Finnish profile for SIP-I interworking

Viestintäviraston suosituksia

201/2014 S

Viestintävirasto
Kommunikationsverket
Finnish Communications
Regulatory Authority

Itämerenkatu 3 A
PL 313
00181 Helsinki
Puhelin 09 69 661
www.viestintävirasto.fi

Östersjögatan 3A
PB 313, FI-00181
Helsingfors, Finland
Telefon +358 9 69 661
www.kommunikationsverket.fi

Itämerenkatu 3A
P.O. Box 313, FI-00181
Helsinki, Finland
Telephone +358 9 69 661
www.ficora.fi

Contents

1 REFERENCES

- ITU-T Q.1912.5 Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part
- GSMA IR.34 Inter-Service Provider IP Backbone Guidelines
- GSMA IR.40 Guidelines for IPv4 Addressing and AS Numbering for GRX/IPX Network Infrastructure and User Terminals
- GSMA IR.77 Inter-Operator IP Backbone Security Requirements For Service Providers and Inter-operator IP backbone Providers
- 3GPP TR 29.802 (G)MSC-S - (G) MSC-S Nc Interface based on the SIP-I protocol
- 3GPP TS 23.231 SIP-I based circuit-switched core network; Stage 2
- 3GPP TS 29.231 Application of SIP-I Protocols to Circuit Switched (CS) core network architecture; Stage 3
- 3GPP TS 29.235 Interworking between SIP-I based circuit-switched core network and other networks
- 3GPP TS 29.163 Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks
- 3GPP TS 29.164 Interworking between the 3GPP CS Domain with BICC or ISUP as Signaling Protocol and external SIP I Networks
- FICORA's regulation 31 on technical aspects of charging in communications networks https://www.viestintavirasto.fi/attachments/maaraykset/Viestintavirasto31E2009M_EN.pdf

2 COMMON CONFIGURATION

- Called party numbers are presented in SUB format in ISUP messages and without plus (+) sign in SIP messages (example: sip:345678@domain;user=phone). Numbers contain the national number portability prefix.
- Calling party number format shall be agreed mutually between interconnecting operators (otherwise assumed to be presented in INT format).
- IP addressing is based on version 4. IP version 6 can be used based on mutual agreement of interconnecting parties.
- Each operator should implement DNS resolving method for FQDNs
- GRX domain name space (i.e. mncxxx.mccxxx.gprs) will be used for FQDNs
- See appendix A - List of IR.67 domain names

- It is assumed that the tests are to be run on each of the test platforms (SP A and SP B).
- The terminating operator is responsible to check validity/correctness of the originating number presentation format.
- Systems clocks must be synchronized correctly as defined by FICORA's regulation 31 on technical aspects of charging in communications networks.
- SIP OPTIONS messages are used to poll availability of call control entities in interconnected network. Call control entities shall respond to OPTIONS messages. It is recommended that call control entities are able to generate OPTIONS messages for polling availability of other call control entities.

3 INTERCONNECTION REQUIREMENTS & DIX

3.1 GENERAL

It is essential that the interconnecting operator advertises only valid networks to its peering partner. More precise information of valid IP's can be found from GSMA IR.40 and IR.34 documents. Default route, loopback and private IP address prefixes are explicitly denied advertisement.

Interconnecting operators are not allowed to advertise routes from one interconnected operator to another interconnected operator. IP addresses used in the peering environment are not allowed to be advertised outside of their own network, in any peering place or any peering connection. Traffic originating from the Internet is not allowed to be routed into the interconnection network.

It is expected that interconnecting operator shall implement proprietary route flapping rules towards themselves to ensure stability of peering connection and whole network infrastructure.

Multicast routes shall not be advertised to interconnection partners and multicast functionality should be disabled from network. Private IP addresses shall not be advertised on DIX broadcast LAN.

Interconnecting operators are not allowed to establish L2 connections between DIX switches.

3.2 PROTOCOLS

Interconnecting parties shall implement following list of settings to its peering router:

- Proxy ARP shall be disabled
- ICMP redirects are not allowed
- IP directed broadcasts are not allowed
- Spanning tree BPDUs are switched of
- Duplex and speed settings of port are not in auto negotiation (except Gigabit Ethernet interfaces)
- Only one globally unique MAC address is used per interface
- Discovery protocols (such as CDP and IRDP) are switched off
- Multicast is switched off
- Peering location IP prefixes shall not be advertised to peers

3.3 ADDRESSING

In DIX environment the link addressed between operators are granted by Suomen Numerot Numpac Oy (http://www.numpac.fi/). Numpac administers an IP address space that can be split into adequate amount of /31 subnets. Primary goal is to use IPv4 addressing, but IPv6 should also be supported.

A sequence number is allocated for each operator that is connected to DIX environment. For the operator with sequence number 1 (the first operator to connect), only one IP address is allocated from Broadcast LAN. For the operator with sequence number 2, Numpac grants the IP address to Broadcast LAN and additionally 1 x /31 address block (for interconnecting with operator 1). Third operator is allocated with IP address and 2 x /31 address blocks (for point to point interconnection with 2 other operators) - and so on. When two operators are starting to exchange traffic between themselves, it is the duty of operator with higher sequence number to allocate necessary IP addresses for the purpose from the granted address block. The IPv6 addressing follows the same scheme.

4 SERVICES

Service specific requirements.

4.1 VOICE CALL

Requirements for signaling level:

- SDP offer/answer mechanism according to Chapter 8
- No support needed for Comfort Noise (IETF RFC 3389)
- No support needed for initial or subsequent INVITE without SDP offer

Requirements for user plane level:

- ITU-T G.711 codec RTP payload format as defined in IETF 3551
- NB-AMR Two modes sets shall be supported in FR-AMR: set 0 (12.2, 7.4, 5.9, 475 kbps) and set 7 (12.2 kbps). This allows to negotiation of the mode set for AMR codecs.
- G.729 (For mobile-to-fixed or fixed-to-fixed scenarios only)
- GSM EFR
- Packetization period: 20ms
- Single sample / RTP packet

5 CODEC NEGOTIATION

Call control entities can negotiate a common codec. SDP offer/answer mechanisms are also used for DTMF.

The following codec negotiation scenarios shall be supported:

Originating call control entity:

- Single offer/answer scheme is used, when the first answer contains only one codec (selected codec SC):
  - Send SDP-offer (SCL1) Receive SDP-answer (SC)
- Second offer/answer is used, when the first answer contains multiple codecs (Supported Codec List SCL):
  - Send SDP-offer (SCL1) Receive SDP-answer (SCL2)

- o Send SDP-offer (SC) Receive SDP-answer (SC)

The second scenario corresponds to the case when the terminating call control entity sends multiple codecs to the originating call control entity. This is an optional behaviour for the terminating call control entity.

Terminating call control entity:

- Offer with codec list received Receive SDP-offer (SCL1) Send SDP-answer (SCL2) Receive SDP-offer (SC) Send SDP-answer (SC)

TrFO can work without multiple offer/answer even though it may benefit from it. Second offer/answer allows for both originating and terminating call control entity to know each other's codec capabilities thus in some scenarios allowing for a better/quicker codec selection in case of mid-call compressed codec to compressed codec change.

In-call codec change to G.711 shall always be possible to allow for in-band fax/modem scenarios.

Codecs of "auxiliary" payload type, i.e. RTP Telephony Event payload type (IETF RFC 2833/4733) or the comfort noise codec (IETF RFC 3389) shall not be considered as a factor when deciding single/multiple codec case as the only deciding factor is the number of speech-codecs.

The list of codecs that have been identified for support are shown below for information.

Payload Type Name	References	Applicable Codecs
audio/AMR	draft-ietf-avt-rtp-amr-bis-06.txt 3GPP TS 29.163 Annex B	all AMR codecs in 3GPP TS 26.103
audio/AMR-WB	draft-ietf-avt-rtp-amr-bis-06.txt 3GPP TS 29.163 Annex B	all AMR-WB codecs in 3GPP TS 26.103
audio/GSM	RFC 3551	GSM FR
audio/GSM-EFR	RFC 3551	GSM EFR
audio/PCMA	RFC 3551	G.711
audio/G723	RFC 3551	MUME audio option G.723.1
audio/G729 audio/G729D audio/G729E	RFC 3551	NGN codecs
audio/telephone-event	4733	DTMF
audio/CN	RFC 3389	comfort noise for CODECs that do not support as part of the CODEC itself such as G.711
image/T38	ITU-T Rec. T.38	G3 facsimile
audio/T38	ITU-T Rec. T.38	G3 facsimile

AMR mode sets shall be agreed mutually between interconnecting parties.

6 ISUP VERSION

ISUP according to ITU-T Q.761-Q.764 -97 is recommended to be used. At least ISUP according to ITU-T Q.761-Q.764 -93 "White book" should be used to have support for the defined supplementary services. Inbuilt Parameter compatibility mechanism helps overcoming the issues with possible national ISUP versions.

Finnish ISUP v3 according to SFS 5869:en specification shall be used, where applicable.

7 SIP PROFILE DEFINITION

Minimum set of features used in SIP-I signaling:

- Support for 100rel/PRACK is required (IETF RFC 3262) Note: optional in Q.1912.5 Profile C
- Support for UPDATE is required (depending on use case, e.g. to test mid-call codec negotiation) (IETF RFC 3311)
- Support for INFO method (IETF RFC 2976) is required (used for ISUP messages that does not cause state changes such as notifications)
- Support for SIP session timers (session expires header) is optional (IETF RFC 4028) Note: not covered by Q.1912.5 Profile C
- Support for privacy mechanism (IETF RFC 3323) and privacy header (Appendix C.2 in Q.1912.5) is required (use with CLIP)
- Also the following RFCs are supported as defined in Q.1912.5 (note: latest RFCs listed here)
  - RFC 2046 (MIME Media Types)
  - RFC 4566 (SDP)
  - RFC 3966 (tel URI)
  - RFC 3204 (MIME media types for ISUP and QSIG Objects)
  - RFC 3261 (SIP)
  - RFC 3264 (Offer/Answer Model with SDP)
  - RFC 3326 (Reason Header Field for SIP)

The basic SIP as defined by IETF RFC 3261 does not include a "keep alive" mechanism. As such, it is possible that one end of a session may fail and be unable to signal the release of the session. Therefore it is recommended that Call control entities support the SIP Session Timer as specified in IETF RFC 4028 as a means to determine whether a SIP session is still active by attempting to perform a session refresh, and therefore as a means to know when resources may be released if one end of the session fails. During call origination each Call control entity negotiates the use of the SIP Session Timer. It is recommended to that session refresh request is an UPDATE request without SDP.

A call control entity not supporting the SIP Session Timer shall accept incoming UPDATE requests without SDP in order to allow the remote node to use the SIP Session Timer. In addition a call control entity not supporting the SIP session timer must provide other mechanisms to detect that a session has failed.

8 DTMF INTERWORKING

DTMF signaling via the RTP telephony-event according to IETF RFC 4733 shall be supported for compressed codecs and is recommended for PCM. In-band DTMF signaling (generation, detection) shall also be supported to ensure open interoperability between two Call control entities from different vendors, when terminating Call control entity selects PCM and does not offer RTP telephony-event in the answer.

A Call control entity initiating an SDP offer shall include the MIME type "telephone events" with default events in the first SDP offer, if it offers any codecs other than the default PCM codec. An Call control entity initiating an SDP offer with only the default PCM codec as speech codec may include the MIME type "telephone events" in the first SDP offer.

A Call control entity terminating an SDP offer shall accept the MIME type "telephone events" with default events in any SDP answer, if it selects any codec other than the default PCM codec. An Call control entity terminating an SDP offer that selects the default PCM codec as speech codec may include the MIME type "telephone events" in the SDP answer, if present in the SDP offer.

- When G.711 is used as speech codec then DTMF transferred as in-band within G.711 coded speech
- RFC 4733 shall be used with all compressed codecs
- RTP payload according to IETF RFC 4733
- INFO method shall not be used for this purpose
- If the call control entity has negotiated DTMF events, then the call control entity is not required to handle in parallel DTMF in-band in G.711

9 TRANSPORT PROTOCOL

TCP transport is preferred. UDP shall also be supported. Details of utilizing SCTP for the purpose of SIP-I are still under discussion in 3GPP. TCP is used when SIP signaling message size exceeds 1300 octets (for requests, UDP responses can carry up-to 1500 octets)

The following rules shall be applied for a SIP-I application using TCP:
- TCP is used to establish a SIP session if INVITE does not fit into previously described size limits. An existing TCP connection can be re-used for any other transaction
- A TCP connection shall be released by the SIP-I node that established the transaction
- A TCP connection shall be kept open as long as it serves an open SIP transaction
- In case of a lost TCP connection, the Via header shall specify where replies are to be addressed to if the TCP connection is terminated prior to all replies being successfully sent
- The SIP-I node which initiates a TCP connection shall use an ephemeral port number for the local port of the connection and the server port number 5060 for the remote port (unless the remote site indicates a different port number to be used instead of the well-known SIP port number 5060)

10 SUPPLEMENTARY SERVICES

Nationally defined supplementary services should be implemented (see GFI 9803).

11 ANNOUNCEMENT AND RINGBACK TONE

Only ISUP codes should be used for the purpose of announcements as stated in Q.1912.5 (SIP codes not correctly mapped in all cases, so SIP codes to be used only as a last resort).

Generation of ring back tone (whether provided by originating or terminating call control entity) is for further study, based on e.g. operator input regarding how this feature is handled at the moment in the existing TDM based inter-operator voice call environment.

12 CAUSE CODES

Cause code fields shall not conflict each other within a SIP-I message. They have the following order of preference:
1. ISUP cause code (preferred cause code)
2. Reason header
3. SIP response code

Note:
ISUP cause codes shall be transported within and between networks end-to-end. If an ISUP cause code cannot be transferred within a network (e.g. IMS), then the Reason header shall be used to transport the ISUP cause code.

13 CHARGING

Charging shall be implemented as defined by FICORA's regulation 31 on technical aspects of charging in communications networks. Details shall be agreed mutually between interconnecting operators.

Finnish ISUP3 MPM messages shall be transported in SIP INFO messages. One INFO message can contain one MPM message. One MPM message can contain a maximum of 15 metering pulses.

14 INTEROPERABILITY TESTING

Test cases and test reporting sheet are defined in GSMA documents IR.86 and IR.87. These documents are largely based on earlier SIP-I tests between Finnish operators. Applicable set of tests shall be agreed between interconnecting parties.

15 Appendix A - IR.67 DOMAIN NAMES

Domain name	Sub-domain(s)	Explanation	Rules of Usage	Resolvability
.gprs	Service Provider domains of the form: <Network_Label>.mnc<MNC>.mcc<MCC>.gprs Where <Network Label> is the Network Label part of the Access Pont Name (APN) as defined in 3GPP TS 23.003 [8], section 9, and <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in GPRS for the Operator ID in APNs..	Each Service Provider is allowed to use only sub-domains consisting of MNC(s) and MCC(s) that are allocated to them by ITU-T and their local national numbering authority. Service Providers should avoid using Network Labels consisting of any of the below defined sub-domains, in order to avoid clashes.	Domain needs to be resolvable by at least all GPRS/PS roaming partners.
	rac<RAC>.lac<LAC>.mnc<MNC>.mcc<MCC>.gprs Where <RAC> and <LAC> are the Routing Area Code and Location Area Code (respectively) represented in hexadecimal (base 16) form, and <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in inter-SGSN handovers (i.e. Routing Area Updates) by the new SGSN (possibly in a new PLMN) to route to the old SGSN (possibly in the old PLMN)..	Each Service Provider is allowed to use only sub-domains consisting of MNC(s) and MCC(s) that are allocated to them by ITU-T and their local national numbering authority.	Domains need to be resolvable by at least all SGSNs to which a UE can hand over (which may be in other networks, if inter network GPRS/PS handovers are supported in a Service Provider's network).
	nri<NRI>.rac<RAC>.lac<LAC>.mnc<MNC>.mcc<MCC>.gprs Where <NRI>, <RAC> and <LAC> are the Network Resource Identifier, Routing Area Code and Location Area Code (respectively) represented in hexadecimal (base 16) form, and <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in Routing Area Updates by the new SGSN (possibly in a new PLMN) to route to the old SGSN (possibly in the old		

Domain name	Sub-domain(s)	Explanation	Rules of Usage	Resolvability
		PLMN), where Intra Domain Connection of RAN Nodes to Multiple CN Nodes (also known as "RAN flex" – see 3GPP TS 23.236 [22]) is applied..		
	rnc<RNC>.mnc<MNC>.mcc<MCC>.gprs Where <RNC> is the RNC ID represented in hexadecimal (base 16) form, and <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in SRNS relocation to route to the target RNC in the new SGSN (possibly in a new PLMN)..		
	mms.mnc<MNC>.mcc<MCC>.gprs Where <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in MMS for the domain name part of the FQDN for MMSCs..		Domain needs to be resolvable by at least all directly connected MMS interworking partners/Service Providers and directly connected MMS Hub Providers.
	<Internet_assigned_domain_name>.gprs Where <Internet_assigned_domain_name> is a domain name reserved by the Service Provider on the Internet. An example is "example.com.gprs"	Used as an alternative Operator ID in APNs (also known as "Human Readable APNs"). See 3GPP TS 23.003 [8], section 9, for more details.	The domain name(s) used must be owned by that Service Provider on the Internet. If the domain name(s) expire on the Internet, they also expire on the GRX/IPX. Care should be taken to ensure there is no clash with the other sub-domains for ".gprs" as	Domain needs to be resolvable by at least all GPRS/PS roaming partners.

Domain name	Sub-domain(s)	Explanation	Rules of Usage	Resolvability
			defined above.	
.3gppnetwork.org	`ims.mnc<MNC>.mcc<MCC>.3gppnetwork.org` Where <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in IMS in SIP addressing; specifically in the Private and Public Identities used in SIP registration. See 3GPP TS 23.003 [8], section 13, for more information.	Each Service Provider is allowed to use only sub domains consisting of MNC(s) and MCC(s) that are allocated to them by ITU T and their local national numbering authority. Sub domains within the Service Provider's domain (i.e. mnc<MNC>.mcc<MCC>) are documented in 3GPP TS 23.003 [8]. It is recommended that Service Providers do not use other sub domains that are not specified in 3GPP, OMA or in this PRD as this could potentially cause a clash of sub domain usage in the future.	Domain needs to be resolvable by at least all SIP/IMS based service inter working partners/Service Providers, as well as roaming partners where a visited P-CSCF is used.
	`epc.mnc<MNC>.mcc<MCC>.3gppnetwork.org` Where <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in the Enhanced Packet Core (EPC) architecture (previously known as Service Architecture Evolution – SAE) for NAIs and FQDNs of EPC related nodes. See 3GPP TS 23.003 [8], section 19, for more information.		Domain and sub-domains need to be resolvable by EPC/SAE roaming partners.
	`ics.mnc<MNC>.mcc<MCC>.3gppnetwork.org` Where <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used in the IMS Centralised Services feature in SIP addressing. See 3GPP TS 23.003 [8], section 20,		Domain should only be resolvable for CS roaming partners where an MSC (Server) enhanced for ICS is allowed to be used in that visited

Domain name	**Sub-domain(s)**	**Explanation**	**Rules of Usage**	**Resolvability**
		for more information.		partner's network.
	`node.mnc<MNC>.mcc<MCC>.3gppnetwork.org` Where <MNC> and <MCC> are the MNC and MCC of the Service Provider represented in decimal (base 10) form, with any 2 digit MNC padded out to 3 digits by inserting a zero ("0") on the beginning e.g. 15 becomes 015.	Used by Service Providers to provide FQDNs to non-service specific nodes/hosts e.g. DNS/ENUM servers, routers, firewalls etc..	Each Service Provider is allowed to use only sub-domains consisting of MNC(s) and MCC(s) that are allocated to them by ITU-T and their local national numbering authority.	Domain needs to be resolvable by at least all roaming/interworking partners for the services used by this domain name.
`.e164enum.net`	The sub-domains of this domain name correspond to reversed ITU-T E.164 numbers (as defined in ITU-T Recommendation E.164 [37]).	Used as the domain name for ENUM queries to the GRX/IPX Carrier ENUM ..	Each Service Provider is allowed to use only sub-domains relating to their subscribers.	
`.in-addr.arpa`	The sub-domains of this domain name correspond to reversed IPv4 addresses that belong to the Service Provider.	Used for reverse lookups for IPv4 addresses i.e. mapping names to IPv4 addresses. This is useful when troubleshooting inter-PLMN connections. Due to available tools being pre-configured to use this hierarchy for reverse look-ups, it would not be feasible to use any	Each Service Provider shall populate this domain for IP addresses assigned to them only (except with permission of the actual owner).	Domain should be resolvable by at least all interworking partners/Service Providers, roaming partners and directly connected GRX/IPX Providers.

Domain name	Sub-domain(s)	Explanation	Rules of Usage	Resolvability
		different TLD.		
`.ip6.arpa`	The sub-domains of this domain name correspond to reversed IPv6 addresses that belong to the Service Provider.	Used for reverse lookups for IPv6 addresses i.e. mapping names to IPv6 addresses. This is useful when troubleshooting inter-PLMN connections. Due to available tools using this hierarchy for reverse look-ups, it would not be feasible to use any different TLD.		

16 Appendix B – GUIDELINES FOR IMPLEMENTATION: ISUP-SIP INTERWORKING PROFILE C

FOREWORD

GFI 0301 Guidelines for implementation, ISUP-SIP Interworking, Profile C contains clarifications and option selections for ITU-T deliverables covering ISUP-SIP interworking and national additions covering areas of national ISUP interworking not included in ITU-T deliverables. The aim of this document is to ensure the interworking between national ISUP and SIP protocols.

This guideline document has been prepared by the members of the national standardization group for network-to-network signaling. The Steering Group for Telecommunications Standardization has discussed this document and recommends it to be followed when implementing ISUP-SIP interworking.

16.1 INTRODUCTION

This specification defines the interworking between IETF SIP-standards and national ISUP standards. The basis of the work has been national ISUP3, but the document is applicable also in case of national ISUP2 and ISUP4 standards.

This version covers only interworking based on the profile C defined in the ITU-T Technical Report TRQ.2815 ´Requirements for Interworking BICC/ISUP Network with Originating/ Destination Networks based on Session Initiation Protocol and Session Description Protocol´. Interworking specified in this document covers only the case where the O-IWU has the knowledge that in the connection between O-IWU and IIWU there are no changes in the SIP or ISUP information. Other profiles will be covered in later versions.

The specification contains some introductory text related to the profile C from the TRQ.2815 and national comments to the ITU-T recommendation Q.1912.5 ´Interworking Between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part´. The comments are made because of national requirements or because they are related to other profiles than profile C.

16.2 REFERENCES

- ITU-T Series Q Supplement 45, Technical report TRQ.2815 - Requirements for Interworking BICC/ISUP Network with Originating/Destination Networks based on Session Initiation Protocol and Session Description Protocol
- ITU-T Recommendation Q.1912.5 - Interworking Between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part
- SFS5869:en - Signaling in the public switched telephone network (PSTN). ISDN User Part ISUP Version 3 of the

national Signaling System No. 7. Application of ITU-T recommendations Q.730 and Q.761 - Q.766 in Finland.
- SFS5779 - Signaling in the public switched telephone network (PSTN). ISDN User Part ISUP Version 2 of the national Signaling System No. 7. Application of ITU-T recommendations Q.730 and Q.761 - Q.766 in Finland.
- SFS5901:en - Signaling in the public switched telephone network (PSTN). ISDN User Part ISUP Version 4 of the national Signaling System No. 7. Application of ITU-T recommendations Q.730 and Q.761 - Q.766 in Finland.

16.3 TRQ.2815

The ITU-T TRQ.2815 is applicable with the following comments:
This GFI covers SIP Profile C for Interworking between SIP with MIME Encoding of ISUP and ISUP according to the following figure:

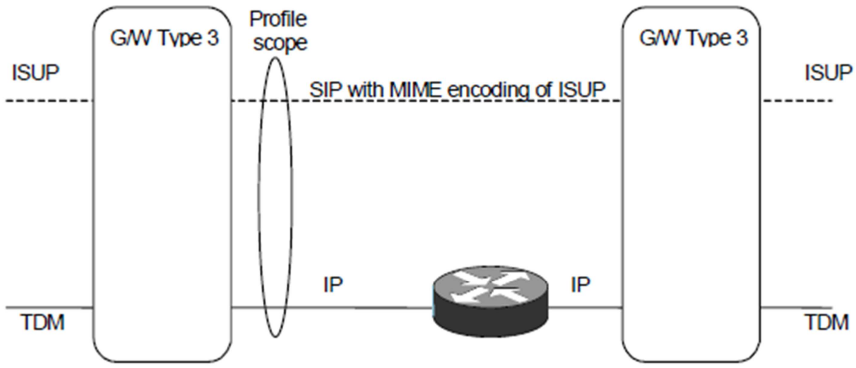

Figure 1 - Profile Scope for SIP with MIME encoding of ISUP Interworking with ISUP with Type 3 Gateways

There should be no misalignment between the MIME'd ISUP and the information in the SIP/SDP headers/fields received at an I-IWF.

The entities in the SIP domain between the O-IWF and the I-IWF do not modify the ISUP or the SIP/SDP headers/fields.

16.4 Q.1912.5

The ITU-T recommendation Q.1912.5 is applicable with the following general comments:
- all procedures and information elements related to BICC are not applicable
- the GFI covers only profile C, i.e. all procedures and information elements related to profiles A and B are not applicable
- the GFI covers only type 3 gateways, i.e. all procedures and information elements related to type 1, 2 and 4 gateways are not applicable

In addition the following comments related to clauses of Q.1912.5 are applicable:

5.3 General principles

The O-IWU and I-IWU shall act as a Type B exchange.

5.3.3 Interworking of ISUP overlap signaling

The text concerning the recommendation of not using overlap signaling is not applicable i.e. in the national network overlap signaling can be used.

5.4.2.1.1 Alignment of SIP headers and ISUP body content

The ISUP field shall be derived form the encapsulated ISUP mime body.

5.4.3 Exclusions/Special considerations

Table 1/Q.1912.5 – ISUP Messages for Special Consideration

Forward Transfer and Continuity Check Request messages are not used in Finland.

Continuity message: The I-IWU can generate COT or it can delay sending the last digit in order to avoid speech clipping. In the interface between operators COT can only be used if it has been agreed by the interconnected operators.

Metering Pulse message: Nationally introduced, transported through the SIP network encapsulated in the INFO message.

Nationally used messages CQM, CQR, OLM, UCICC and SDM are locally terminated.

Nationally used messages INR and INF are encapsulated according to the clause 5.4.3.2.

6.1.1 INVITE received without an SDP offer

Not applicable (SDP offer always included).

6.1.2 INVITE received with an SDP offer or continuation from clause 6.1.1 (1)

SIP preconditions are in use.

6.1.3.3 Nature of Connection Indicators (Mandatory)

The bits AB are coded always 00 (no satellite circuit in the connection) in the national network.

6.1.3.6 ISUP Calling Line Identification (CLI) parameters

The Calling Party Number and Generic Number included in encapsulated ISUP message are transferred as such to the ISUP procedures.

Table 7/Q.1912.5 - Mapping of SIP From/P-Asserted-Identity/Privacy headers to ISUP CLI parameters

Calling Party Number parameter
Address signals

A network provided E.164 number is included if no "P-Asserted-Identity" header field containing a URI with an identity in the format "+CC"+"NCD"+"SN" is received.

Calling Party Number parameter
APRI
APRI is set to "presentation restricted" in case of a network provided E.164 number.

6.1.3.9 Hop Counter (Optional)

In interconnection applications the use of Hop counter and the value of the Max-Forwards Factor is based on bilateral agreement.

6.11.2 Receipt of REL
Table 21/Q.1912.5 - Receipt of the Release message (REL)

Cause values No. 8 (Preemption) and No. 9 (Preemption-circuit reserved for reuse) are not used.

Cause value 24 and the corresponding SIP status code 503 Service unavailable are added.

Cause values 39, 40 and 46 are not applicable.

7.1 Sending of the first INVITE

Only the case B) Sending INVITE with precondition for ISUP IAM/SAM is applicable.

7.1.1 Coding of SDP Media Description Lines from TMR/USI

Table 26/Q.1912.5 - Coding of SDP Media Description Lines from TMR/USI: ISUP to SIP

Rows related to G.711 m-law are not applicable.
Rows related to 1536 kbit/s unrestricted and N x 64 kbit/s unrestricted, N from 3 to 29 are not applicable.

In the cases related to T.38 (Group 3 fax), T.38 or G.711 can be used in accordance with the agreement by the interconnected operators.

CLEARMODE is applicable.

ISUP TMR value ´64 kbit/s preferred´ is handled like value ´64 kbit/s unrestricted´.

7.1.3 P-Asserted-Identity, From and Privacy header fields
Table 27/Q.1912.5 - Mapping ISUP CLI Parameters to SIP Header fields

The encapsulated ISUP message contains all the relevant information related to the Calling Party Number and Generic Number so that there is no need to include any optional INVITE headers related to them.

The case, where a Calling Party Number parameter with complete E.164 number, with Screening Indicator = UPVP or NP, and with APRI = "presentation allowed" or "presentation restricted" has not been received, is applicable only in international calls.

7.1.4 Hop Counter (Optional)

In interconnection applications the use of Hop counter and the value of the Max-Forwards Factor is based on bilateral agreement.

7.1.5.1 Nature of Connection Indicators

Satellite indicator is not incremented in the O-IWU.

7.1.5.2 Propagation delay counter

Propagation delay counter is not required.

7.7.6 Receipt of 4XX, 5XX, 6XX Responses To INVITE

Table 40/Q.1912.5 - Receipt of 4XX, 5XX or 6XX at O-IWU

SIP status code 503 Service unavailable is mapped to REL cause value 127 interworking also in case of ACR.

ANNEX-B.1~B.N
Interworking for ISDN Supplementary Services

ANNEX B.17
Interworking of Multi-Level Precedence and Preemption (MLPP) Supplementary service to SIP networks.
Not applicable.

ANNEX B.18
Interworking of Global Virtual Network Service (GVNS) Supplementary service to SIP networks.

Not applicable.

ANNEX B.19
Interworking of International telecommunication charge card (ITCC) Supplementary service to SIP networks.

Not applicable.

ANNEX B.20
Interworking of Reverse charging (REV) Supplementary service to SIP networks.

Not applicable.

APPENDIX I
Interworking scenarios between SIP and BICC

Not applicable.

APPENDIX II
Interworking scenarios between SIP and ISUP

Not applicable.