ATIS-1000009.200

IP NETWORK-TO-NETWORK INTERFACE (NNI) STANDARD FOR VOIP

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Published by
Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005

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American National Standard for Telecommunications

IP NETWORK-TO-NETWORK INTERFACE (NNI) STANDARD FOR VOIP

Secretariat

Alliance for Telecommunications Industry Solutions

Approved May 16, 2006

American National Standards Institute, Inc.

Abstract

This document defines a standard approach to support IP-IP interconnection for VoIP between carriers.

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FOREWORD

The information contained in this Foreword is not part of this American National Standard (ANS) and has not been processed in accordance with ANSI's requirements for an ANS. As such, this Foreword may contain material that has not been subjected to public review or a consensus process. In addition, it does not contain requirements necessary for conformance to the Standard.

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J. Zebarth, PTSC Vice-Chair
S. Carioti, ATIS Disciplines
S. Barclay, ATIS Secretariat
C. Underkoffler, ATIS Chief Editor
M. Dolly & V. Shaikh, PTSC Technical Editors

Organization Represented	Name of Representative
AcmePacket	Kevin Klett
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Tellabs Operations, Inc.	William A. Walker
Tridea Works	Greg Ratta
VerisSign, Inc.	Anthony M. Rutkowski
Verizon Communications	Thomas Helmes Christine Huff

The Signaling, Architecture, and Control (SAC) Subcommittee was responsible for the development of this document.

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American National Standard for Telecommunications –

IP Network-to-Network Interface (NNI) Standard for VoIP

SUMMARY

This document defines a standard approach to support IP-IP interconnection for VoIP between carriers.

1 SCOPE, PURPOSE, AND APPLICATION

This standard defines the IP Network-to -Network Interface (NNI) for VoIP between carriers. It addresses the need for a standard interface as telecom networks migrate the NNI from TDM circuit-switched to IP. The focus of this standard is to support VoIP. This standard defines:

- Interconnection architecture;
- SIP call/session control signaling;
- Signaling and media transport;
- Quality of Service (QoS);
- Association between call control and media control; and
- Mandatory SIP URIs to be Supported.

There is also an informative annex on items for consideration in SLAs.

The following related topics are not defined in this document:

- Call Routing;
- Security;
- Session Border Controller Functions; or
- Call Admission Control and Traffic Management.

Figure 1 illustrates the relationship of this document to other related IP-NNI documents.

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