**ATIS-xxxxxxx**

ATIS Technical Report on

**VoIP Interconnection over the Public Internet**

**Alliance for Telecommunications Industry Solutions**

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**Abstract**

This document describes an "OTT Interconnection" model, where IP connectivity between peer SPs is established over the public internet.

**Foreword**

The Alliance for Telecommunications Industry Solutions (ATIS) serves the public through improved understanding between carriers, 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. International Telecommunication Union Telecommunication Sector (ITU-T) and U.S. ITU Radiocommunication Sector (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 SIP Forum is an IP communications industry association that engages in numerous activities that promote and advance SIP-based technology, such as the development of industry recommendations, the SIPit, SIPconnect-IT, and RTCWeb-it interoperability testing events, special workshops, educational seminars, and general promotion of SIP in the industry. The SIP Forum is also the producer of the annual SIP Network Operators Conference (SIPNOC), focused on the technical requirements of the service provider community. One of the Forum's notable technical activities is the development of the SIPconnect Technical Recommendation – a standards-based SIP trunking recommendation for direct IP peering and interoperability between IP Private Branch Exchanges (PBXs) and SIP-based service provider networks. Other important Forum initiatives include work in Video Relay Service (VRS) interoperability, security, Network-to-Network Interoperability (NNI), and SIP and IPv6.

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 an optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

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

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# Executive Summary

## Scope

This Technical Report describes an interconnect profile for VoIP Service Providers who choose to peer over the public internet. It recommends mechanisms for establishing IP connectivity, securing the signaling and media, and providing reliable quality-of-service for real-time media traversing the unmanaged public internet. The report does not describe the SIP interworking procedures on the VoIP interconnection between peering Service Providers.

## Purpose

# References

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

SIPconnect 2.0, *SIPconnect 2.0 Technical Recommendation, SIP Forum Document Number: TWG-11*

ATIS-1000074-E, *Signature-based Handling of Asserted Information using Tokens (SHAKEN).*

# Definitions, Acronyms, & Abbreviations

For a list of common communications terms and definitions, please visit the *ATIS Telecom Glossary*, which is located at < <http://www.atis.org/glossary> >.

## Definitions

**<term>:** <meaning>.

## Acronyms & Abbreviations

|  |  |
| --- | --- |
| ATIS | Alliance for Telecommunications Industry Solutions |

# Overview

Peering VoIP Service Providers traditionally interconnect through a carrier hotel, where the managed IP networks of the two peering partners are connected via private dedicated facilities. The carrier hotel model has good security and quality-of-service characteristics due to the physical security provided by the carrier hotel building, and the direct non-shared facility connecting the managed networks of the two peering partners.

This document describes an alternative "OTT VoIP Interconnection" model, where IP connectivity between peer SPs is established over the public internet. Since calls traverse the public internet in this case, special measures must be taken so that calls are delivered securely and with adequate quality. First, strong authentication mechanisms must be in place to ensure that peering partners can identify each other. Second, call signaling and media must be protected from eavesdropping or manipulation via man-in-the-middle attacks while traversing the open internet. Finally, media encode/decode across the OTT VoIP interconnection interface should utilize modern codec technology that incorporate the ability to survive packet-loss and packet congestion with the use of adaptive bit-rate support and forward error correction techniques to tolerate the potential of varying congestion levels encountered on the public internet.

## Reference Architecture

Figure 4.1 shows the reference architecture for the OTT Peering model. Peering partners SP-1 and SP-2 each deploy a Session Border Controller (SBC) at their peering interconnect point to support SIP signaling and media on the OTT VoIP Interconnect interface. SIP signaling across the interconnect interface is protected by TLS with mutual authentication, while the media is protected by SRTP. Opus is a good example of a royalty-free codec with very good adaptive characteristics and is in wide usage for OTT solutions, although a number of other codecs may also be similarly supported.



Figure 4.1 – OTT VoIP Interconnection Reference Architecture

# OTT VoIP Interconnection Procedures

## Preconfigured Information to support OTT VoIP Interconnection

Some level of information exchange must occur between two SPs who wish to establish a VoIP interconnection over the public internet. For example, each SP must provide an interface or mechanism to provide its interconnection partner with the IP addresses of the SBCs that terminate the OTT VoIP interconnect interface.

Additionally, there needs to be an interface to setup and update the set of OCNs that the interconnection partner will terminate directly to its customers; i.e., an interconnection partner would typically only want to receive traffic for the OCNs and associated LRNs that have been assigned to it by the numbering authority. This report proposes the following procedure for exchanging OCNs:

1. Each SP hosts a server that contains the list of OCNs assigned to that SP.
2. Each SP fetches the list of OCNs from its peering partner by sending an HTTP GET to the IP address of peering partner's SBC (all SBC IP addresses for a SP return the same list of OCNs). The body of the 200 OK response contains a JSON array of OCNs.
3. The SP updates its local routing database so that calls destined to the set of OCNs and associated LRNs obtained from the LERG are routed to the appropriate peering partner. The SP must portability correct the called TN during call origination processing before routing the call to the terminating interconnection service.

## Procedures to Establish/Use the OTT VoIP Interconnection Interface

This report leverages the procedures described in the SIP Forum document "SIPConnect 2.0 Technical Recommendation". SIPConnect 2.0 defines an interworking profile for the SIP Trunk interface between a SIP-PBX and its host VoIP Service Provider. The SIPconnect 2.0 reference architecture is shown in Figure 5.1, where interface (1) is the SIP signaling interface between the SP and SIP-PBX, while interface (2) is the media interface between the Enterprise and Service Provider media endpoints. SIPconnect 2.0 addresses all aspects of these interfaces, including transport, security, and authentication, plus a SIP profile for interface (1) and a media profile for interface (2). This report will reference only a subset of the procedures described in SIPconnect 2.0; primarily the procedures associated with transport, security, and authentication.

Diagram

Description automatically generated

Figure 5.1 – SIPconnect 2.0 Reference Architecture

SIPconnect 2.0 defines two modes of operation that differ in how the host SP obtains the SIP signaling address of the SIP-PBX:

* Registration Mode: the SIP-PBX provides its address dynamically using the SIP registration procedure,
* Static Mode: the SIP-PBX provides its address via DNS or static configuration.

As stated in SIPconnect 2.0, an Enterprise and Service Provider Network operating in the Static Mode can view each other as peer networks. Therefore, the transport and security mechanisms described by SIPconnect 2.0 for interfaces (1) and (2) can be directly applied to the OTT VoIP interconnection model.

The following clauses identify the specific SP-SSE and media endpoint procedures from SIPconnect 2.0 that must be supported by a VoIP SP at its OTT VoIP interconnection interface.

### Locating SIP Servers

The procedures to locate SIP servers using DNS described in section 17.1.2 of SIPconnect 2.0 shall not be applied to the OTT VoIP interconnection interface. Instead, interconnection partners shall exchange the public IP addresses of their SBCs that terminate the OTT VoIP interconnection interface.

### Signaling Transport, Security and Authentication

A VoIP SP shall support the SIP signaling transport, security and authentication procedures defined in sections 8, 17.2 and 17.5 of SIPconnect 2.0, with the exception that TLS with mutual is not a configuration option but is always enabled.

### Media and Session Interactions

Section 14 of SIPconnect 2.0 describes the procedures related to support of media on the SIP Trunk interface.

#### SDP Offer/Answer

While the SDP offer/answer procedures in section 14.1 are not in-scope of this report, it is assumed that they would generally apply to the OTT VoIP interconnection interface.

#### Media Transport

A VoIP SP shall support the media transport procedures described in section 14.2 of SIPconnect 2.0, with the exception that support for SRTP is mandatory.

#### Audio Profile

The audio profile described in section 14.3 of SIPconnect 2.0 shall not be applied to OTT VoIP interconnection interface. Instead, a VoIP SP shall support the variable bit rate audio codec Opus as specified in [RFC6716], with the payload format specified in [RFC7587]. Since Opus provides its own Comfort Noise (CN) mechanism, the use of [RFC3389] CN with Opus is NOT RECOMMENDED.

#### Media Security

A VoIP SP shall support the media security procedures described in section 14.4 of SIPconnect 2.0, with the exception that support for SRTP is mandatory.

#### Transport of DTMF Tones

A VoIP SP shall support the [RFC 4733] media transport of DTMF tones as described in section 14.5 of SIPconnect 2.0.

#### Echo Cancellation

The echo cancellation procedures described in section 14.6 do not apply to the OTT VoIP interconnection interface, since echo cancellation occurs at the user CPE device.

#### Fax Calls

A VoIP SP shall support the T.38 FAX transmission procedures described in section 14.7 of SIPconnect 2.0.

#### Other Media Requirements

The media procedures described in SIPconnect 2.0 sub-sections 14.8 through 14.10 do not apply to the OTT VoIP interconnection interface.