Finnish profile for SIP interworking

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202/2014 S

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Contents

1 Background and motivation

The purpose of this document is to give practical guidance for selecting parameters to be used within SIP NNI interconnection.

2 Introduction

This document discusses following services

- Voice call
- DTMF within a voice call
- Telefax
- Video call
- Routing to emergency service
- Call forwarding

3 Terminology and definitions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC 2119].

CIC carrier identification code

NP Number Portability

NPDB Number Portability Database

npdi NP Database Dip Indicator

SDP Session Description Protocol

SI Service Indicator (same as palvelutyyppi, y) [FIN-NP-FIXED]

SIP Session Initiation Protocol [RFC 3261]

SP service provider

SSE SIP Signaling Entity [SIPconnect 1.1]

URI Uniform Resource Identifier

4 General principles

Every MUST in this document is a strict requirement.

Aspects not covered in this document or given reference can be agreed based on bilateral operator agreement.

This document describes conventions to be used within SIP NNI interconnection in Finnish environment.

SIP defined in RFC 3261 MUST be supported.

4.1 URI scheme

SIP URI scheme [RFC 3261 19.1.1] MUST be supported.

4.2 Network and user entities

4.2.1 Calling party identity

P-Asserted-Identity header [RFC 3325] MUST be used to indicate calling party URI in initial request.

P-Asserted-Identity URI MUST contain registered and syntactically correct domain part.

When using E.164 number [E.164] based user part, number MUST be presented as full international number with leading plus '+'.

4.2.2 Diverting party identity

Diversion header [RFC 5806] MUST be supported to indicate diverting party URI.

Diversion URI MUST contain registered and syntactically correct domain part.

When using E.164 number [E.164] based user part, number MUST be presented as full international number with leading plus '+'.

Diversion header privacy parameter MUST contain value "full" when privacy is requested.

Any other value or missing privacy parameter means privacy is not requested.

NOTE: History-Info -header interworking with Diversion header is specified in [RFC 6044].

4.2.3 Called party identity

Request URI [RFC 3261] MUST be used to indicate called party URI.

Number portability lookup result is added as per [section "route parameters"].

4.2.4 Connected party identity

When connected line identity is transferred to calling party, it MUST be transported in P-Asserted-Identity header of 200 response to initial request. [RFC 3325 #5]

4.3 Privacy

Privacy header field value 'id' [RFC 3323] [RFC 3325] MUST be used to indicate calling or connected party anonymization. From URI MUST be set to sip:anonymous@anonymous.invalid when anonymization is used.

Originating network MUST ensure that identity is carried only in P-Asserted-Identity header.

4.4 Signaling layer security

When direct IP peering is not used, encryption of signaling and media packets is RECOMMENDED.

5 Route parameters

5.1 Route parameter transfer methods

SP SSE MUST support number portability prefixes as described in documents [FIN-NP-FIXED] and [FIN-NP-MOBILE].

SP SSE SHOULD support request URI parameters /npdi/, /cic/ and /rn/ as described in [RFC 4694].

Interconnecting networks MUST agree method of route parameter transfer.

Parameter construction rules for method RFC 4694

5.2 NP database dip indicator

Existence of URI user parameter /npdi/ acts as an indicator for further processing that there is no need to perform the NP database dip again.

5.3 Destination carrier identification code

Destination carrier identification code is carried in request URI parameter /cic/ as per [RFC 4694].

Ficora assigned operator code (teleyritystunnus, TY) [Regulation 32]is converted to "global-cic" by concatenating plus '+', country code '358' and zero padded four digit TY.

5.4 Service indicator

Service indicator is constructed as per [FIN-NP-FIXED].

Service indicator is carried in request URI parameter /rn/ as per [RFC 4694].

Service indicator is converted to "global-rn" by concatenating plus '+', country code '358', double zero '00' and service indicator value.

In addition to the service indicator 7, it is possible to use an additional hexadecimal character D. In this case service indicator is converted to "global-rn" by concatenating plus '+', country code '358', zero '0', service indicator value '7' and hexadecimal character 'D'.

6 Signaling

6.1 SIP signaling endpoint

Reply to OPTIONS request MUST be supported for interconnection monitoring.

6.2 SIP methods

Pass through of the following SIP requests and their responses MUST be supported: ACK, BYE, CANCEL, INVITE, OPTIONS, PRACK and UPDATE.

6.3 SIP services

SIP session timer pass through [RFC 4028] MUST be supported.

6.4 Media negotiation

Late SDP offer SHOULD be supported.

Re-negotiation SHOULD be supported.

7 Media

7.1 Media codecs

7.1.1 Voice

G.711 A-law codec MUST be supported.

It is RECOMMENDED that negotiation of wideband codecs is not prevented.

7.1.2 DTMF

DTMF transport method MUST be agreed between operators.

DTMF transport in-band within voice is NOT RECOMMENDED.

DTMF transport using RTP telephony event [RFC 4733] is RECOMMENDED.

7.1.3 Video calls

It is RECOMMENDED that negotiation of video codecs is not prevented.

7.1.4 Telefax

T.38 fax support [T.38] is RECOMMENDED.

NOTE: sip.fax feature tag is introduced in [RFC 6913].

8 General IP network considerations

8.1 For NNI signaling element:

Use of private addresses [RFC 1918] is NOT RECOMMENDED in network layer, as part of SIP header value or in media negotiation.

8.2 For NNI media element:

Use of private addresses [RFC 1918] is NOT RECOMMENDED in media transport.

9 Routing to emergency service

Routing number defined in [Regulation 32] MUST be used in request URI user part.

10 Future work

Following issues are identified, but not considered in current revision:

- Online charging evolution from ISUP MPM
- SCTP as signaling transport protocol
- Video codec recommendation

11 Examples

11.1 Setting /global-cic/

- TY=42 → cic=+3580042
- TY=901 → cic=+3580901

11.2 Setting /global-rn/

- SI=7D → rn=+35807D
- SI=E → rn=+35800E

11.3 Setting Request URI

Example 1: B=+358942411234, lookup=done, ported=no, method=FIN-NP-FIXED / FIN-NP-MOBILE

- URI user: 3BD10942411234
- URI host: proxy.example.com

→ request URI: sip:3BD10942411234@proxy.example.com

Example 2: B=+3584577501234, lookup=done, ported=yes, method=FIN-NP-FIXED / FIN-NP-MOBILE

- URI user: 1D42504577501234
- URI host: proxy.example.com

→ request URI: sip:1D42504577501234@proxy.example.com

Example 3: B=+358942411234, lookup=done, ported=no, method=RFC 4694

- URI user: +358942411234
- URI user parameters: npdi;cic=+3580042;rn=+358001
- URI host: proxy.example.com

→ request URI:
sip:+358942411234;npdi;cic=+3580042;rn=+358001@proxy.example.com

Example 4: B=+3584577501234, lookup=done, ported=yes, method=RFC 4694

- URI user: +3584577501234
- URI user parameters: npdi;cic=+3580042;rn=+358005
- URI host: proxy.example.com

→ request URI:
sip:+3584577501234;npdi;cic=+3580042;rn=+358007@proxy.example.com

Initial INVITE request with diverter, privacy and diverter privacy,

1. voice call

- INVITE sip:+358942411234;npdi;cic=+3580042;rn=+358001@B.example.com SIP/2.0
- From: sip:+358942700000@enterprise.example;tag=abc
- To: sip:+358942411234@example.com
- P-Asserted-Identity: sip:+358942700000@enterprise.example

2. diverted voice call 1

- INVITE sip:+358942419999;npdi;cic=+3580042;rn=+358001@C.example.com SIP/2.0
- From: sip:+358942700000@enterprise.example;tag=def
- To: sip:+358942411234@example.com
- P-Asserted-Identity: sip:+358942700000@enterprise.example
- Diversion: sip:+358942411234@B.example.com;reason=unconditional

3. voice call with privacy

- INVITE sip:+3584577501234;npdi;cic=+3580042;rn=+358005@B.example.com SIP/2.0
- From: sip:anonymous@anonymous.invalid;tag=ghi
- To: sip:+3584577501234@anonymous.invalid

- P-Asserted-Identity: sip:+3584577501111@enterprise.example
- Privacy: id

4. diverted voice call 3

- INVITE sip:+358942419999;npdi;cic=+3580042;rn=+35800E@C.example.com SIP/2.0
- From: sip:anonymous@anonymous.invalid;tag=jkl
- To: sip:+3584577501234@anonymous.invalid
- P-Asserted-Identity: sip:+3584577501111@enterprise.example
- Diversion: sip:+3584577501234@B.example.com;reason=unconditional
- Privacy: id

5. diverted voice call 1 with diverter privacy

- INVITE sip:+358942419999;npdi;cic=+3580042;rn=+358001@C.example.com SIP/2.0
- From: sip:+358942700000@enterprise.example;tag=mno
- To: sip:+358942411234@example.com
- P-Asserted-Identity: sip:+3584577501111@enterprise.example
- Diversion: sip:+358942411234@B.example.com;reason=unconditional;privacy=full

6. diverted voice call 3 with diverter privacy

- INVITE sip:+358942419999;npdi;cic=+3580042;rn=+35800E@C.example.com SIP/2.0
- From: sip:anonymous@anonymous.invalid;tag=pqr
- To: sip:+3584577501234@anonymous.invalid
- P-Asserted-Identity: sip:+3584577501111@enterprise.example
- Diversion: sip:+3584577501234@B.example.com;reason=unconditional;privacy=full
- Privacy: id

12 References

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