



SIP FORUM

ATIS-1000063 – SIP Forum TWG-6

JOINT ATIS/SIP FORUM TECHNICAL REPORT – IP NNI PROFILE

JOINT TECHNICAL REPORT



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IP NNI Profile

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Abstract

This document specifies an NNI profile applicable to the interface between the home network of the originating party and the home network of the terminating party; or between the home network of either party, and a transit network. The interface between the home and visited network of a roaming mobile user is out of scope.

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

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Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005, and/or to the SIP Forum, 733 Turnpike Street, Suite 192, North Andover, MA, 01845.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes a optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

The **ATIS/SIP Forum IP-NNI Task Force** under the **ATIS Packet Technologies and Systems Committee (PTSC)** and under the **SIP Forum Technical Working Group (TWG)** were responsible for the development of this document.

Table of Contents

1	Scope, Purpose, & Application	1
1.1	Scope	1
1.2	Purpose	1
1.3	Application	2
2	Normative References	2
3	Definitions, Acronyms, & Abbreviations	6
3.1	Acronyms & Abbreviations	6
4	Reference Model for Interconnection	8
4.1	Current US Telephony PSTN Interconnect Model	8
4.2	VoIP Interconnection Basic Configuration	9
4.3	Trust Model	9
5	General Procedures	11
5.1	Extension Negotiation	11
5.2	Public User Identities	12
5.2.1	<i>Identifying the Called User</i>	12
5.2.2	<i>Identifying the Calling User</i>	13
5.2.3	<i>Numbering & Addressing</i>	13
5.3	IPv4/6 Interworking	15
5.4	Fault Isolation & Recovery	15
5.4.1	<i>Interface Failure Detection</i>	15
5.4.2	<i>Congestion Control</i>	16
5.4.3	<i>Session Timer</i>	16
5.4.4	<i>RTP Loopback Test</i>	16
5.5	Media Transport	16
5.5.1	<i>Codecs</i>	16
5.5.2	<i>Codec/Packetization Period Use & Transcoding Guidelines</i>	18
5.5.3	<i>General Guidelines</i>	18
5.5.4	<i>Voice-band Data Transport Mechanisms</i>	18
5.5.5	<i>DTMF Digit Transport Mechanisms</i>	19
5.6	IP Packet Marking	20
5.6.1	<i>Distinguishing Traffic Classes</i>	20
5.6.2	<i>IP Marking Table</i>	20
5.6.3	<i>Traffic Treatment</i>	21
6	Call Features	21
6.1	Basic Call Setup	21
6.1.1	<i>SDP Requirements</i>	21
6.2	Ringback Tone vs. Early Media	22
6.3	Early-Media	22
6.3.1	<i>Terminating Network Procedures</i>	22
6.3.2	<i>Originating Network Procedures</i>	23
6.4	Forking the INVITE	23
6.5	Redirecting the INVITE	24
6.6	Call Hold	24
6.7	Calling Number & Name Delivery	24
6.8	Call Forwarding	24
6.9	Emergency Telecommunications Service (ETS)	25
7	NNI Signaling Profile	25
7.1	SIP Methods & Header Fields	25
7.1.1	<i>SIP Methods</i>	25

7.1.2	SIP Header Fields.....	26
7.1.3	SDP Protocol.....	28
7.1.4	Major Capabilities.....	28
7.2	Control Plane Transport.....	32
7.3	SIP Timers.....	33
8	Security.....	33
Annex A – Response Codes.....		34

Table of Figures

Figure 4.1	– Current US Telephony PSTN Interconnect Model.....	8
Figure 4.2	– Bilateral Carrier VoIP Interconnections.....	9
Figure 4.3	– Carrier Interconnection Trust Relationship.....	10

Table of Tables

Table 5.1	– Called User Identities.....	13
Table 7.1	– Key to notation codes for SIP messages.....	25
Table 7.2	– Supported SIP methods.....	26
Table 7.3	– Management of SIP header fields over NNI in presence or not of a trust relationship.....	27
Table 7.4	– Major capabilities over NNI.....	29
Table 7.5	– Key to notation codes for major capabilities.....	32
Table A.1	– Response Codes.....	34

ATIS Standard on –

IP Interconnection

1 Scope, Purpose, & Application

1.1 Scope

This document was developed under a joint ATIS and Session Initiation Protocol (SIP) Forum collaboration. The document defines an Internet Protocol (IP) Network-to-Network Interface (NNI) profile with an emphasis on Voice over IP (VoIP). Other Multimedia services will be addressed in subsequent releases.

This document specifies an NNI profile applicable to the interface between the home network of the originating party and the home network of the terminating party; or between the home network of either party, and a transit network. The interface between the home and visited network of a roaming mobile user is out of scope.

The scope of this documented is limited to the information exchanged at the reference points illustrated in Figure 4.1. The behavior of network elements upon receipt of such information is governed by other specifications.

The scope of this profile document is to:

1. Define a reference architecture that sets forth the common functional entities for Carrier to Carrier Interconnection. This reference architecture will be from the perspective of the interconnection points between carriers and will not deal with implementation details inside the networks on either side of the IP NNI.
2. Define the normative standards (including IETF RFCs, 3GPP, and other existing standards) associated with these protocols that are supported by each element of the reference architecture. Where required, the options that MUST or SHOULD be supported within a given standard will also be defined for this profile.
3. Define for this profile the customary methods for negotiating protocols, protocol extensions, and exchanging capability information between carriers. The methods of formulating SIP protocol messages are where multiple options exist in standards.
4. Define for this profile the presentations of Fully Qualified Domain Names in “From:” and “To:” fields, including P-Asserted Identity (PAI).
5. Define support for underlying transport [e.g., User Datagram Protocol (UDP), Transmission Control Protocol (TCP), and Stream Control Transmission Protocol (SCTP)].
6. Define an audio codec selection strategy that minimizes the need for transcoding and a transcoding strategy that balances the workload between originating and terminating carrier.
7. Define strategies for Dual-Tone Multi-frequency (DTMF) and facsimile (Fax) support.
8. Specify call loop detection and avoidance methods.
9. Include recommendations for network overload and congestion notification and processing mechanisms.

1.2 Purpose

IP Interconnection among service providers is significantly increasing as the transition of the Public Switched Telephone Network (PSTN) from Signaling System No. 7 (SS7)/ Time-Division Multiplex (TDM) to SIP/IP networks progresses. Current deployments of SIP/IP in the core carrier networks have exposed operational and implementation differences on how IP for SIP traffic works ‘on the wire’. These differences complicate interconnection, and in some cases require ‘protocol normalization’ to achieve full interoperability. The call control protocol SIP [RFC 3261] is defined in the IETF and is further refined by 3GPP or ATIS specifications. There are hundreds of IETF SIP and 3GPP specifications that are open to interpretation, creating ambiguity in the detailed options that are implemented. This often requires Session Border Controllers or Interconnection Border Control Function (IBCF) proxies to reconcile the signaling between service providers and resolve those ambiguities. Time