for voice services. It is based on following Inter IMS Interconnect (II-Interconnect) specifications:

- GSMA IR.95, “SIP-SDP Inter-IMS NNI Profile"
- 3GPP TS 29.165, “Inter-IMS Network to Network Interface (NNI)”

Only voice services are in scope for this interconnect and only for the non-roaming II-NNI.

Even though the SIP interconnect specified in this document is based on the II-Interconnect NNI specifications, SIP is to large extent an end-to-end protocol and it is acknowledged by Telia that there will be various endpoints involved in this interconnect, also non-IMS based.

This is illustrated in Figure 1.

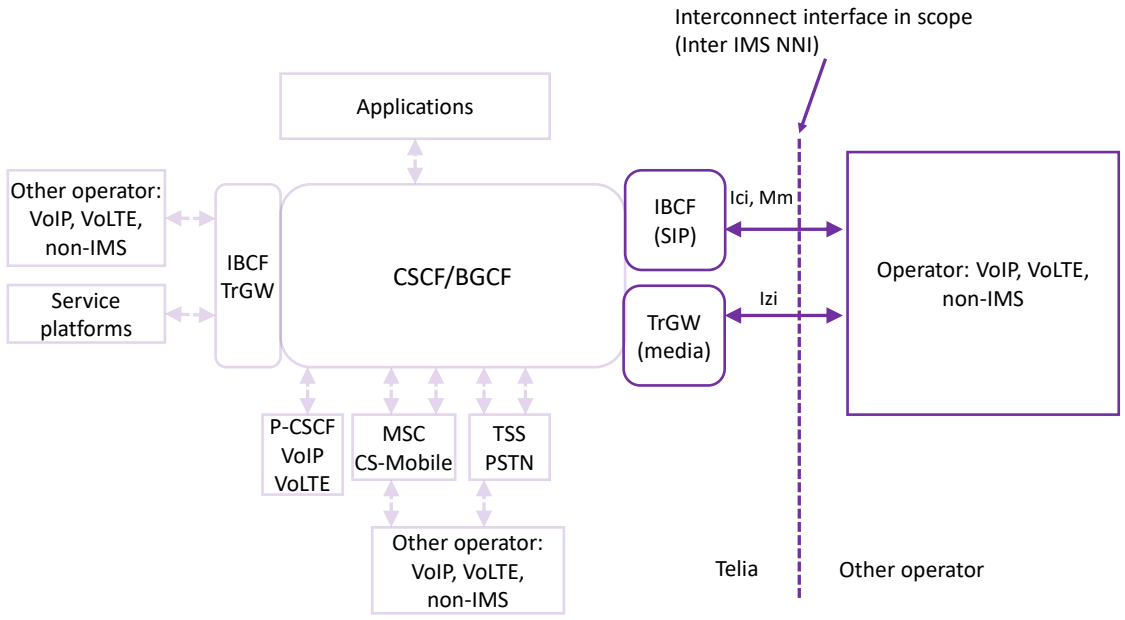

Figure 1: SIP interconnect and surroundings

1.3 Limitations (out of scope)

The following is not in scope for the SIP interconnect described in this document:

- SIP-I interconnect
- Roaming NNI
- RCS
- SMS

SIP over TLS is for the time being not supported (nor SRTP).

1.4 Target

This document serves as input to operators that will connect to Telia IMS based interconnect.

2 IP

This document holds information regarding SIP, SDP and RTP/RTCP. Details related to IP is described in *"8211-A 355"*. That document includes information related to e.g.:

- Physical network/connection points
- VPN details
- IP-version
- MTU

- TCP/UDP port range
- QoS/DSCP

3 SIP signalling

3.1 IP and connectivity

Both UDP (User Datagram Protocol) and TCP (Transmission Control Protocol) transport are mandatory and shall be supported. If UDP is the transport of choice, then TCP shall be used for large messages to avoid fragmentation as described in clause 18.1.1 of IETF RFC 3261.

OPTIONS ping:
Support for this is required and valid responses shall be provided if requests are received. If no response or SIP response code 503 is received from remote peer, this specific IP-address should be considered unavailable and should not be used for other requests than OPTIONS ping. Reason for no response or SIP response code 503 could be either due to network issues or for maintenance reasons.

Initial INVITES:
It is expected that only sip error response code 503 shall generate re-try (to other IP-address) at the Ici/Mm interface.

3.2 SIP methods

Following methods shall always be allowed:

- INVITE
- ACK
- BYE
- CANCEL
- OPTIONS
- PRACK
- UPDATE

3.3 SIP header fields and parameters

History-Info header is mandatory to be supported and shall be included in INVITE if call is forwarded.

Diversion header may be included in addition to History-Info

The use of P-Early-Media header shall be supported and used to indicate how the flow of early media is enabled or disabled.

Session timer may be used

The x- Header, X-From-Abroad: true, may be used to indicate call from abroad that is routed onwards to another national interconnect operator.

3.4 SIP body content

Multipart body may appear and therefore needs to be accepted
E.g., the use of PIDF in emergency VoLTE calls.

3.5 SDP and supported codecs

Whitelisted codecs:

Following codecs are allowed. Any set of them may show up in SDP offer and are possible to use depending on support at media end points.

- Audio
  - evs
  - amr-wb
  - amr
  - gsm-efr
  - g722
  - g723
  - g726-40
  - g728
  - g729
  - pcma
  - pcmu
  - telephone-event
  - clearmode
- Video
  - h263-1998
  - h263-2000
  - mp2p
  - mp2t
  - mp4v-es
- Image
  - t38
- Text
  - t140
  - red

Guaranteed codecs:

In case offer includes none-WB-codec, following codecs needs to be included in initial offers (transcode when needed):

- amr
- pcma

In case offer includes any WB-codec, following WB-codecs needs to be included in initial offers (transcode when needed):

- g722
- amr-wb

3.6 Numbering, naming and addressing

Subscriber number information should be used and transferred in accordance with ITS Application Guide 32.

4 Media

RTCP shall always be sent for a session with negotiated RTP stream, also when media are put on hold.

5 Terms and abbreviations

Terms and abbreviations	Definition
BGCF	Break-out Gateway Control Function
IBCF	Interrogating Border Control Function
II-Interconnect	Inter IMS Interconnect
II-NNI	Inter IMS Network to Network Interface
OIP	Originating Identity Presentation
OIR	Originating Identity Restriction
PIDF	Presence Information Data Format
PIDF-LO	Presence Information Data Format Location Object
RCS	Rich Communication Suite
RTP	Real Time Protocol
RTCP	Real Time Control Protocol
RTT	Real Time Text
SRTP	Secure Real Time Protocol
TLS	Transport Layer Security
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SMS	Short Message Services
WB Codec	Wide-Band Codec

6 References

Appendices, document ID	Name or web link of the document
GSMA IR.95	“SIP-SDP Inter-IMS NNI Profile"
3GPP TS 29.165	“Inter-IMS Network to Network Interface (NNI)”
8211-A 355	“IP interconnect interface for SIP/SIP-I”
ITS Application Guide 32	“Transfer of number information in national interconnections based on SIP and SIP-I”

7 Version history

Versions	Status	Date	Modified by	Comments
1.00	Approved	2022-04-05	Telia	Reviewed and approved
2.00	Approved	2025-01-24	Telia	Updates in 3.5 regarding media and payload types due to RTT, and in 3.6 due to release of ITS ApG32. Updates in 3.1 regarding Options ping.
3.00	Approved	2025-08-19	Telia	Updates in 3.3 regarding x- Header: X-From-Abroad