SHAKEN
STI-AS and STI-VS Overview with API

ATIS-1000074
ATIS-1000080
IPNNI-2017-00021R004
IPNNI-2017-00089R000
Overview

• Architecture
• API for Authenticator and Verifier <-> SSVS
• Detailed call flows using API
• Backup – call flows without API
Signing and Signature Validation Server
Call Flow Steps

1. The originating SIP UA, which first REGISTERs and is authenticated to the CSCF, creates a SIP INVITE with a telephone number identity.
2. The CSCF of the originating provider adds a P-Asserted-Identity header field asserting the telephone number of the originating SIP UA. The CSCF then triggers the Authenticator in the STI-AS.
3. The Authenticator in the originating SP network (i.e., Service Provider A) first determines through service provider specific means the legitimacy of the telephone number identity being used in the INVITE. The Authenticator sends a signing request to the STI-SSVS.
4. The STI-SSVS then securely requests its private key from the SKS.
5. The SKS provides the private key in the response, and the STI-SSVS signs the Identity header field per RFC draft-ietf-stir-4474bis using the originating telephone number in the P-Asserted-Identity header field.
6. The STI-SSVS returns the signed SIP Identity header field to the Authenticator.
7. The Authenticator passes the INVITE back to the SP A’s CSCF.
8. The originating CSCF, through standard resolution, routes the call to the egress IBCF.
9. The INVITE is routed over the NNI through the standard inter-domain routing configuration.
10. The terminating SP’s (Service Provider B) ingress IBCF receives the INVITE over the NNI.
11. The terminating CSCF triggers the STI-VS Verifier. The STI-VS Verifier must be invoked before terminating call processing.
12. The Verifier sends a verification request to the STI-SSVS.
13. The STI-SSVS uses the “info” parameter information in the Identity header field per RFC 4474bis to determine the STI-CR URI and makes an HTTPS request to the STI-CR.
14. The STI-SSVS validates the certificate received from the STI-CR and then extracts the public key. It constructs the RFC 4474bis 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.
15. The STI-SSVS sends a response to the Verifier indicating whether the identity has been verified.
16. 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 service provider network by a third party.
17. Depending on the result of the STI validation, the STI-VS determines that the call is to be completed with an appropriate indicator and the INVITE is passed back to the terminating CSCF which continues to set up the call to the terminating SIP UA.
18. 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.
STI-AS & STI-VS RESTful web services

• STI-AS and STI-VS expose a RESTful web service - Signing and Signature Verification Service (SSVS).
• Implemented using HTTP and aligned with the principles of RESTful API.
• Only JSON based data format is supported.
  – APIs use “application/json” content type
• POST HTTP request is used for the both APIs.
• HTTP 1.1 protocol version has to be supported by server side.
REST resource structure

• REST resources are defined with respect to a “server Root”:

  “serverRoot” = http://{hostname}:{port}/{optionalRoutingPath}

• The resource structure is provided below:
## HTTP Request (POST) headers

<table>
<thead>
<tr>
<th>Header Name</th>
<th>Mandatory?</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>X-RequestID</td>
<td>N</td>
<td>According to the general agreement the transaction UUID should be published by component calling an exposed by other component API in order to make possible the transaction traceability in case of troubleshooting and fault analysis. Generated UUID should be compliant with RFC 4122. If received will not be validated explicitly by server. If not received it will be automatically generated by STI-AS/VS service on request receipt. Received/Generated transaction UUID will be returned back in the corresponding HTTP response in “X-RequestID” header.</td>
</tr>
<tr>
<td>X-InstanceID</td>
<td>N</td>
<td>For auditing purpose each component calling the API should identify itself by sending its identity ( e.g. Instar name, VNFC name/UUID, VM name/UUID ... ) in &quot;X-InstanceID&quot; header.</td>
</tr>
<tr>
<td>Content-Type</td>
<td>Y</td>
<td>Determines the format of the request body. Valid value is: “application/json”. Requests with other types will be rejected with “415 Unsupported Media type” HTTP status code.</td>
</tr>
<tr>
<td>Accept</td>
<td>N</td>
<td>If specified has to contain “application/json” content type, otherwise HTTP request will be rejected with “406 Not Acceptable” HTTP Status Code. If not specified will be y default handled as “application/json”.</td>
</tr>
</tbody>
</table>
# HTTP Response headers

<table>
<thead>
<tr>
<th>Header Name</th>
<th>Mandatory?</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>X-RequestID</td>
<td>Y</td>
<td>Received/Generated transaction UUID will be returned back in the corresponding HTTP response.</td>
</tr>
<tr>
<td>Content-Type</td>
<td>Y</td>
<td>Determines the format of the response body. Valid value is “application/json”</td>
</tr>
</tbody>
</table>
Datatype: signingRequest

<table>
<thead>
<tr>
<th>Key Name</th>
<th>Key Value Type</th>
<th>Required?</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>attest</td>
<td>String</td>
<td>Y</td>
<td>SHAKEN extension to PASSporT. Indicator identifying the service provider that is vouching for the call as well as a clearly indicating what information the service provider is attesting to. \nSHAKEN spec requires “attest” key value be set to uppercase characters “A”, “B”, or “C”.</td>
</tr>
<tr>
<td>dest</td>
<td>destTelephoneNumber</td>
<td>Y</td>
<td>Represents the called party. Array containing one or more identities of telepnoneNumber type.</td>
</tr>
<tr>
<td>iat</td>
<td>Integer</td>
<td>Y</td>
<td>“Issued At Claim”: Should be set to the date and time of issuance of the PASSporT Token. \nThe time value should be in the Numeric Date format defined in RFC 7519: number of seconds elapsed since 00:00:00 UTV, Thursday, 1 January 1970 not including leap seconds.</td>
</tr>
<tr>
<td>orig</td>
<td>origTelephoneNumber</td>
<td>Y</td>
<td>Represents the asserted identity of the originator of the personal communications signaling.</td>
</tr>
<tr>
<td>origid</td>
<td>String</td>
<td>Y</td>
<td>The unique origination identifier (“origid”) is defined as part of SHAKEN extension to PASSporT. This unique origination identifier should be a globally unique string corresponding to a UUID (RFC 4122). \nNote: VM UUID can be used as a unique originator identifier.</td>
</tr>
</tbody>
</table>

Note: API assumes ES256 as the algorithm
Datatype: origTelephoneNumber

<table>
<thead>
<tr>
<th>Field</th>
<th>Type</th>
<th>Required</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tn</td>
<td>String</td>
<td>Y</td>
<td>Telephone Number of Originating/Destination identity. Server will remove all non-numeric characters if received except start (*) and pound (#) characters. Ex. : (+1)235-555-1212 → 12355551212</td>
</tr>
</tbody>
</table>
# Datatype: destTelephoneNumber

<table>
<thead>
<tr>
<th>Field</th>
<th>Type</th>
<th>Required</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tn</td>
<td>List of Strings [1 ... unbounded] Allowed Characters: [0-9] , *, #, +, and visual separators defined in RFC 3966: &quot;:&quot;, &quot;,&quot;, &quot;(&quot;&quot;, &quot;)&quot;.</td>
<td>Y</td>
<td>List containing <strong>one or more</strong> identities of String type.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Server will remove all non-numeric characters if received except start (*) and pound (#) characters.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Ex.: (+1)235-555-1212 → 12355551212</td>
</tr>
</tbody>
</table>
## Datatype: signingResponse

<table>
<thead>
<tr>
<th>Key Name</th>
<th>Key Type</th>
<th>Value</th>
<th>Required?</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>identity</td>
<td>String, Cannot be NULL</td>
<td>Y</td>
<td>Identity header value as defined in RFC4474bis with “identityDigest” in full format and mandatory “info” header parameter. “info” parameter will contain the public key URL of the certificate used during STI signing.</td>
<td></td>
</tr>
</tbody>
</table>
### Datatype: verificationRequest

<table>
<thead>
<tr>
<th>Key Name</th>
<th>Key Value Type</th>
<th>Required?</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>identity</td>
<td>String</td>
<td>Y</td>
<td>Identity header value as defined in RFC4474bis with &quot;identityDigest&quot; in full format and mandatory &quot;info&quot; header parameter.</td>
</tr>
<tr>
<td>dest</td>
<td>destTelephoneNumber</td>
<td>Y</td>
<td>Represents the one or more identities. This should be the value of “To:” header field parameter in the incoming Invite.</td>
</tr>
<tr>
<td>iat</td>
<td>Integer</td>
<td>Y</td>
<td>“Issued At Clue” time of issuance. The time value format defined as elapsed since 1970 not including leap seconds.</td>
</tr>
<tr>
<td>orig</td>
<td>origTelephoneNumber</td>
<td>Y</td>
<td>Represents the personal contact of the &quot;P-Asserted-Identity&quot; or “From” header field parameter in the incoming Invite.</td>
</tr>
</tbody>
</table>
# Datatype: verificationResponse

<table>
<thead>
<tr>
<th>Key Name</th>
<th>Key Value Type</th>
<th>Required</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reasoncode</td>
<td>Integer</td>
<td>N</td>
<td>Reason Code to be used in case of failed verification by STI-VS to build SIP Reason header if required. Currently possible values are defined as follows (please pay attention they can be extended/changed in the future): 403,428 (will not be returned in the initial release), 436,437,438 403 – “Stale Date header received” 436 – Bad Public Key Certificate URI</td>
</tr>
<tr>
<td>reasontext</td>
<td>String</td>
<td>N</td>
<td>Reason Text to be used in case of failed verification by STI-VS to build SIP Reason header if required. Currently possible values are defined as follows (please pay attention they can be extended/changed in the future): 403 - “Stale Date” 428 - “Use Identity Header” (will not be returned in the initial release) 436 – “Bad Identity Info” 437 – “Unsupported Credential” 438 – “Invalid Identity Header”</td>
</tr>
<tr>
<td>reasondesc</td>
<td>String</td>
<td>N</td>
<td>Reason details description. Can be used for logging and troubleshooting.</td>
</tr>
<tr>
<td>verstat</td>
<td>String</td>
<td>Y</td>
<td>Verification Status: <strong>TN-Validation-Passed</strong> - The calling number passed the validation  <strong>TN-Validation-Failed</strong> - The calling number failed the validation  <strong>No-TN-Validation</strong> - No validation number was performed</td>
</tr>
</tbody>
</table>
End-to-End STI Call Origination & Termination

1. SIP INVITE
2. SIP INVITE
3. signingRequest
4. Get Private Key
5. Private Key
6. signingResponse
7. SIP INVITE (with Identity)
8. SIP INVITE
9. SIP INVITE
10. SIP INVITE
11. SIP INVITE
12. verificationRequest
13. Get Certificate
14. Certificate
15. verificationResponse
16. Invoke Analytics
17. Result of Analytics
18. SIP INVITE (with Verification Status)
19. SIP INVITE (with Verification Status)
20. 200 OK
21. 200 OK
22. 200 OK
23. 200 OK
24. 200 OK
Information/control flow for CM

1. The originating SIP UA sends a SIP INVITE request with a tel URI in the From header field parameter.
2. The CSCF of the originating provider adds a P-Asserted-Identity header field for the originating SIP UA. After initial call processing, the CSCF then initiates an originating trigger to the STI-AS for the INVITE.
3. The authenticator parses the SIP INVITE and sends a signingRequest to the SSVS.
4. The SSVS then securely requests its private key from the SKS.
5. The SKS provides the private key in the response, and the SSVS builds the PASSporT using the JWT populated from the information in the signingRequest received from the Authenticator. The SSVS signs the PASSport using the private key.
6. The SSVS sends the PASSporT and a link to the public key certificate in a signingResponse to the Authenticator.
7. The Authenticator builds the SIP Identity header field using the Identity information returned from the SSVS. The Authenticator then returns the INVITE back to SP A’s CSCF.
8. The originating CSCF, through standard resolution, routes the call to the egress IBCF.
9. The INVITE is routed over the NNI through the standard inter-domain routing configuration.
10. The terminating SP’s (Service Provider B) ingress IBCF receives the INVITE over the NNI and forwards to the CSCF.
11. The terminating CSCF initiates a terminating trigger to the Verifier in the STI-VS for the INVITE before terminating call processing.
12. The Verifier extracts the Identity header field from the SIP INVITE and sends a verificationRequest to the SSVS.
Information/control flow for CM

13. The SSVS uses the “info” parameter header field parameter in the Identity header field per draft-ietf-stir-4474bis to determine the STI-CR URI and makes an HTTPS request to the STI-CR to get the public key certificate.

14. The SSVS validates the certificate (see Section 5.3.1 of the SHAKEN framework for details) and then extracts the public key. It constructs the format as defined in draft-ietf-stir-4474bis 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.

15. The SSVS returns the results of the verification by sending a verificationResponse sent to the Verifier.

16. The CVT is an optional function that can be invoked by the Verifier to perform call spam analytics or other mitigation techniques. The CVT may be integrated in the service provider network or outside the service provider network by a third party.

17. The CVT returns a response related to what should be signaled to the user for a legitimate or illegitimate call.

18. Depending on the result of the STI validation, the Verifier determines that the call is to be completed using the “verstat” URI parameter and the SIP INVITE is passed back to the terminating CSCF, including any Call-Info values as defined in draft-ietf-sipcore-callinfo-spam. The CSCF continues to set up the call to the terminating SIP UA.

19. The terminating SIP UA receives the INVITE and uses the Call-Info values to display an indication on the user’s device with reflective of the values in the Call-Info. Normal SIP processing of the call continues, returning a SIP 200 OK response.

20-24. The SIP 200 OK response is sent to the originating UA and media is setup end-to-end.
Step 5

a) Construct the PASSporT JWS elements
   - Construct the JWS Protected header
   - Construct the JWS Payload

b) Generate JSON Web Signature over the PASSporT using the private key and the specified algorithm.
   - See draft-ietf-stir-passport Appendix A for a step by step example.
Step 7

Build SIP Identity header field (example on the next slide):

- signed PASSporT from step 5b.
- “info” header field parameter MUST match the “x5u” parameter in the PASSport
- “alg” header field parameter MUST match the “alg” in the PASSporT
Example SIP INVITE

INVITE sip:+12155551213@tel.example1.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:+19299100814@provteam.in>
From: <sip:+19299100813@provteam.in>;tag=614bdb40
Call-ID: 79048YzkkNDA-NTI1MzA0OWFjOTFkMfIODhiNTI2OWQ1ZTI
P-Asserted-Identity: <sip:+19299100814@provteam.in>
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

Identification: eyJhbGciOiJFUzI1NiIsInR5cCI6IkpXVCJ9.eyJhdHRlc3QiOiJBIiwiZGVzdCI6eyJ0biI6IisxMjE1NTU1M
TIxMyI6IjEyM2U0N
J9._28kAwRwnheXyA6nY4vmK5JKZH9hSYkWi4g75m
q9TjI2IW4WPM0PlvudoGaj7wM5xujZUTb_3MA4modoDIvCA;
info=<http://cert.provteam.in/example.cert>;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

Step 5a. JWS Token/PASSporT

JWS Protected header :
{"alg":"es256","typ":"passport","ppt":"shaken","x5u":"https://cert.cr.provteam.in/example.cert"}

JWS Payload (JWT claims ) :
{"attest":"A","dest":{"tn":["19299100814"]},"iat":1443208345,"orig":{"tn":"19299100813"},"origid":"123e4567-e89b-12d3a456-426655440000"}
Backup
STI Call Origination & Termination – Target for Prototype

Service Provider A
Originating/Authorization

Service Provider B
Terminating/Verification
Call Flow Steps

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

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. The STI-AS is invoked after originating call processing.

3. The **STI-AS** in the originating SP (i.e., Service Provider 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.

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 draft-ietf-stir-4474bis using the Caller ID in the P-Asserted-Identity header field.

5. The STI-AS passes the INVITE back to the SP A’s 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’s (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. The STI-VS must be invoked before terminating call processing.

10. The terminating SP STI-VS uses the “info” parameter information in the Identity header field per RFC 4474bis to determine the STI-CR URI and makes an HTTPS request to the STI-CR.

11. The STI-VS validates the certificate and then extracts the public key. It constructs the RFC 4474bis 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.

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 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 an appropriate indicator and the INVITE is passed back to the terminating CSCF which continues to set up the call to the terminating SIP UA.

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.
End-to-End STI Call Origination & Termination
Information/control flow for CM

1. The originating SIP UA sends a SIP INVITE request with a tel URI in the From header field parameter.
2. The CSCF of the originating provider adds a P-Asserted-Identity header field for the originating SIP UA. After initial call processing, the CSCF then initiates an originating trigger to the STI-AS for the INVITE.
3. The STI-AS then securely requests its private key from the SKS.
4. The SKS provides the private key in the response, and the STI-AS adds an Identity header field per RFC per draft-ietf-stir-4474bis. The From header field parameter from the initial SIP INVITE is used in the P-Asserted-Identity header field.
5. The STI-AS passes the INVITE back to the SP A’s 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’s (Service Provider B) ingress IBCF receives the INVITE over the NNI and forwards to the CSCF.
9. The terminating CSCF initiates a terminating trigger to the STI-VS for the INVITE before terminating call processing.
10. The terminating SP STI-VS uses the “info” parameter header field parameter in the Identity header field per draft-ietf-stir-4474bis to determine the STI-CR URI and makes an HTTPS request to the TN-CR to get the public key certificate.
11. The STI-VS validates the certificate (see Section 5.3.1 of the SHAKEN framework for details) and then extracts the public key. It constructs the format as defined in draft-ietf-stir-4474bis 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.
12. The CVT is an optional function that can be invoked to perform call spam analytics or other mitigation techniques. The CVT may be integrated in the service provider network or outside the service provider network by a third party.
13. The CVT returns a response related to what should be signaled to the user for a legitimate or illegitimate call.
14. Depending on the result of the STI validation, the STI-VS determines that the call is to be completed with an appropriate indicator and the SIP INVITE is passed back to the terminating CSCF, including any Call-Info values as defined in draft-sipcore-callinfo-spam. The CSCF continues to set up the call to the terminating SIP UA.
15. The terminating SIP UA receives the INVITE and uses the Call-Info values to display an indication on the user’s device with reflective of the values in the Call-Info. Normal SIP processing of the call continues, returning a SIP 200 OK response.
16-20. The SIP 200 OK response is sent to the originating UA and media is setup end-to-end.