# DSP-1282 & DSP-1283 Crestron Avia™ DSP (Secure) with Avaya Aura® 7.1 Platform

## Configuration Guide

Crestron Electronics, Inc.

The product warranty can be found at www.crestron.com/warranty.

The specific patents that cover Crestron products are listed at patents.crestron.com.

Certain Crestron products contain open source software. For specific information, visit www.crestron.com/opensource.

# Contents

# DSP-1282 & DSP-1283: Secure SIP Endpoint with Avaya Aura® 7.1 Platform

## Introduction

This configuration guide describes the procedures required to configure Crestron Avia™ Digital Signal Processor (DSP) devices in a secure mode. The devices operate on the Avaya Aura® Communications Manager as Session Initiation Protocol (SIP) endpoints .

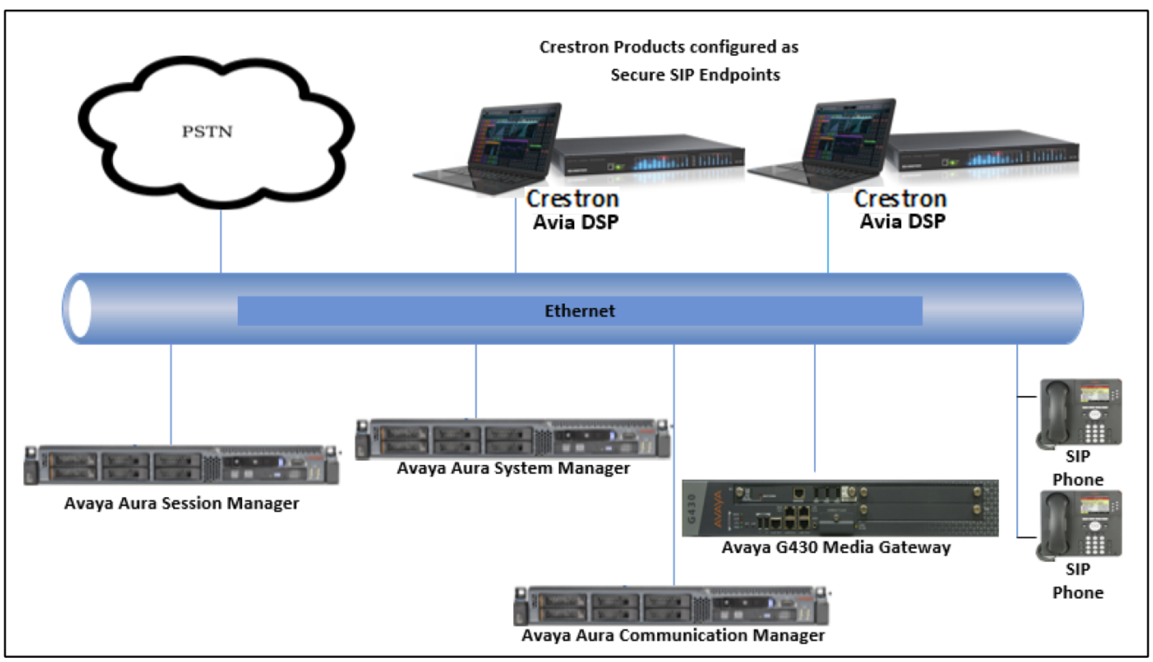
### Audience

The intended audience includes those attempting to configure and use Crestron Avia DSP devices as secure SIP endpoints registered to Avaya Aura Communication Manager 7.1.

### Topology

The diagram below shows the network topology for integration of a Crestron Avia DSP endpoint with Avaya Aura.

**Secure SIP Endpoint Integration with Avaya Aura - Reference Network**



The lab network consists of the following components:

- Avaya Aura Communication Manager
- Avaya Aura Session Manager
- Avaya Aura System Manager
- Avaya® SIP phones
- Avaya G430 Media Gateway
- Crestron Avia DSP as SIP endpoints

## Software Requirements

- Avaya Aura Communication Manager v7.1.2.0.0.532.24184
- Avaya Aura Communication Manager Messaging v7.0.0.1.441.1
- Avaya Aura System Manager v7.1.2.0.057353
- Avaya Aura Session Manager v7.1.2.0.712004
- Avaya g430 Media Gateway v39.5.0/2
- Crestron Avia DSP-128 v 1.00.262.005

## Hardware Requirements

- Cisco UCS-C240-M3S VMWare Host running ESXi 5.5
- Avaya Components either in a virtual environment or with separate hardware servers:
  - Avaya Aura Communication Manager
  - Avaya Aura Session Manager
  - Avaya Aura System Manager
  - Avaya G430 Media Gateway
  - Avaya Aura Communication Manager Messaging
- Public Switched Telephone Network (PSTN) gateway (Cisco 3845)
- Avaya Phones (2) in SIP
- Crestron Avia DSP devices (2):
  - Microphones for the DSP (2)
  - Speakers for the DSP (2)
  - Amplifiers for the DSP (2)
  - Appropriate cables for the above

## Product Description

The Crestron Avia DSP products (DSP-1282 and DSP-1283, specifically) consist of a family of programmable digital audio signal processors intended for the commercial sound market. Each version provides 12 analog mic/line inputs and eight analog line outputs. The devices include a Local Area Network (LAN) connection and a Universal Serial Bus (USB) connection for programming and control. The programmable signal flow is a fixed topology with user-configurable input and output processing chains using a library of preset signal-specific DSP blocks.

Use the Crestron Avia too to:

- Discover the device on the network
- Configure the SIP parameters
- Configure the mixers to allow 2-way communication on a SIP call

Save the audio configuration along with the SIP configuration as a project file. The project file can be loaded onto all of the DSPs that receive similar settings on a given project. Minor modifications may be necessary.

Use the Crestron Toolbox™ software to discover and control all Crestron devices on the network.

During the integration test, Crestron Toolbox can:

- Discover devices on the network
- Console connect to the devices
- Configure the Ethernet settings
- Upgrade firmware

## Summary

This document describes how to configure the Crestron Avia DSP devices, in secure mode, as basic SIP endpoints. It also provides information on how to register devices to the Avaya Aura Session Manager with digest authentication.

Supported features include:

- Registration with digest authentication
- Basic calls with G711u and G711a codecs
- Dual-Tone Multi-Frequency (DTMF) support
- Early media support
- Retrieval of a parked call
- Transferee in a call transfer
- Conference participant
- Member of hunt group

Unsupported features include:

- Calls with non-secure (Real-time Transport Protocol (RTP) only) devices
- Caller ID presentation
- Call hold and resume
- Call forwarding on the device (forwarding can be configured on the Private Branch Exchange (PBX) for the Domain Name (DN) assigned to the endpoint)
- Call waiting
- Conference
- Attended call transfer
- Early attended call transfer
- Blind call transfer
- Shared line (configuration of shared line on device)
- Initiating call park
- Voice mail access and interaction

Known issues and limitations include:

- No support for caller ID on the Crestron Avia DSP.
- No support for MWI on the Crestron Avia DSP.
- The DSP does not support Music on Hold when integrated with the Avaya Aura PBX.
- The DSP does not support changes to DNS management when configured without DHCP settings via Toolbox.
- Intermittent issue with DTMF sent from Crestron Avia DSP to Avaya Media Gateway. The far end indicates duplicate and missing DTMF events.
- During a call from a secure DSP to the Avaya Communication Manager Messaging (CMM), there is incompatibility in Real-time Transport Control Protocol (RTCP) support between devices. The Avaya CMM only supports RTCP and does not support Secure Real-time Transport Control Protocol (SRTCP, sent by the DSP in secure mode). The Avaya CMM rejects SRTCP calls with a 488 "Not Acceptable Here" message.

# Crestron Avia DSP Configuration

This section provides the following details:

- How to set up connections to the amplifier and speaker
- How to access the DSP on the network (once powered)
- How to configure the DSP for registration and integration with the PBX

## Connections

Make the following connections:

- Connect microphone to DSP MIC/LINE INPUTS port 1
- Connect DSP LINE OUTPUTS port 1 to "Audio In" on amplifier
- Connect "Audio Out" of amplifier to speaker
- Connect LAN port to network

## Device Discovery/Access

Use the Crestron Toolbox and the Crestron Avia tool to discover and access the connected LAN and/or VOIP ports) DSP devices.

Use the Help menu to assist when performing the discovery and configuration procedure.

### Set Up SIP Interface

The DSP units have separate network interfaces for Voice over Internet Protocol (VoIP) and LAN on the rear panel. Configure either one for SIP calling. The default configuration binds SIP calling to the LAN interface. An optional console command binds the SIP interface to the VoIP connector. Configure all VoIP connections on a separate Virtual Local Area Network (VLAN) or subnet. VoIP connections cannot be on the same subnet as the LAN connection.

#### Ethernet

Use the **Ethernet** command to turn the VoIP port on/off.

```
DSP-1281>Ethernet ?
ETHERNET [<device_num> ON | OFF [/now]]
Device_num - 0  n
ON - enables VoI
OFF - disables VoIP
/now - take effect without a reboot
No parameter - displays the current setting
```

The VoIP port is off by default. The LAN port is not selectable.

```
<device_num> = 0 selects the LAN port
<device_num> = 1 selects the VoIP port
```

### SIP Interface

Use the **sipinterface** command to bind all SIP activity, data, and traffic to the selected port. If a VLAN or exclusive VoIP network is available, bind to the VoIP port (recommended).

```
DSP-1281>sipinterface ?
Get or Set SIP Interface
SIPINTERFACE [LAN | VOIP]
LAN - normal LAN port
VOIP - VOIP port
No Parameter - Displays current setting
```

## Set Up Routes

If the configured VoIP port is the SIP interface, add a static route to ensure that all SIP routing is via the VoIP port.

The following console commands (**routeadd**, **routedel**, **routeprint**, and **routetrace**) support the static IP routing configuration:

```
DSP-1282>routeadd ?
ROUTEADD <destination> <netmask> <gateway> [/FORCE]
   destination - destination IP address in dot decimal notation
   netmask - netmask in dot decimal notation
   gateway - gateway in dot decimal notation
   /FORCE - force to add/delete even if failed to persist to NVRAM

DSP-1282>routedel ?
ROUTEDELETE <destination> <netmask> <gateway> [/FORCE]} | </ALL>
   destination - destination IP address in dot decimal notation
   netmask - netmask in dot decimal notation
   gateway - gateway in dot decimal notation
   /FORCE - force to add/delete even if failed to persist to NVRAM
   /ALL - delete all routes from NVRAM

DSP-1282>routeprint ?
ROUTEPRINT - shows current routes

DSP-1282>routetrace ?
ROUTETRACE <IPaddress>
   IPaddress - IP address in dot decimal notation
```

# Device Configuration

The basic setup for a phone call requires:

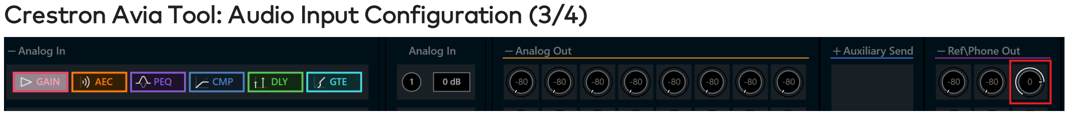
- An analog input (such as from a microphone) routed out through the phone line
- Audio coming in from the phone line routed to an analog output (such as to an amplifier or speaker)

## Configure the DSP Device

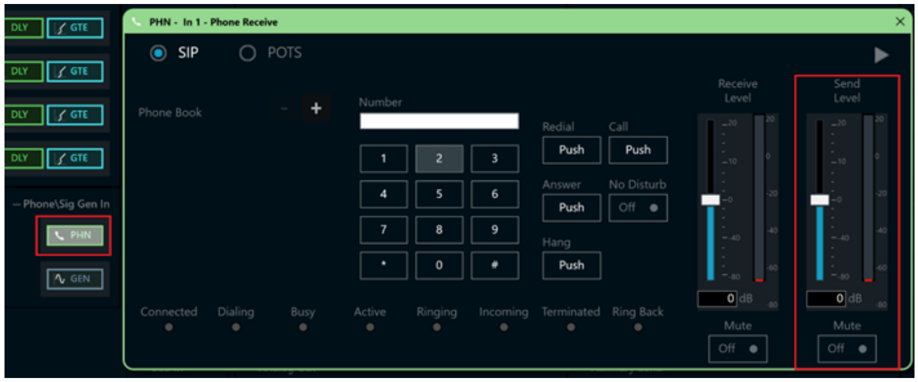
Use the Crestron Avia tool to select and configure the DSP device.

### Input Configuration

To configure the analog input:

1. Click **Signal**.
   **Crestron Avia tool: Audio Input Configuration (1/4)**
   
   
2. Under **Analog In 1** (first row), double click **Gain**. In the new window set the following:
   a. Click **Mute** to **Off**.
   b. Select **33** for the **Analog Gain**.
   c. If a condenser microphone is being used, click **+48V** (phantom power) to **On**.
   **Crestron Avia Tool: Audio Input Configuration (2/4)**
   
   
3. Under **Analog In 1** (first row), click **Ref/Phone Out** (right-most column) and enter **0** as the decibel value.
   **Crestron Avia Tool: Audio Input Configuration (3/4)**
   
   

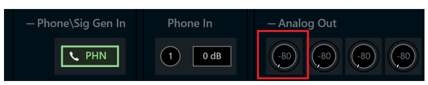
4. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
    a. Move the **Send Level** slider to **0 db**.
    b. Click **Mute** to **Off**.
    **Crestron Avia Tool: Audio Input Configuration (4/4)**



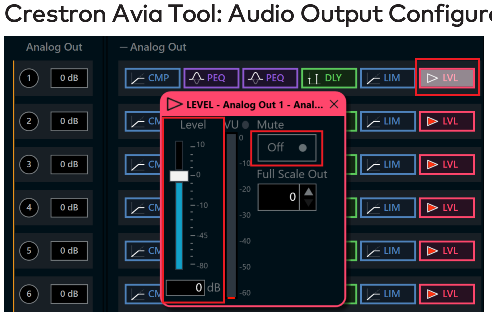

## Output Configuration

To configure the analog output:

1. Under **Phone In 1** (first row), click **Analog Out** (left-most column) and enter **0** as the decibel value.
    **Crestron Avia Tool: Audio Output Configuration (1/3)**




2. Under **Analog Out 1**, double click **LVL**. In the new window set the following:
    a. Move the **Level** slider to **0 db**.
    b. Click **Mute** to **Off**.
    **Crestron Avia Tool: Audio Output Configuration (2/3)**



3. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
   a. Move the **Receive Level** slider to **0 db**.
   b. Click **Mute** to **Off**.

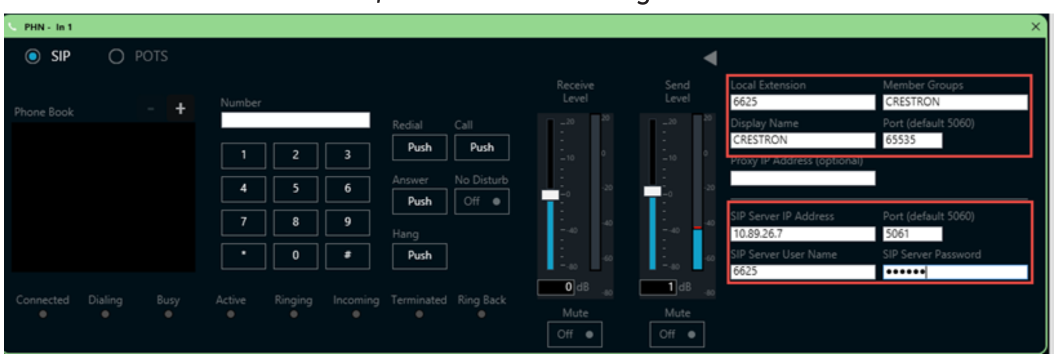
   **Crestron Avia Tool: Audio Output Configuration (3/3)**


## Configure the SIP Parameters

From the open **PHN - In 1 - Phone Receive** window, select and configure the SIP parameters.

1. With **SIP** selected, click the chevron at the right top corner to expand the window.

   **Crestron Avia Tool: Phone Dialer, SIP Parameters Configuration**



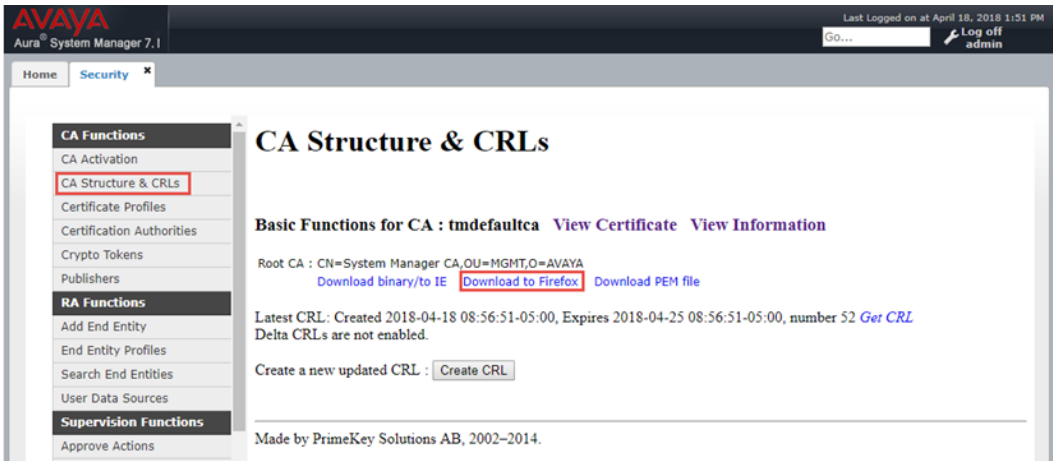

2. Enter the extension configured on Avaya Aura CM for the **Local Extension** for this device. This example uses **6625**.
3. Enter the Avaya Aura Session Manager PBX for the **SIP Server IP Address**. This example uses **10.89.26.7**.
4. Enter the SIP server port (**5061**) for the **Port**.
5. Enter the same end user name configured for the Avaya Aura Session Manager with the digest authentication credentials for the **SIP Server User Name**..
6. Enter the same password as configured for the Avaya Aura Session Manager end user digest credentials for the **SIP Server Password**.

## Certificates

For a successful TLS handshake between the DSP device and the Avaya Aura, add the following root certificate to the DSP: A rootCA certificate (systemmanagerca.cer).

Download this certificate from the Avaya System Manager. The DSP requires the certificate to validate the Avaya Aura when the "Enable Server Validation" is enabled.

To download Avaya Aura CA from Avaya Aura System Manager:

1. Click **Home** > **Services** > **Security**.
   **Download Root Certificate from Avaya System Manager**
   
   
2. Click **CA Structure & CRLs**.
3. From the CA Structure & CRLs window, click on **Download to Firefox** (for this example).
4. Save the file as **systemmanager.cer** (for this example).

### Copy Certificates

Copy the Root Certificate into the DSP device under the directory /user/cert **by doing SFTP**.

## Import and Assign Root Certificate

Use the Crestron Toolbox text console utility (in the Crestron Avia DSP console) to import and assign the root certificate.

1. Type the command `certificate addf systemmanagerca.cer root` (where `RootCA` is the name of the root certificate uploaded in the previous section).

   **Root Certificate - Successful Import**

```
DSP-1282>certificate addf systemmanagerca.cer root
Import successful.
```

2. Verify the uploaded root certificate and note the # of the certificate.

   **List of Trusted Root Certificates in the Device**

```
DSP-1282>Console command returned error.
DSP-1282>siptrustedcas listn
TableStart: [Sip Trusted Certification Authorities]

#   | Name                                          | UID                                        | In Use |
----+-----------------------------------------------+--------------------------------------------+--------+
145 | EC-ACC                                        | -11D4C2142BDE21EB579D53FB0C223BFF          | No     |
146 | thawte Primary Root CA                        | 344ED55720D5EDEC49F42FCE37DB2B6D           | No     |
147 | Visa eCommerce Root                           | 1386354D1D3F06F2C1F96505D5901C62           | No     |
148 | skypelabsj-DC01-CA                            | 343F63A0387235A1449ABB5676612403           | No     |
149 | System Manager CA                             | 0853D8A9BB8E1C09                           | No     |
```

   **Root Certificate - Assignment**

```
DSP-1282>siptrustedcas use 149
Sip Trusted Certification Authorities USING System Manager CA                    |0853D8A9BB8E1C09

DSP-1282>|
DSP-1282>siptrustedcas listu
TableStart: [Sip Trusted Certification Authorities]

#   | Name                                          | UID                                        | In Use |
----+-----------------------------------------------+--------------------------------------------+--------+
149 | System Manager CA                             | 0853D8A9BB8E1C09                           | Yes    |
```

# Avaya Aura Communication Manager Configuration

This section describes the Avaya Aura Communication Manager (Avaya CM) configuration necessary to integrate the Crestron Avia DSP devices in secure mode.

**NOTE:** Confirm that the general installation and basic Avaya CM configuration have been administered.

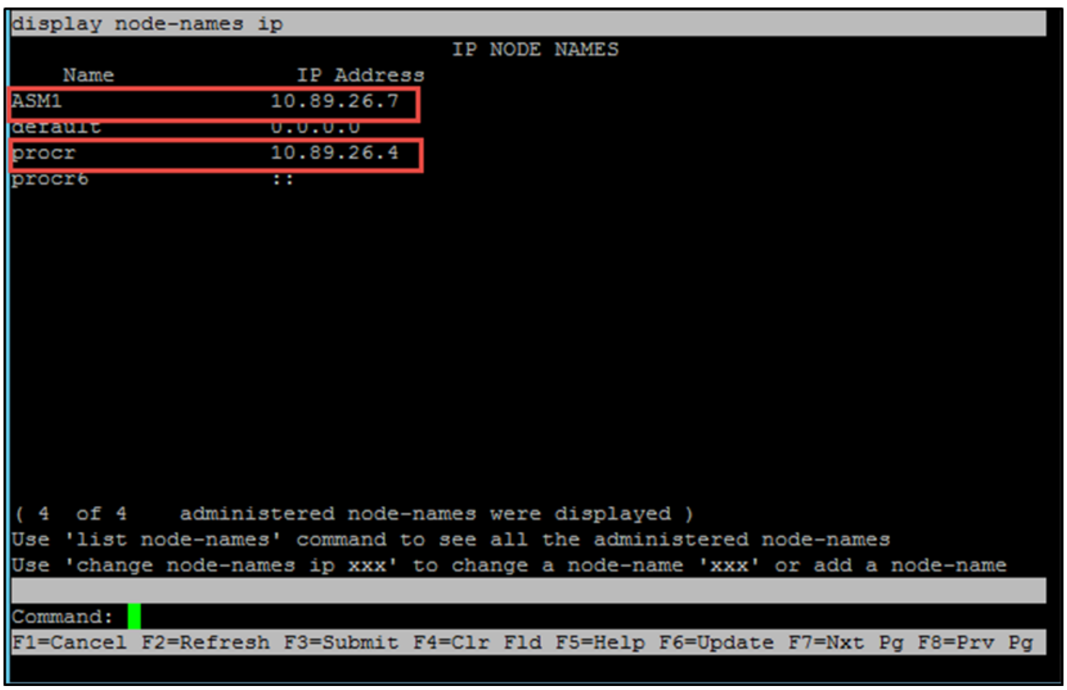
## Node Names

Configure the node IP for Avaya Aura Session Manager (SM) and Avaya CM.

Use the **change name-names ip** command to add the node name. This example adds **ASM1** and **procr** with their respective IPs.

- Use **ASM1**, an Avaya Aura Session Manager, to register the SIP phones and third-party SIP devices.
- User **procr** to register SIP trunk between Avaya CM and Avaya SM.

**Avaya Aura CM: Configure Node**


```
display node-names ip
                                   IP NODE NAMES
    Name                  IP Address
ASM1                    10.89.26.7
default                 0.0.0.0
procr                   10.89.26.4
procr6                  ::

( 4  of 4    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

Command:
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

## Dial Plan Analysis

Configure several dial strings to ensure complete test coverage. This example includes calling between stations, calling to PSTN, and accessing PBX features.

Use the **change dialplan analysis** command to configure the following dial patterns for this example:

1. Enter **5** for the mail number **Dialed string**.
2. Enter **6** for the station number **Dialed string**.
3. Enter **8** for the feature access code **Dialed string**.
4. Enter **9** for the feature access code **Dialed string**.
5. Enter * for the feature access code **Dialed string**.
6. Enter **#** for the dial access code **Dialed string**.

Use the **display dialplan analysis** command to view the configured dial strings/codes.

Avaya Aura CM: Dial Plan Analysis

```
display dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 3

    Dialed    Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String    Length Type      String    Length Type      String    Length Type
   0             1   attd
   1             4   ext
   2             4   ext
   5             4   ext
   6             4   ext
   7             4   ext
   8             1   fac
   9             1   fac
   *             3   fac
   #             4   dac

F1=Cancel  F2=Refresh  F3=Submit  F4=Clr Fld  F5=Help  F6=Update  F7=Nxt Pg  F8=Prv Pg
```

## IP Network Region

The example configures all the SIP phones in ip-network-region-1. Configure the **Domain** name and **Codec Set** parameters.

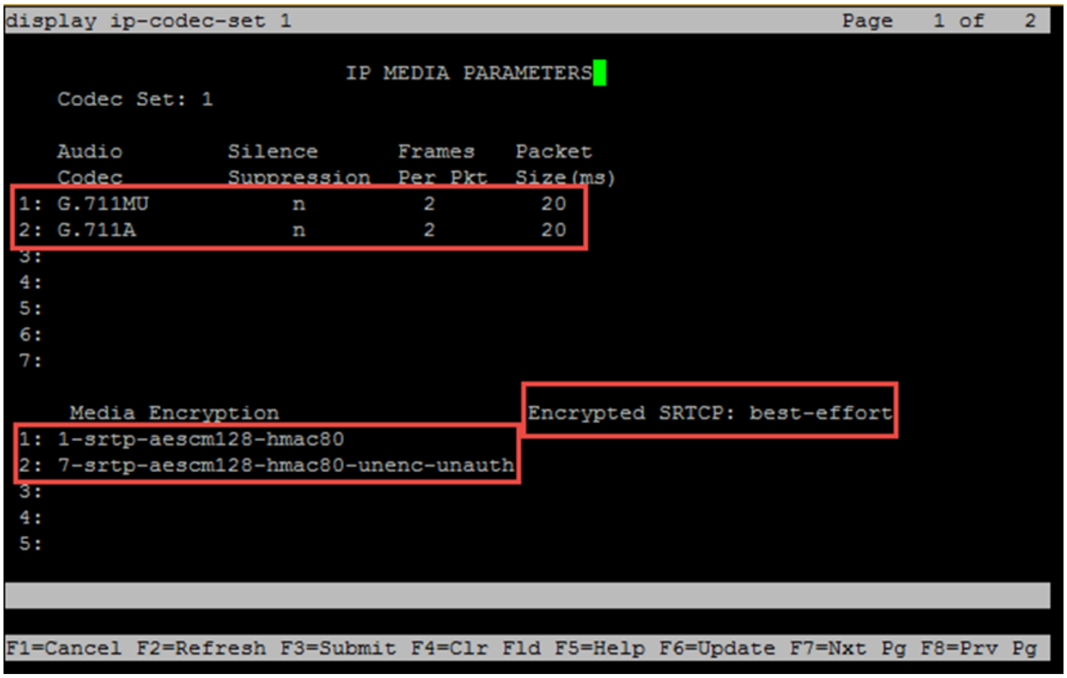
**Avaya Aura CM: ip-network-region**

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1         NR Group: 1
Location:           Authoritative Domain: lab.tekvizion.com
    Name:                       Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

## Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the PBX and PSTN.

**Avaya Aura CM: Codec Configuration**



```
display ip-codec-set 1                                          Page   1 of   2

                          IP MEDIA PARAMETERS

    Codec Set: 1

    Audio          Silence         Frames    Packet
    Codec          Suppression     Per Pkt   Size(ms)
 1: G.711MU             n             2         20
 2: G.711A              n             2         20
 3:
 4:
 5:
 6:
 7:

     Media Encryption                       Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: 7-srtp-aescm128-hmac80-unenc-unauth
 3:
 4:
 5:

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

This example uses **1** for the **Codec Set**. The Crestron DSP device supports and includes G.711A and G.711MU in this set. To test with the DSP, enter **G.711A** and **G.711MU** in the **Audio Codec** column of the table. Use default values for all other fields. Configure media encryption (SRTP and SRTCP). The **Media Encryption** section shows sample values.

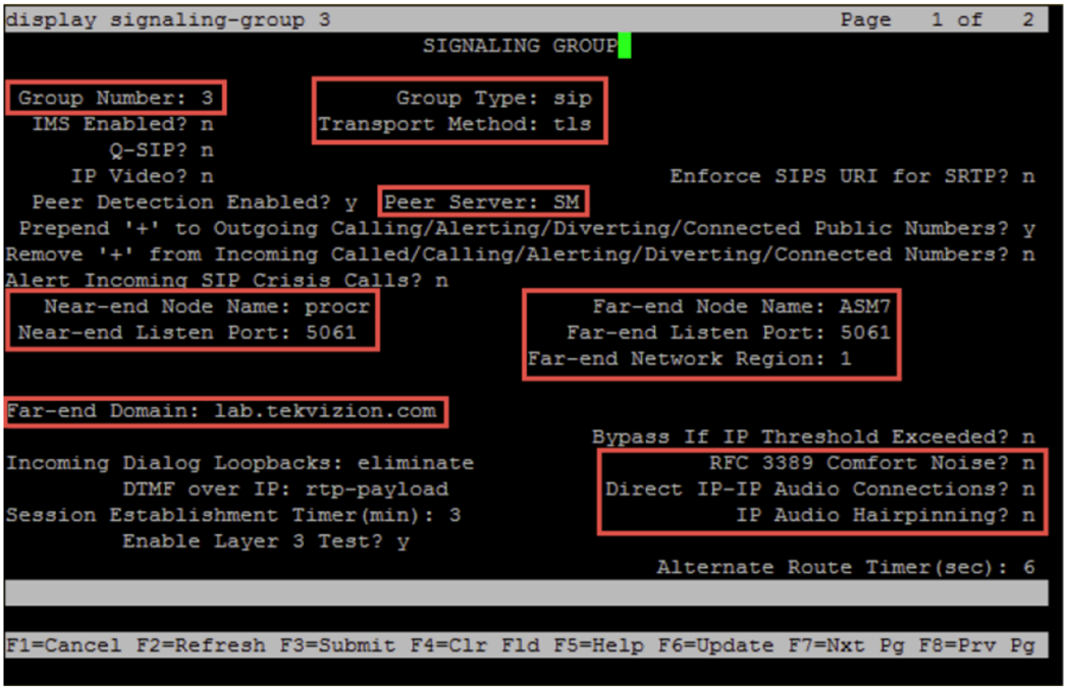
## Signaling Group

This example configures two signaling groups.

- Signaling Group 3
  This group supports communication between SM and CM for SIP phone registration and features.
- Signaling Group 10
  This group supports PSTN calling on ISDN-PRI.

Use the **add signaling group n** command to create a signaling group system (where **n** represents the signaling group number for this example).

Avaya Aura CM: Signaling Group Configuration for Phones

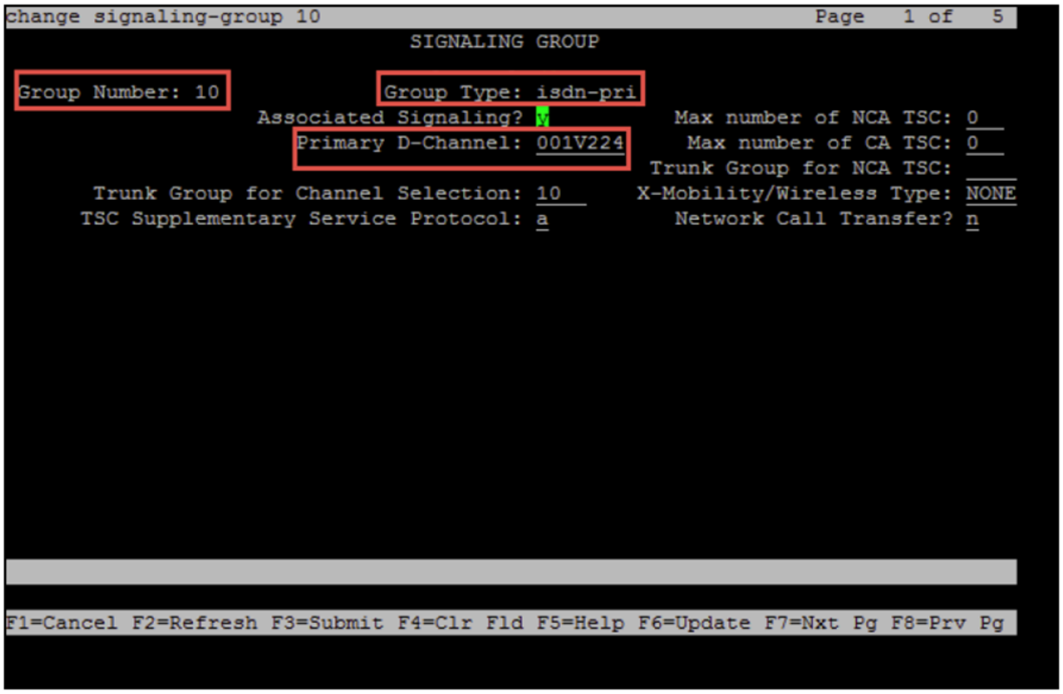


```
display signaling-group 3                                       Page   1 of   2
                                SIGNALING GROUP

Group Number: 3              Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                    Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: ASM7
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: lab.tekvizion.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                   Alternate Route Timer(sec): 6

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

To configure Signaling Group 3 (for this example):

1. Enter **3** for the **Group Number**.
2. Enter **sip** for the **Group Type**.
3. Enter **tls** for the **Transport Method**.
4. Enter **SM** for the **Peer Server**.
5. Enter **procr** for the **Near-end Node Name**.
6. Enter **5061** for the **Near-end Listen Port**.
7. Enter **ASM7** for the **Far-end Node Name**.
8. Enter **5061** for the **Far-end Listen Port**.
9. Enter **1** for the **Far-end Network Region**.
10. Enter **lab.tekvizion.com** for the **Far-end Domain**.
11. Enter **n** for **Direct IP-IP Audio Connections**.

Avaya Aura CM: Signaling Group Configuration for PSTN




To configure Signaling Group 10 (for this example):

1. Enter **10** for the **Group Number**.
2. Enter **isdn-pri** for the **Group Type**.
3. Enter **001V224** for the **Primary D-Channel**.

## Trunk Groups

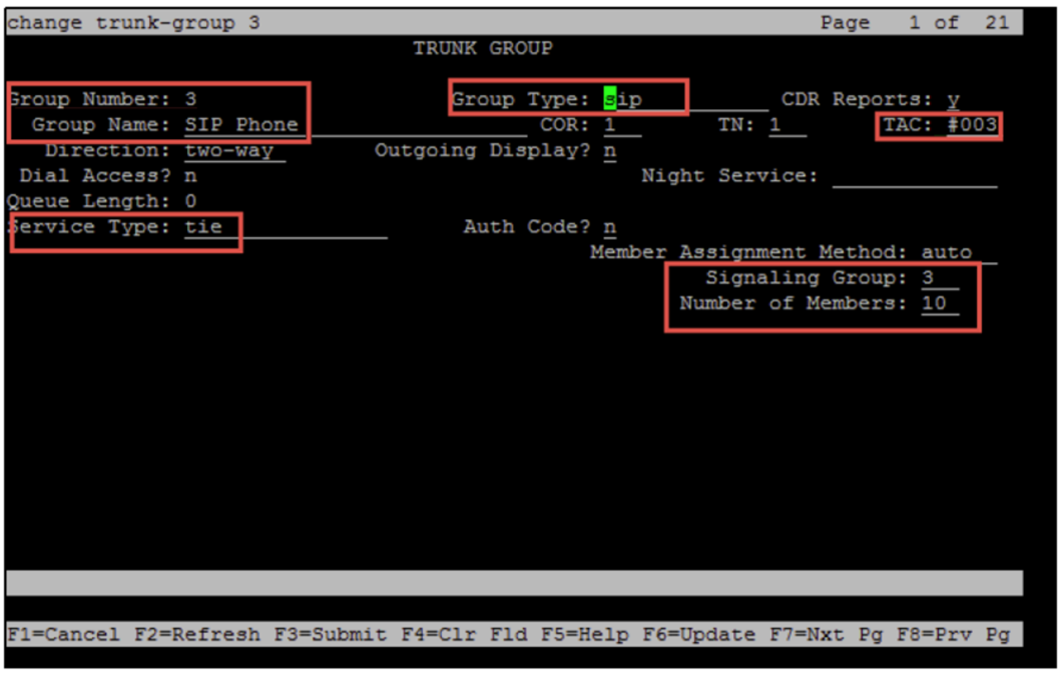
Configure two trunk groups (for this example):

- Trunk Group 3
  This group accesses the stations registered to the Avaya Session Manager.
- Trunk Group 10
  This group sends a 10/11-digit calling number to PRI trunk or PSTN.

Use the **add trunk group n** command to add a new trunk group (where **n** represents the trunk group number).

### Trunk Group 3

**Avaya Aura CM: Trunk Group to Session Manager - Trunk Group 3 (1/4)**




Configure Trunk Group 3 (entries for steps 1 through 7 are for this example):

1. Enter **3** for the **Group Number**.
2. Enter **SIP Phone** for the **Group Name**.
3. Enter **sip** for the **Group Type**.
4. Enter **tie** for the **Service Type**.
5. Enter **#003** for the **TAC**.
6. Enter **3** for the **Signaling Group**.
7. Enter **10** for the **Number of Members**.

Avaya Aura CM: Trunk Group to Session Manager - Trunk Group 3 (2/4)

```
change trunk-group 3                                           Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

           SCCAN? n                                   Digital Loss Group: 18
                     Preferred Minimum Session Refresh Interval(sec): 1800

 Disconnect Supervision - In? y  Out? y

            XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n




 Caller ID for Service Link Call to H.323 1xC: station-extension


F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

8. Enter **1800** for the **Preferred Minimum Session Refresh Interval (sec.)**.

Avaya Aura CM: Trunk Group to Session Manager - TrunkGroup 3 (3/4)

```
display trunk-group 3                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

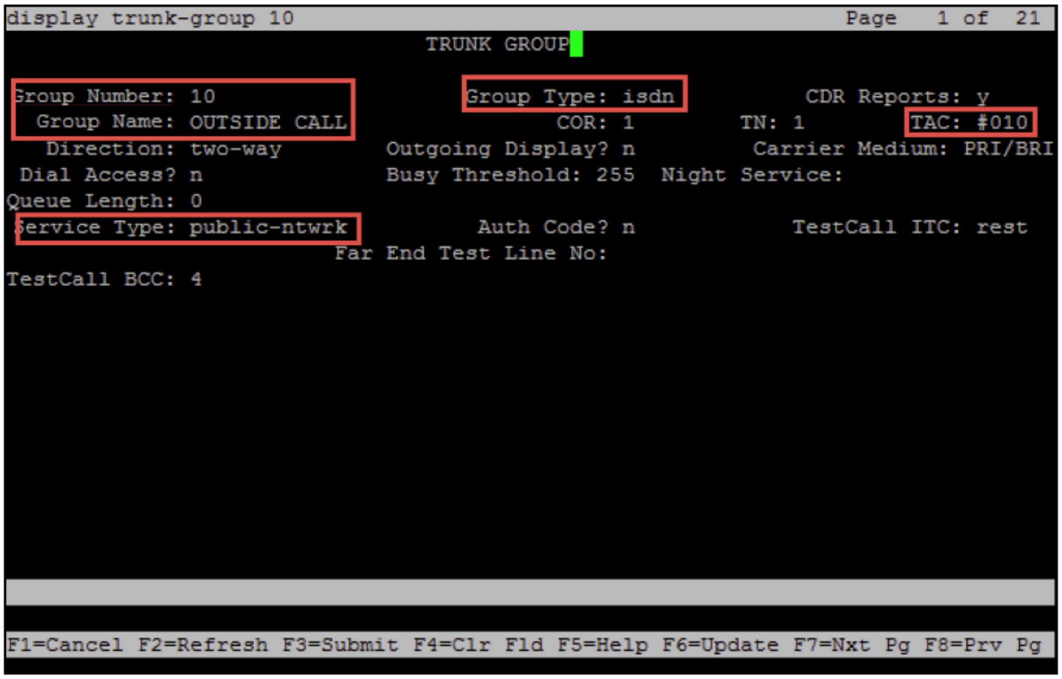
9. Enter **private** for the **Numbering Format**.

Avaya Aura CM: Trunk Group to Session Manager - Trunk Group 3 (4/4)

```
display trunk-group 3                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 96


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? y
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits


F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

## Trunk Group 10

**Avaya Aura CM: Trunk Group to PRI/PSTN - Trunk Group 10**




Configure Trunk Group 10 (for this example):

1. Enter **10** for the **Group Number**.
2. Enter **OUTSIDE CALL** for the **Group Name**.
3. Enter **isdn** for the **Group Type**.
4. Enter **public-ntwrk** for the **Service Type**.
5. Enter **#010** for the **TAC**.

## Route Pattern

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route pattern n** command (where **n** represents the route pattern number to configure the parameters for the PSTN trunk route pattern).

Use Route Pattern 3 for calling extensions via Avaya Aura Session Manager.

**Avaya Aura CM: Route Pattern for SIP Phones**

```
display route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3      Pattern Name: SIP Phone
    SCCAN? n     Secure SIP? n      Used for SIP stations? y
    Primary SM: ASM7              Secondary SM:
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 3     0                                                            n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W      Request                                Dgts Format
 1: y y y y y n  n            rest                               unk-unk    none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

Use Route Pattern 10 for calling PSTN.

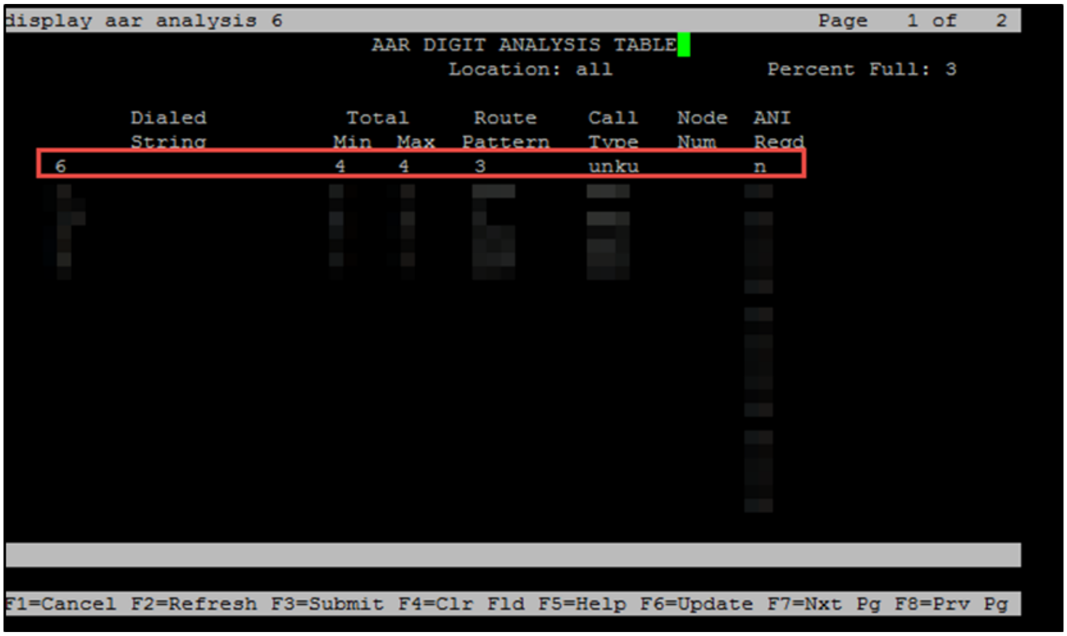
**Avaya Aura CM: Route Pattern for PSTN (PRI)**

```
display route-pattern 10                                         Page   1 of   3
                    Pattern Number: 10     Pattern Name: PRI
    SCCAN? n     Secure SIP? n      Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 10    0                                                            n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W      Request                                Dgts Format
 1: y y y y y n  n            rest                               unk-unk    none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

## Auto Alternative Routing

Use the **change aar analysis n** command (where **n** represents the first digit of the extension numbers for making calls).

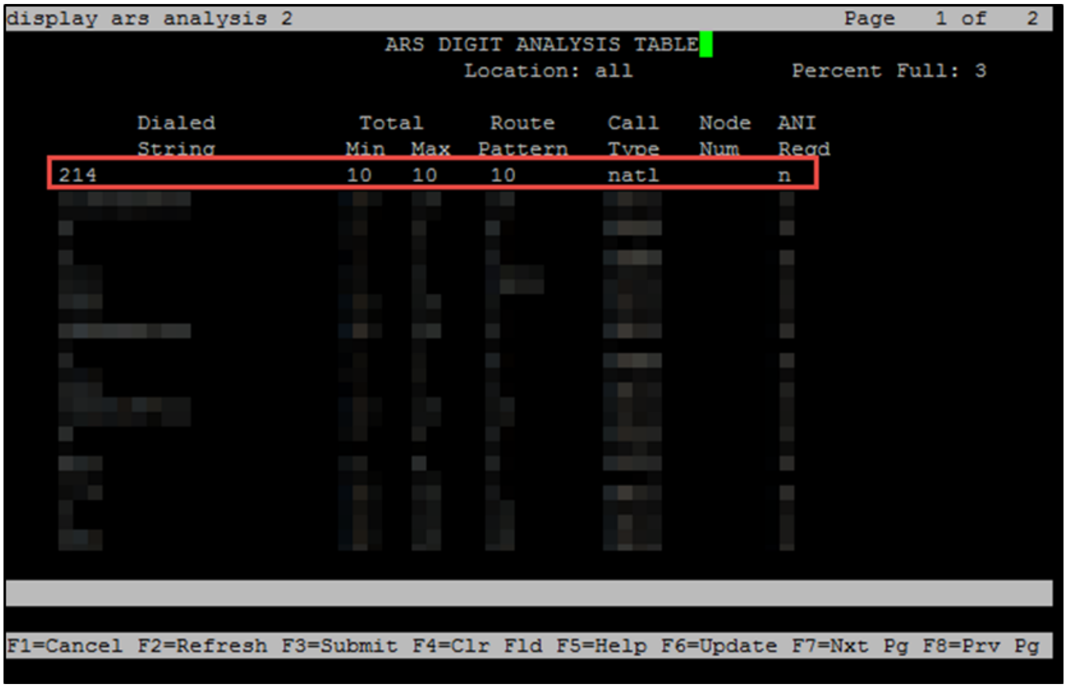
**Avaya Aura CM: Auto Alternative Routing Analysis 6**


```
display aar analysis 6                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 3

          Dialed               Total     Route     Call    Node   ANI
          String             Min  Max   Pattern    Type    Num    Reqd
    6                         4    4     3         unku           n
[redacted]

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

## Automatic Route Selection

Use the **change ars analysis n** command (where **n** represents the pattern for making PSTN calls).

**Avaya Aura CM: Auto Routing Selection Analysis**



```
display ars analysis 2                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 3

          Dialed              Total     Route     Call    Node   ANI
          String             Min  Max  Pattern    Type    Num    Reqd
    214                      10   10    10        natl           n
[redacted]

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

# Avaya Aura Session Manager

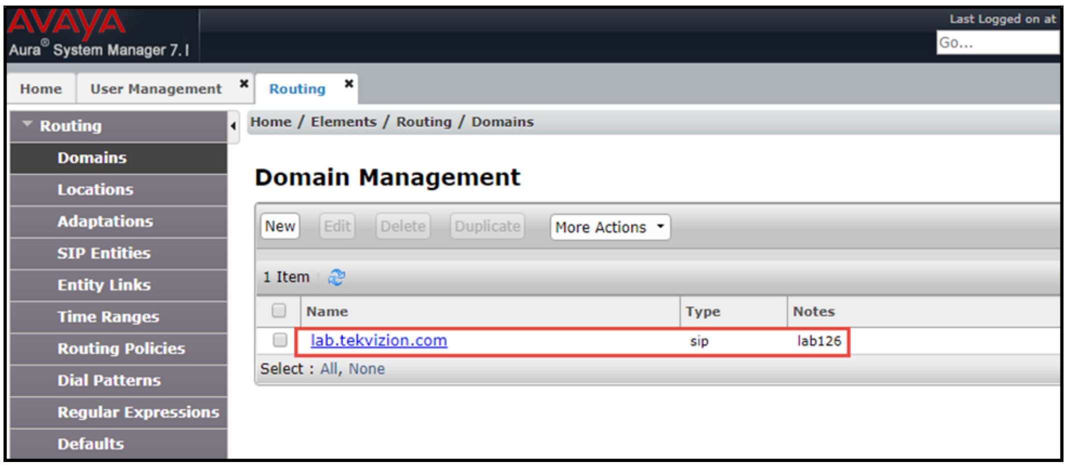
## Domain

To route calls, create a SIP domain for each domain administered by the Session Manager.

To configure a domain:

1. Click **Home** > **Routing** > **Domains**.
2. Click **New**.

**Avaya Aura SM: Domain Configuration**




3. Enter the domain name for the **Name**. This example uses **lab.tekvizion.com**.
4. Select **sip** for the **Type**.
5. Enter a brief description for the **Notes** (optional).
6. Click **Commit** to save (not shown).

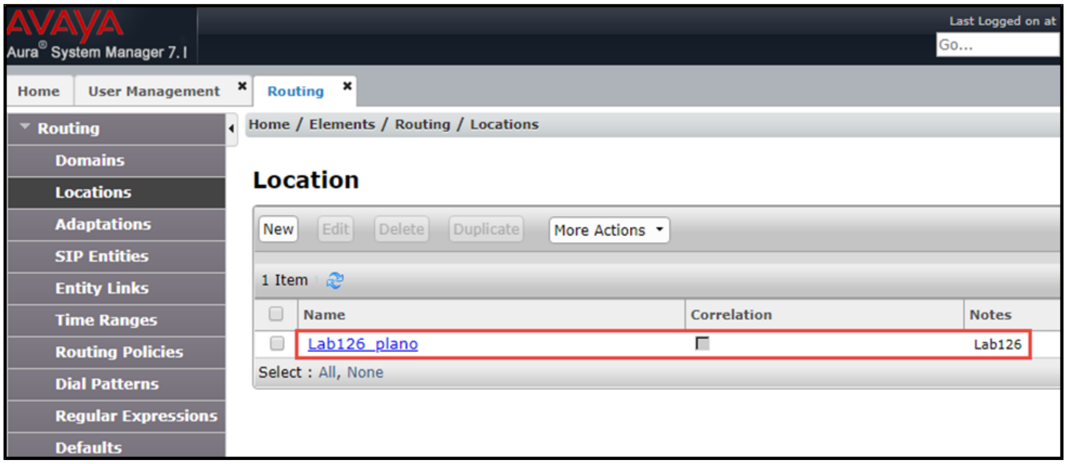
## Location

Use locations to identify logical and/or physical locations where SIP entities reside for the purposes of bandwidth management and call admission control.

To add a location:

1. Click **Routing** > **Locations**.
2. Click **New**.

   **Avaya Aura SM: Location Configuration**

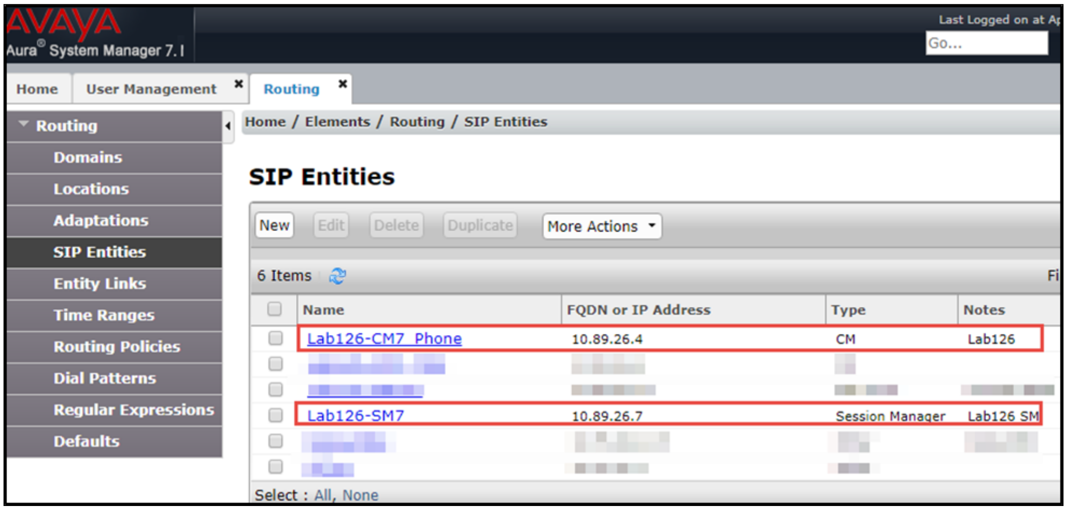
   
   

3. In the **General** section, do the following:
   a. Enter a descriptive name for the **Name** of the location. This example uses **Lab126-plano**.
   b. Enter a brief description for the **Notes** (optional).
   c. Use the default values for all remaining fields.
4. Click **Commit** to save (not shown).

## SIP Entity

Add a SIP entity for each SIP telephony system connected to the Session Manager, which includes Communication Manager and Avaya Communication Manager Messaging Component.

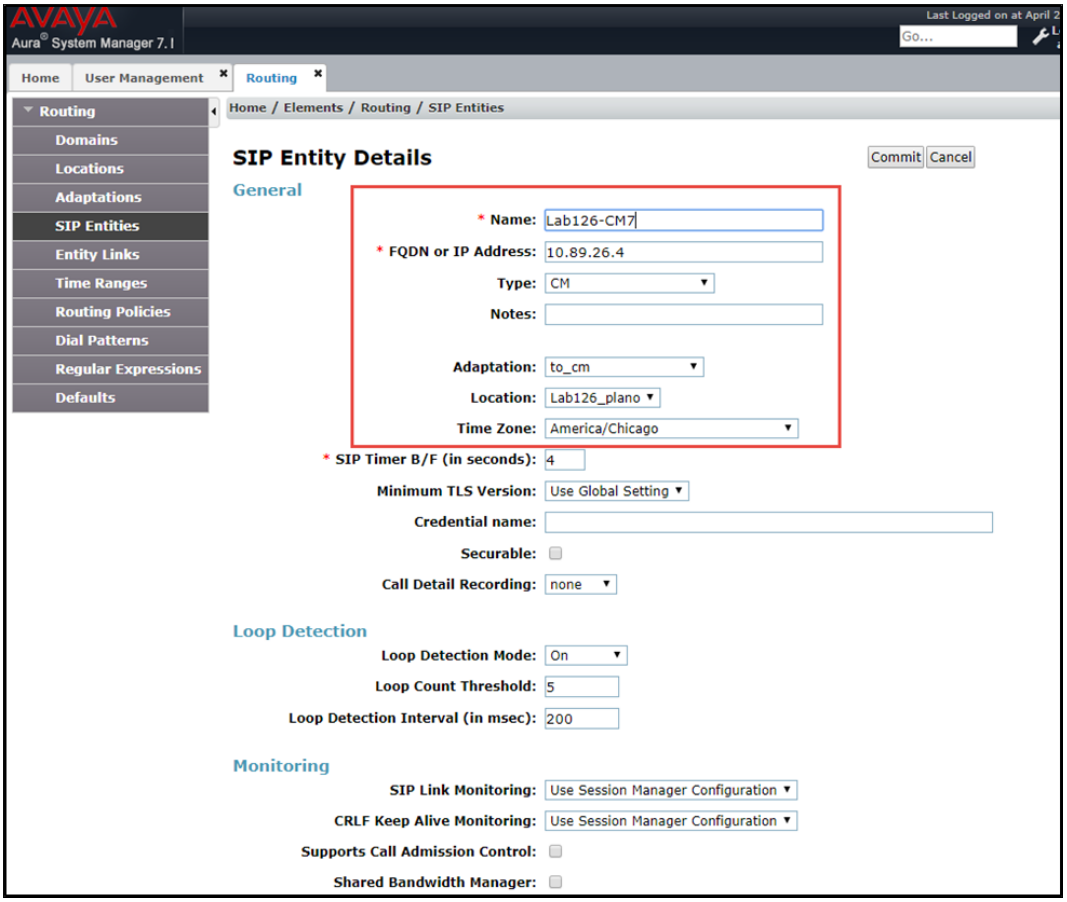
**Avaya Aura SM: SIP Entity**




To add a SIP entity:

1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: Sip Entity - CM Configuration

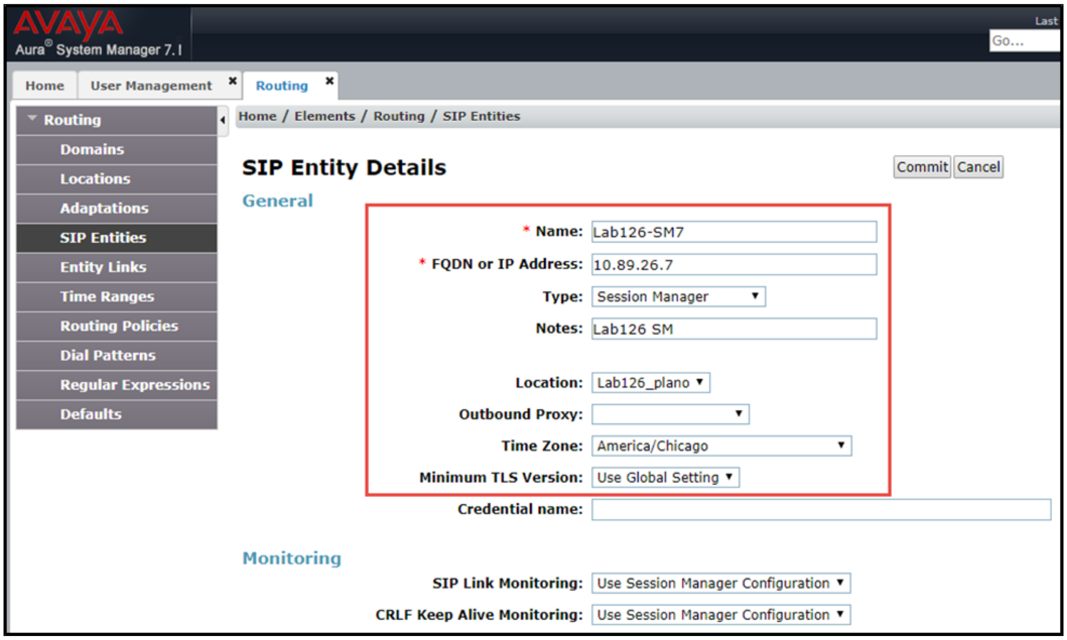



3. In the **General** section, do the following:
    a. Enter a descriptive name for the **Name**. This example uses **Lab126-CM7** for the Avaya CM.
    b. Enter the FQDN or IP address of the SIP entity interface used for SIP signaling for the **FQDN or IP Address**. This example uses **10.89.26.4**.
    c. Select **Session Manager** (for Session Manager), **CM** (for Communication Manager), and **Other** (for the Avaya SBCe) for the **Type**.
    d. Select **Lab126-plano** (a location previously defined) for the **Location**.
    e. Select the time zone for the location in the previous step for the **Time Zone**.
    f. Scroll to the **Port** section of the **SIP Entity Details** screen to define the ports used by Communication Manager. Click **Add** and enter the following values:
        i. Enter the port number on which the CM listens for SIP requests for the **Port**. This example uses **5060**.
        ii. Select the protocol used to send SIP requests for the **Protocol**. This example uses **TCP**.
        iii. Use the default values for all remaining fields.

To add a SIP entity for the Avaya SM:

1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: SIP Entity - SM Configuration




3. In the **General** section, do the following (for this example):
   a. Enter **Lab126-SM7** for the **Name** (for a SIP entity of Avaya SM) .
   b. Enter **10.89.26.7** for the **FQDN or IP Address**.
   c. Select **Session Manager** for the **Type**.
   d. Enter **Lab126 SM** for the **Notes**.
   e. Select **DomainAdapter** for the **Adaptation**.
   f. Select **Lab126-plano** for the **Location**.
   g. Select **America/Chicago** for the **Time Zone**.

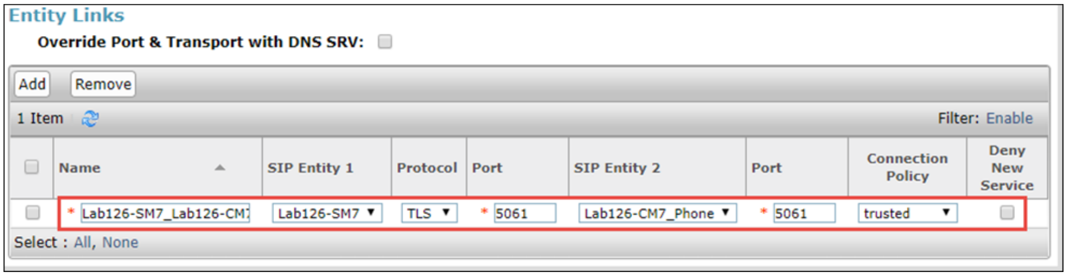
## Entity Links

A SIP trunk between Avaya Session Manager and a telephony system is an entity link. This example creates an entity link.

To add Avaya CM as an entity link:

1. Click **Routing** > **Entity Links**.
2. Click **New**.

   **Avaya Aura SM: Avaya CM Entity Link Configuration**

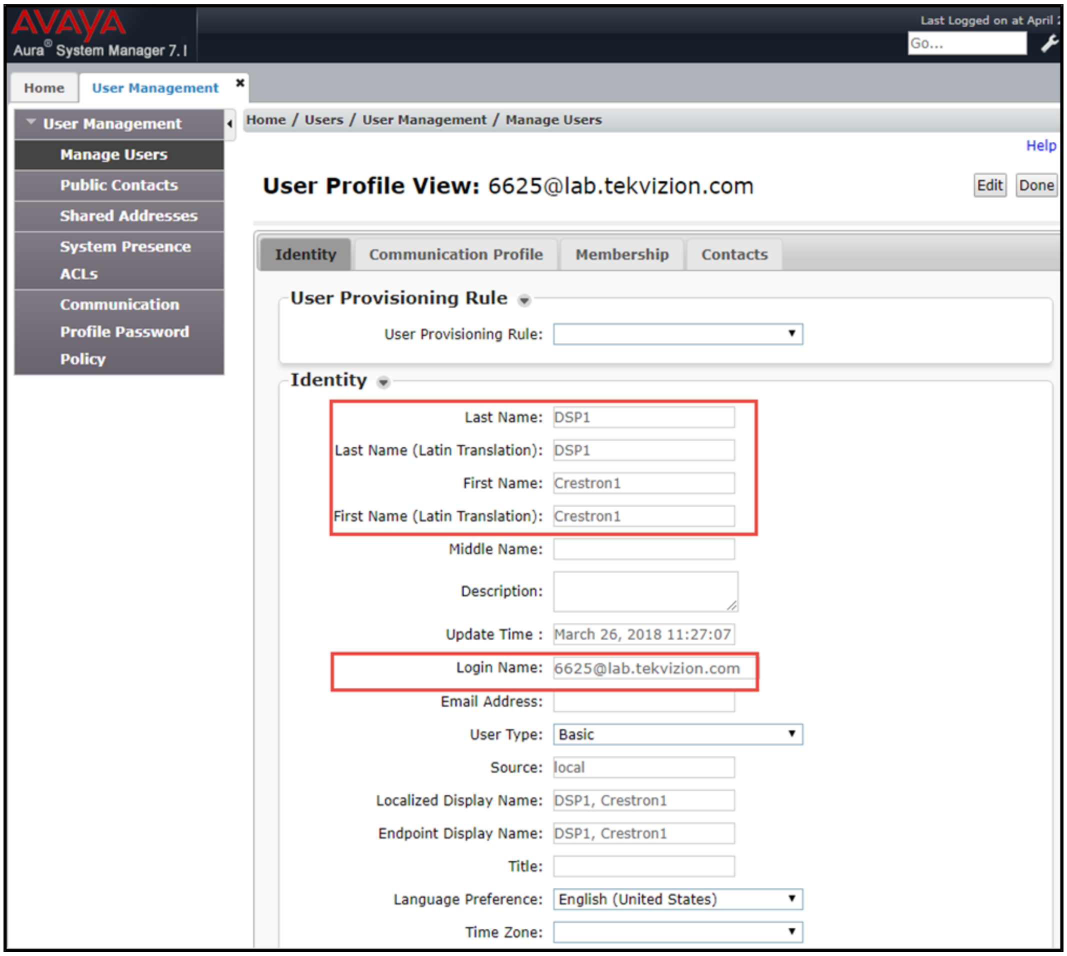
3. In the new row that is displayed, do the following:
   a. Enter a descriptive name for the **Name**.
   b. Select the Session Manager for **SIP Entity 1**.
   c. Select **TLS** (for the **Protocol** for this example).
   d. Enter **5061** for the **Port** (for this example).
   e. Select the Communication Manager for **SIP Entity 2**.
   f. Enter **5061** for the **Port** (for this example).
   g. Select **trusted** for the **Connection Policy**.
4. Click **Commit** to save.

## User Configuration for Each Device/Phone

Configure a user for each phone and Crestron device:

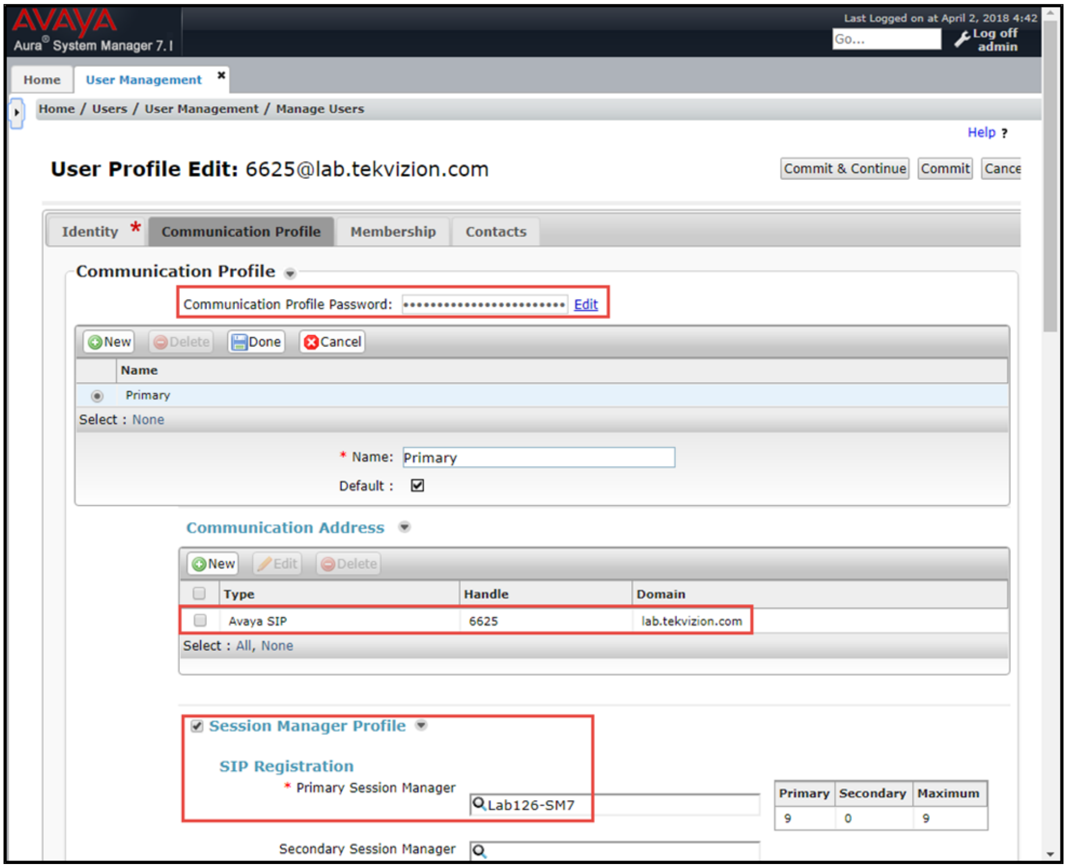
1. Click **Home** > **User Management** > **Manage Users**.
2. Click **Add New**. The **User Profile View** window appears.

**Avaya Aura SM: User Configuration (1/3)**



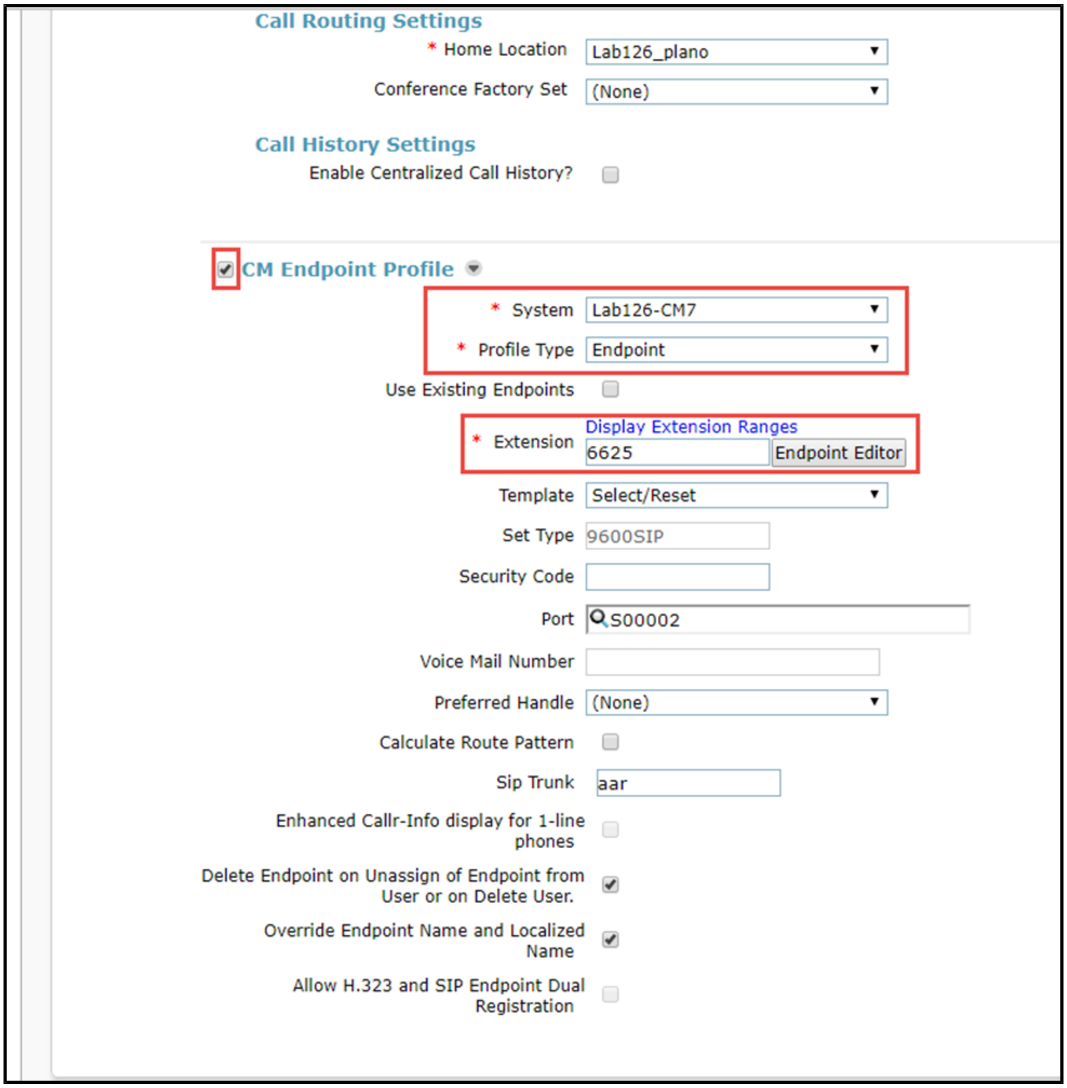

3. Enter **DSP1** for the **Last Name** (for this example).
4. Enter **Crestron1** for the **First Name** (for this example).
5. Enter **6625@lab.tekvizion.com** for the **Login Name** (for this example).
6. Click the **Communication Profile** tab.

Avaya Aura SM: User Configuration (2/3)




7. Enter the desired SIP user registration password for the **Communication Profile Password**.
8. Confirm the password.
9. Scroll down to the **Communication Address** subsection and click **New** to add a new address.
10. Enter **Avaya SIP** for the **Type**.
11. Under **SIP Registration**, enter **Lab126-SM7** for the **Primary Session Manager**.

Avaya Aura SM: User Configuration (3/3)




12. Check **CM Endpoint Profile**.
13. Select **Lab126-CM7** for the **System** (for this example).
14. Select **Endpoint** for the **Profile Type** (for this example).
15. Enter **6625** for the **Extension** (for this example).
16. Click **Done**.

**Crestron Electronics, Inc.**
15 Volvo Drive, Rockleigh, NJ 07647
Tel: 888.CRESTRON
Fax: 201.767.7656
www.crestron.com

**Configuration Guide — 8340B**
**2052160**
**10.18**
Specifications subject to
change without notice.