ATIS-0300116

ATIS Standard on -

Interoperability Standards between Next Generation Networks (NGN) for Signature-based Handling of Asserted information using ToKENs (SHAKEN)

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Interoperability Standards between Next Generation Networks (NGN) for Signature-based Handling of Asserted information using ToKENs (SHAKEN)

Alliance for Telecommunications Industry Solutions

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Abstract

This document provides Next Generation Network (NGN) service providers (SPs) with a framework and guidance for interoperability as calls process through their networks implementing Signature-based Handling of Asserted information using ToKENs (SHAKEN) technologies to ensure the validation as well as the completion of legitimate calls and the mitigation of illegitimate spoofing of telephone identities.

Foreword

The Alliance for Telecommunications Industry Solutions (ATIS) serves the public through improved understanding between carriers, customers, and manufacturers. The Next Generation Interconnection Interoperability Forum (NGIIF) addresses next generation network interconnection and interoperability topics associated with emerging technologies. Specifically, it develops operational procedures that involve the network aspects of architecture, disaster preparedness, installation, maintenance, management, reliability, routing, security, and testing between network operators. In addition, the NGIIF addresses issues that impact the interconnection of existing and next generation networks and facilitate the transition to emerging technologies.

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, NGIIF, 1200 G Street NW, Suite 500, Washington, DC 20005.

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

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1.1 Scope

This document provides Next Generation Network (NGN) service providers (SPs) with a framework and guidance for interoperability as calls process through their networks implementing Signature-based Handling of Asserted information using ToKENs (SHAKEN) technologies to ensure the validation as well as the completion of legitimate calls and the mitigation of illegitimate spoofing of telephone identities.

1.2 Purpose

This document defines the operationalization between NGN SPs, for SHAKEN framework.

2 Normative References

The following standards contain provisions that, 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-1000074, *Signature-based Handling of Asserted information using ToKENs (SHAKEN).*[1]

ATIS-0300106, *Intercarrier Call Completion/Call Termination Handbook.*[1]

Report and Order and *Further Notice of Proposed Rulemaking* in Federal Communications Commission (FCC) 13-135 and WC Docket No.13-39, adopted October 28, 2013 and released November 8, 2013.[2]

Rules and Regulations Implementing the Telephone Consumer Protection Act of 1991, CG Docket No. 02-278, WC Docket No. 07-135, *Declaratory Ruling and Order, FCC 15-72*, (released July 10, 2015).[2]

RFC 8224, *Authenticated Identity Management in the Session Initiation Protocol (SIP)*.[3]

RFC 8588, *Persona Assertion Token*.[3]

3 Definitions, Acronyms, & Abbreviations

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

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[1] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS) at < https://www.atis.org/docstore/>.

[2] This document is available from the Federal Communications Commission (FCC) at: < https://www.fcc.gov/ >.

[3] This document is available from the Internet Engineering Task Force (IETF) at: < https://datatracker.ietf.org/doc/draft-ietf-stir-rfc4474bis/ >.

3.1 Definitions

Trusted Network: A trusted network is a network that follows the Third Generation Partnership Project (3GPP) Trust Model.[4]

Untrusted Network: An untrusted network is a network that does not follow the 3GPP Trust Model.[5]

Local Policy: Factors that could influence "local policy" include state, national, international regulation, operator preferences (consistent with applicable regulation) and user preferences.

3.2 Acronyms & Abbreviations

ATIS	Alliance for Telecommunications Industry Solutions
CRL	Certificate Revocation List
CSCF	Call Session Control Function
CVT	Call Validation Treatment
FCC	Federal Communications Commission
HTTPS	Hypertext Transfer Protocol Secure
IBCF	Interconnection Border Control Functions
ID	Identification
IETF	Internet Engineering Task Force
IMS	IP-Multimedia Subsystem
JSON	JavaScript Object Notation
NGIIF	Next Generation Interconnection Interoperability Forum
NGN	Next Generation Network
NNI	Network-to-Network Interface
PASSporT	Persona Assertion Token
RFC	Request for Comments
SHAKEN	Signature-based Handling of Asserted information using ToKENs
SIP	Session Initiation Protocol
SIP UA	Session Initiation Protocol User Agent
SKS	Secure Private Key Store
SP	Service Provider
STI	Secure Telephone Identity
STI- AS	Secure Telephone Identity- Authentication Service
STI- CR	Secure Telephone Identity- Certificate Repository

[4] 3GPP TS 33.234 V023.0 (2002-11); 3GPP TS 29.165 V11.5.0 (2012-12).

[5] 3GPP TS 33.234 V023.0 (2002-11); 3GPP TS 29.165 V11.5.0 (2012-12).

STI-PA	Secure Telephone Identity - Policy Administrator
STI-VS	Secure Telephone Identity -Verification Service
STIR	Secure Telephone Identity Revisited
3GPP	Third Generation Partnership Project
TN	Telephone Number
TrGW	Transition Gateway
UA	User Agent
URI	Uniform Resource Identifier
VoIP	Voice over Internet Protocol
VS	Verification System

4 Overview

This interoperability standards document provides NGN SPs with the information to interoperate while utilizing the SHAKEN framework for the authentication and assertion of a telephone identity on origination and the validation of the telephone identity of calls across networks.

4.1 Operationalization Assumptions

- All calls must attempt to complete to the end user.[6]
- SPs that sign Persona Assertion Token (PASSporTs) will have the means to determine the legitimacy of the calling party identity.
- Today, assertion of telephone identity in Voice over Internet Protocol (VoIP) networks between peering SPs, particularly in a 3GPP Internet Protocol Multimedia Subsystem (IMS) environment, typically uses the P-Asserted-Identity as defined in RFC 3325 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. Currently, the P-Asserted-Identity header field can be populated by an enterprise Private Branch Exchange and passed on without validation by the SP.[7]
- Use of standardized cryptographic digital signatures to validate the originator of a signed identity can provide a verifiable mechanism to identify the authorized originator of a call into the VoIP network with non-repudiation. Further, the use of an assigned attestation indicator and a unique origination identifier depending on how and where the call is originated in the VoIP network represents the originating signer's ability to vouch for the accuracy of the source of origin of the call. For example, if the SP 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 SP may have received from an unauthenticated network or that are unsigned. Verifiers of signatures will use these attestations as information to provide trace back mechanisms, as well as information to feed into any call spam identification solution enabled on behalf of their customer.[8]

[6] Report and Order and Further Notice of Proposed Rulemaking in FCC 13-135 and WC Docket No.13-39, adopted October 28, 2013 and released November 8, 2013.

[7] ATIS-1000074, *Signature-based Handling of Asserted information using Tokens (SHAKEN).*

[8] ATIS-1000074, *Signature-based Handling of Asserted information using Tokens (SHAKEN).*

- PASSporTs signed by trusted SPs will be accepted by the terminating provider even if the call is routed/handed off from any intermediate provider.
- Common procedures will be established by the industry to deal with cases when a PASSporTs cannot be signed by the originating SP due to a network event or traffic congestion.
- Common procedures will be established by the industry to deal with cases when a received PASSporT cannot be validated by the terminating SP due to a network event or traffic congestion.
- Only the end user can choose to block the call or delegate that authority to block the call or use a call-blocking mitigation technique.[9] Calls that are sent to an intermediate provider and are "cranked back" (a term used to describe the means by which SPs routed around congestion) must continue processing to the next step in an attempt to complete the call to the end user of the dialed digits.[10]
- Terminating SPs shall complete the call to the end user (i.e., the called party) unless otherwise specified by the end user to block the call.
- Terminating SPs, while processing calls without signed PASSporTs in their networks, will follow SP specific processes for processing the Caller Identification (ID).
- Terminating SPs, while processing calls with PASSporTs signed by an SP other than the originating SP, will follow SP specific processes for processing the Caller ID.
- SPs will follow SP specific processes to address call handling if the certificate validation fails.
- Terminating SPs will query the Certificate Revocation List (CRL), maintained by the Secure Telephone Identity - Policy Administrator (STI-PA), to determine if an originating SP's certificate has been revoked.
- Intermediate SPs are expected to pass all signaling without modifications. In cases where intermediate SPs legitimately (via contract or government mandate) make changes, they will apply their own signatures to the result (such as via government contract for certain national security services).
- Terminating SPs will follow SP specific processes to address the presentation of Caller ID.

4.2 STIR/SHAKEN Overview

4.2.1 Secure Telephone Identity Revisited (STIR) Overview

The documents RFC 8224 and RFC 8588 define a set of protocol level tools that can be used in Session Initiation Protocol (SIP) for applying digital signatures to the Caller ID or telephone number of the calling party.[11]

4.2.2 PASSporT

The document RFC 8588 defines a token-based signature that combines the use of JavaScript Object Notation (JSON) Web Tokens, JSON Web Signatures, and X.509 certificate key pairs, or Public Key Infrastructure, 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 includes a number of claims the signer of the PASSporTs is asserting. The associated public certificate is used to verify the digital signature and the claims included in the PASSporT. The public certificate is also used to validate the entity that signed the PASSporTs. 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 originating entity and their asserted calling party information. Note that PASSporTs and signatures themselves are agnostic to network signaling protocols but are used in RFC 8224 to define specific SIP usage as described in the next section.[12]

4.2.3 IETF RFC 8224

The document RFC 8224 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

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[9] Rules and Regulations Implementing the Telephone Consumer Protection Act of 1991, CG Docket No. 02-278, WC Docket No. 07-135, Declaratory Ruling and Order, FCC 15-72, (released July 10, 2015).

[10] ATIS-0300106, *Intercarrier Call Completion/Call Termination Handbook*, Clause 6.

[11] ATIS-1000074, *Signature-based Handling of Asserted information using Tokens (SHAKEN).*

[12] Ibid.

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 verify the signature using the identity in the P-Asserted-Identity header field or From header field.[13]

4.3 SHAKEN Architecture

There are a number of architectural components required for an end-to-end Secure Telephone Identity (STI) framework.

The figure below shows the SHAKEN reference architecture. This is a logical view of the architecture and does not 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 provided as an example to set the context for the functionality described in this document. The diagram shows the two IMS instances that comprise the IMS half-call model; an originating IMS network hosted by SP A, and a terminating IMS network hosted by SP B.[14]

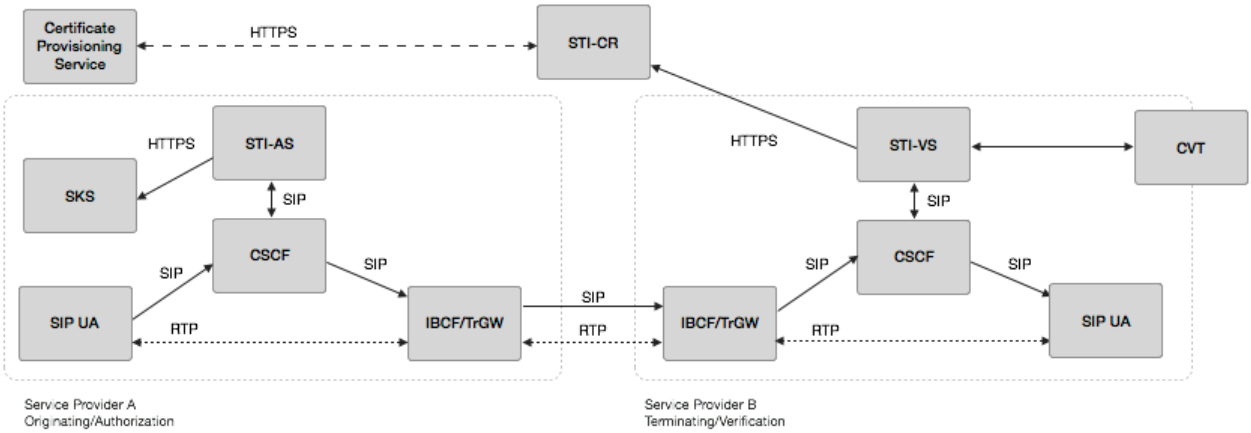

Figure 1- SHAKEN Reference Architecture[15]

This SHAKEN reference architecture includes the following elements:

- Session Initiation Protocol User Agent (SIP UA) – The SIP UA authenticated by the SP network. When the SIP UA is under direct management control of the SP, the SP network can assert the calling party identity in originating SIP INVITE requests initiated by the SIP UA.
- IMS/Call Session Control Function (CSCF) – This component represents the SIP registrar and routing function. It also has a SIP application server interface.
- Interconnection Border Control Function (IBCF)/Transition Gateway (TrGW) – This function is at the edge of the SP network and represents the Network-to-Network Interface (NNI) or peering interconnection point between SPs. It is the ingress and egress point for SIP calls between SPs.
- Secure Telephone Identity-Authentication Service (STI-AS) – The SIP application server that performs the function of the authentication service defined in RFC 8224. It should either itself be highly secured and contain the Secure Key Store (SKS) of secret private key(s) or have an authenticated, Transport Layer Security - encrypted interface to the SKS that stores the secret private key(s) used to create PASSporT signatures.

[13] Ibid.

[14] Ibid.

[15] Ibid.

- Secure Telephone Identity-Verification Service (STI-VS) – The SIP application server that performs the function of the verification service defined in RFC 8224. It has a Hypertext Transfer Protocol Secure (HTTPS) interface to the STI Certificate Repository that is referenced in the Identity header field to retrieve the SP's 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. The CVT can also provide information in its response that indicates how the results of the verification should be displayed to the called user.
- SKS – The SKS is a logical highly secure element that stores secret private key(s) for the STI-AS to access.
- Certificate Provisioning Service – A logical service used to provision certificate(s) used for STI.
- Secure Telephone Identity-Certificate Repository (STI-CR) – This represents the publicly accessible store for public key certificates. This should be an HTTPS web service that can be validated back to the owner of the public key certificate.[16]

4.4 SHAKEN Call Flow

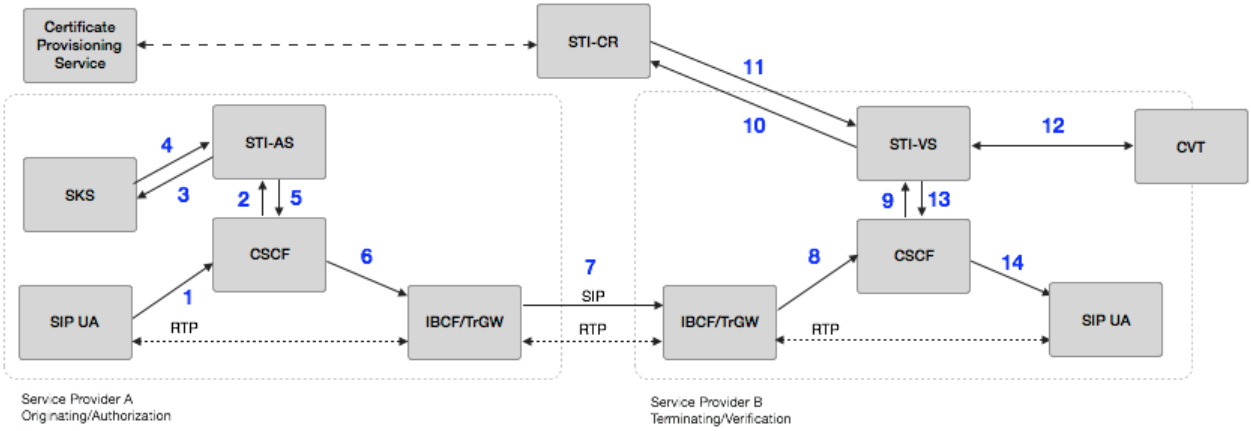

Figure 2- SHAKEN Reference Call Flow[17]

Step 1. The originating SIP UA, which first REGISTERs and is authenticated to the CSCF, creates a SIP INVITE with a telephone number identity.

Step 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.

NOTE: The STI-AS must be invoked after originating call processing.

Step 3. The STI-AS in the originating SP (i.e., SP A) first determines through service provider-specific means the legitimacy of the telephone number identity being used in the INVITE. The STI-AS then securely requests its private key from the SKS.

Step 4. The SKS provides the private key in the response, and the STI-AS signs the INVITE and adds an Identity header field per RFC 8224 using the Caller ID in the P-Asserted-Identity header field.

Step 5. The STI-AS passes the INVITE back to the SP A's CSCF.

Step 6. The originating CSCF, through standard resolution, routes the call to the egress IBCF.

Step 7. The INVITE is routed over the NNI through the standard inter-domain routing configuration.

Step 8. The terminating SP's (SP B) ingress IBCF receives the INVITE over the NNI.

Step 9. The terminating CSCF initiates a terminating trigger to the STI-VS for the INVITE.

NOTE: The STI-VS must be invoked before terminating call processing.

[16] Ibid.

[17] Ibid.

Step 10. The terminating SP STI-VS uses the "info" parameter information in the Identity header field per RFC 8224 to determine the STI-CR Uniform Resource Identifier (URI) and makes an HTTPS request to the STI-CR.

Step 11. The STI-VS validates the certificate (see Section 5.3.1 [of ATIS-1000074] for details) and then extracts the public key. It constructs the RFC 8224 format and uses the public key to verify the signature in the Identity header field, which validates the Caller ID used when signing the INVITE on the originating service provider STI-AS.

Step 12. The CVT is an optional function that can be invoked to perform call spam analytics or other mitigation techniques and return a response related to what should be signaled to the user for a legitimate or illegitimate call. The CVT may be integrated in the service provider network or outside the SP network by a third party.

Step 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 [of ATIS-1000074 and Section 6.2 of this document].

Step 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.[18]

4.5 Verification Procedures

RFC 8224 defines the procedures for verification services including the methods used to verify the signature contained in the Identity header field.[19]

4.5.1 PASSporT & Identity Header Verification

The certificate referenced in the "info" parameter of the Identity header field shall be validated by performing the following:

- Check the certificate's validity using the Basic Path Validation algorithm defined in the X.509 certificate standard (RFC 5280).
- Check that the certificate is not revoked by querying the CRL.

The verifier validates that the PASSporT provided in the Identity header of the INVITE includes all of the baseline claims, as well as the SHAKEN extension claims. The verifier shall also follow the RFC 8224-defined verification procedures to check the corresponding date, originating identity (i.e., the originating telephone number) and destination identities - i.e., the terminating telephone numbers.

The "orig" claim and "dest" claim shall be of type "tn".

The "orig" claim "tn" value validation shall be performed as follows:

- The P-Asserted-Identity header field value shall be checked as the telephone identity to be validated if present, otherwise the From header field value shall also be checked.
- If there are two P-Asserted-Identity values, the verification service shall check each of them until it finds one that is valid.

  NOTE: As discussed in RFC 8224, call features such as call forwarding can cause calls to reach a destination different from the number in the To header field. The problem of determining whether or not these call features or other B2BUA functions have been used legitimately is out of scope of STIR. It is expected that future SHAKEN documents will address these use cases.[20]

[18] Ibid.

[19] Ibid.

[20] Ibid.

4.5.2 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 RFC 8224. 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[21] while providing diagnostic information back to the signer.

There are five main procedural errors defined in RFC 8224 that can identify issues with the validation of the Identity header field. The error conditions and their associated response codes and reason phrases are as follows:

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. The same response may be used when the "iat" has a value 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 to be signed either by local or global policy.

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 does not support it or it does not contain the proper certificate chain in order to trust the credentials.

438: 'Invalid Identity Header' – This occurs if the signature verification fails.

If any of the above error conditions are detected, the terminating network shall convey the response code and reason phrase back to the originating network, indicating which one of the five error scenarios has occurred. How this error information is signaled to the originating network depends on the disposition of the call as a result of the error. If local policy dictates that the call should not proceed due to the error, then the terminating network shall include the error response code and reason phrase in the status line of a final 4xx error response sent to the originating network. On the other hand, if local policy dictates that the call should continue, then the terminating network shall include the error response code and reason phrase in a Reason header field (defined in [RFC 3326]) in the next provisional or final response sent to the originating network as a result of normal terminating call processing.

Example of Reason header field:

```
Reason: SIP ;cause=436 ;text="Bad Identity Info"
```

In addition, if any of the base claims or SHAKEN extension claims are missing from the PASSporT claims, the verification service shall treat this as a 438 'Invalid Identity Header' error and proceed as defined above.[22]

5 Certificates

5.1 Certificate Assertion in Calls Processed Across Multiple Networks

With the implementation of SHAKEN, it is expected that there will be the addition of the identity header with the signature. ATIS-1000074 expects that for the majority of calls, the originator network will sign and authenticate the calling party telephone number (TN) where the originating SP holds the telephone number and has explicitly authenticated the origination of the telephone call from the device. There are, however, other scenarios to consider:

- The originating network provider signs and indirectly authenticates the calling party TN (where they have a third party (e.g., reseller) to whom they have provided their numbering resources).
- The originating network provider signs the call but does not have any authority for calling party TN - e.g., roaming.

[21] Report and Order and Further Notice of Proposed Rulemaking in FCC 13-135 and WC Docket No. 13-39, adopted October 28, 2013 and released November 8, 2013 ("Rural Call Completion").

[22] ATIS-1000074, *Signature-based Handling of Asserted information using Tokens (SHAKEN).*

- Calls that originate on untrusted networks - e.g., legacy time division multiplexing networks or networks where the provider is not certified.

5.2 Attestation Indicator ("attest")

This indicator allows for both identifying the SP that is vouching for the call as well as clearly indicating what information the SP is attesting.

In the SHAKEN framework we define the following three levels of attestation:

1. Full Attestation

The signing SP shall satisfy all of the following conditions:

- Is responsible for the origination of the call onto the IP based SP voice network.
- Has a direct authenticated relationship with the customer and can identify the customer.
- Has established a verified association with the telephone number used for the call.

NOTE: The signing SP is asserting that their customer can "legitimately" use the number that appears as the calling party - i.e., the Caller ID. The legitimacy of the telephone number(s) the originator of the call can use is subject to signer-specific policy, but could use mechanisms such as the following:

- The number was assigned to this customer by the signing SP.
- This number is one of a range of numbers assigned to an enterprise or wholesale customer.
- The signing SP has ascertained that the customer is authorized to use a number -e.g., by business agreement or evidence the customer has access to use the number. This includes numbers assigned by another SP.
- The number is not permanently assigned to an individual customer but the signing SP can track the use of the number by a customer for certain calls or during a certain timeframe.

NOTE: Ultimately it is up to SP policy to decide what constitutes "legitimate right to assert a telephone number" but the SP's reputation may be directly dependent on how rigorous they have been in making this assertion.

2. Partial Attestation

The signing SP shall satisfy all of the following conditions:

- Is responsible for the origination of the call onto its IP-based voice network.
- Has a direct authenticated relationship with the customer and can identify the customer.
- Has NOT established a verified association with the telephone number being used for the call.

NOTE: When partial attestation is used, each customer will have a unique origination identifier created and managed by the SP, but the intention is that it will not be possible to reverse engineer the identity of the customer purely from the identifier or signature. As described in Section 5.2.4, the unique origination identifier allows data analytics to establish a reputation profile and assess the reliability of information asserted by the customer assigned this unique identifier. The identifier also provides a reliable mechanism to determine the customer for forensic analysis or legal action where appropriate.

3. Gateway Attestation

The signing SP shall satisfy all of the following conditions:

- Is the entry point of the call into its VoIP network.
- Has no relationship with the initiator of the call - e.g., international gateways.

NOTE: The PASSporTs will provide a unique origination identifier of the node in the "origid" claim. (The signer is not asserting anything other than "this is the point where the call entered my network".)

For the PASSporT extension claim, the “attest” key value pair shall be set to uppercase characters “A”, “B”, or “C” corresponding to the appropriate attestation defined above.[23]

6 Operational Considerations

6.1 Exception Processing

The SHAKEN framework depends on the ability of the originating SP, after verifying the veracity of the signaled data, to sign the PASSporT using a private encryption key. Similarly, the SHAKEN framework requires the terminating SP to verify the signature using the corresponding public key, obtained from the Certificate Repository.

Certificates in the Certificate Repository have expiration dates and may expire without being renewed. Additionally, under exceptional circumstances such as network outages, physical damage, disasters (natural or otherwise), or simple traffic congestion, it is possible that access to the encryption functionality will not be available to one or both carriers. SPs will use SP specific processes to address exception cases without either disrupting the flow of network traffic or compromising network integrity and security.

6.2 Error Handling

The philosophy behind error handling is that, whenever possible and where not otherwise authorized by the terminating subscriber, call processing should continue to call completion regardless of the state of the Caller ID processing. Error handling procedures should consider how to deliver the call, whenever possible, per current regulatory requirements, while providing diagnostic information back to the originating SP.

The normative handling of the call set up where the SHAKEN framework is implemented will be for the originating SP, having registered and been authenticated to the CSCF, to then process the call and send the call setup information, including the certificate information in the INVITE and the attestation indication. The terminating SP receives the INVITE and validates the certificate, performs the termination functions, and returns the “200 OK” to the originating SP.

When the terminating SP determines that there is an issue with the INVITE or certificate, the error handling processes are implemented. Appropriate response codes, as listed below, should be sent to inform the originating SP of the anomaly. The following are the error messages and the means for handling those error messages.

6.2.1 403 – 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. The same response may be used when the "iat" has a value older than the local policy for freshness permits.[24]

In the case of a ‘Stale Date’ error, call processing should continue to call completion. The Caller ID should be treated as an un-verified Caller ID.

Upon receiving this response code, the originating SP is expected to initiate off-line activity, such as opening a trouble ticket, to address the cause of this error.

6.2.2 428 – Use Identity Header

‘Use Identity Header’ is not recommended for SHAKEN until a point where all calls on the VoIP network are mandated to be signed either by local or global policy.[25]

[23] Ibid.

[24] Ibid.

[25] Ibid.

6.2.3 436 – 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.[26]

In the case of a ‘Bad-Identity-Info’ error, call processing should continue to call completion. The Caller ID should be treated as an un-verified Caller ID.

Upon receiving this response code, the originating SP is expected to initiate off-line activity, such as opening a trouble ticket, to address the cause of this error.

6.2.4 437 – Unsupported Credential

This error occurs when a credential is supplied by the “info” parameter, but the verifier does not support it or it does not contain the proper certificate chain in order to trust the credentials.[27]

In the case of an ‘Unsupported Credential’ error, call processing should continue to call completion. The Caller ID should be treated either as an un-verified Caller ID or as ‘Unknown’.

Upon receiving this response code, the originating SP is expected to initiate off-line activity, such as opening a trouble ticket, to address the cause of this error.

6.2.5 438 – Invalid Identity Header

This occurs if the signature verification fails.[28]

In the case of an ‘Invalid Identity Header’ error, call processing should continue to call completion. The Caller ID should be treated as an invalid (spoofed) Caller ID.

Upon receiving this response code, the originating SP is expected to initiate off-line activity, such as opening a trouble ticket, to address the cause of this error.

[26] Ibid.

[27] Ibid.

[28] Ibid.