**ATIS-0x0000x**

ATIS Standard on

**Signature-based Handling of Asserted Information Using Tokens**

**Alliance for Telecommunications Industry Solutions**

Approved Month DD, YYYY

**Abstract**

Signature-based Handling of Asserted information using Tokens (SHAKEN) is an industry framework for managing the deployment of Secure Telephone Identity (STI) technologies with the purpose of providing end-to-end cryptographic authentication and verification of the telephone identity and other information in an IP-based service provider voice network. This specification defines the framework for telephone service providers to create signatures in SIP and defines the key Network-to-Network Interface (NNI) requirements, Network Elements, the X.509 certificate framework to validate the initiator of the signature, and the various classes of signers and how the validation of a signature can be used towards the mitigation and identification of illegitimate use of national telecommunications infrastructure and protecting its users.

**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 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, [**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 **[SUBCOMMITTEE NAME]** Subcommittee was responsible for the development of this document.

**Revision History**

| **Date** | **Version** | **Description** | **Author** |
| --- | --- | --- | --- |
| March 24, 2016 | 0.1 | Initial Draft | Chris Wendt |
| August 25, 2016 | 0.2 | Baseline Draft | Chris Wendt |

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# Scope & Purpose

## Scope

This document is intended to provide telephone services providers with a framework and guidance on how to utilize Secure Telephone Identity (STI) technologies toward the validation of legitimate calls and the mitigation of illegitimate spoofing of telephone identities on the VoIP Telephone Network.

## Purpose

Using the protocols defined in draft-ietf-stir-rfc4474bis, draft-ietf-stir-passport and draft-ietf-stir-certificate, this document will define the signature-based handling of asserted information using tokens (SHAKEN) framework. This framework is targeted at telephone service providers delivering telephone calls over VoIP, addressing the implementation and usage of the IETF STIR WG protocols and the architecture and management of STI-related certificates on VoIP networks. This includes definition of what STI certificates represent, as well as how they should be managed and distributed. It also discusses the general architecture of service provider authentication and verification services and identifies NNI and peering impacts and dependencies. Finally, it provides guidance on the use of positive or negative verification of the signature to mitigate illegitimate telephone identity in general, and also in the context of different call origination and destination scenarios.

Illegitimate CallerID spoofing is a growing concern for North American telephone service providers and their customers. There are many Caller ID spoofing mechanisms, and illegitimate spoofing can evolve to evade mitigation techniques. Service provider solutions must therefore be flexible to respond to evolving threats in much the same way as cybersecurity solutions. In addition, the integration of new technologies into established VoIP networks imposes many interoperability and interworking challenges. As a result, this document specifically focuses on a short term path for implementing STI in a progressive, practical, and realistic manner, with the initial steps defined in detail and the evolution path described in broad terms. The objective is to provide an approach that can evolve over time, incorporating more comprehensive functionality and a broader scope in a backward compatible and forward looking manner.

Editor’s Note: reconsider addressing short term path

# 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.

ATIS-0x0000x, *Technical Report*.

ATIS-0x0000x.201x, *American National Standard*.

draft-ietf-stir-passport

draft-ietf-stir-rfc4474bis

draft-ietf-stir-certificates

IETF RFC 3325 - Private Extensions to SIP for Asserted Identity within Trusted Networks

# 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

**Caller ID**: the originating or calling parties telephone number used to identify the caller carried either in the P-Asserted ID or From header.

## Acronyms & Abbreviations

|  |  |
| --- | --- |
| ATIS | Alliance for Telecommunications Industry Solutions |
| NNI | Network-to-Network Interface |
| PSTN | Public Switched Telephone Network |
| STI | Secure Telephone Identity |
| VoIP | Voice over Internet Protocol |

# Overview

This document presents the SHAKEN framework. SHAKEN is defined as a framework that utilizes protocols defined in the IETF STIR working group (WG) that work together in an end-to-end architecture for the authentication and assertion of a telephone identity by an originating service provider and the validation of the telephone identity by the terminating service provider.

Today, assertion of telephone identity in VoIP networks between peering service providers, particularly in a 3GPP IMS environment, typically uses the P-Asserted-ID as defined in RFC3325 as a network self-asserted identity. This usage assumes an inherent trust model between peering providers. However, in many telephone calling scenarios where there are many indirect call path relationships between the originating and terminating providers, these trust relationships are often simply not verifiable and do not allow for identification of the true origination of the call. In addition, P-Asserted-ID can be populated by an enterprise PBX and passed on without verification by the service provider. Secure Telephone Identity (STI) as defined in the STIR WG and the usage of cryptographic digital signatures to verify the originator of a signed identity can provide a verifiable mechanism to identify the authorized originator of a call into the telephone network with non-repudiation and assignment of an attestation indicator and a unique ID depending on how and where the call is originated or received. This attestation and identifier represents the originating signers ability to vouch for the accuracy of the source of origin of the call. For example, if the service provider has an authenticated direct relationship with the origination of the call this attestation is categorized differently than calls that are originated from different networks or gateways that the service provider may have received from an unauthenticated network or that are unsigned. Verification of signatures will use these attestations as information to provide trace back mechanisms as well as information to feed into any call spam identification techniques the service provider has enabled on behalf of their customer.

## STIR Overview

The documents draft-ietf-stir-rfc4474bis and draft-ietf-stir-passport define a set of protocol level tools that can be used in SIP for applying digital signatures to the CallerID or telephone number of the calling party.

### PASSporT Token

The document draft-ietf-stir-passport defines a token based signature that combines the use of JSON Web Tokens, JSON Web Signatures, and X.509 certificate key pairs, or PKI, to create a trusted signature. The authorized owner of the certificate used to generate the signature can be validated and traced back to the known trust anchor who signed the certificate. The PASSporT token includes a number of "claims" the signer of the token is passing with non-repudiation. The associated public certificate is used to verify the digital signature and the “claims” included in the PASSporT token. The public certificate is also used to validate the entity that signed the token. The validated claims, and the validated identity of the entity signing the claims, can both be used to determine the level of trust in the calling party information. Call blocking applications could use this information over time to determine “reputation” of the entity signing the token, which could provide further input to determine the level of trust for the calling party information. Note that PASSporT signatures are agnostic to network signaling protocols.

### RFC4474bis

The document draft-ietf-stir-rfc4474bis defines a SIP based framework for an authentication service and verification service for using the PASSporT signature in a SIP INVITE. It defines a new Identity header field that delivers the PASSporT signature and other associated parameters. The authentication service adds the Identity header field and signature to the SIP INVITE generated by the originating provider. The INVITE is delivered to the destination provider which uses the verification service to validate the signature using the asserted identity in the P-Asserted-ID header field or From header field.

P-Asserted-ID must be used as the telephone identity if present, otherwise the From header field should be used. This is true both on the Authentication side for the telephone identity verified as well as on the verification side when validation of the INVITE and Identity header field occurs.

## SHAKEN Architecture

There are a number of required architectural components required for an end-to-end framework for STI.

The figure below shows the SHAKEN reference architecture. This is a logical view of the architecture and doesn’t mandate any particular deployment and/or implementation. For reference, this architecture is specifically based on the 3GPP IMS architecture with an IMS application server, and is only done as an example for discussion in this document.

 

Figure : SHAKEN reference architecture

This SHAKEN reference architecture includes the following elements:

* SIP UA - SIP User Agent that is authenticated by the service provider network is considered secure and the calling party identity is “known” since it is under direct management by the telephone service provider. It initiates the SIP INVITE as the calling party.
* IMS/CSCF - This component represents the SIP registrar and routing function. It also has a SIP application server interface.
* IBCF/TrGW - This function is at the edge of the service provider network and represents the NNI or peering interconnection point between telephone service providers. It is the ingress and egress point for SIP calls between providers.
* Authentication Service (STI-AS) - The SIP application server that performs the function of the authentication service defined in 4474bis. It has an HTTPS interface to the Secure Private Key Store (SKS) which stores the secret private key certificate used to create the PASSporT signature.
* Verification Service (STI-VS) - The SIP application server that performs the function of the verification service defined in 4474bis. It has an HTTPS interface to the Certificate Repository that is referenced in the identity header field to retrieve the provider public key certificate.
* Call Validation Treatment (CVT) - This is a logical function that could be an application server function or a third party application for applying anti-spoofing mitigation techniques once the signature is positively or negatively verified and then provides a response to signal the display response for the end user.
* SKS – Secure Key Store is a logical place to store private keys for the authentication service to access.
* Certificate Provisioning Portal – The Certification Authority (CA) or Telephone Authority (TA) equivalent in SHAKEN validates requests for telephony certificates and represents the mechanism the originating service provider uses to get its public key certificate signed via a Certificate Signing Request (CSR).
* TN Certificate Repository (TN-CR): This represents the publically accessible store for public key certificates maintained by the service provider. This should be an HTTPS web service that can be validated back to the owner of the public key certificate.

## SHAKEN call flow



Figure : SHAKEN reference call flow

1. The originating SIP UA, which first REGISTERs and is authenticated to the CSCF, creates a SIP INVITE with an E.164 calling number.
2. The CSCF of the originating provider adds a P-Asserted-Identity header field asserting the Caller ID of the originating SIP UA. The CSCF then initiates an originating trigger to the STI-AS for the INVITE.
3. The STI-AS in the originating SP (i.e., Service Provider A) retrieves its private key from the SKS.
4. The SKS provides the private key, and the STI-AS signs the INVITE and adds an Identity header field per RFC 4474bis using the Caller-ID in the P-Asserted-Identity header field.
5. The STI-AS passes the INVITE back to the SP A CSCF.
6. The originating CSCF, through standard resolution, routes the call to the egress IBCF.
7. The INVITE is routed over the NNI through the standard inter-domain routing configuration.
8. The terminating SP (Service Provider B) ingress IBCF receives the INVITE over the NNI.
9. The terminating CSCF initiates a terminating trigger to the STI-VS for the INVITE.
10. The terminating SP STI-VS uses the Info header field parameter in the Identity header field per RFC 4474bis to determine the TN-CR URI and the originating TN.
11. The STI-VS validates the certificate, which can include these steps: check the validity dates, check the certificate’s signature, check chain of trust, and check certificate validity via CRLs and/or OCSP. It then extracts the public key. It constructs the RFC 4474bis format and uses the public key to validate the signature in the Identity header field, which validates the Caller ID used when signing the INVITE on the originating service provider STI-AS.
12. The CVT is and optional function that can be invoked to perform call spam analytics or other mitigation techniques and return a response related to what is displayed to the user for legitimate or illegitimate call determination. Note: In some implementations, the CVT may be physically integrated with the STI-VS. The CVT may be integrated in the service provider network or outside the service provider network by a third party.
13. Depending on the result of the STI validation, the STI-VS determines that the call is to be completed with any appropriate indicator (that may be defined outside of this document) and the INVITE is passed back to the terminating CSCF which continues to set up the call to the terminating SIP UA. Note: Error cases where verification fails are discussed in Section 6.
14. The terminating SIP UA receives the INVITE and normal SIP processing of the call continues, returning “200 OK”, or optionally setting up media end-to-end.

Editor’s Note: will modify the diagram to have the CVT straddle the service provider boundary

# STI SIP Procedures

Both draft-ietf-stir-4474bis and draft-ietf-stir-passport define a base set of procedures for how STI fits into the SIP call flow. 4474bis defines an authentication service, corresponding to STI-AS in the SHAKEN reference architecture, as well as a verification service or STI-VS. This section will detail the procedures required for the STI-AS to create the identity header required.

## PASSporT Token Overview

PASSporT tokens have the following form:

* A protected header with the value BASE64URL(UTF(JWS Protected Header))
* A payload with the value BASE64URL(JWS Payload)
* A signature with the value BASE64URL(JWS Signature)

An example of each is as follows:

*Protected Header*

{

 "typ":"passport",

 "alg":"ES256",

 "x5u":"https://cert.example.org/passport.crt"

}

*Payload*

{

"iat":"1443208345",

 "orig":{“tn”:"12155551212"},

 "dest":{“uri”:"sip:alice@example.com"}

}

draft-ietf-stir-passport has specific examples of a Passport token.

## 4474bis Authentication procedures

### PASSporT extension ‘shaken’

The base passport set of claims cover the assertion of the telephone number along with date and destination telephone numbers to avoid replay attacks using valid identity headers. This section will detail a specific extension to the PASSporT to cover the following requirements of SHAKEN.

1. The ability to provide an attestation indicator for the context of how the call was originated.
2. The ability to provide a unique identifier that can service as an opaque indication of where in the service provider network the call was originated.
	1. This identifier MUST be globally unique and consistent so can be used in analytics for tracking the reputation of a particular originating service.
	2. This identifier MUST be globally unique and consistent so it can be used for any traceback efforts if a particular originator is a consistent or pervasive “bad actor”.

Draft-wendt-stir-shaken-passport-extension defines a PASSporT extension IANA registered names for “shaken” extension and the claims that represent both an attestation indicator (attest) and Origination Identifier (origID). The format of these claims are defined in the following sections, but the PASSporT token would have the form:

*Protected Header*

{

 "alg":"ES256",

 "typ":"passport",

 “ppt”:”shaken”,

"x5u":"https://cert.example.org/passport.crt"

}

*Payload*

{

 “attest”:”A”

 "dest":{“uri”:"sip:alice@example.com"}

"iat":"1443208345",

 "orig":{“tn”:"12155551212"},

 “origid”:”123e4567-e89b-12d3-a456-426655440000”

}

### Attestation Indicator

e

For the PASSporT extension claim, the “attest” key value pair MUST be set to “A”, “B”, or “C” corresponding to the appropriate attestation listed above.

TBD: Should there be a case for no attest value? Or should we forbid it.

### Origination Identifier (origID)

 (origID)

## 4474bis Verification procedures

Draft-ietf-stir-rfc4474bis defines the procedures for verification services including the methods used to verify the signature contained on the identity header.

### Verification Error conditions

If the authentication service functions correctly, and the certificate is valid and available to the verification service, the SIP message can be delivered successfully. However, if these conditions are not satisfied, errors can be generated as defined in rfc4474bis. This section identifies important error conditions and specifies procedurally what should happen, if they occur. Error handling procedures should consider how best to always deliver the call per current regulatory requirements, while providing diagnostic information back to the signer.

There are four main procedural errors defined in rfc4474bis that can identify issues with allowing the validation of the Identity header field to occur. They are:

**436** – ‘Bad-Identity-Info’ – the URI in the info parameter cannot be dereferenced (i.e., the request times out or receives a 4xx or 5xx error)

**437** – ‘Unsupported credential’ – this error occurs when a credential is supplied by the info parameter but the verifier doesn’t support it or it doesn’t contain the proper certificate chain in order to trust the credentials.

**438** – ‘Invalid Identity Header’ – this occurs if the signature validation fails

**403** - ‘Stale Date’ – sent when the verification service receives a request with a Date header field value that is older than the local policy for freshness permits.

**428** – ‘Use Identity Header’ is not recommended for SHAKEN until a point where all calls on the VoIP network are mandated either by local or global policy to be signed.

If any of these error conditions occur, a reason code MUST be passed in the 18x provisional response that indicates one of the four above scenarios.

Example:

Reason: SIP ;cause=436 ;text=“Bad Identity Info”

Optionally, the provision response could be followed up by Authentication Service with a RE-INVITE in the same dialog with a “repaired” identity header field.

### Call rejection

TBD: consider “call rejected by user – spam” or “call rejected by network – spam” scenerios

Assuming there is any additional call spam analytics functions being performed by a CVT that decides based on service provider policy to reject the call. The STI-VS should send a BYE with a reason code …

TBD: should this be reason code or call-info, if reason code do we need a draft to define these or maybe there is existing SIP errors we can map these too.

## Use of canon parameter

For initial SHAKEN deployment, canon must be included to avoid any potential SBC interaction with headers, especially the Date header field, which could lead to large numbers of 438, Invalid Identity Header errors.

TBD text to explain further

## SIP Identity Header

Draft-ietf-stir-rfc4474bis defines the identity header field for SIP. It uses the PASSporT token as a basis for creation of the Identity header field for INVITE, MESSAGE, and NOTIFY SIP messages.

The procedure is detailed in rfc4474bis, but an example of an INVITE with an Identity header field is as follows:

INVITE sip:+12155551212@tel.example.net SIP/2.0
Via: SIP/2.0/UDP 10.36.78.177:60012;branch=z9hG4bK-524287-1---77ba17085d60f141;rport
Max-Forwards: 69
Contact: <sip:+12155551212@69.241.19.12:50207;rinstance=9da3088f36cc528e>
To: <sip:+12155551213@tel.example.net>
From: "Alice"<sip:+12155551212@siptest.example.net>;tag=614bdb40
Call-ID: 79048YzkxNDA5NTI1MzA0OWFjOTFkMmFlODhiNTI2OWQ1ZTI
CSeq: 2 INVITE
Allow: SUBSCRIBE, NOTIFY, INVITE, ACK, CANCEL, BYE, REFER, INFO, MESSAGE, OPTIONS
Content-Type: application/sdp
Date: Tue, 16 Aug 2016 19:23:38 GMT
Identity: eyJ0eXAiOiJwYXNzcG9ydCIsImFsZyI6IkVTMjU2IiwieDV1IjoiaHR0cHM6Ly9jZXJ0LWF1dGgucG9jLnN5cy5jb21jYXN0Lm5ldC9leGFtcGxlLmNlcnQifQ.eyJkZXN0Ijp7InVyaSI6WyJzaXA6MTAwMEBzaXB0ZXN0LmNvbWNhc3QubmV0Il19LCJpYXQiOiIxNDcxMzc1NDE4Iiwib3JpZyI6eyJ1cmkiOiJzaXA6NTcxMjIyMzMzM0BzaXB0ZXN0LmNvbWNhc3QubmV0In19;info=<https://cert.example.net/example.crt>;alg=ES256
Content-Length: 153
v=0
o=- 13103070023943130 1 IN IP4 10.36.78.177
c=IN IP4 10.36.78.177
t=0 0
m=audio 54242 RTP/AVP 0
a=sendrecv

Editor’s Note: discuss NNI implications

# STI Certificate Management Model

Management of certificates for TLS and HTTPS based transactions on the Internet is well defined and common practice for website and internet applications. Generally, there are recognized certification authorities that can "vouch" for the authenticity of a domain owner based on some out-of-band verification techniques like e-mail and unique codes in DNS.

The certificate management model for SHAKEN is based on Internet best practices to the extent possible, but is modified where appropriate to reflect unique characteristics of the service provider based telephone network. Certificates are initially expected to take advantage of service providers’ recognized ability to legitimately assert telephone identities on a VoIP network. The following sections detail the SHAKEN approach to the certificate management model that allows telephone authorities to sign certificates for use on the telephone network. The roles involved in the management of certificates are described, along with a governance model. The process for acquiring certificates is described in the context of an architecture within the scope of the SHAKEN framework.

## Certificate Management Roles

The SHAKEN certificate management model is illustrated in the following diagram.



This diagram defines the following roles in the certificate management model:

* Governance Authority (GA)
* Administration and Verification Authority (AVA)
* Telephone Authority (TA)
* Service Provider (SP)

The Governance Authority and the Administration and Verification Authority are distinct roles in this model, though in practice both roles could be performed by a single entity. This entity is the root of trust for all STI certificates within a given area. For example, all certificates in the United States would be associated with a single root of trust, although other countries could have a different root of trust. It is also worth noting that although the Telephone Authority and Service Provider are distinct roles, it would also be possible for a Service Provider to establish an internal Telephone Authority for their own use.

The following sections describe these roles in more detail.

### Governance Authority

The Governance Authority is responsible for defining and modifying the rules that the Administration and Verification Authority will use to authorize Telephone Authorities and to verify Service Providers. It is anticipated that the Governance Authority would be structured as a Committee or as a Board of Directors. The criteria for membership / participation in the Governance Authority is out of scope for SHAKEN.

### Administration and Verification Authority

The Administration and Verification Authority will apply the rules defined by the Governance Authority to verify service providers are authorized to request certificates and to authorize telephone authorities to provide the certificates.

### Telephone Authority (TA)

In X.509, there is the concept of Certification Authorities (CA). There are two flavors of CAs - a root CA and an intermediate CA. The root CA represents the Trust Anchor in a X.509 certificate. When constructing a public key certificate, a certificate chain is created that represents a chain from the domain owner to the trust anchor. This generally can include the domain owner, multiple intermediate CAs and the root CA.

Analogous to the concept of Certificate Authorities, SHAKEN defines the concept of a Telephone Authority (TA). A Telephone Authority acts as a root certificate provider to validate authorized signatures for telephone numbers on a VoIP network.

In the North American telephone network, it is anticipated that the number of entities that should act as an authority is a relatively limited number. Certificate signing requests (CSRs) will be directly validated and processed by TAs and will be linked to the Administration and Verification Authority which is the trust anchor represented in the certificate chain. Note, that this makes the SHAKEN model slightly different than the X.509 model whereby the root CA is the trust anchor.

### Service Provider

The Service Provider obtains certificates from the Telephone Authority and uses the certificate to sign and verify calling party information in SIP calls.

## Governance Model

This section describes the process for establishing Telephone Authorities and validating legitimate service providers. This section provides details on how this process will work.

Editor’s Note: the text from this section may be pulled out into a separate document in the future

### Service Provider Criteria

Ultimately this is the responsibility of the Governance Authority, but the initial criteria for obtaining Service Provider certificates will be having an OCN (Operating Company Number) as administered by the National Exchange Carrier Association. The OCN is proposed as an objective mechanism to determine that an entity is a service provider and entitled to sign calling party information. Initially there will not be a mechanism to revoke service provider certificates, although the Governance Authority will have the ability to define criteria for revoking certificates (e.g., signing invalid numbers) if it is determined to be appropriate. In addition, as a condition of being validated as a service provider for SHAKEN, service providers would need to commit to signing calling party information for all calls where it is technically and economically feasible.

### Service Provider Validation Process

Note: this section will outline the process used by a service provider to be validated. This will include:

* The interface between the service provider and the SP Administrator/Validator (e.g., API)
* Details of the “token” the service provider obtains that will allow the service provider to request STI certificates from a TA (e.g., a certificate signed by the Administration and Verification Authority root).

### Telephone Authority Criteria

Ultimately this is the responsibility of the Governance Authority, but the initial criteria for becoming a Telephone Authority would be:

* Have the necessary certificate management expertise
* Have an in-market presence (e.g., be incorporated in the U.S.)
* Having a service provider express interest in using their service could be a useful but is not considered to be a mandatory requirement.

### Telephone Authority Approval Process

Note: this section will outline the process used by a Telephone Authority to obtain approval to operate as a Telephone Authority and to obtain a certificate that can be used to sign STI certificates for service providers. This will include:

* The interface between the Telephone Authority and the SP Administrator/Validator (e.g., API)
* Details of how the Telephone Authority obtains a certificate signed by the Administration and Verification Authority root.

## Certificate Management Architecture

The following figure represents the certificate management architecture for SHAKEN.



Figure : SHAKEN Certificate Management Architecture

The SHAKEN certificate management architecture defines the following elements:

* Telephone Authority Management Server (TAMS) - The telephone authority server that processes the Certificate Signing Request (CSR) following a service provider verification process.
* Service Provider Key Management Server (SP-KMS) - The service provider server that generates private/public key pair for signing, submits to Telephone Authority Management Server, and receives the TA signed public key certificate.
* Secure Key Store (SKS) - The store for private keys used by the originating service provider Authentication Service.
* Certificate Repository (TN-CR) - The HTTPS server that hosts the public key certificates used by destination service provider Verification Service to validate signatures.

## Certificate Management Process

### ACME based Certificate Management Flow

ACME (draft-ietf-acme-acme) provides a more automated framework and set of protocols for acquiring a telephone authority signed public key certificate. ACME allows a client to request certificate management actions using a set of JSON messages carried over HTTPS, much like a traditional CA.

ACME enables the following certificate management functions:

* Account Key Registration
* Application for a Certificate
* Account Key Authorization (Service Provider Verification)
* Certificate Issuance
* Lifecycle Management of certificates (including Revocation)

Prior to being able to request certificates from a specific TA, an ACME client needs to first be registered with that TAMS per the procedures in draft-ietf-acme-acme.

The ACME flow for a telephone authority is as follows:

* The ACME client on the Service Provider Key Management Server prompts the operator for the service provider domain the Authentication Service is to represent.
* The ACME client presents the operator with a list of TAs from which it could get a certificate.
* The operator selects a TA.
* In the background, the ACME client contacts the TAMS and requests that a certificate be issued for the intended domain.
* Once the TAMS is satisfied that the requestor is authorized to manage certificates for the requested domain per section 8.4.3, the certificate is issued and the ACME client automatically downloads and installs it, potentially notifying the operator via e-mail, SMS, etc.
* The ACME client periodically contacts the TAMS to get updated public key certificates, CRLs, or whatever else would be required to keep the server functional and its credentials up-to-date as described in section 8.4.4**.**.

### Service Provider verification

A process is required that allows the telephone authority to verify that the service provider has the authority to manage certificates for the domain for which a certificate is being requested. In the context of ACME, the ACME client fetches the challenges after the request for a new certificate and then answers the challenges.

ACME uses an extensible challenge/response framework for identifier validation. For this initial deployment of the SHAKEN framework, it is recommended to use HTTP validation per draft-ietf-acme-acme. The ACME client proves its control over the domain by proving that it can provision resources on the Authentication Service server. The TAMS challenges the ACME client to provision a file at a specific path, with a specific string as its content. Note that longer term the use of the approach as described in section 8.2.2 is recommended.

### Certificate updates/rotation best practices

Consideration of impact of switching certificates and other certificate management impacts while there is in flight calls should be considered. Standard CRL techniques should be considered the initial preferred way of signaling the expiry of a certificate. OCSP techniques could be considered in the future.

### Evolution of STI certificates

SHAKEN proposes starting with service provider level certificates. There are important use cases that may require telephone number level certificates including School District, Police and government agencies, where calls should be validated in order to guarantee delivery through the potential use of anti-spoofing mitigation techniques.