**ATIS-0x0000x**

ATIS Standard on

**Use Case and Requirements for STIR/SHAKEN Data Communications between Service Provider Networks and IP-PBX, UCaaS Systems etc.**

**Alliance for Telecommunications Industry Solutions**

Approved Month DD, YYYY

**Abstract**

This technical report defines mechanisms that enable a Service Provider to delegate STI authentication authority for a subset of its TNs to another entity. This delegation capability is needed to support STI for cases such as multi-homed SIP-PBXs, where the authorized owner of a TN does not provide originating call services for that TN.

**Foreword**

The Alliance for Telecommunications Industry Solutions (ATIS) serves the public through improved understanding between carriers, customers, and manufacturers. The [**COMMITTEE NAME**] Committee [**INSERT MISSION**]. [**INSERT SCOPE**].

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.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, [**COMMITTEE NAME**], 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, [**COMMITTEE NAME**], which was responsible for its development, had the following leadership:

[**LEADERSHIP LIST**]

The ATIS SIP Forum **IPNNI**  Joint Task Force Subcommittee was responsible for the development of this document.

**Revision History**

| **Date** | **Version** | **Description** | **Author** |
| --- | --- | --- | --- |
| February 5, 2018 | Initial | Baseline | Richard Shockey |

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# Scope, Purpose, & Application

## Scope

* The STIR/SHAKEN framework provides attestation about the nature of a call that is transversing the network across AS boundaries. Existing models of deployment have focused on individual end user deployments in mobile and cable networks.
* There is no current model on how STIR/SHAKEN might work in the critical landline Enterprise IP SIP PBX or SIP based call center networks utilizing modern SIP based Automatic Call Distribution Networks [ACD]. There is ample evidence that the data analytics generated by the framework is mission critical to multiple industries including Financial Services, Health Care and Utilities.

## Use Case

There is ample evidence that there is a market place for customized data analytics collected by service providers or others needs to be delivered to the call center or PBX utilizing well understood and widely deployed SIP trunking methods.

## Requirements

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# Normative References

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

SIP Forum SIP Connect 2.0

<https://www.sipforum.org/download/sipconnect-technical-recommendation-version-2-0/?wpdmdl=2818>

VERISTAT Paramater 3GPP

Enhanced CNAM ATIS

# 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

**AAA**: xxxx.

**Bbbb**: xxxx.

## Acronyms & Abbreviations

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

# Overview of SIP Connect

T**he SIP Forum SIP Connect Technical Recommendation** is an industry-wide, standards-based approach to direct IP peering between SIP-enabled IP PBXs and VoIP service provider networks. SIP Connect has been under continuous development and revision since 2005.

The current version SIP Connect 2.0 effectively extends SIP Connect 1.1. Where SIP Connect 1.0, and 1.1, focused primarily on basic network registration, identity/privacy management, call originations, call terminations, and advanced services, this version adds additional guidance on Security, Emergency Calling, and IPv6.

## The SIP Connect Architecture



SIP Connect only defines the “on the wire” interface between the service provider network and the enterprise.

## 

Here we have the classic example of how STIR/SHAKEN is deployed.

## 

The proposal would extends the SIP Connect model to STIR/SHAKEN interface between the service provider and the enterprise incorporating what ever data parameters are necessary to assist the enterprise in determining the trustworthiness of the call.

It is presumed that the veristat parameter would be passed but other parameters could be passed as well including proprietary data gathered by the service provider and call data analysts engines.

This proposal specifically excludes as **“out of scope**” any discussion or attempt to standardize how the STIR/SHAKEN data would be displayed within the enterprise or call center.

## 

Is such a model out of scope?

## One Proposal Combine VERISTAT with eCNAM

We understand how veristat is passed in the INVITE. That’s done. What other data paramaters are needed?

We would clearly redocument the tel URI parameter in the P-Asserted-Identity or FROM header field in a SIP requests P-Asserted-Identity: <tel:+14085264000;verstat=TN-Validation-Passed> or redefine this as needed.

Combine with Data extensions in the Call-Info header? [RFC 3261] Defined in ATIS eCNAM ?

## 

(normative/informative)

# A Annex Title

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