OpenCNAM Integration with SIP Interface

A. Introduction

OpenCNAM provides several data channels through which customers can query its Caller ID Name lookup products. One of these is the SIP redirect-based query interface.

In this scenario, the customer sends a SIP INVITE to OpenCNAM, and OpenCNAM returns a SIP redirect containing the calling party name. The network element issuing the query, such as a session border controller (SBC), a convergent softswitch, a SIP proxy, etc., is configured by the customer to extract this value and populate it into the appropriate display field(s). The call is then sent onward to its final destination(s) with the calling party name value provided by OpenCNAM incorporated into the SIP message.

SIP redirect is an industry-standard approach to supplemental data queries, but effectively interfacing with OpenCNAM this way requires some technical work. This query interface is most appropriate for service providers, wholesalers, application service providers, and other actors who have industrial SIP equipment capable of consuming values in SIP redirects in a flexible, operator-defined manner. It also works well with the OpenSER-descended family of high-performance open-source SIP proxies (Kamailio and OpenSIPS).
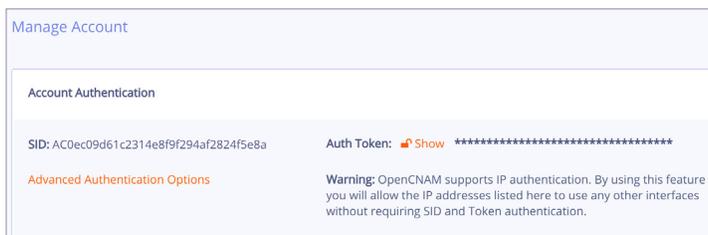
B. Prerequisites

First, the customer must add a static IP address from which SIP queries will be issued to the "whitelist" for this product. This can be accomplished in the OpenCNAM customer portal as follows:
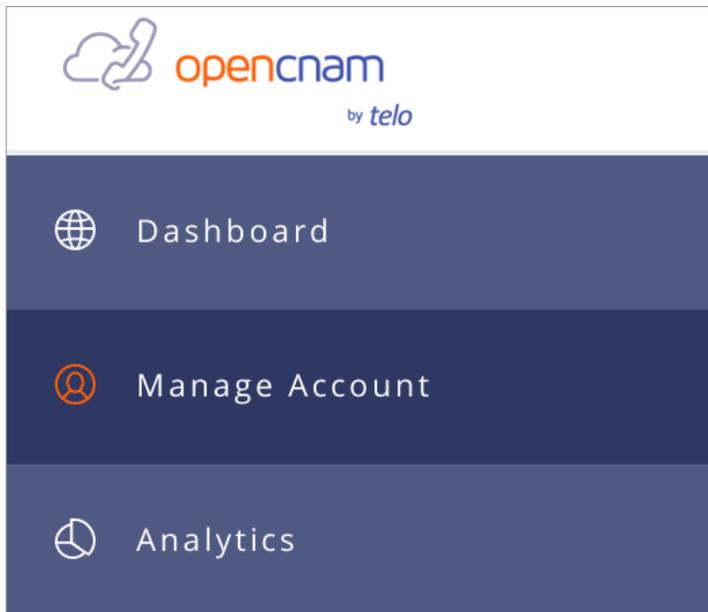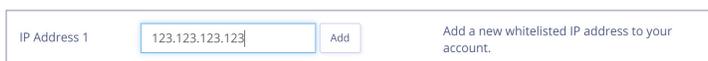

1 | Log into the customer dashboard at **www.opencnam.com/dashboard**

2 | Click on **Manage Account** on the left side bar

3 | Click **Advanced Authentication Options** under the Account Authentication section to expose the IP address whitelist
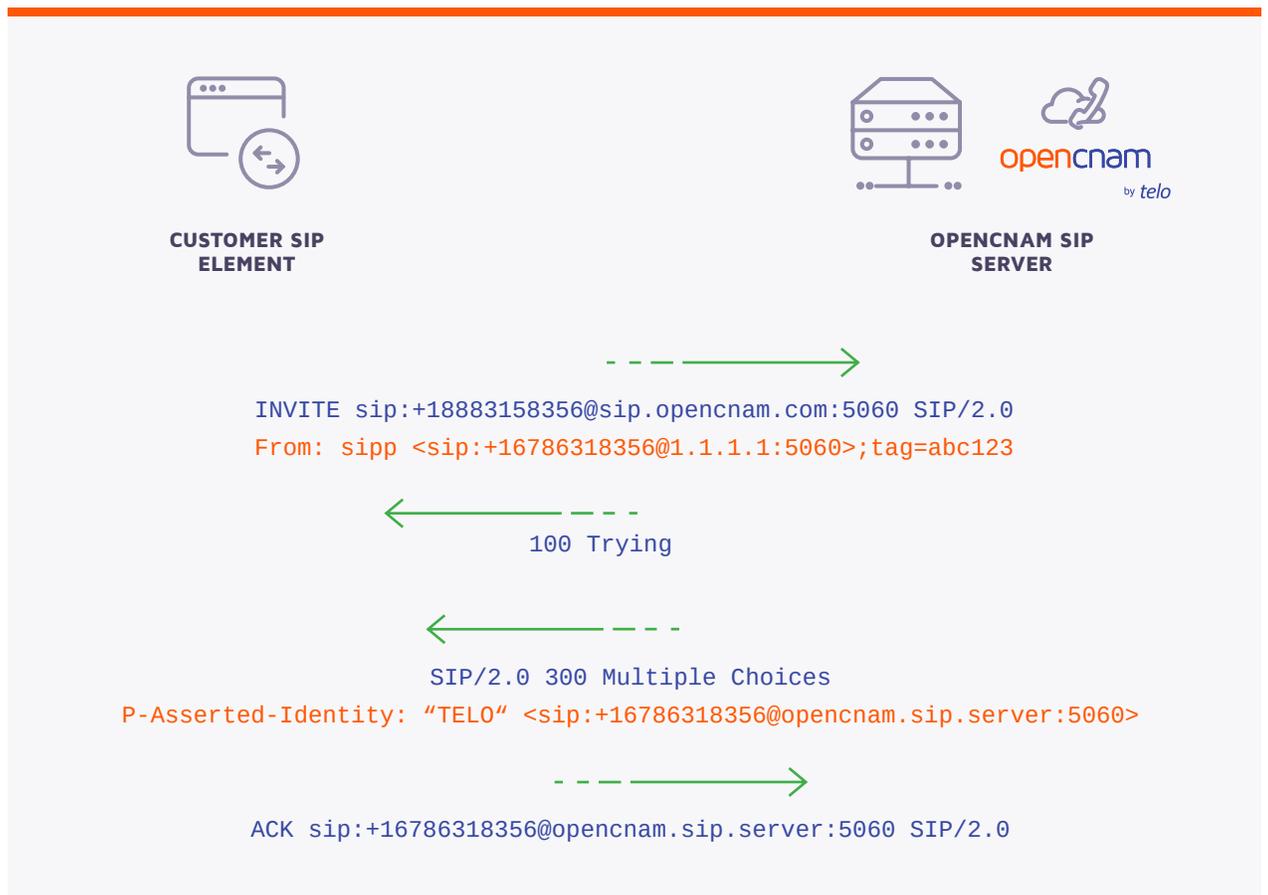

4 | Add the IP address(es) of the SIP signalling endpoint(s) from which your queries will be issued here

Queries from dynamic IP addresses are not supported.

C. Technical operation

An initial INVITE message is forwarded by the customer's SIP network element to OpenCNAM. OpenCNAM uses the calling party number (retrieved from the user part of the From URI) to look up the calling party name, and then in turn responds with a 300 Multiple Choices reply containing a P-Asserted-Identity header with that calling party name value in the display name field.

Here is an illustration of the call flow:


```
INVITE sip:+18883158356@sip.opencnam.com:5060 SIP/2.0
From: sipp <sip:+16786318356@1.1.1.1:5060>;tag=abc123
```

```
100 Trying
```

```
SIP/2.0 300 Multiple Choices
P-Asserted-Identity: "TELO" <sip:+16786318356@opencnam.sip.server:5060>
```

```
ACK sip:+16786318356@opencnam.sip.server:5060 SIP/2.0
```

Grammatically, a P-Asserted-Identity header value is a "name-address pair", which means that it can contain both a SIP URI and a quoted "display name" value (in which case the URI must mandatorily be in angle brackets). This is not unlike the To and From headers.

For purposes of the OpenCNAM service, the returned URI is irrelevant, though it is grammatically required. The calling party name is contained in the display name portion.

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Putting it all together, a typical SIP INVITE message sent to OpenCNAM would look something like this:

```
INVITE sip:+18883158356@sip.opencnam.com:5060 SIP/2.0

Via: SIP/2.0/UDP 1.1.1.1:5065;branch=z9hG4bK-19363-1-0

From: sipp <sip:+16786318356@1.1.1.1:5065>;tag=19363SIPpTag001

To: +18883158356 <sip:+18883158356@sip.opencnam.com:5060>

Call-ID: 1-19363@1.1.1.1

CSeq: 1 INVITE

Contact: sip:sipp@1.1.1.1:5065

Max-Forwards: 70

Subject: Performance Test

Content-Type: text/plain

Content-Length: 0
```

And the 300 Multiple Choices response from OpenCNAM for this request would look like this:

```
SIP/2.0 300 Multiple Choices

Via: SIP/2.0/UDP 1.1.1.1:5065;branch=z9hG4bK-19363-1-0;rport=5065

From: sipp <sip:+16786318356@1.1.1.1:5065>;tag=19363SIPpTag001

To: +18883158356 <sip:+18883158356@opencnam.server:5060>;tag=81a909aaaecb6605fd
ed3476e0d312ef-fe54

Call-ID: 1-19363@1.1.1.1

CSeq: 1 INVITE

P-Asserted-Identity: "TELO" <sip:+16786318356@172.30.0.220:5060>

Server: Telo OpenCNAM SIP query interface

Content-Length: 0
```

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D. Implementation

Per the foregoing example response, it is necessary for the customer to configure their SIP network element to either:

1 Transfer the "TELO" portion of the P-Asserted-Identity header returned in the 300 Multiple Choices response into the From header of an outgoing INVITE, or

2 Copy the returned P-Asserted-Identity header verbatim and append it to an outgoing INVITE. If this is done, any existing P-Asserted-Identity header will need to be replaced, and the success of this approach depends on the receiving SIP endpoint implementing the widespread preference in for sourcing calling party ID from P-Asserted-Identity over Remote-Party-ID over From, in that order.

E. Success

A successful CNAM query will return a 300 Multiple Choices message that looks something like this:

```
SIP/2.0 300 Multiple Choices

Via: SIP/2.0/UDP 1.1.1.1:5065;branch=z9hG4bK-19363-1-0;rport=5065

From: sipp <sip:+16785318356@1.1.1.1:5065>;tag=19363SIPpTag001

To: +18883158356 <sip:
+18883158356@sip.opencnam.com:5060>;tag=81a909aaaecb6605fd ed3476e0d312ef-fe54

Call-ID: 1-19363@1.1.1.1

CSeq: 1 INVITE

P-Asserted-Identity: "TELO" <sip:+16785318356@172.30.0.220:5060>

Server: Telo OpenCNAM SIP query interface

Content-Length: 0
```

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The "TELO" portion of the message contains the result of the CNAM query, which should be extracted and displayed to the SIP callee or otherwise consumed by a SIP endpoint.