Finnish profile for SIP interworking

Recommendation 202/2025 S

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Revision History

Version	Date	Description	Author
1.0	31.3.2014	Finnish profile for SIP interworking	Viestintävirasto
2.0	30.6.2025	Finnish profile for SIP interworking	Liikenne- ja viestintävirasto

Table of Contents

1 Background and motivation

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

2 Introduction

This document discusses following services

- Voice call
- DTMF within a voice call
- Telefax
- Routing to emergency service
- Call forwarding
- Real Time Text

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
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 MAY 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.

When anonymization is required then SIP URI MUST be set to sip:anonymous@anonymous.invalid or sip:unavailable@unknown.invalid.

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

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

SP SHOULD support request URI parameter /rn/ as described in [RFC 4694].

Interconnecting networks MUST agree method of route parameter transfer.

5.2 Service indicator in IMS-NNI or terminating in IMS

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.

Calls originating from abroad MUST be identified by rn +35800x, x being service indicator 6 or E. Other service indicators are NOT identified/used.

5.3 Service indicator in SIP-NNI

Usage of Service indicator in the beginning of B-number (as example 3BDx091234567 or 1Dtyx091234567) x being service indicator, MUST be agreed between the parties.

6 Signalling

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, INFO 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 Telefax

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

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

7.1.4 Real-time text

T.140 real-time text (RTT) in IMS-NNI MUST be supported. [RFC 4103] In case of Mobile Virtual Network Operator (MVNO) the supporting Mobile Network Operator (MNO) shall ensure the implemented setup is compliant with requirements related to RTT.

8 SIP based Tariff

For calls made to premium rate numbers with tariff, recommendation "Finnish profile for SIP tariff interworking (Liikenne- ja viestintäviraston suosituksia 217/2025 S)" MUST be supported.

9 General IP network considerations

9.1 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.

9.2 For NNI media element

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

10 Routing to emergency service

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

11 References

[RFC 2119] Key words for use in RFCs to Indicate Requirement Levels, https://datatracker.ietf.org/doc/rfc2119/

[FIN-NP-FIXED] TR 5/2004 Puhelinnumeron siirrettävyys, kiinteä puhelinverkko, tekninen verkkototeutus, https://www.traficom.fi/sites/default/files/media/regulation/Puhelinnumeron%20siirrettavyys_kiinteapuhelinverkko_tekninenverkkototeutus_5_2004.pdf

[RFC 3261] SIP: Session Initiation Protocol, https://datatracker.ietf.org/doc/rfc3261/

[RFC 3325] Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks, https://datatracker.ietf.org/doc/rfc3325/

[E.164] ITU-T recommendation E.164 (11/10): The international public telecommunication numbering plan, https://www.itu.int/rec/t-rec-e.164/en

[RFC 5806] Diversion Indication in SIP, https://datatracker.ietf.org/doc/rfc5806/

[RFC 6044] Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP), https://datatracker.ietf.org/doc/rfc6044/

[RFC 3323] A Privacy Mechanism for the Session (SIP), https://datatracker.ietf.org/doc/rfc3323/

[FIN-NP-MOBILE] TR 10/2002 Matkapuhelinnumeron siirrettävyys, tekninen verkkototeutus, https://www.traficom.fi/sites/default/files/media/regulation/Matkapuhelinnumeron%20siirrettavyys_tekninenverkkototeutus.pdf

[RFC 4694] Number Portability Parameters for the "tel" URI, https://datatracker.ietf.org/doc/rfc4694/

[Regulation 32] on numbering in a public telephone network, https://traficom.fi/en/regulations/regulation-32-numbering-public-telephone-network

[RFC 4028] Session Timers in the Session Initiation Protocol (SIP), https://datatracker.ietf.org/doc/rfc4028/

[RFC 4733] RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals, https://datatracker.ietf.org/doc/rfc4733/

[T.38] ITU-T recommendation T.38 (11/2015): Procedures for real-time Group 3 facsimile communication over IP networks, https://www.itu.int/rec/T-REC-T.38/en

[T.140] ITU-T recommendation T.140 (02/2000): Protocol for multimedia application text conversation, https://www.itu.int/rec/T-REC-T.140

[Recommendation 217/2025 S]: Finnish profile for SIP tariff interworking, (Liikenne- ja viestintäviraston suosituksia 217/2025 S)

[RFC 1918] Address Allocation for Private Internets, https://datatracker.ietf.org/doc/rfc1918/

Finnish Transport and Communications Agency Traficom
PO Box 320, FI-00059 TRAFICOM
Switchboard: +358 29 534 5000

traficom.fi

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