ATIS-1000038

TECHNICAL PARAMETERS FOR
IP NETWORK TO NETWORK INTERCONNECTION
RELEASE 1.0

TECHNICAL REPORT

ATIS is the leading technical planning and standards development organization committed to the rapid development of global, market-driven standards for the information, entertainment and communications industry. More than 250 companies actively formulate standards in ATIS' 18 Committees, covering issues including: IPTV, Service Oriented Networks, Energy Efficiency, IP-Based and Wireless Technologies, Quality of Service, and Billing and Operational Support. In addition, numerous Incubators, Focus and Exploratory Groups address emerging industry priorities including "Green", IP Downloadable Security, Next Generation Carrier Interconnect, IPv6 and Convergence.

ATIS is the North American Organizational Partner for the 3rd Generation Partnership Project (3GPP), a member and major U.S. contributor to the International Telecommunication Union (ITU) Radio and Telecommunications' Sectors, and a member of the Inter-American Telecommunication Commission (CITEL). For more information, please visit < http://www.atis.org >.

ATIS-1000038, *Technical Parameters for IP Network to Network Interconnection Release 1.0*

Is an ATIS Standard developed by the **Next Generation Carrier Interconnection (NG-CI)** Task Force under the **ATIS Packet Technologies and Systems Committee (PTSC)**.

ATIS-1000038

ATIS Standard on

TECHNICAL PARAMETERS FOR
IP NETWORK TO NETWORK INTERCONNECTION
RELEASE 1.0

Alliance for Telecommunications Industry Solutions

Approved August 2010

Abstract

This TR specifies the "Interconnection Technical Parameters" that need to be collected and eventually exchanged between two service providers so that they can successfully interconnect IP-based facilities and VoIP services at an NNI.

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.

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.

Shall indicates a strict requirement to be exercised and validated. *May* indicates an optional feature that may be supported, but is not required and will not be validated.

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.

The Next Generation Carrier Interconnection (NG-CI) Task Force was responsible for the development of this document.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following leadership:

> M. Dolly, PTSC Chair (AT&T)
> V. Shaikh, PTSC Vice-Chair (Telcordia)
> G. Yum, PTSC NG-CI Chair (Verizon)
> M. Hammer, PTSC NG-CI Vice-Chair (Cisco)
> M. Desterdick, Technical Editor (Verizon)

TABLE OF CONTENTS

ATIS Standard on –

Technical Parameters for
IP Network to Network Interconnection
Release 1.0

1 SCOPE, PURPOSE, & APPLICATION

The ATIS Next Generation Carrier Interconnection (NG-CI) Task Force (TF) Technical Reports (TRs) specify a standards-based interface for network-to-network interconnection for next generation network services.

This TR specifies the "Interconnection Technical Parameters" that need to be collected and eventually exchanged between two service providers so that they can successfully interconnect IP-based facilities and VoIP services at an NNI.

The purpose of this TR is to document the set of technical specifications pertaining to the parameters and attribute settings to be discussed and agreed upon for SIP-based voice services.

This TR may be used to develop an interconnection ordering template upon which providers can exchange all the information needed to ensure interconnection planning, testing, provisioning, and ordering of circuits/facilities leading to VoIP interoperability.

It is a component of a family of documents that, taken as a whole, provides an IP network to network interconnection (NNI) guideline that supports next generation service interoperability.

This Release 1.0 addresses Basic VoIP service, Basic VoIP service with DTMF support, and Basic VoIP service with T.38 Fax support, for native IP endpoints.

Post release 1.0 will address support for RPH as well as services and capabilities such as domestic mobile-to-mobile voice service interconnect over IP and international mobile and nomadic endpoints. Other services like wireline-TDM, managed peer-to-peer, mobile-to-mobile peering, IPTV, multimedia services, gaming, fixed-mobile service convergence, etc., may also be considered.

The ATIS NG-CI TRs cover four aspects of interconnection and this specific TR has three other companion TRs that together address VoIP services Interconnection over IP-based links/networks.

The companion TRs include the following:

♦ The "Protocol Suites" TR (ATIS-1000040) provides the service specific protocol requirements to support service interoperability. While the other documents in this Release 1.0 set make reference to the specifications covered, the "Protocol Suites" TR identifies the specific details of protocol elements to be tested.

♦ The "Test Suite" TR (ATIS-1000041) provides the tests used for IP NNI interconnection testing in order to support service interoperability.

♦ The "Test Configuration" TR (ATIS-1000039) provides the network interface configuration in order to support service interoperability.

Establishing and specifying the fixed protocol and configuration profiles, and the variable or selectable parameters that can then be applied uniformly and consistently for interconnects, will ensure a reliable level of conformance to a standard that supports the establishment of successful interoperability.

1.1 Assumptions

1. Due to the variety of security configurations and credentials possible between providers, the use of IPSEC or TLS to support signaling or media streams will be subject to agreement between those providers and will not be defined within this document.

2. The level of information presented here is intended to be sufficient to support interoperability testing events; however, additional work will be required to develop actual test scripts based on the test scenarios, configurations and protocol suites presented.

3. It is understood that test SIP device endpoint E.164 addresses will need to be exchanged prior to testing. SIP URIs converted from TEL URI format will be used to convey the E164 addresses. (See section of ATIS-1000009.2006 for example URIs.)

4. IPv4 is assumed unless otherwise stated.

5. The term Provider is used to generically represent all types of parties.

2 NORMATIVE REFERENCES

The following standards contain provisions which, through reference in this text, constitute provisions of this ATIS Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this ATIS Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

ATIS-1000007.2006, *Generic Signaling and Control Plane Security Requirements for Evolving Networks*.[1]

ATIS-1000008.2006, *ANSI Extensions to Q.1980.1 - The Narrowband Signaling Syntax (NSS) – Syntax Definitions*.[1]

ATIS-1000009.2006, *IP Network -To-Network Interface Standard for VoIP*.[1]

ATIS-1000013.2007, *Lawfully Authorized Electronic Surveillance (LAES) for Internet Access and Services*.[1]

ATIS-1000013.a.2009, *Supplement to ATIS-1000013.2007, LAES for Internet Access and Services*.[1]

ATIS-1000014, VoIP *Network-to-Network Interface Testing Framework*.[1]

ATIS-1000017.2008, *Network Interface Protocol and the Session Initiation Protocol (SIP) with ANSI Extensions to the Narrowband Signaling Syntax (NSS)*.[1]

ATIS-1000026.2008, *Session/Border Control Functions and Requirements*.[1]

ATIS-1000028.2008, *IP Device (SIP for UA) to Network Interface Standard*.[1]

ATIS-1000029.2008, *NGN Security Requirements*.[1]

---

[1] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/default.aspx >

ATIS-1000030.2008, *NGN Authenticating Requirements.*[1]

ATIS-1000035.2009, *NGN Identity Management (IdM) Framework.*[1]

ATIS-1000036, *NGN Operator Services Architectures and Interfaces.*[1]

ATIS-1000104.1991 (R2008), *Exchange-Interexchange Carrier Interfaces - Individual Channel Signaling Protocols.*[1]

ATIS-1000018, *NGN Architecture.*[1]

ATIS-1000019.2007, *Network to Network (NNI) Standard for Signaling & Control Security for Evolving VoIP/Multimedia Networks.*[1]

ATIS-1000607.a.2006, *Supplement to T1.607-2000 (R2004).*[1]

ATIS-1000619.a.1994 (R2007), *Integrated Services Digital Network (ISDN) – Multi-Level Precedence and Preemption (MLPP) Service Capability (MLPP service domain and cause value changes).*[1]

ATIS-1000625.a.1998 (R2008), *Supplement to ATIS-1000625.1993 (R2008).*[1]

ATIS-1000627.1993 (R2009), *Broadband ISDN - ATM Layer Functionality and Specification.*[1]

ATIS-1000678.2006, *Lawfully Authorized Electronic Surveillance (LAES) for Voice over Packet Technologies in Wireline Telecommunications Networks, Version 2.*[1]

ATIS-1000678.a.2007, *Supplement to ATIS-1000678.2006.*[1]

ATIS-1000679.2004 (R2010), *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part.*[1]

T1.TR.76-2001, *Operation of the BICC Protocol with IP Bearer control Protocol (WD).*[1]

T1.TR.82-2003, *SIP Network Operators Implementers Guide for Pre-Paid Calling Cards, with DTMF Detection at the PSTN-IP Gateway.*[1]

3 ACRONYMS & ABBREVIATIONS

ANSI	American National Standards Institute
ATIS	Alliance for Telecommunications Industry Solutions
DTMF	Dual-Tone Multi-Frequency
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSec	Internet Protocol Security
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
NG-CI TF	Next Generation Carrier Interconnection Task Force
NNI	Network to Network Interface
PTSC	Packet Technologies and Systems Committee
RFC	Request for Comments
RTCP	Real-time Control Protocol
RTP	Real-time Transport Protocol

SDP	Session Description Protocol
SIP	Session Initiated Protocol
SP	Service Provider
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UDP	User Datagram Protocol
URI	Universal (or Uniform) Resource Identifier (or Indicator)
VoIP	Voice over Internet Protocol

4 DEFINITIONS

4.1 N/A – Signifies the field is intentionally void of information where there is no relevant information and therefore is "Not Applicable."

4.2 Private Interconnection Mode – Facilities/circuits used for interconnection are privately owned by the interconnecting providers (or private circuits leased to the interconnecting providers) so that all interconnect traffic is segregated from and not reachable by traffic from the public internet.

4.3 Public Interconnection Mode – Physical and virtual facilities/circuits used for interconnection consist of parts also used for transmitting traffic to/from the public internet.

5 FIELD DESCRIPTIONS

The identification and specification of the technical parameters to be collected for exchange between providers are described in this section.

5.1 Signaling Parameters

♦ *Purpose*: This section provides information regarding the network element to which SIP signaling is to be directed. This will generally be a port(s) on a Border Element and shall be in the Advertised Routing Domain (ARD).

♦ *ATIS Reference(s)*: For more information on the SIP Signaling Port covered in this section, please refer to the following references, specifically to ATIS-1000039, *Testing Configuration for IP Network to Network Interconnection, Release 1.0*, for a description and diagrams of the possible physical configurations, and the corresponding logical port used for SIP Signaling, as identified in the figures of ATIS-1000039 as "Port B."

♦ *Specified in this Section*: The identification of the border element (hosting the signaling ports) Type, Manufacturer, Software Release, and IP Address to which signaling is sent/received.

5.1.1 Manufacturer/Device Type

♦ *Field Description*: Specify the border element (hosting the signaling ports) Type and Manufacturer's Name and Model Number of the device to which SIP signaling is sent/received.

♦ *Acceptable Input*: Freeform (Various Manufacturer Device Types and Models).

♦ *Field Type*: Mandatory and freeform of non-specific format and length.

♦ *Reference(s)*: N/A.

5.1.2 Software Version

♦ *Field Description*: Specify the software Version/Release of the specified Manufacturer border element (hosting the signaling ports) to which signaling is sent/received.

♦ *Acceptable Input(s)*: Freeform with examples being "Software Version X.X" and "Release X.X".

♦ *Field Type*: Mandatory and freeform of non-specific format and length.

♦ *Reference(s)*: N/A.

5.1.3 IP Address(es) & Port(s)

♦ *Field Description*: Specify the IP address(es) and Port(s) (or their range) in the peer network to which the SIP signaling is sent/received.

♦ *Acceptable Input(s)*:

   o "XXX.XXX.XXX.XXX" and "XXX.XXX.XXX.XXX   - XXX.XXX.XXX.XXX " for IP Address(es)

   o "XXXXX" and "XXXXX – XXXXX" for Port(s)

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*: IETF RFC 791, *Internet Protocol*, 1981-09.

5.1.4 Other Signaling Parameters

♦ *Field Description*: Specify the information requested, per the respective numbered fields in this section.

♦ *Acceptable Input(s)*: Mandatory and of specific format and length as specified in the respective numbered.

♦ *Field Type*: Various, per the respective numbered field..

♦ *Reference(s)*: N/A.

5.1.5 Transport Protocol Parameter for SIP Signaling Traffic

♦ *Field Description*: Specify the transport layer protocol to carry SIP signaling traffic.

♦ *Acceptable Input(s)*: TCP or UDP.

♦ *Field Type*: Mandatory and freeform of non-specific format and length.

♦ *Reference(s)*: ATIS-1000040, *Protocol Suite Profile for IP Network-to-Network Interconnection, Release 1.0.*

5.2 Media Parameters

♦ *Purpose*: This section provides information regarding the ports to which the media packets are directed. This shall be in the advertised routing domain and may be a Border Element or another Network Element if the port is not associated with a provider's Border Element.

♦ *ATIS Reference(s)*: For more information on the Media Port covered in this section, please refer to the following references, specifically to ATIS-1000039, *Testing Configuration for IP Network to Network Interconnection, Release 1.0*, for a description and diagrams of the possible physical configurations, and the corresponding logical port used for Media, as identified in the figures of ATIS-1000039 as "Port C."

♦ *Specified in this Section*: The identification of the Border Element or Network Element (hosting the media ports) Type, Manufacturer, Software Release, and IP addresses to which voice media is sent / received.

5.2.1 Manufacturer/Device Type

♦ *Field Description*: Specify the Device Type and Manufacturer's Name and Model Number of the Border Element or Network Element (hosting the media ports) to which media is sent/received.

♦ *Acceptable Input(s)*: Freeform (Various Manufacturer Media Device Types and Models).

♦ *Field Type*: Mandatory and freeform of non-specific format and length.

♦ *Reference(s)*: N/A.

5.2.2 Software Version

♦ *Field Description*: Specify the Software Version/Release of the specified Manufacturer Border Element or Network Element (hosting the media ports) to which media is sent/received.

♦ *Acceptable Input(s)*: Freeform (Examples being "Software Version X.X" and "Release X.X").

♦ *Field Type*: Mandatory and freeform of non-specific format and length.

♦ *Reference(s)*: N/A.

5.2.3 Integrated or Distributed

♦ *Field Description*: Identify whether the Media and Signaling port functionality are combined (Integrated) in a single Border Element.

♦ *Acceptable Input(s)*: "Yes" (Integrated) or "No" (Distributed).

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*: N/A.

5.2.4 IP Address(es) & Port(s)

♦ *Field Description*: Specify the IP address(es) and Port(s) (or their range) in the peer network to which RTP/RTCP and RTP/UDP media is sent/received.

♦ *Acceptable Input(s)*:

o "XXX.XXX.XXX.XXX" or "XXX.XXX.XXX.XXX - XXX.XXX.XXX.XXX "

o "XXXXX" and "XXXXX – XXXXX" for Port(s)

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*: IETF RFC 791, *Internet Protocol*, 1981-09.

5.3 Peering Edge Router (PER) & Network Parameters

♦ *Purpose*: To specify the parameters relevant to the network and/or peering edge router used for voice interconnection.

♦ *ATIS Reference(s)*: For more information on the Peering Edge Router Port covered in this section, please refer to the following references, specifically to ATIS-1000039, *Testing Configuration for IP Network to Network Interconnection, Release 1.0*, for a description and diagrams of the possible physical configurations, and the corresponding logical port used for this IP Transport Layer 3 port, as identified in the figures of ATIS-1000039 as "Port A."

♦ *Specified in this Section*: This section specifies parameters relative to the network and/or peering edge router.

5.3.1 Layer 1 & 2

♦ *Field Description*: Optional for provider use to identify or note Layer 1 and/or Layer 2 information.

♦ *Acceptable Input(s)*: N/A (Freeform).

♦ *Field Type*: Freeform of non-specific format and length.

♦ *Reference(s)*: N/A.

5.3.2 IP Address for PING

♦ *Field Description*: The IP Address used to ensure IP connectivity using the PING (Packet Internet Groper) TCP/IP utility that tests for network IP Reachability/connectivity of each provider's peering edge router. [The PING utility is based on the Internet Control Message Protocol (ICMP) Echo Request and Echo Reply messages.]

♦ *Acceptable Input(s)*: XXX.XXX.XXX.XXX

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*: IETF RFC 791, *Internet Protocol*, 1981-09.

5.3.3 Signaling & Control IP Address(es)

♦ *Field Description*: Identify the IP address(es) (or range of IP Addresses) in the peer network to which the PER Signaling and Control Layer 3 Control Messages are sent/received.

♦ *Acceptable Input(s)*: "XXX.XXX.XXX.XXX" or "XXX.XXX.XXX.XXX - XXX.XXX.XXX.XXX "

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*:

  o IETF RFC 791, *Internet Protocol*, 1981-09.

  o IETF RFC 4271, *A Border Gateway Protocol 4 (BGP-4)*, 2006-06.

5.3.4 Forwarding IP Address(es)

♦ *Field Description*: Identify the IP address(es) (or range of IP Addresses, i.e., advertised routing domain) in the peer network to which the PER Forwarding messages are sent / received.

♦ *Acceptable Input(s)*: "XXX.XXX.XXX.XXX" or "XXX.XXX.XXX.XXX - XXX.XXX.XXX.XXX "

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*: IETF RFC 791, *Internet Protocol*, 1981-09.

5.4 Basic Voice – Service Parameters

♦ *Purpose*: Parameters identifying media specifications for voice inter-working to ensure successful voice media transmission.

♦ *ATIS Reference(s)*:

  o ATIS-1000039, *Testing Configuration for IP Network to Network Interconnection, Release 1.0*.

  o ATIS-1000040, *Protocol Suite Profile for IP Network-to-Network Interconnection, Release 1.0*.

  o ATIS-1000041, *Test Suites for IP Network-to-Network Interconnection, Release 1.0*.

♦ *Specified in this Section*: This section specifies Voice Codec(s) parameters for basic voice service.

5.4.1 Voice Codec(s)

♦ *Field Description*: Specify the voice codec(s) to be supported and available for negotiation during call set-up message exchange across the interface.

♦ *Acceptable Input(s)*: "Codec/Packetization Period (pp)" (sample formats: G.711mu/20ms; G.729A/20ms; G.729/20ms).

♦ *Field Type*: Mandatory and of specific format (but not length).

♦ *Reference(s)*: ITU-T "G Series".

5.4.2 Codec Policy

♦ *Field Description*: Specify if your SDP Answer to an SDP Offer contains only the preferred codec, or will it contain a list of codecs with the preferred codec listed first?

♦ *Acceptable Input(s)*: "Yes" or "No".

♦ *Field Type*: Answering each question in this section is Mandatory and of specific format and length.

♦ *Reference(s)*: N/A.

5.5 Features Ancillary to Voice Service

♦ *Purpose*: Parameters identifying specifications for inter-working of features ancillary to voices services to ensure successful interworking of the features.

♦ *ATIS Reference(s)*:

    o ATIS-1000039, *Testing Configuration for IP Network to Network Interconnection, Release 1.0*.

    o ATIS-1000040, *Protocol Suite Profile for IP Network-to-Network Interconnection, Release 1.0*.

    o ATIS-1000041, *Test Suites for IP Network-to-Network Interconnection, Release 1.0*.

♦ *Specified in this Section*: This section specifies ancillary features of transmission of DTMF digits, and Fax support.

5.5.1 DTMF Digits

♦ *Field Description*: Specify if the transmission of DTMF digits using telephone-event RTP payload format type is supported.

♦ *Acceptable Input(s)*: "Supported " or "Not Supported".

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*: IETF RFC 4733, *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*, 2006-12.

5.5.2 FAX

♦ *Field Description*: Specify here if G.711, *Pass Thru*, and/or T.38, *Fax*, is supported.

♦ *Acceptable Input(s)*: "Yes" or "No" (for each of the fields corresponding to "G711 Supported" and "T.38 Supported").

♦ *Field Type*: Mandatory and of specific format and length.

♦ *Reference(s)*:

    o ITU-T Recommendation T.38, *Fax*.

    o IETF 4734, *Definition of Events for Modem, Fax, and Text Telephony Signals*, 2006-12.

Annex A

A TECHNICAL PARAMETERS TABLE

Text Section	Technical Parameter
5.1	Signaling Parameters
5.1.1	Manufacturer/Type
5.1.2	Software Version
5.1.3	IP Address(es) & Port(s) 1. SIP Signaling
5.1.4	Other Signaling Parameters 1. Ring, No-Answer Timer (Seconds)
5.2	Media Parameters
5.2.1	Manufacturer/Type
5.2.2	Software Version
5.2.3	Integrated? ("Yes" or "No")
5.2.4	IP Address(es) & Port(s) 1. RTP/UDP 2. RTP/RTCP
5.3	Peering Edge Router (PER) and Network Parameters
5.3.1	Layer 1 and 2 (Optional)
5.3.2	IP Address for PING
5.3.3	Signaling/Control IP Address(es)
5.3.4	Forwarding IP Address(es)
5.4	Basic Voice – Service Parameters
5.4.1	Voice Codec(s) Accepted and Packetization Period (pp)
5.4.2	Codec Policy
5.5	Features Ancillary to Voice Service Parameters
5.5.1	DTMF Digits ("Supported" or "Not Supported")
5.5.2	FAX Supported ("Yes" or "No" for each) 1. G.711 Pass Thru 2. T.38 Fax (Specify Version)