RedSky SIP (Session Initiation Protocol) Specification

Version 1.3 | March 6, 2024

RedSky Technologies, Inc.
333 North Michigan Avenue, Suite 1600
Chicago, IL 60601
redskye911.com

Revision History

Date	Version	Revision	Made By
04/06/2021	0.1	Initial Draft	Katrina Vlasich
04/12/2021	0.2	Various content edits	Cruz Tovar, Ryan Olsen, Mike Koepke
04/12/2021	1.0	Final Release	Katrina Vlasich
04/22/2021	1.1	Updated Callback Number Flow Chart to include new E911-Callback-Number SIP Header.	Katrina Vlasich
04/01/2023	1.2	- Added References section - Added Error Scenario section - Clarified PIDF-LO Support - Clarified Called Party/To: handling	Mike Koepke
03/06/2024	1.3	- Callback number selection with Alternate ID functionality - E911-User-Info now supports bssid.	Mike Koepke

Table Of Contents

1. INTRODUCTION 5

2. SCOPE 5

3. TECHNICAL SPECIFICATION 6

4 SECURITY 7

5 TECHNICAL NOTES 8

6 SUPPORTED SIP INVITE REQUEST HEADERS 12

7 SIP INVITE REQUEST HEADER PRIORITY 19

8 DETERMINING CALLBACK NUMBER 20

9 ERROR SCENARIO REPONSES 22

10 REFERENCES 23

1. Introduction

This document defines the 9-1-1 SIP messaging interface used by the RedSky 9-1-1 Voice Gateway as part of the Horizon Mobility® and E911 Anywhere® service. Outlined within are the SIP header fields, or attributes, that are available for an emergency call SIP INVITE Request, how they are defined, and usage examples. Additionally, the order of precedence the information in a SIP INVITE Request will be processed in, as well as how a callback number will be derived, will be covered in detail.

2. Scope

This specification describes the RedSky 9-1-1 Voice Gateway interface. RedSky processes incoming emergency calls received over the Session Initiation Protocol (SIP) interface, and using information extracted from the SIP message contents, routes the 9-1-1 emergency call to the appropriate Public Safety Answering Point (PSAP) serving the caller's location.

RedSky has been providing emergency call routing for the past 15 years and this specification is meant to encapsulate the many different solutions RedSky has built for its customers and partners. The information below is all encompassing, and different headers are used depending on the solution being offered.

As no complete or formal documentation had existed previously, that encompassed the implementation of the RedSky 9-1-1 Voice Gateway, we have decided to start anew and version this document as Version 1 of the SIP interface.

3. Technical Specification

3.1 Technical Information

Supported Codecs: ulaw - G.711/PCMU

Supported Transport Protocols: TCP, UDP, TLS (Transport Layer Security) 1.2+

Supported Ports: 5060 (TCP/UDP),5061 (TLS), 5062 (TCP/UDP), RTP Ports 30000 – 60000
Note: 5062 is used for outgoing SIP traffic between the voice gateway and 9-1-1 VPC Providers only.

3.2 Supported SIP Methods

The RedSky 9-1-1 Voice Gateway supports the following SIP Request methods:

- SIP **INVITE** Method – Refer to RFC 3261 Initiating a Session for more information.
- SIP **re-INVITE** Method – Refer to RFC 6141 Re-INVITE Handling in SIP for more information.
- SIP **OPTIONS** Method – Refer to RFC 3261 Querying for Capabilities topic for more information.
- SIP **CANCEL** Method – Refer to RFC 3261 Canceling a Request for more information.
- SIP **BYE** Method – Refer to RFC 3261 Terminating a Session for more information.
- SIP **PRACK** Method – Refer to the RFC 3262 Provisional Response Acknowledgement for more information.

3.3 Unsupported SIP Methods

The RedSky 9-1-1 Voice Gateway does NOT support the following SIP Request methods:

- SIP **INFO** Method – Refer to RFC 6086 The SIP INFO Method and Package Framework for more information.
- SIP **SUBSCRIBE** and **NOTIFY** Methods – Refer to RFC 3265 Specific Event Notification for more information.
- SIP **UPDATE** Method – Refer to RFC 3311 SIP UPDATE Method for more information.
- SIP **REGISTER** Method – Refer to RFC 5626 Managing Client-Initiated Connections for more information.
- SIP **MESSAGE** Method – Refer to RFC 3428 Extension for Instant Messaging for more information.
- SIP **PUBLISH** Method – Refer to RFC 3903 Extension for Event State Publication for more information.
- SIP **SUBSCRIBE** Method – Refer to RFC 3265 Specific Event Notification for more information.

4 Security

4.1 Whitelisting

Organizations desiring their call servers to route emergency calls to a RedSky 9-1-1 Voice Gateway will need to have all external IP addresses of said Call Servers whitelisted by RedSky. If a SIP INVITE Request is received for an unknown SIP client, a "403", "Forbidden" will be sent back.

4.2 TLS Support

An organization should ensure that a Root SSL Certificate, from the Sectigo Certificate Authority, is installed on each call server routing emergency calls to a RedSky 9-1-1 Voice Gateway for them to route successfully.

Additionally, Call Servers will need to send a valid SSL Certificate, that has been signed by a trusted Root Certificate Authority, in order establish an encrypted connection with RedSky 9-1-1 Voice Gateways.

5 Technical Notes

5.1 RedSky Custom 9-1-1 Headers

RedSky has built 9-1-1 emergency support directly into its 9-1-1 Voice Gateway by inclusion of custom headers to deliver 9-1-1 Caller information. Any call server is free to use these headers to deliver 9-1-1 information to the RedSky 9-1-1 Voice Gateway to route the emergency call. These headers are:

- **E911-Organization-ID –** Identifier of the organization sending the 9-1-1 call.
- **E911-User-ID – Non-DID** Identifier for users.
- **E911-Location-ID – Non-DID Identifier for physical phones**
- **E911-Callback-Number –** Phone number to be used as the callback number to send to the PSAP.
- **E911-User-Info –** Network discovery information (IP Address, BSSID, MAC) of the 9-1-1 caller's device. It is used to locate the caller's location.

5.2 Phone Number Format

The RedSky 9-1-1 Voice Gateway can accept phone numbers in either the North American Numbering Plan (NANP) format, the ITU E.164 format, or both. In the NANP format phone numbers will be 10-digit in length, such as *312-555-1212* or *3125551212*. In the E.164 format, the number will be proceeded by the country code, with or without the preceding '+'. The Country Code for North America is 1. In the E.164 format, the phone numbers, mentioned above, would be *+13125551212* or *13125551212*.

5.3 SIP Request-Line

The RedSky Voice Gateways will only accept incoming calls destined for 9-1-1, 9-3-3 or defined 10-digit DID number in either North American Numbering Plan or E.164 formats. Calls received from endpoints with unexpected destinations in the To: header, will simply be disconnected with a 404 Not Found response code.

Supported Request URIs:

- 911@<gatewayip:port>
- 933@<gatewayip:port>
- 1911@<gatewayip:port>
- 1933@<gatewayip:port>
- +1911@<gatewayip:port>

- +1933@<gatewayip:port>
- xxx-xxx-xxxx@<gatewayip:port>
- +1xxx-xxx-xxxx@<gatewayip:port>

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:3123210000@5.4.3.1:5060>;tag=as520a8116
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
E911-Location-ID: AcmeIPUSA1424@voip.acmeip.net
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

5.4 Response Codes

Call Servers connecting to RedSky 9-1-1 Voice Gateways are expected to handle standard SIP response codes, such as 100 (Trying), 180 (Ringing), 200 (OK), and 403 (Forbidden). Please refer to the Response Codes topic in RFC 3261 for more information.

5.5 Presence Information Data Format Location Object (PIDF-LO)

PIDF-LO is the Location-by-Value (LbV) contained in a SIP INVITE Request message body. When a RedSky 9-1-1 Voice Gateway receives a SIP INVITE Request, it will first look for a PIDF-LO in determining the location of the caller, so that the location and the call can be routed to the appropriate PSAP.

Location in the SIP Invite is represented by content in the PIDF-LO document, RFC 4119 (A Presence-based GEOPRIV Location Object Format). updated by RFCs 5139 (Revised Civic Location Format for Presence Information Data Format Location Object) and 5491 (GEOPRIV Presence Information Data Format Location Object (PIDF-LO) Usage Clarification, Considerations, and Recommendations).

The RedSky Voice Gateways only supports the revised Civic Location Format defined in RFC 5139. A PIDF-LO using the RFC 4119 format and tags, will be ignored and treated as not present.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip: 15557772255@5.4.3.1:5060>;tag=as520a8116
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: sip:18.190.127.223;did=44c.b777e821
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
P-Asserted-Identity: <tel:+15557772255>
P-Asserted-Identity: <sip:5557772255@1.2.3.4>
Content-Type: multipart/mixed; boundary=border
Content-Length: 1554

Content-ID: <acmecorp_geolocation>
Content-Type: application/pidf+xml

<?xml version="1.0" encoding="UTF-8"?>
<presence xmlns="urn:ietf:params:xml:ns:pidf" xmlns:ca="urn:ietf:params:xml:ns:pidf:geopriv10:civicAddr"
xmlns:conf="urn:ietf:params:xml:ns:geopriv:conf" xmlns:gbp="urn:ietf:params:xml:ns:pidf:geopriv10:basicPolicy"
xmlns:gml="urn:opengis:specification:gml:schema-xsd:feature:v3.0" xmlns:gp="urn:ietf:params:xml:ns:pidf:geopriv10"
entity="pres:528041362179366@127.0.0.1">
  <tuple id="eee0d930-f021-443d-b94a-323110097991">
    <status>
      <gp:geopriv>
        <gp:location-info>
          <ca:civicAddress>
            <ca:country>US</ca:country>
            <ca:a1>IL</ca:a1>
            <ca:a2>Cook</ca:a2>
            <ca:a3>Chicago</ca:a3>
            <ca:prd>N</ca:prd>
            <ca:rd>Michigan</ca:rd>
            <ca:sts>Ave</ca:sts>
            <ca:hno>180</ca:hno>
            <ca:pc>60601</ca:pc>
            <ca:bld>Corporate Headquarters</ca:bld>
            <ca:nam>Acme Corporation</ca:nam>
            <ca:loc>Fl 16 SW Corner</ca:loc>
          </ca:civicAddress>
          <gml:point>
            <gml:pos>41.88515 -87.62457</gml:pos>
          </gml:point>
        </gp:location-info>
        <gp:usage-rules>
          <gbp:retransmission-allowed>yes</gbp:retransmission-allowed>
          <gbp:retention-expires />
        </gp:usage-rules>
        <method>LIS</method>
      </gp:geopriv>
    </status>
    <timestamp>2020-01-30T23:33:38.632Z</timestamp>
  </tuple>
</presence>
```

5.6 Organization Identifier

A unique identifier is assigned to an organization at the time of being onboarded onto the Horizon Mobility® / E911 Anywhere® platform. This identifier is used in tandem with other information to look up the location of the emergency caller.

It is also used to identify what organization will incur the charge for an emergency call being routed to the RedSky Relay center due to a dispatchable location not being found for the caller. For this reason, it is recommended that the E911-Organization-ID header always be included when a SIP trunk is servicing multiple customers.

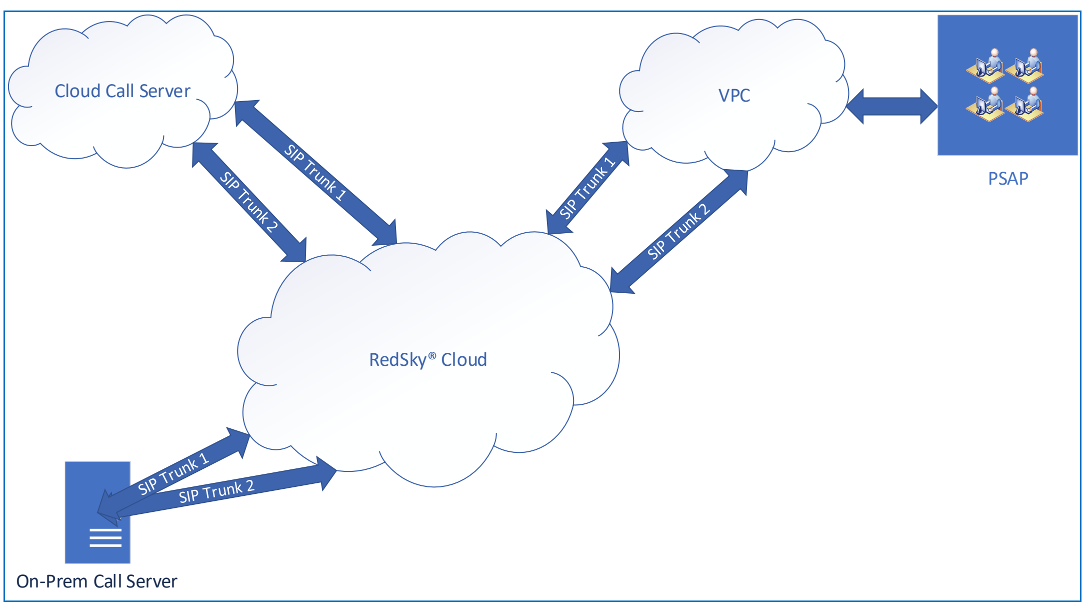
5.7 Redundancy

RedSky strongly recommends that organizations connecting to our cloud have a pair of SIP Trunks available to them, for the sake of redundancy.

RedSky has several voice gateways operating in its cloud to allow for redundancy and load balancing, maximizing service availability to our customers. Customers should stand up two SIP trunks, with one trunk going to Gateway 1 and the other going to Gateway 2. In this configuration, should some problem occur on the SIP trunk, or the Voice Gateway is not functional, due to an error or planned maintenance, SIP trunk can still be sent to the other gateway.

5.8 SIP Trunking Diagram

Below is a network diagram depicting SIP Trunking between a customer, RedSky, and a VoIP Positioning Center (VPC).



6 Supported SIP INVITE Request Headers

6.1 SIP INVITE Request E911-Organization-ID Header

A RedSky proprietary SIP **INVITE** Request header that contains the Organization Identifier of the organization the caller belongs to.

If the **E911-Organization-ID** header is not available, the RedSky 9-1-1 Voice Gateway will then use the IP Address of the SIP Trunk that the call came in on, to identify the organization.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-User-ID: 34De65a@acme.com
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.2 SIP INIVITE Request RedSky-CustomerID Header (Deprecated)

A SIP **INVITE** Request header that contains header that contains the Organization Identifier of the organization the caller belongs to. This header was only used by Cisco. **The E911-Organization-ID header should be used in lieu of this header.**

In the case where an **RedSky-CustomerID** header is not available, the RedSky 9-1-1 Voice Gateway will then use the IP Address of the SIP Trunk that the call came in on, to identify the organization.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-User-ID: 34De65a@acme.com
RedSky-CustomerID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.3 SIP INVITE Request Geolocation Header

A Location-by-Reference (LbyR) value; typically, a URL that resolves to location information of a caller. If a RedSky 9-1-1 Voice Gateway does not find a PIDF-LO in a SIP **INVITE** Request, it will next look for the **Geolocation** header to determine the location, so that the location and the call can be routed to the appropriate PSAP.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:3123210000@5.4.3.1:5060>;tag=as520a8116
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
Geolocation: <https://api.acme.cloud.com/held-service/heldref/?token=48d58993-b03b-4a85-a35a-
ca4c815e68dc&companyID=81a1d268-6d8e-450a-92aa-0d5420989772>
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.4 SIP INVITE Request E911-User-ID Header

An alternate identifier, in lieu of a Direct Inward Dial (DID) number that will be used to identify a Device User and determine their location, so that the location and the call can be routed to the appropriate PSAP. If the value in the **E911-User-ID** header is not valid, the call will be routed to the RedSky Relay Center (RSRC).

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-User-ID: 34De65a@acme.com
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.5 SIP INVITE Request E911-Location-ID Header

An alternate identifier, in lieu of a Direct Inward Dial (DID) number that will be used to identify the location a caller is calling from, so that the location and the call can be successfully routed to the correct PSAP. If the value in the **E911-Location-ID** header is not valid, the call will be routed to the RedSky Relay Center (RSRC).

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
E911-Location-ID: AcmeIPUSA1424@voip.acmeip.net
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.6 SIP INVITE Request ININAttr Header

A Genesys Pure Connect PBX SIP **INVITE** Request header that contains the IP Address and MAC Address of a caller's device. As the values contained in the **ININAttr** header may not be globally unique, the **E911-Organization-ID** header will be used to look up the organization the **ININAttr** header values belong to with the goal of obtaining the location of a caller. The location and the call can then be successfully routed to the appropriate PSAP. If the **E911-Organization-ID** header is not available, the RedSky 9-1-1 Voice Gateway will then use the IP Address of the SIP Trunk that the call came in on, to identify the organization.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
ININAttr: "uu_StationMacAddress=0004f215c285;uu_StationIPAddress=192.168.1.4"
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.7 SIP INVITE Request E911-User-Info Header

A RedSky proprietary SIP **INVITE** Request header containing the IP Address, MAC Address or BSSID of a caller's device.

The contents of this header will be used to establish the location of the caller based on the network element provided. The call can then be successfully routed to the appropriate PSAP based on the caller's location. The E911-User-Info header is "freeform" in that it can contain multiple network elements. The 3 elements are IP Address (ip=), MAC Address (mac=) or BSSID (bssid=).

Rules for this header are:

- 1 or more of the elements can be submitted
- If multiple elements are submitted the elements need to be semicolon separated
- The order the ip, mac, bssid elements does not matter
- IP, mac, bssid names are case-insensitive
- Spaces are ignored (ip = 192.192.0.1 is the same as ip=192.192.0.1)

As the network elements contained in the **E911-User-Info** header may not be globally unique, the **E911-Organization-ID** / **RedSky-CustomerID** header will be used to determine what organization the values in the **E911-User-Info** header belong to and the corresponding dispatchable location. If the **E911-Organization-ID** header is not available, the RedSky 9-1-1 Voice Gateway will use the IP Address of the SIP Trunk that the call was delivered over to identify the organization.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
E911-User-Info: ip=172.16.154.150;mac=00E0A70ECD65
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.8 SIP INVITE Request X-Genesys-ELIN Header

A Genesys Pure Engage PBX SIP INVITE Request header that contains the ELIN of the caller. The value will be used to identify the caller's location, so that the location and the call can be routed to the correct PSAP.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
X-Genesys-ELIN: 3120001234
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.9 SIP INVITE Request AP-Loc Header

An Avaya® PBX SIP INVITE Request header that contains the ELIN of the caller. The value will be used to identify the caller's location, so that the location and the call can be routed to the correct PSAP.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
AP-Loc: elin=3120001234
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.10 SIP INVITE E911-Callback-Number Header

A header containing the callback number to be used be used for the 9-1-1 caller. If this header is present, the RedSky 9-1-1 Voice Gateway will use this number to send to the PSAP and will forego looking at the **P-Asserted-Identity** or **From** headers for the callback number to be used.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
P-Asserted-Identity: <sip:3120001234@172.20.20.255>
P-Asserted-Identity: <tel:+13120001234>
E911-Callback-Number: 3120001234
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.11 SIP INVITE Request P-Asserted-Identity (PAI) Header

The standard SIP **INVITE** Request header that contains the ELIN of the caller. The value will be used to identify the caller's location, so that the location and the call can be routed to the correct PSAP.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:50015@45.25.216.1;transport=TCP>;tag=b639fceb
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
P-Asserted-Identity: <sip:3120001234@172.20.20.255>
P-Asserted-Identity: <tel:+13120001234>
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

6.12 SIP INVITE Request From Header

A header containing the contact information of the initiator of the SIP INVITE Request. The information is used to help identify the organization in cases when a RedSky 9-1-1 Voice Gateway does not receive any of the aforementioned headers, or a valid DID in X-Genesys-ELIN / AP-Loc or in the P-Asserted-Identity header.

EXAMPLE:

```
INVITE sip:911@18.190.128.222;transport=TCP SIP/2.0
From: <sip:3123210000@5.4.3.1:5060>;tag=as520a8116
To: <sip:911@18.190.127.223;transport=TCP>
Call-ID:
DLGCH_LjtQH2ogE34sQV0KMC16YCpFAF9hfGN2PhERATwlMjw3XhcdOickNj0eBAJ9JT8wLhwlHzI9PzwjEgoWfTsjOiEEABw9KDx9I
x8GDz8-
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:18.190.127.223;did=44c.b777e821>
E911-User-ID: 34De65a@acme.com
E911-Organization-ID: 9ae5df2b-83e9-4e4c-bdd0-74316e36eae6
Via: SIP/2.0/UDP 18.190.127.223:5060;branch=z9hG4bKb7cc.79a18ff1.0;i=80b9f064
Content-Length: 88
```

7 SIP INVITE Request Header Priority

In determining the location of the caller, when PIDF-LO content is not directly contained within the SIP INVITE Request, the RedSky 9-1-1 Voice Gateway will process the supported SIP INVITE Request headers in the following order to determine the currently known location of the 9-1-1 caller:

1) Geolocation Header
2) E911-User-ID Header
3) E911-Location-ID Header
4) ININAttr Header
5) E911-User-Info Header
6) X-Geneys-ELIN Header
7) AP-Loc Header
8) P-Asserted-Identity Header
9) From Header

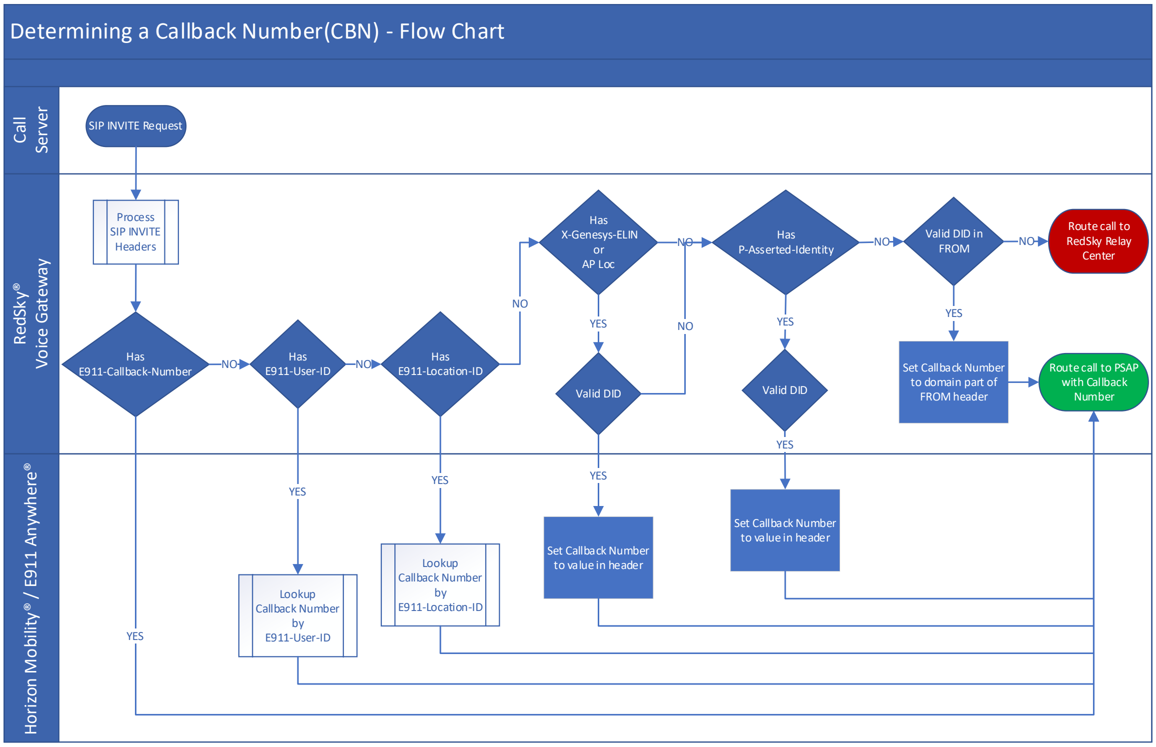
Should the location of the caller not be able to be determined by any of the methods above, the caller will be sent to an Emergency Relay Center (ERC), to manually gain the caller's location.

8 Determining Callback Number

A RedSky 9-1-1 Voice Gateway will process information sent in a SIP **INVITE** Request in the following manner to determine what value should be presented as the callback number to the PSAP. The PSAP will use this number to call back the 9-1-1 caller in the event the call gets disconnected prematurely:

1) If a SIP **INVITE** Request is sent with a valid **E911-Callback-Number** header value, the RedSky 9-1-1 Voice Gateway will use this number as the caller's callback number on Horizon Mobility® / E911 Anywhere®.
2) If a SIP **INVITE** Request is sent with a valid **E911-User-ID** header value, the RedSky 9-1-1 Voice Gateway will obtain the callback number from the corresponding Device User record on Horizon Mobility® / E911 Anywhere®.
3) If a SIP **INVITE** Request is sent with a valid **E911-Location-ID** header value, the RedSky 9-1-1 Voice Gateway will obtain the callback number from the corresponding Location record on Horizon Mobility® / E911 Anywhere®.
4) If a SIP **INVITE** Request is sent with a valid **X-Genesys-ELIN** header value (valid DID), it will be set as the callback number in the outgoing PIDF-LO.
5) If a SIP **INVITE** Request is sent with a valid **AP-Loc** header value (valid DID), it will be set as the callback number in the outgoing PIDF-LO.
6) If none of the aforementioned headers are sent in a SIP **INVITE** Request, or the **X-Genesys-ELIN** / **AP-Loc** does not contain a valid Direct Inward Dial (DID), the RedSky 9-1-1 Voice Gateway will take the following steps, in determining a callback number for the caller:

    a. The calling party number from the **P-Asserted-Identity** header will be used as the callback number in the outgoing PIDF-LO.
    b. If the **P-Asserted-Identity** does not contain a valid DID, the calling party number from the **From** header will be used as a callback number.
    c. If the calling party number in the **From** header is not a valid DID, the call will be eventually routed to the RedSky Relay Center (RSRC).



Determining a Callback Number(CBN) - Flow Chart
Call Server
RedSky® Voice Gateway
Horizon Mobility® / E911 Anywhere®
SIP INVITE Request
Process SIP INVITE Headers
Has E911-Callback-Number
Has E911-User-ID
Has E911-Location-ID
Has X-Genesys-ELIN or AP Loc
Has P-Asserted-Identity
Valid DID in FROM
Route call to RedSky Relay Center
Valid DID
Valid DID
Set Callback Number to domain part of FROM header
Route call to PSAP with Callback Number
Lookup Callback Number by E911-User-ID
Lookup Callback Number by E911-Location-ID
Set Callback Number to value in header
Set Callback Number to value in header
YES
NO

9 Error Scenario Reponses

Below are application specific error scenarios and expected responses return from the voice gateways.

9.1 Unsupported To: Destination

If an incoming call is received and the destination is not 9-1-1, 9-3-3 or a known DID number, the Voice Gateway will return a 404 Not Found response code.

9.2 Unknown Endpoint IP Address

If an incoming call is received from an IP address that has not been whitelisted in the system, a 403 Forbidden response code is returned.

9.3 Unsupported SIP Methods

For a REGISTER, MESSAGE, PUBLISH, or SUBSCRIBE method, a SIP Response of 503 is returned.

For an INFO or NOTIFY method, no response code is returned.

9.4 System Error

If an operational error occurs with the Voice Gateway and it cannot process the incoming SIP message, a 408 Request Timeout or a 5xx series response code will be returned.

10 References

- RFC 3261 – SIP: Session Initiation Protocol
- RFC 3265 – Session Initiation Protocol (SIP)-Specific Event Notification
- RFC 5411 – A Hitchhiker's Guide to the Session Initiation Protocol (SIP)
- RFC 6442 – Location Conveyance for the Session Initiation Protocol
- RFC 8787 – Location Source Parameter for the SIP Geolocation Header Field

- RFC 4119 – A Presence-based GEOPRIV Location Object Format
- RFC 5139 – Revised Civic Location Format for Presence Information Data Format Location Object (PIDF-LO)
- RFC 5491 – GEOPRIV Presence Information Data Format Location Object (PIDF-LO) Usage Clarification, Considerations, and Recommendations
- RFC 7459 – Presence Information Data Format Location Object (PIDF-LO)

- NENA i3 Standard for Next Generation 9-1-1
- NG9-1-1 Civic Location Data Exchange Format