DSP-1282 & DSP-1283 Crestron Avia™ DSP with Avaya Aura® 7.0 Platform

Configuration Guide

Crestron Electronics, Inc.

Contents

DSP-1282 & DSP-1283: SIP Endpoint with Avaya Aura® 7.0 Platform

Introduction

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

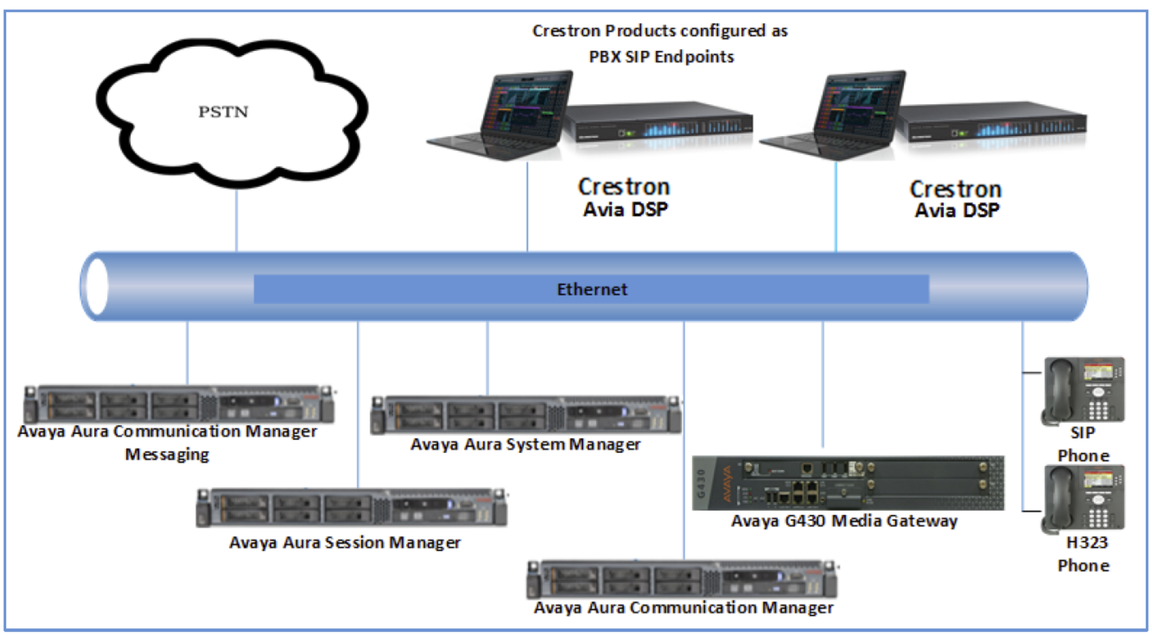
Audience

The intended audience includes those attempting to configure and use Crestron Avia DSP devices as SIP endpoints registered to Avaya Aura Communication Manager 7.0.

Topology

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

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 H323 and SIP phones
- Avaya® G430 Media Gateway
- Avaya Aura Communication Manager Messaging as the voice mail system
- Crestron Avia DSP as the SIP endpoints

Software Requirements

- Avaya Aura Communication Manager v7.0.1.1.0.441.23169
- Avaya Aura Communication Manager Messaging v7.0-28.0
- Avaya Aura System Manager v7.0
- Avaya Aura Session Manager v7.0.1.1.701114
- Avaya g430 Media Gateway v37 .39 .0 /2
- Crestron Avia DSP: v1.00.121

Hardware Requirements

- Avaya Components either in a virtual environment or with separate hardware servers:
    - Avaya Aura Communication Manager
    - Avaya Aura Session Manager
    - Avaya G430 Media Gateway
    - Avaya Aura Communication Manager Messaging
    - Avaya Aura Session Manager
- Public Switched Telephone Network (PSTN) gateway
- Avaya Phones (3) in SIP and H323 mode
- 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 as basic SIP endpoints, since they support a single line/extension. 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

- Member of shared line configuration
- Voice mail access and interaction
- DND (Do Not Disturb)

Unsupported features include:

- 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
- Message Waiting Indicator (MWI)

Known issues and limitations include:

- No support for caller ID on the Crestron Avia DSP. (This issue was tracked via Bugzilla™ software defect: 115708.
- No support for MWI on the Crestron Avia DSP. (this issue was tracked via Bugzilla defect: 118991.
- The DSP does not support Music on Hold when integrated with the Avaya Aura PBX. This issue was tracked via Bugzilla defect: 116049.
- The DSP fails to play a reorder tone when a call from the DSP to a PBX extension times out after the called party does not answer. This issue was tracked via Bugzilla defect: 120378.

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 Avaya Aura Communications Manager (CM)

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
- Connect VOIP 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

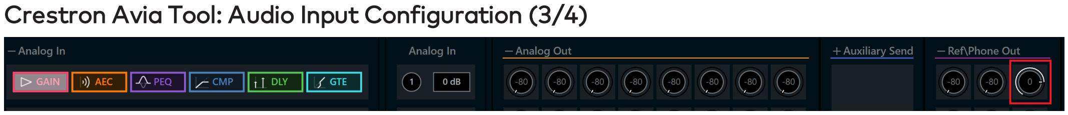
Use the Crestron Avia tool to select and configure the DSP device.

Input Configuration

To configure the analog input:

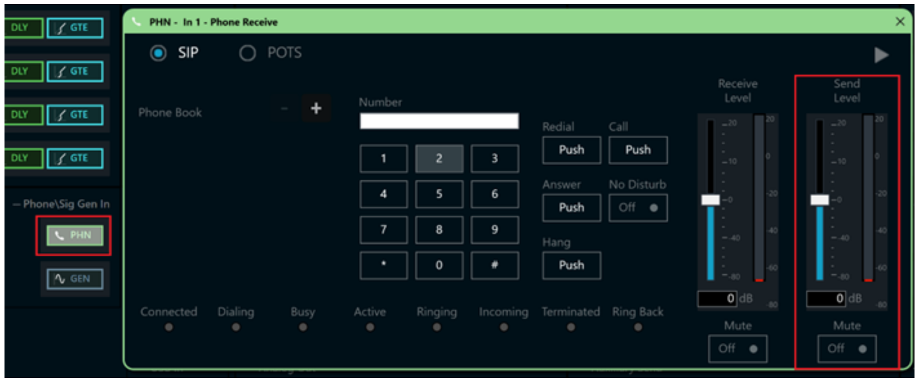
1. Click **Signal**.
   Crestron Avia tool: Audio Input Configuration (1/4)
   

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)

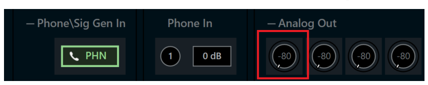


Output Configuration

To configure the analog output:

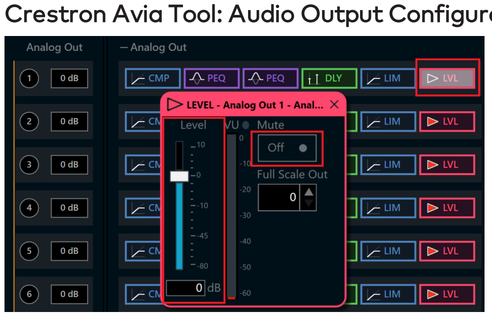
1. Under **Phone In 1** (first row), click **Analog Out** (left-most column) and enter **0** as the decibel value.

    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 **2101**.
3. Enter the Avaya Aura Session Manager for the **SIP Server IP Address**. This example uses **10.89.26.7**.
4. Enter the SIP server port (**5060**) 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**.

Avaya Aura Communication Manager Configuration

This section describes the Avaya Aura Communication Manager (Avaya CM) configuration necessary to support registration of Crestron devices and connectivity to the Public Switched Telephone Network (PSTN).

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

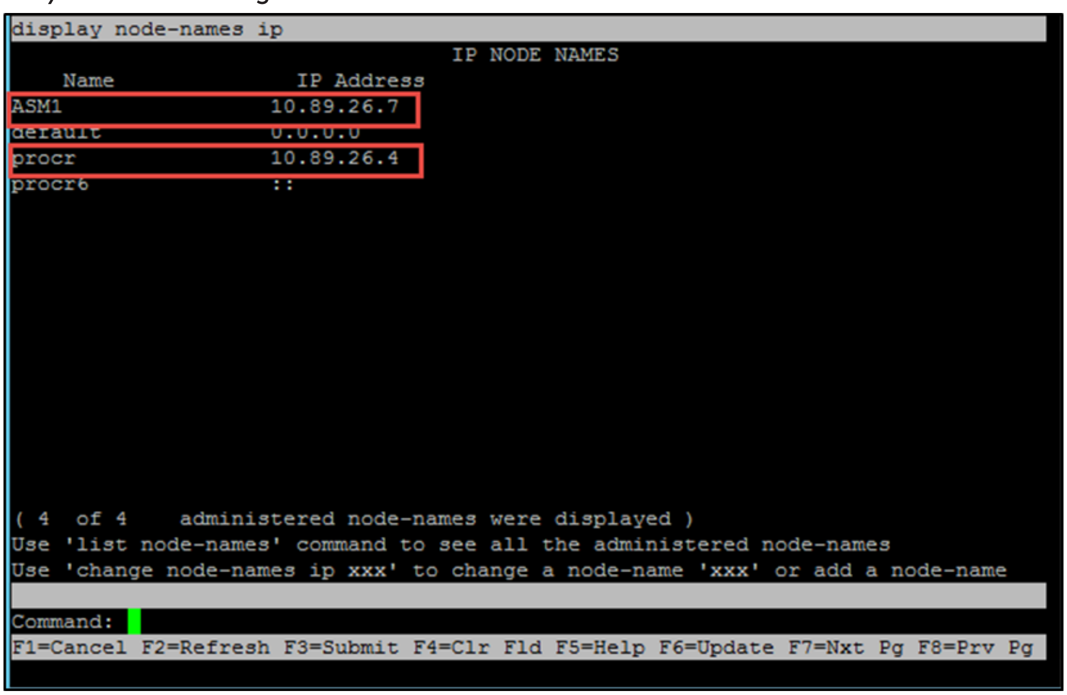
Node Names

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 H323 phones and SIP trunk.

Avaya Aura CM: Configure Node



Media Gateway

Add the G430 media gateway for DSP resource utilization in Avaya CM.

The G430 provides VoIP services over the LAN and Wide Area Network (WAN). The G430 has an on-board VoIP DSP providing 20 VoIP channels, and optionally supports an additional DSP board providing 10, 20, or 80 VoIP channels.

Avaya Aura CM: Media Gateway Configuration (1/3)

```
list media-gateway

                                MEDIA-GATEWAY REPORT

Num Name                     Serial No/      IPV4 Address/      Type     NetRgn    Reg?
                            FW Ver/HW Vint/ IPV6 Address/
                            RecRule         Cntrl IP Addr

1    G430                    160L17035780    10.89.26.14        g430       1         y
                            37 .39 .0  /2
                            none            10.89.26.4
```

Use the **add media-gateway** command to configure the media gateway.

Avaya Aura CM: Media Gateway Configuration (2/3)

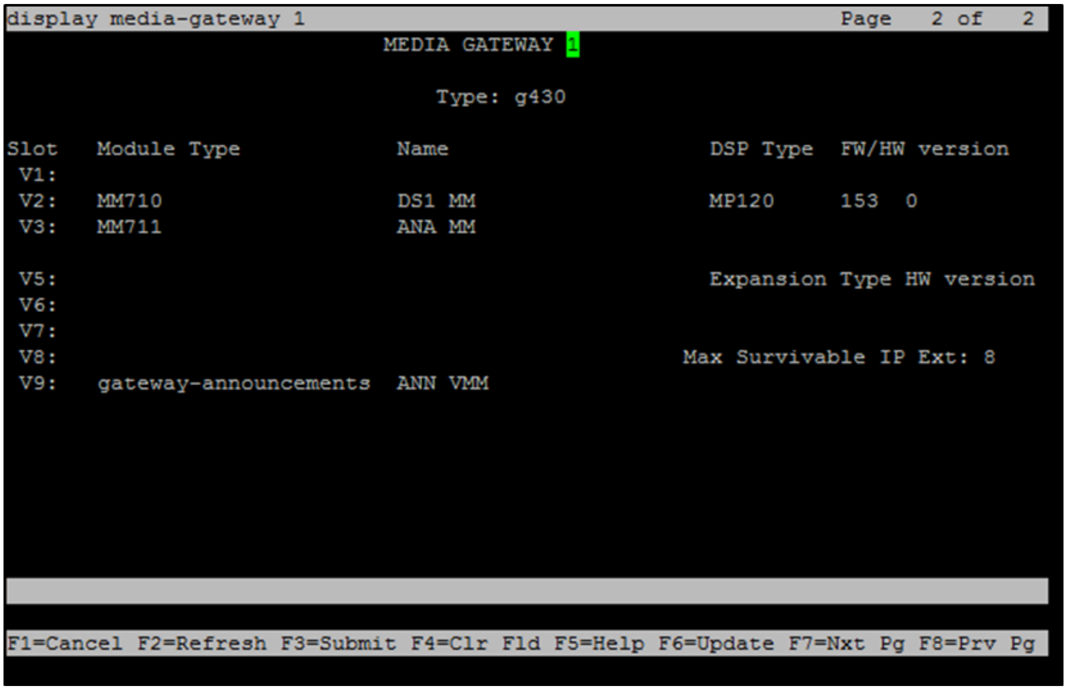
```
display media-gateway 1                                                Page   1 of   2
                                MEDIA GATEWAY 1

                         Type: g430
                         Name: G430
                    Serial No: 160L17035780
         Link Encryption Type: any-ptls/tls        Enable CF? n
               Network Region: 1                    Location: 1
                                                   Site Data:
                Recovery Rule: none

                  Registered?  y
        FW Version/HW Vintage: 37 .39 .0  /2
            MGP IPV4 Address: 10.89.26.14
            MGP IPV6 Address:
        Controller IP Address: 10.89.26.4
                 MAC Address: a4:25:1b:a7:b5:91

        Mutual Authentication? optional



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

To configure the media gateway for this example:

1. Enter **g430** for the **Type**.
2. Enter **G430** for the **Name**.
3. Enter **14TG41631501** for the **Serial No**.
4. Enter **Y** for **Registered**.

5. Enter **10.89.26.14** for the **MGP IPV4 Address**.
6. Enter **10.89.26.4** (the procr IP) for the **Controller IP Address**.

Avaya Aura CM: Media Gateway Configuration (3/3)



```
display media-gateway 1                                          Page   2 of   2
                             MEDIA GATEWAY 1

                                 Type: g430

Slot   Module Type             Name                    DSP Type  FW/HW version
 V1:
 V2:   MM710                   DS1 MM                  MP120     153  0
 V3:   MM711                   ANA MM

 V5:                                                   Expansion Type HW version
 V6:
 V7:
 V8:                                                 Max Survivable IP Ext: 8
 V9:   gateway-announcements   ANN VMM



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

Dial Plan Analysis

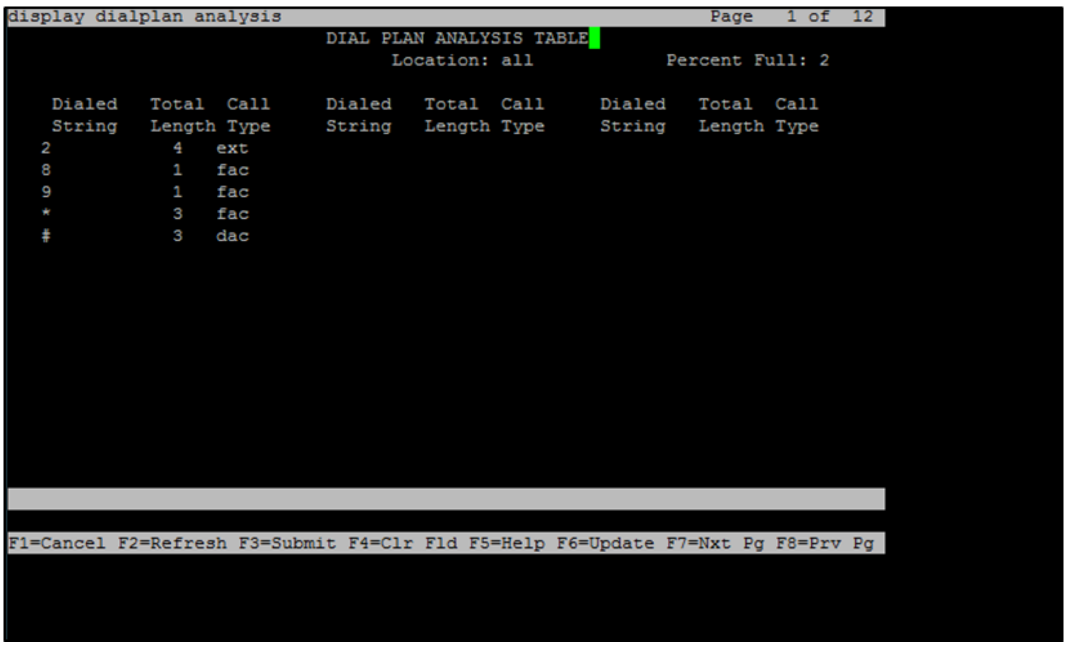
Configure several dial strings to ensure complete test coverage. This example includes calling between stations, calling to PSTN, and accessing PBX features.

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

1. Enter **2** for the station number **Dialed string**.
2. Enter **8** for the feature access code **Dialed string**.
3. Enter **9** for the feature access code **Dialed string**.
4. Enter ***** for the feature access code **Dialed string**.
5. 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: 2

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   2            4    ext
   8            1    fac
   9            1    fac
   *            3    fac
   #            3    dac

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

Uniform Dial Plan

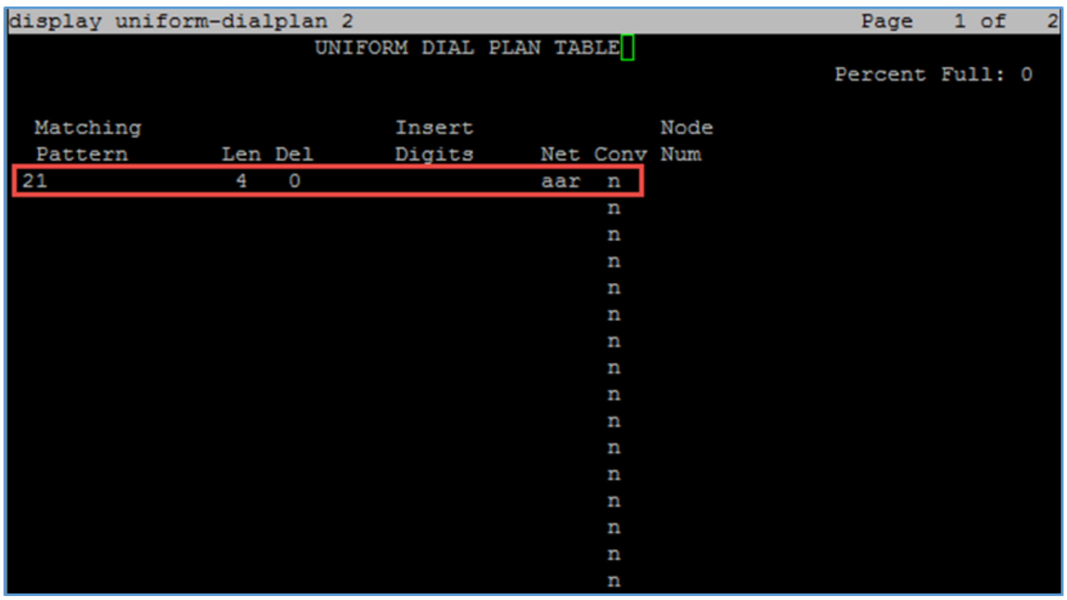
Configure a dial plan using the **change uniform-dialplan n** command (where **n** represents the first digit of the extension numbers used for SIP stations in the system).

NOTE: This example uses a 4-digit extension starting with "21xx" for extensions associated with the Avaya SIP phones and Crestron SIP devices.

Issue the **change uniform-dialplan n** command and configure the dial plan (for this example):

1. Enter **21** (the starting digits for the extension number used for the SIP) for the **Matching Pattern**.
2. Enter **4** for the **Len**.
3. Enter **0** for the **Del**.
4. Enter **aar** for the **Net**.

Avaya Aura CM: Uniform Dial Plan Configuration



Inbound Routing

DID numbers received from PSTN are mapped to extensions using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt** command to create an entry for each DID number.

Avaya Aura CM: Inbound Routing

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number       Del Insert
 Feature         Len    Digits
 tie             10  9722657277      10  2102
 tie             10  9722657278      10  2103
 tie             __  _____________   __  _____________
 tie             __  _____________   __  _____________
 tie             __  _____________   __  _____________
```

Outbound Routing

Automatic Route Selection (ARS)

Use the Automatic Route Selection (ARS) feature to route outbound calls via the SIP trunk to PSTN. This example uses the single digit **9** as the ARS access code. PBX users initially dial "9" to initiate a call to PSTN. Use the **change dialplan analysis** command to define a dialed string (shown after this paragraph) beginning with **9** and length **1** as a feature access code (**fac**).

Avaya Aura CM: Outbound Routing - Configure Dial Plan Analysis Table

```
change 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       [redacted]                 _________  __  ______
   [redacted]                                             _________  __  ______
                               9            1    fac      _________  __  ______
                               [redacted]                 _________  __  ______
```

Configure the following feature access codes (for this example):

1. Enter **8** for the **Auto Alternate Routing (AAR) Access Code**.
2. Enter **9** for the **Auto Route Selection (ARS) - Access Code 1**.

Avaya Aura CM: Outbound Routing - Configure Feature Access Codes

```
display feature-access-codes                                    Page   1 of  10
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code: *11
                         Attendant Access Code:
   Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9       Access Code 2:
             Automatic Callback Activation:              Deactivation:
Call Forwarding Activation Busy/DA: *14     All: *10     Deactivation: *15
   Call Forwarding Enhanced Status:         Act: *16     Deactivation: *17
                         Call Park Access Code: *01
                       Call Pickup Access Code: *02
CAS Remote Hold/Answer Hold-Unhold Access Code:
                    CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:          Deactivation:
                   Contact Closure    Open Code:           Close Code:

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

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. This example adds an entry for the number beginning with **214242**.

Avaya Aura CM: Outbound Routing - Auto Route Selection

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

          Dialed            Total     Route     Call    Node   ANI
          String          Min  Max   Pattern    Type    Num    Reqd
    2                      7    7     2         hnpa           n
    214242                10   10     1         natl           n
    3                      7    7     2         hnpa           n
    4                      7    7     2         hnpa           n
```

Configure the following auto route selection (for this example):

1. Enter **214242** for the **Dialed string**g (to call PSTN numbers).
2. Enter **10** for the **Min**.
3. Enter **10** for the **Max**.
4. Enter **1** for the **Route Pattern**.
5. Enter **natl** for the **Call Type**.

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).

Avaya Aura CM: PSTN Route Pattern Configuration

```
display route-pattern 1                                             Page   1 of   3
                    Pattern Number: 1      Pattern Name:  Trunk Group 1
    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: 1     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                               lev0-pvt  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
```

Configure a public numbering plan for calling PSTN via the Avaya Aura Session Manager:

1. Enter **1** for the **Route Pattern**.
2. Enter **1** for the **Grp No** (for this example).

Avaya Aura CM: Voice Mail Route Pattern Configuration

```
display route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2      Pattern Name: Trunk Group 2
    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: 2     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                               lev0-pvt  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
```

Configure a private numbering plan for the voice mail feature offered by the Avaya Communication Manager Messaging via the Avaya Aura Session Manager:

1. Enter **2** for the **Route Pattern**.
2. Enter **2** for the **Grp No** (for this example).

Auto Alternative Routing

Use the **change aar analysis n** command (where **n** represents the first digit of the extension numbers used for SIP stations in the system.

Avaya Aura CM: Modify AAR Digit Analysis Table

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

        Dialed              Total     Route    Call    Node   ANI
        String             Min  Max  Pattern   Type    Num    Reqd
   21                       4    4    2        aar             n
   2222                     4    4    2        aar             n
   28                       4    4    2        aar             n
   3                        7    7    254      aar             n
   4                        7    7    254      aar             n
   5                        7    7    254      aar             n
   6                        7    7    254      aar             n
   7                        7    7    254      aar             n
   8                        7    7    254      aar             n
   9                        7    7    254      aar             n
```

Configure the following (for this example):

1. Enter **21** for the **Dialed number** (used for Avaya SIP phones and Crestron DSP SIP devices).
2. Enter **2222** for the **Dialed number** (used for voice mail access).

Trunk Groups

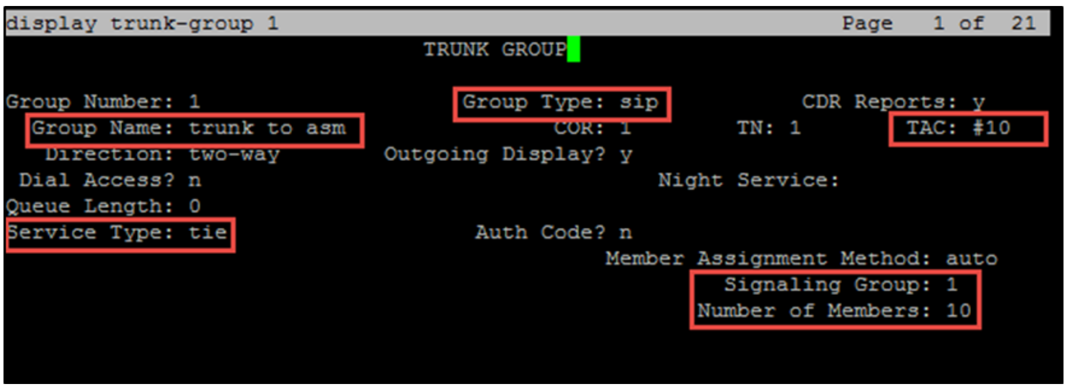
Configure two trunk groups (in this example):

- Trunk Group 1
  This group utilizes a public numbering plan to access the stations registered to the Avaya Session Manager.
- Trunk Group 2
  This group utilizes a private numbering plan to send a 4-digit calling number to Avaya Communication Manager Messaging or voice mail access.

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

Trunk Group 1

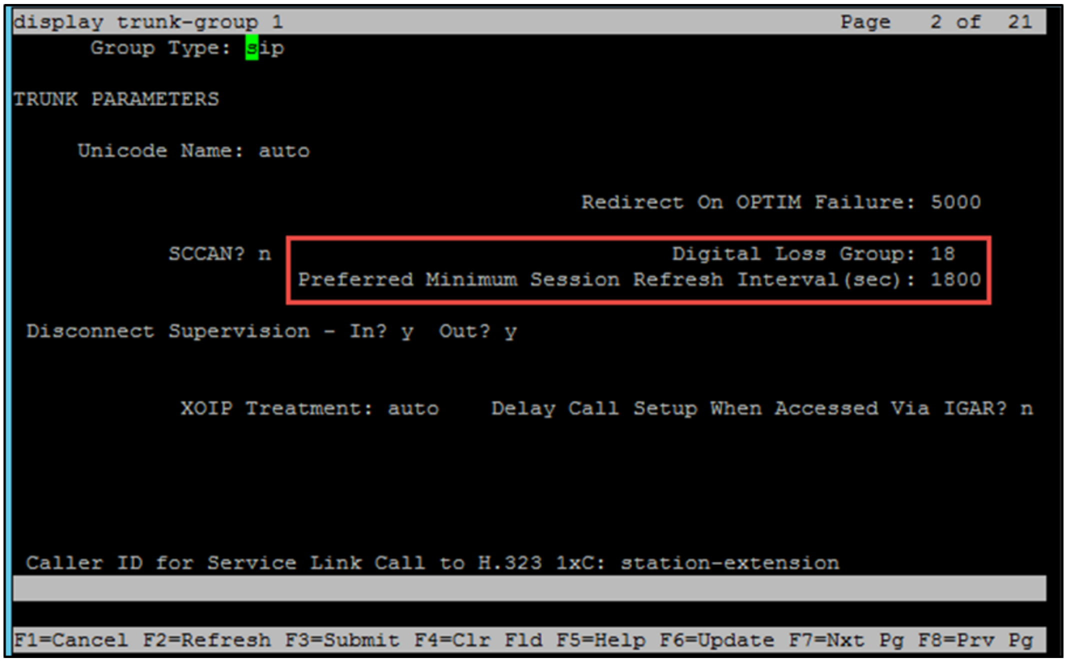
Avaya Aura CM: Trunk Configuration to Session Manager - Trunk Group 1 (1/4)



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

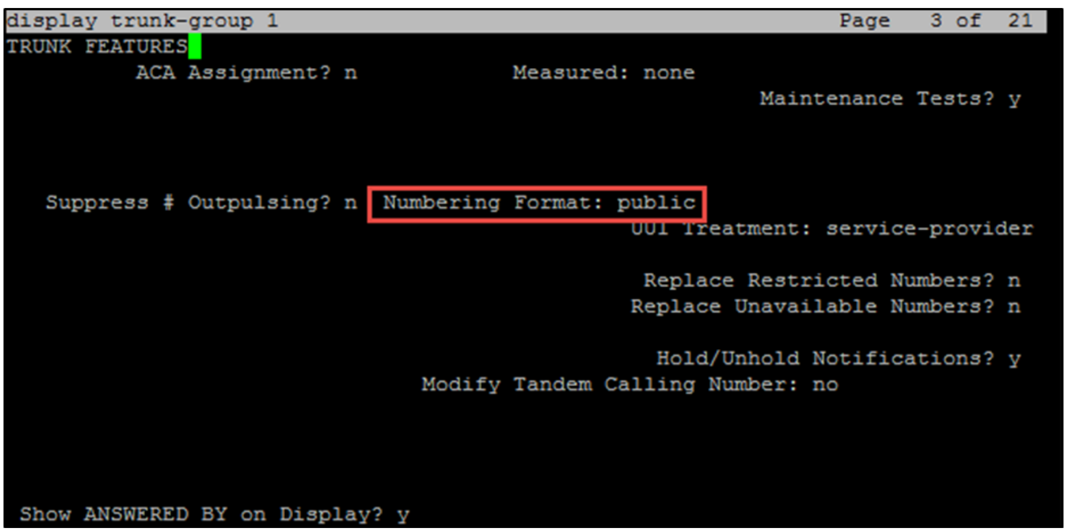
1. Enter **1** for the **Group Number**.
2. Enter **trunk to asm** for the **Group Name**.
3. Enter **sip** for the **Group Type**.
4. Enter **tie** for the **Service Type**.
5. Enter **#10** for the **TAC**.
6. Enter **1** for the **Signaling Group**.
7. Enter **10** for the **Number of Members**.

Avaya Aura CM: Trunk Configuration to Session Manager - Trunk Group 1 (2/4)



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

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



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

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

```
display trunk-group 1                                           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:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: From
            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 2

Avaya Aura CM: Trunk Configuration to Session Manager - Trunk Group 2 (1/4)

Configure Trunk Group 2 (entries for steps 1 through 7 are for this example):

1. Enter **2** for the **Group Number**.
2. Enter **CM Messaging** for the **Group Name**.
3. Enter **sip** for the **Group Type**.
4. Enter **tie** for the **Service Type**.
5. Enter **#002** for the **TAC**.
6. Enter **1** for the **Signaling Group**.
7. Enter **5** for the **Number of Members**.

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

```
display trunk-group 2                                           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 Configuration to Session Manager - Trunk Group 2 (3/4)

```
display trunk-group 2                                           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 Configuration to Session Manager - Trunk Group 2 (4/4)

```
display trunk-group 2                                           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:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: From
            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
```

Signaling Group

Use the add signaling group command to create a signaling group between Communication Manager and Session Manager for use by the PSTN SIP trunk. Use the signaling group for inbound and outbound calls between the PBX and PSTN.

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

Avaya Aura CM: Signaling Group Configuration

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

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tcp
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  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: ASM1
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          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? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             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 a signaling group (for this example):

1. Enter **1** for the **Group Number**.
2. Enter **sip** for the **Group Type**.
3. Enter **tcm** for the **Transport Method**.
4. Enter **SM** for the **Peer Server**.
5. Enter **procr** for the **Near-end Node Name**.
6. Enter **5060** for the **Near-end Listen Port**.
7. Enter **ASM1** for the **Far-end Node Name**.
8. Enter **5060** 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**.

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 CODEC SET

    Codec Set: 1

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

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.

Hunt Group

Configure two hunt groups (for this example):

- Hunt group extension 2200 used for the group hunt feature
- Hunt group extension 2222 used for the voice mail feature

Avaya Aura CM: Hunt Group Configuration (1/2)

```
list hunt-group

                               HUNT GROUPS

Grp  Grp
No.  Name/           Grp      ACD/                No. Cov  Notif/  Dom   Message
     Ext             Type     MEAS Vec MCH   Que Mem Path Ctg Adj Ctrl  Center

1    Crestron HG
     2200            circ     n/-  n   none n   0   1    n              n
2    cmm_hunt
     2222            ucd-mia  n/-  n   none n   0        n              S
```

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

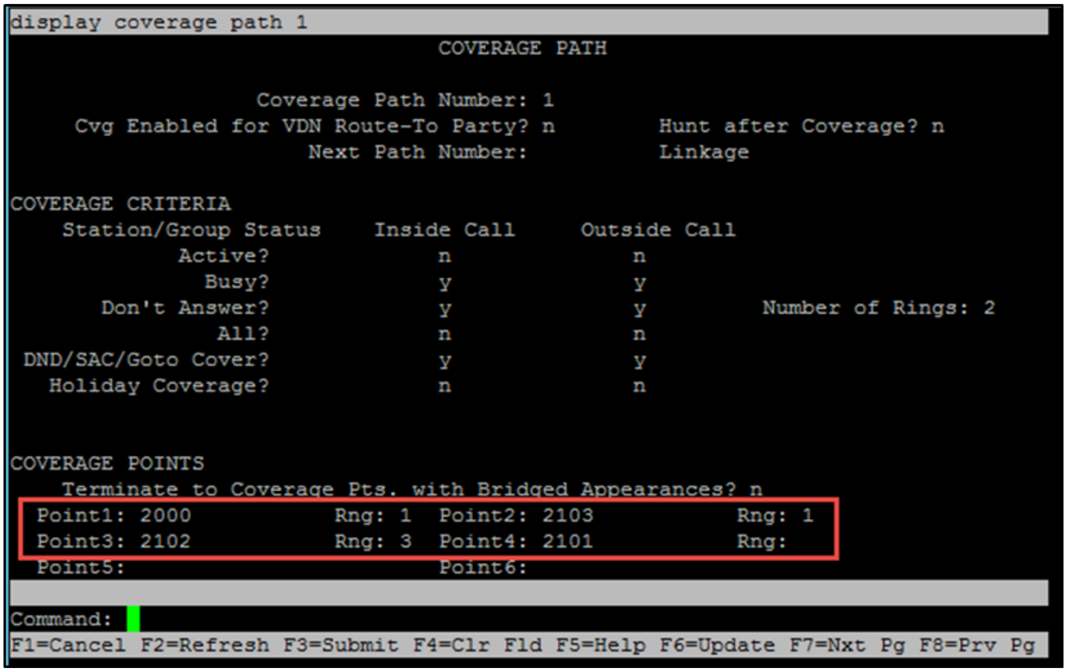
Avaya Aura CM: Hunt Group Configuration (2/2)

Configure the hunt group (for this example):

1. Enter **1** for the **Group Number**.
2. Enter **Crestron HG** for the **Group Name**.
3. Enter **2200** for the **Group Extension**.
4. Enter **circ** for the **Group Type** to enable sequential ringing on the hunt group members.
5. Enter **1** for the **Coverage Path** to include sequentially alerted hunt group members.

Use the **add coverage path n** command (where **n** represents the available coverage path number) to add the coverage path, which includes members of the hunt group. Configure **Coverage Path 1** for Hunt Group 1 for this example.

Avaya Aura CM: Hunt Group Coverage Path Configuration

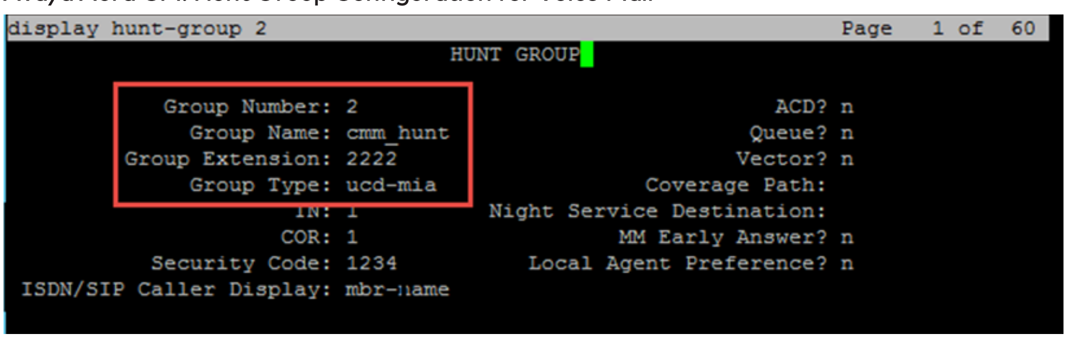


```
display coverage path 1
                                COVERAGE PATH

                  Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n          Hunt after Coverage? n
                      Next Path Number:             Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
        Don't Answer?            y               y        Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: 2000              Rng: 1  Point2: 2103              Rng: 1
  Point3: 2102              Rng: 3  Point4: 2101              Rng:
  Point5:                           Point6:

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

Configure the hunt group coverage path (for this example):

1. Enter **2000** for **Point1**.
2. Enter **2103** for **Point2**.
3. Enter **2102** for **Point3**.
4. Enter **2101** for **Point4**.

Avaya Aura CM: Hunt Group Configuration for Voice Mail



```
display hunt-group 2                                            Page   1 of  60
                                  HUNT GROUP

            Group Number: 2                                 ACD? n
              Group Name: cmm_hunt                        Queue? n
         Group Extension: 2222                           Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1              Night Service Destination:
                     COR: 1                        MM Early Answer? n
           Security Code: 1234              Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

Configure the voice mail hunt pilot (for this example):

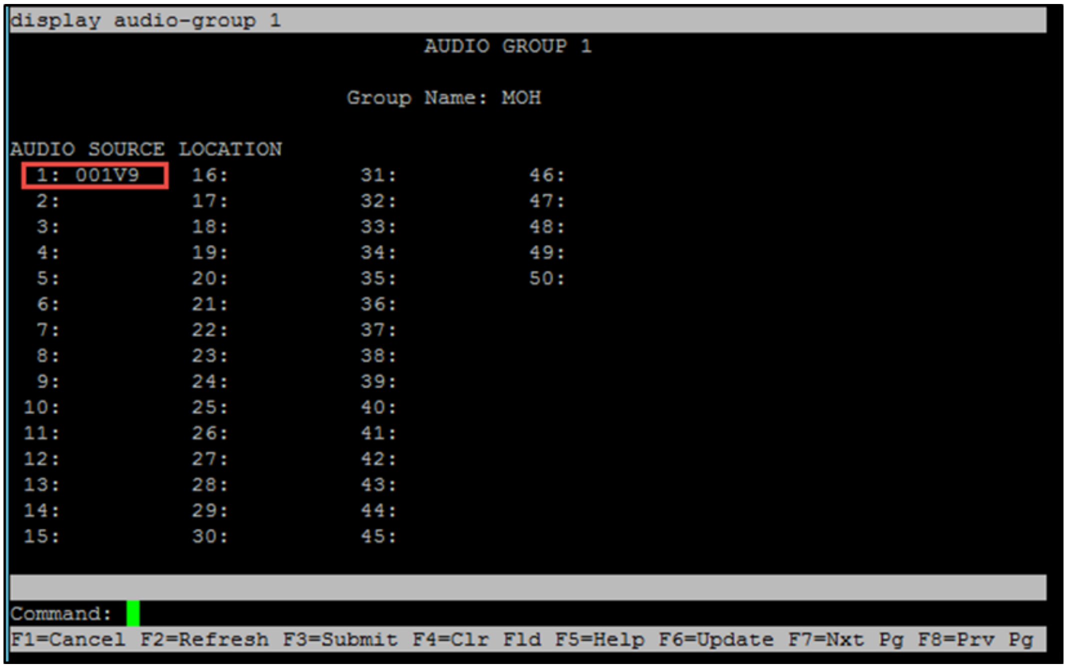
1. Enter **2** for the **Group Number**.
2. Enter **cmm_hunt** for the **Group Name**.
3. Enter **2222** for the **Group Extension**.
4. Enter **ucd_mia** for the **Group Type**.

Music on Hold

Use the following commands to configure the Music on Hold (MoH) source:

- **enable announcement board 1v9**
  Enables music source 1v9.
- **add audio group n**
  Adds the audio source. This example uses **001v9**.

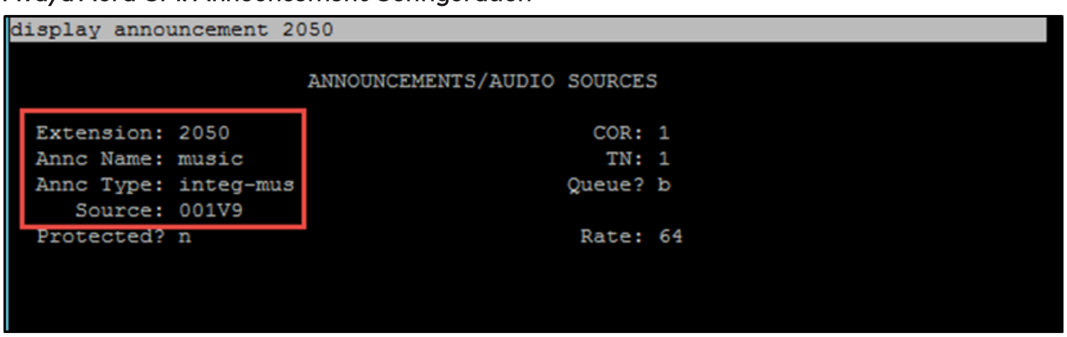
Avaya Aura CM: MoH Configuration



```
display audio-group 1
                                   AUDIO GROUP 1

                             Group Name: MOH

AUDIO SOURCE LOCATION
  1: 001V9      16:          31:          46:
  2:            17:          32:          47:
  3:            18:          33:          48:
  4:            19:          34:          49:
  5:            20:          35:          50:
  6:            21:          36:
  7:            22:          37:
  8:            23:          38:
  9:            24:          39:
 10:            25:          40:
 11:            26:          41:
 12:            27:          42:
 13:            28:          43:
 14:            29:          44:
 15:            30:          45:

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

- **add announcement n**
  Adds the new announcement associated with a station number.

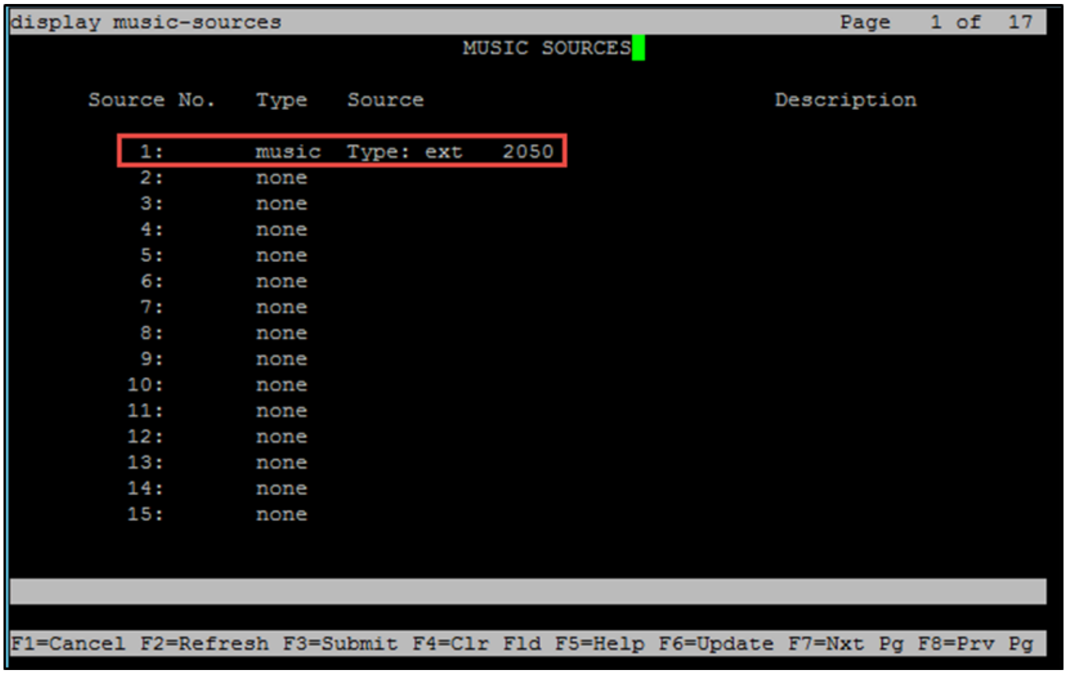
Avaya Aura CM: Announcement Configuration



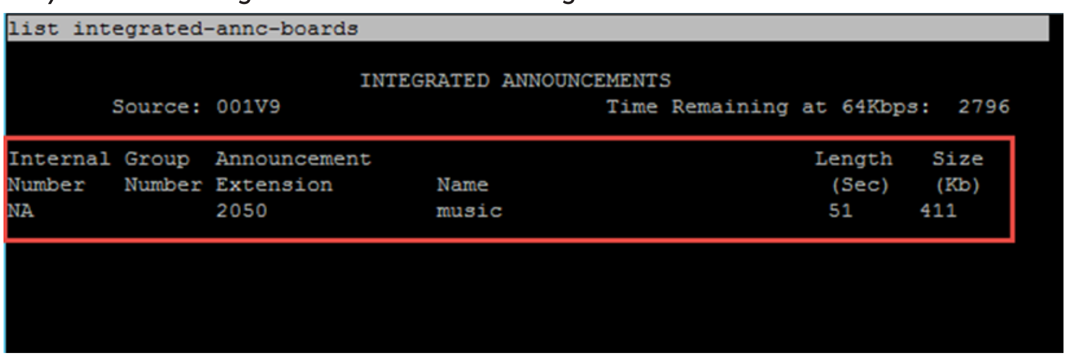
```
display announcement 2050

                       ANNOUNCEMENTS/AUDIO SOURCES

  Extension: 2050                          COR: 1
  Annc Name: music                          TN: 1
  Annc Type: integ-mus                  Queue? b
     Source: 001V9
 Protected? n                             Rate: 64
```

- **display music scores**
  Displays the list of music sources configured on the system. This example uses **1** for the **Source No**, **music** for the **Type**, and **ext 2050** for the **Source**.

  Avaya Aura CM: Music Source Configuration



Use the list integrated-announcement board command to verify the configuration.
Avaya Aura CM: Integrated Announcement Configuration

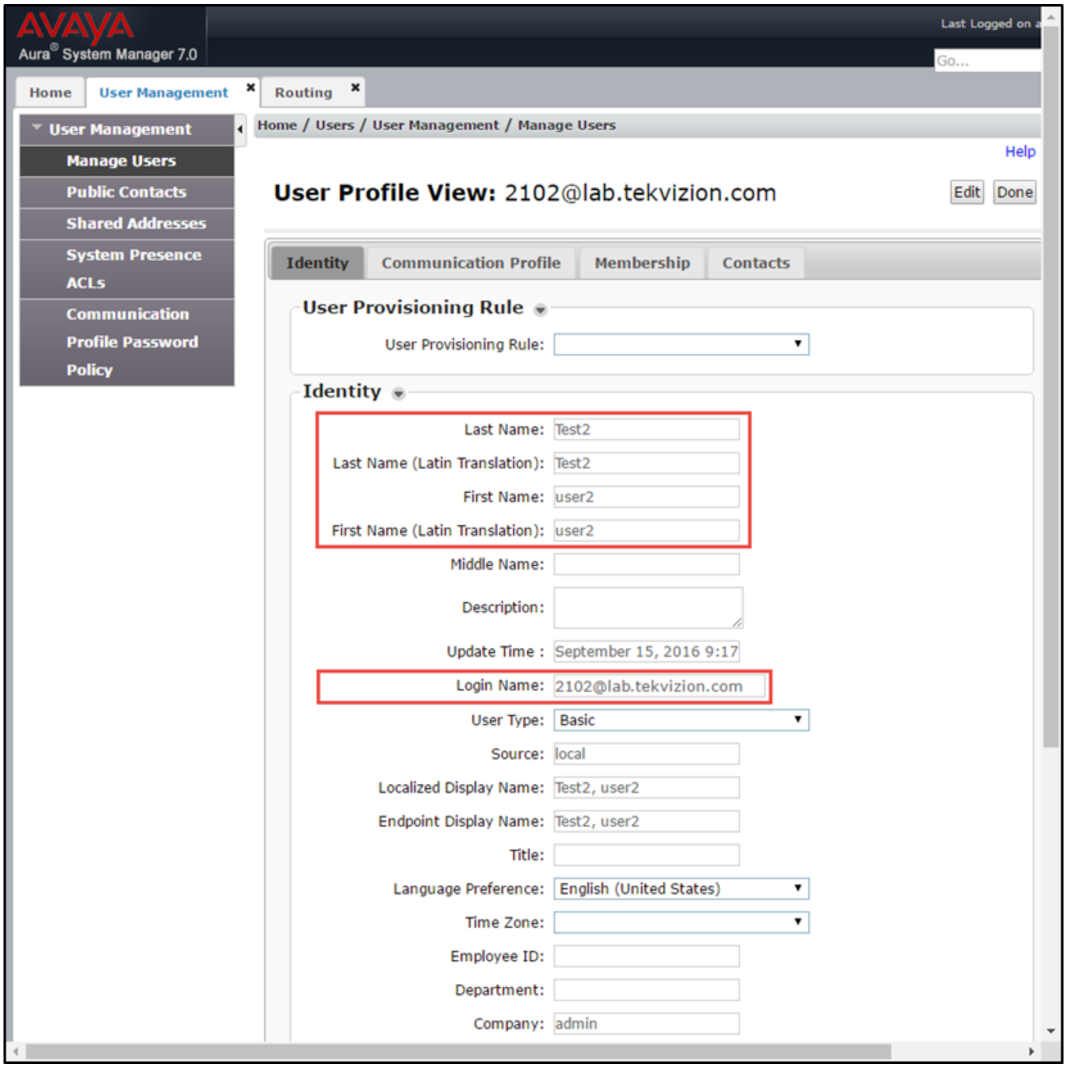


User Configuration for Each Device/Phone

Configure a user for each phone and Crestron device:

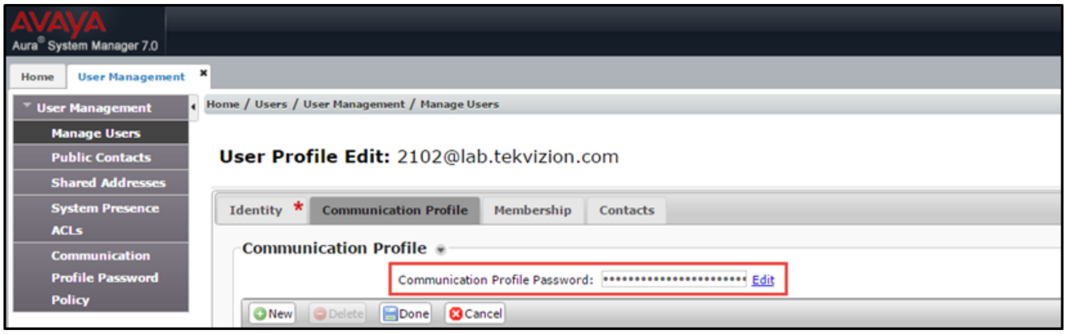
1. Click **Home** > **User Management** > **Manage Users**.
2. Click **Add New**. The **User Profile View** window appears.

Avaya Aura CM: Phone Configuration (1/4)



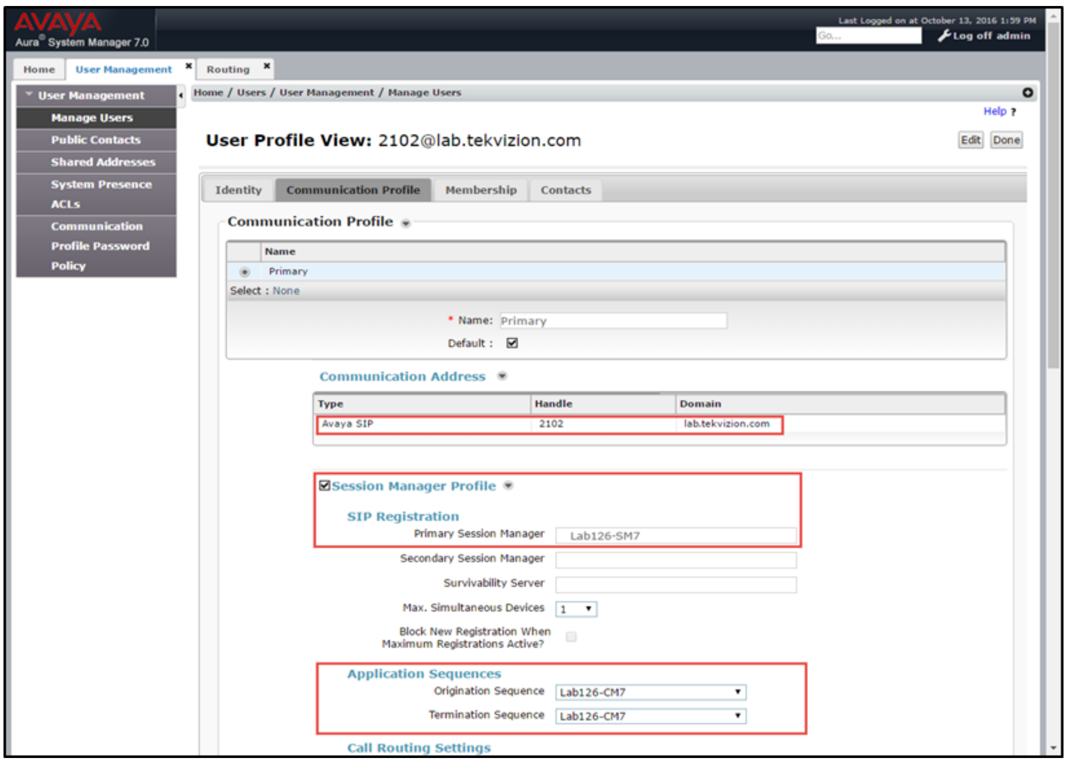
3. Enter **Test2** for the **Last Name** (for this example).
4. Enter **Test2** for the **First Name** (for this example).
5. Enter **2102@lab.tekvizion.com** for the **Login Name** (for this example).
6. Click the **Communication Profile** tab.

Avaya Aura CM: Phone Configuration (2/4)



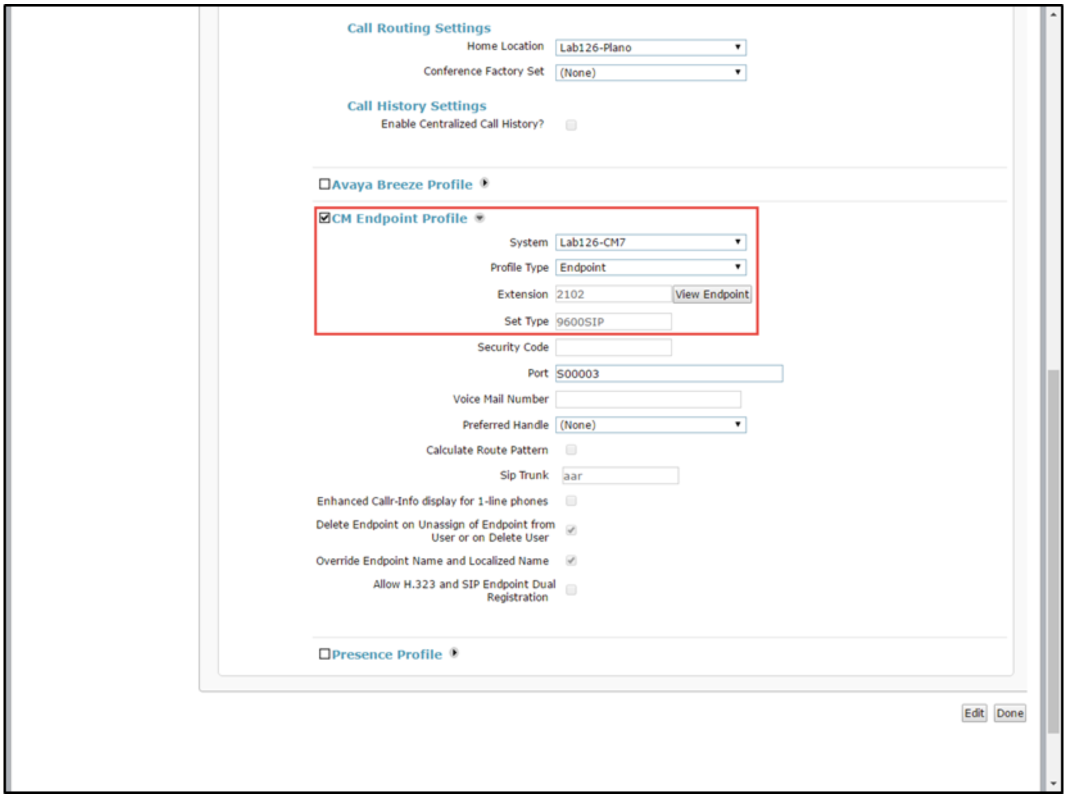
7. Enter the desired SIP user registration password for the **Communication Profile Password**.
8. Confirm the password.
9. Scroll down to the **Communication Address** subsection and click **New** to add a new address.

Avaya Aura CM: Phone Configuration (3/4)



10. Set the Communication Manager **Type** to **Avaya SIP**.
11. Enter **Lab126-SM7** for the **Primary Session Manager**.

Avaya Aura CM: Phone Configuration (4/4)



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 **2102** for the **Extension** (for this example).
16. Click **Done**.

Avaya Aura Session Manager

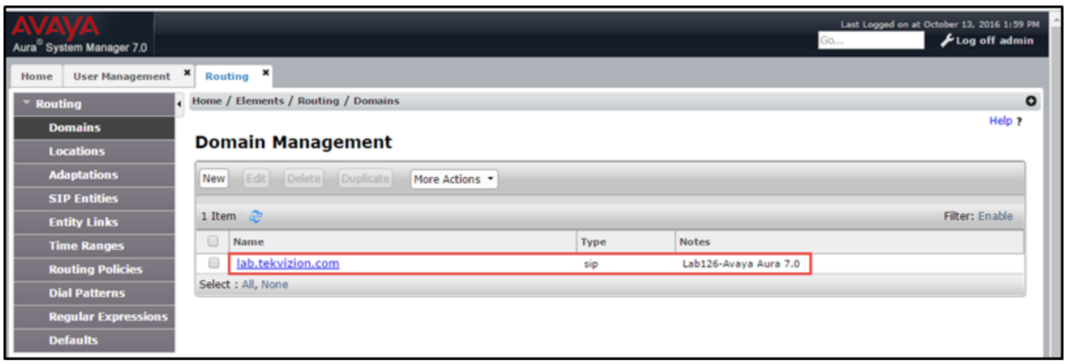
Domain

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).

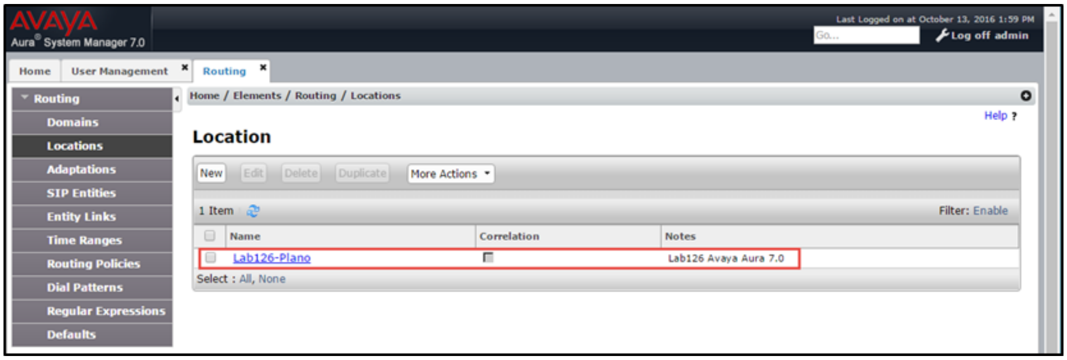
Location

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).

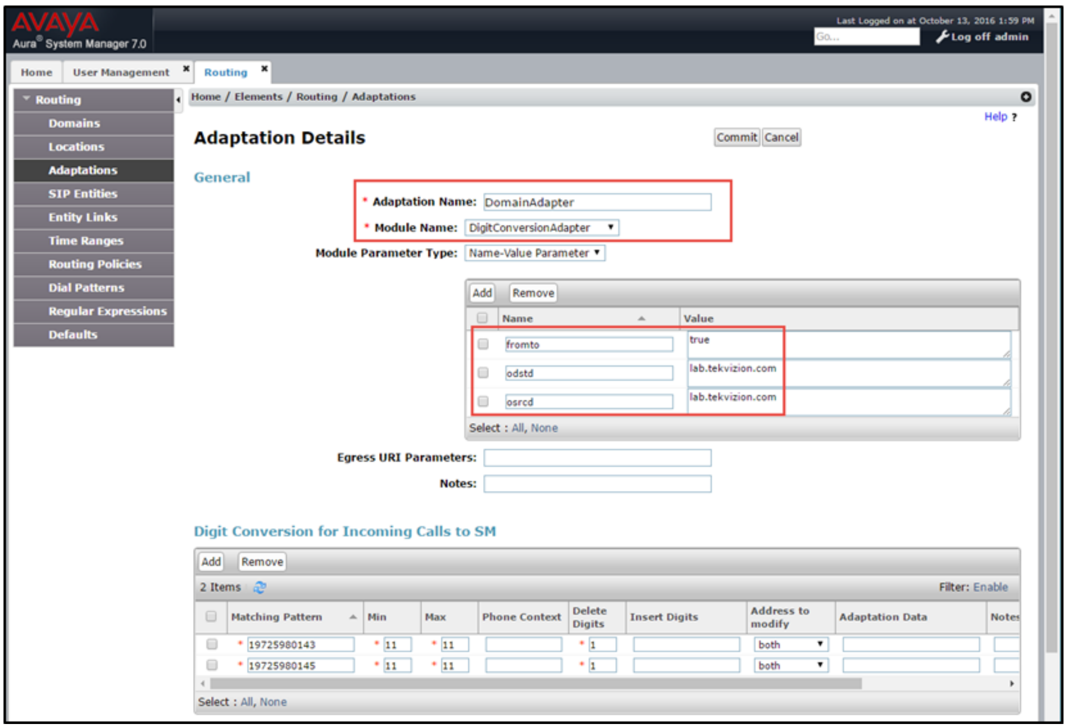
Adaptation

Use adaptations to modify administered SIP messages. A SIP entity can have its own unique adaptation, or multiple entities can share one adaptation. Session Manager includes the DigitalConversionAdapter module, which can convert digit strings in various message headers as well as hostnames in the Request-URI and other headers.

To configure an adaptation:

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

Avaya Aura SM: Adaptation Configuration

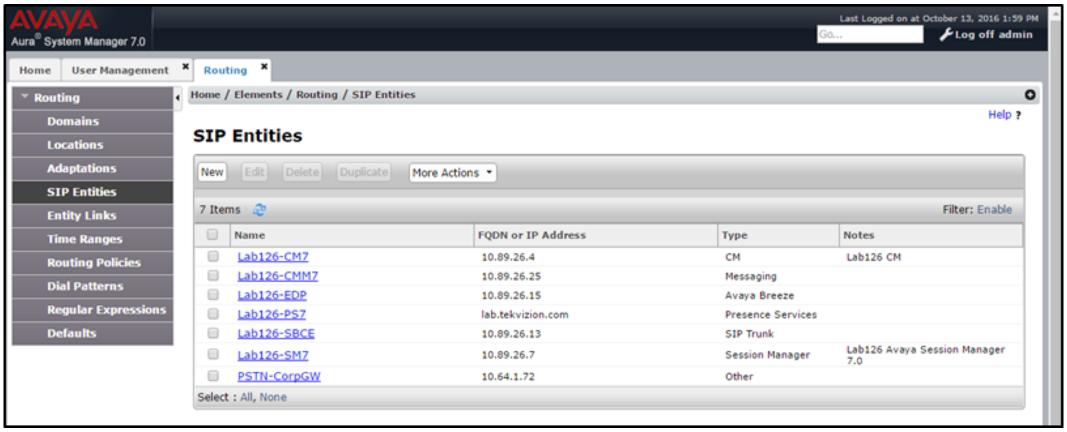


3. Enter **DomainAdapter** for the **Adaptation Name** (for this example).
4. Select **DigitConversionAdapter** for the **Module Name**.
5. Select **Name-Value Parameter** for the **Module Parameter Type**.
6. In the **Name** and **Value** columns, enter the following:
   a. **fromto** : **true**
   b. **odstd** : **lab.tekvizion.com**
   c. **osrcd** : **lab.tekvizion.com**
7. Type a brief description for the **Notes** (optional).
8. Click **Commit** to save.

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.

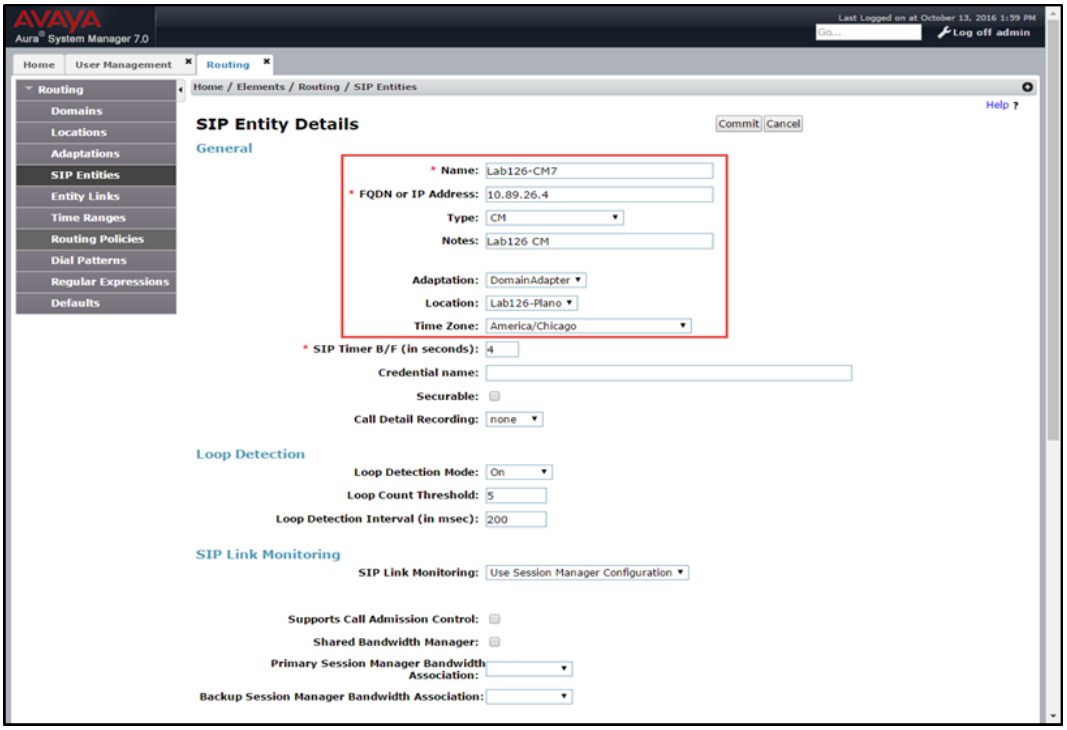
Avaya Aura SM: SIP Entity



To add a SIP entity:

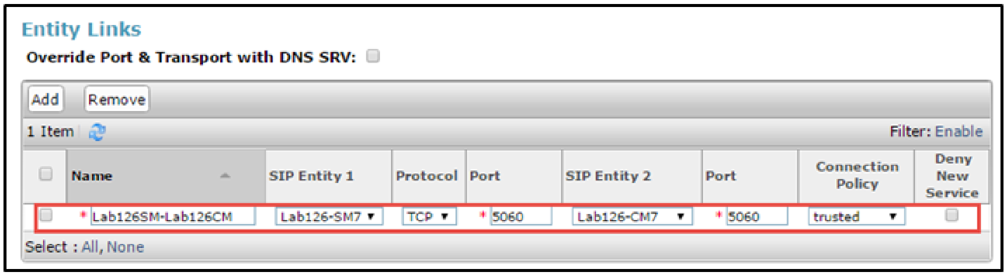
1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: Sip Entity - CM Configuration (1/2)



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 **DomainAdapter** for the **Adaptation** (for this example).
   e. Select **Lab126-Plano** (a location previously defined) for the **Location**.
   f. Select the time zone for the location in the previous step for the **Time Zone**.
   g. Scroll to the **Port** section of the **SIP Entity Details** screen to define the ports used by Communication Manager.

Avaya Aura SM: SIP Entity - CM Configuration (2/2)



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**.

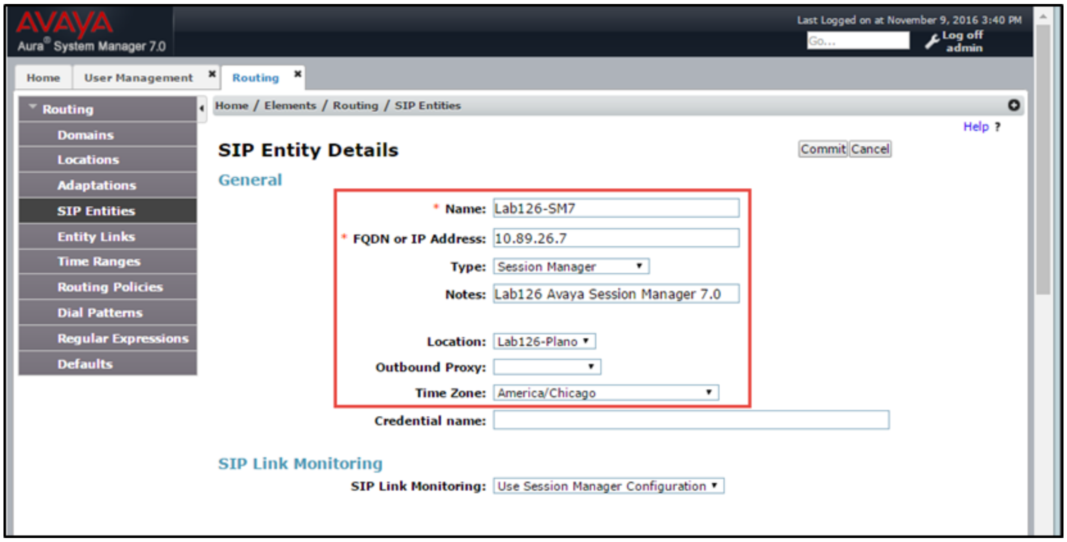
ii. Select the protocol used to send SIP requests for the **Protocol**. This example uses **TCP**.

iii. Use the default values for all remaining fields.

To add a SIP entity for the Avaya SM:

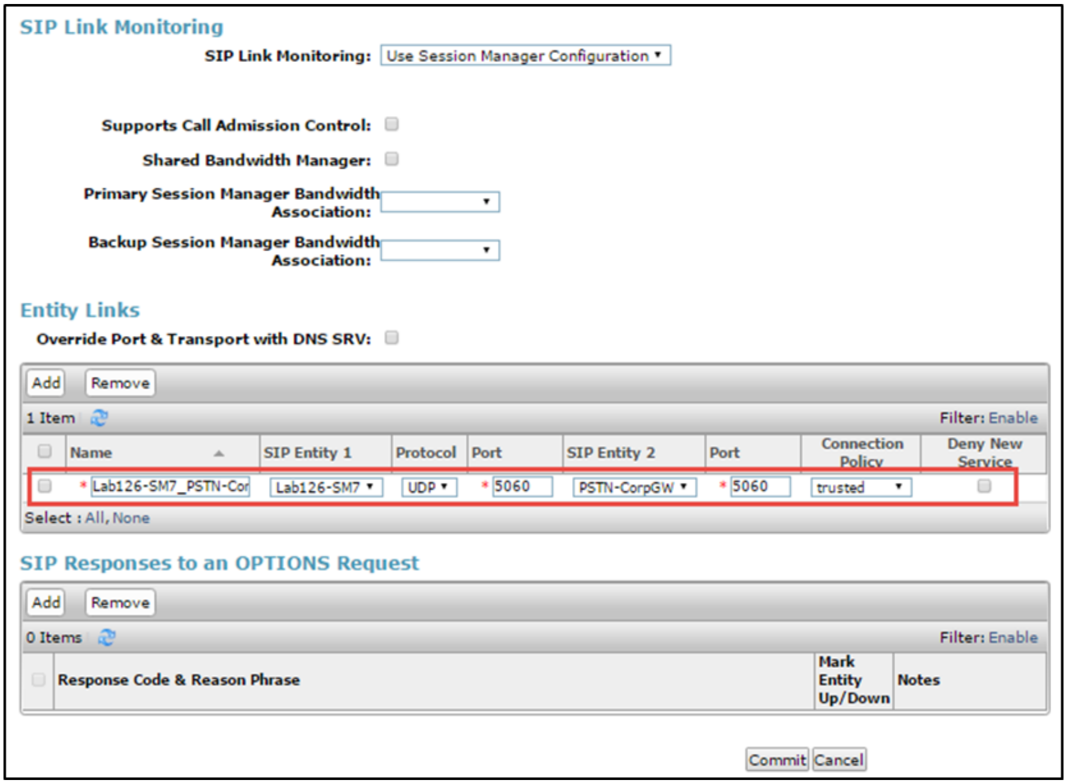
1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: SIP Entity - SM Configuration (1/2)



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 Avaya Session Manager 7.0** for the **Notes**.
   e. Select **DomainAdapter** for the **Adaptation**.
   f. Select **Lab126-Plano** for the **Location**.
   g. Select **America/Chicago** for the **Time Zone**.

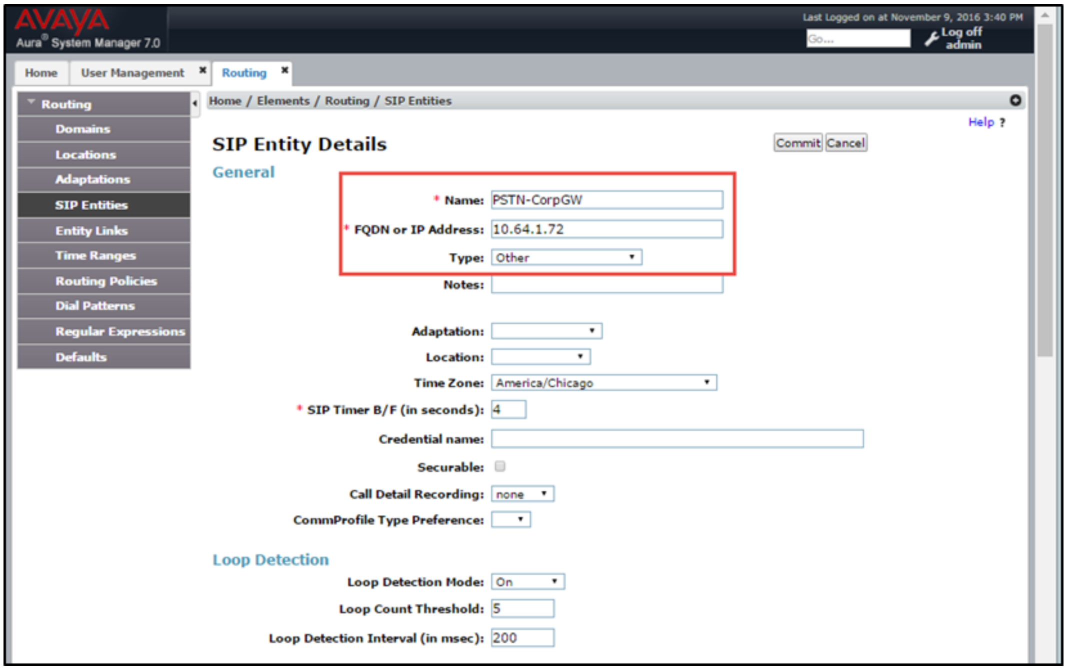
Avaya Aura SM: SIP Entity - SM Configuration (2/2)



To add a SIP entity for the PSTN gateway:

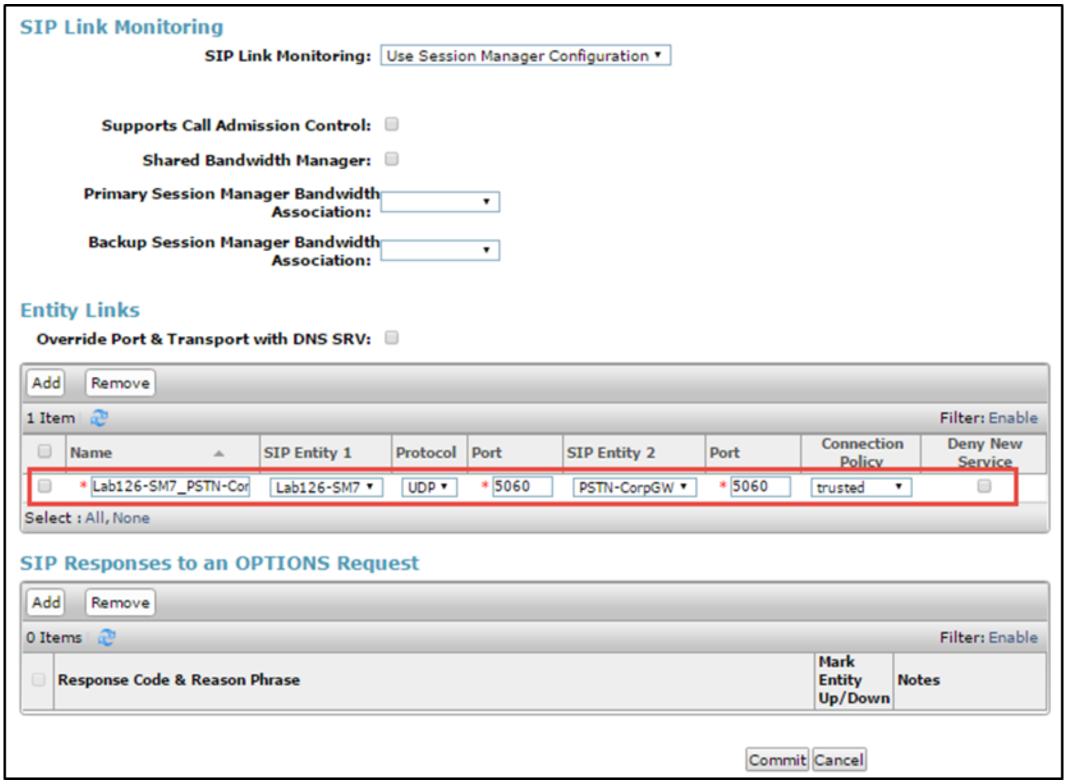
1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: SIP Entity - PSTN GW Configuration (1/2)



3. In the **General** section, do the following (for this example):
   a. Enter **PSTN-CorpGW** for the **Name** (for a SIP entity of PSTN gateway) .
   b. Enter **10.64.1.72** for the **FQDN or IP Address**.
   c. Select **Other** for the **Type**.
   d. Select **America/Chicago** for the **Time Zone**.

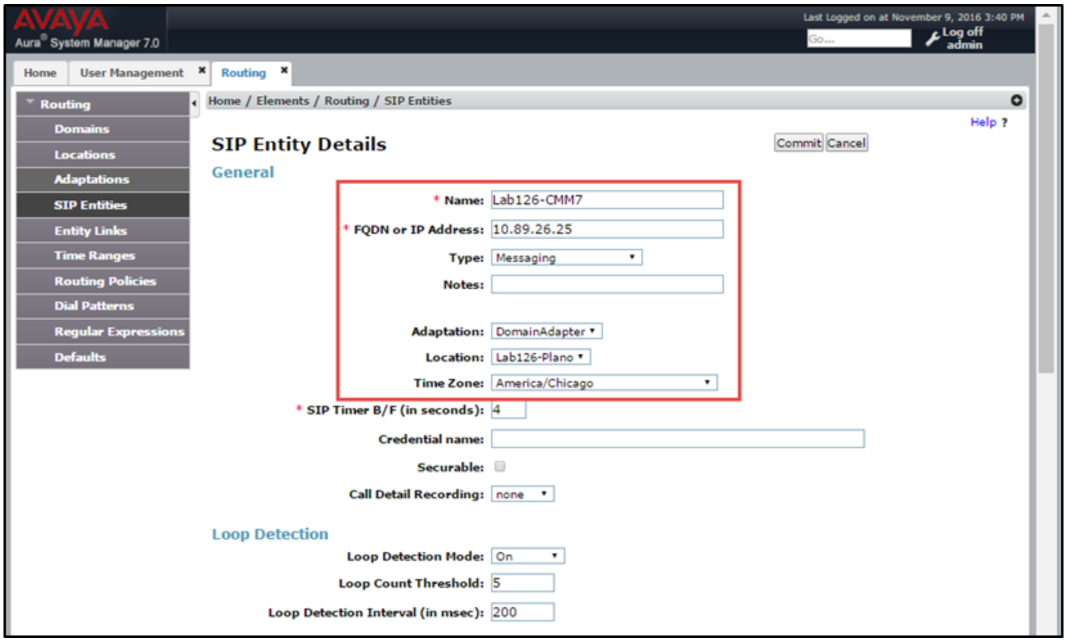
Avaya Aura SM: SIP Entity - SIP Entity Configuration (2/2)



To add a SIP entity for the Avaya Communication Manager Messaging:

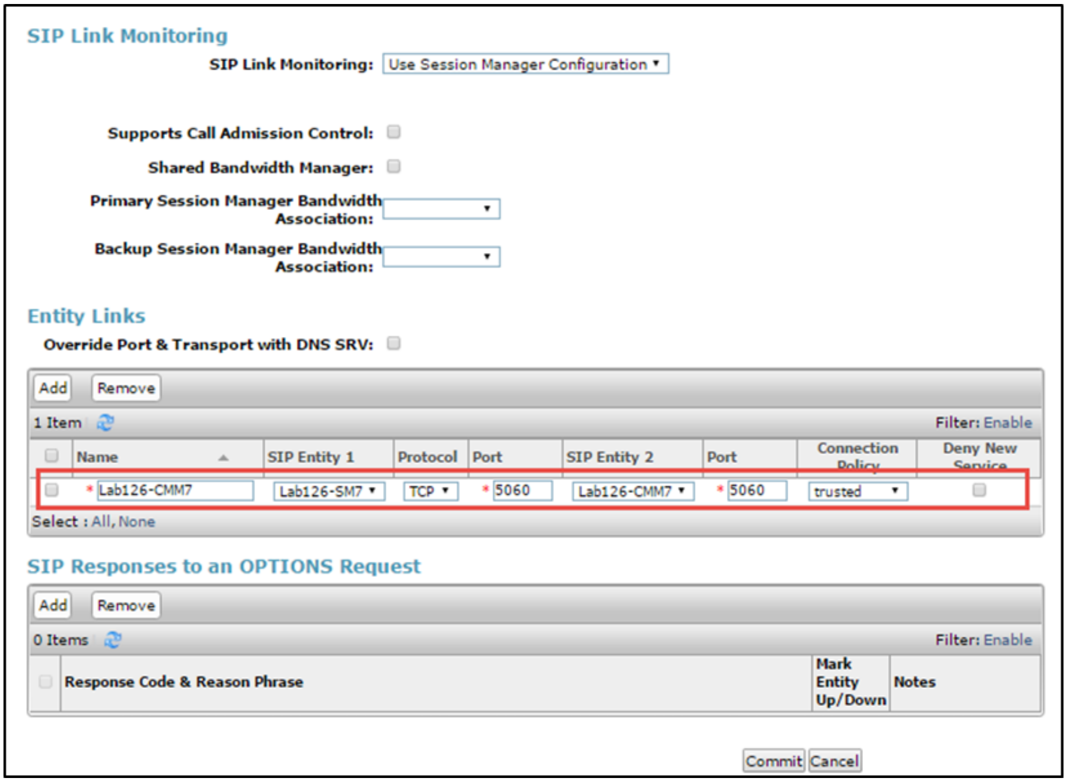
1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: SIP Entity - Avaya Communication Manager Messaging Configuration (1/2)



3. In the **General** section, do the following (entries for steps b through f are for this example):
   a. Enter **Lab126-CMM7**for the **Name** (for a SIP entity of Avaya Communication Manager Messaging) .
   b. Enter **10.89.26.25** for the **FQDN or IP Address**.
   c. Select **Messaging** for the **Type**.
   d. Select **DomainAdapter** for the **Adaptation**.
   e. Select **Lab126-Plano** for the **Location**.
   f. Select **America/Chicago** for the **Time Zone**.

Avaya Aura SM: SIP Entity - Avaya Communication Manager Messaging Configuration (2/2)

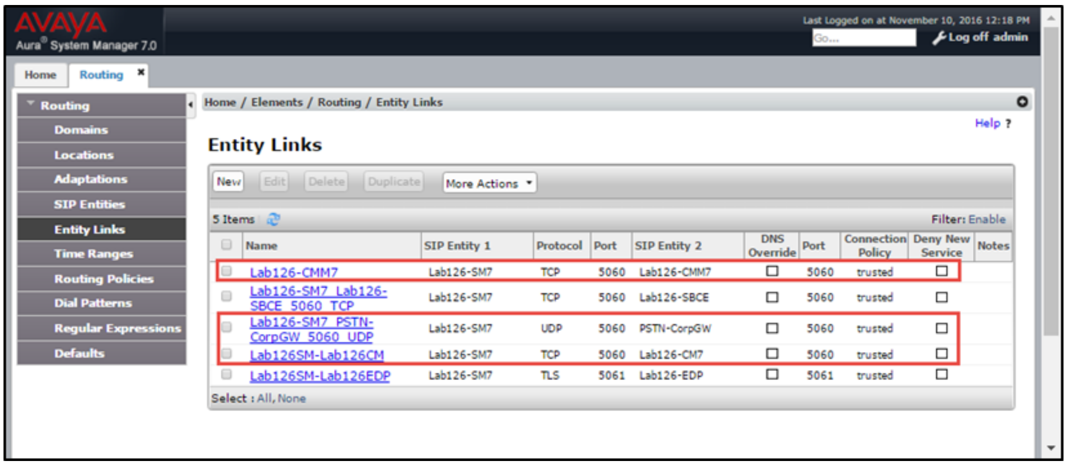


Entity Links

A SIP trunk between Avaya Session Manager and a telephony system is an entity link. This example creates an entity link to each of the following:

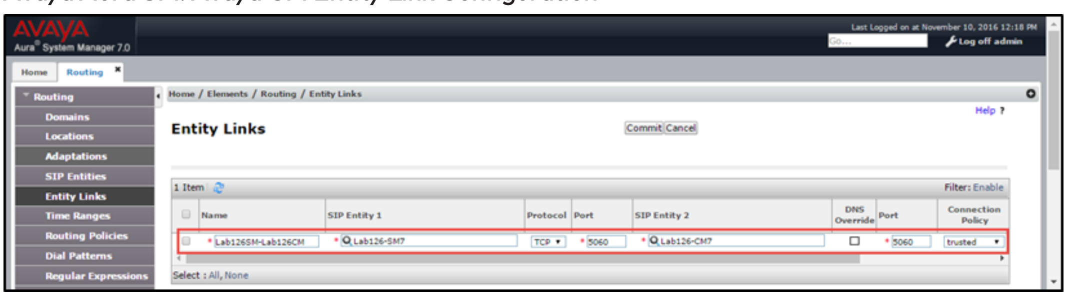
- Communication Manager
- Avaya Communication Manager Messaging
- PSTN gateway

Avaya Aura SM: Entity Links



To add Avaya CM as an entity link:

1. Click **Routing** > **Entity Links**.
2. Click **New**.
   Avaya Aura SM: Avaya CM Entity Link Configuration

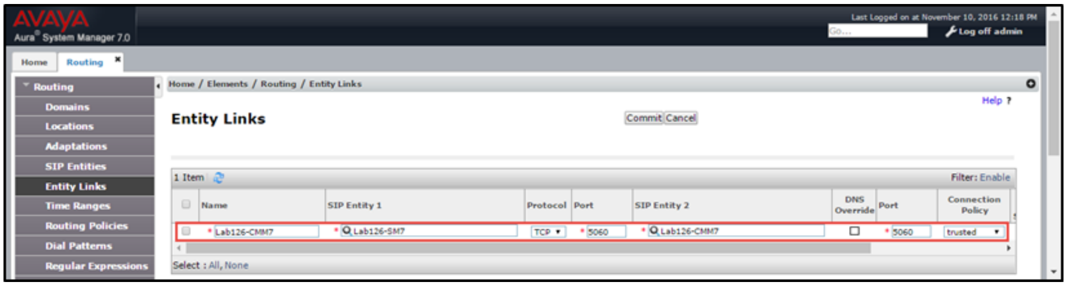


3. In the new row that is displayed, do the following:
   a. Enter a descriptive name for the **Name**.
   b. Select the Session Manager for **SIP Entity 1**.
   c. Select **TCP** (for the **Protocol** for this example).
   d. Enter **5060** for the **Port** (for this example).
   e. Select the Communication Manager for **SIP Entity 2**.
   f. Enter **5060** for the **Port** (for this example).
   g. Select **trusted** for the **Connection Policy**.
4. Click **Commit** to save.

To add Avaya Communication Manager Messaging as an entity link:

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

   Avaya Aura SM: CMM Entity Link Configuration

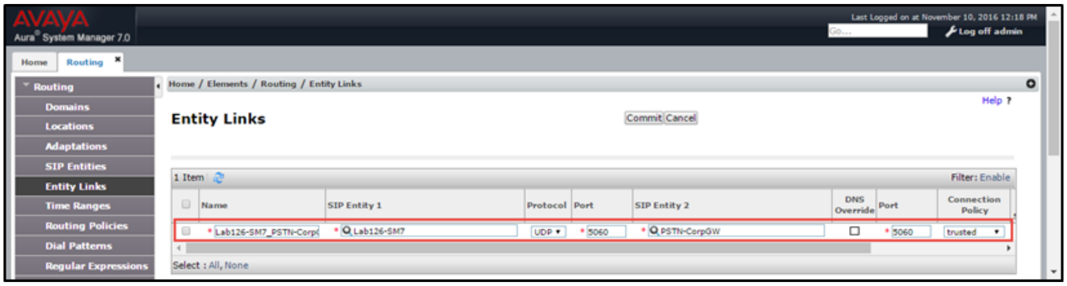
   

3. In the new row that is displayed, do the following:
   a. Enter a descriptive name for the **Name**.
   b. Select Avaya Communication Manager Messaging for **SIP Entity 1**.
   c. Select **TCP** (for the **Protocol** for this example).
   d. Enter **5060** for the **Port** (for this example).
   e. Select the Session Manager for **SIP Entity 2**.
   f. Enter **5060** for the **Port** (for this example).
   g. Select **trusted** for the **Connection Policy**.
4. Click **Commit** to save.

To add PSTN GW as an entity link:

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

Avaya Aura SM: PSTN GW 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 PSTN GW for **SIP Entity 1**.
   c. Select **TCP** (for the **Protocol** for this example).
   d. Enter **5060** for the **Port** (for this example).
   e. Select the Session Manager for **SIP Entity 2**.
   f. Enter **5060** for the **Port** (for this example).
   g. Select **trusted** for the **Connection Policy**.
4. Click **Commit** to save.

Routing Policy

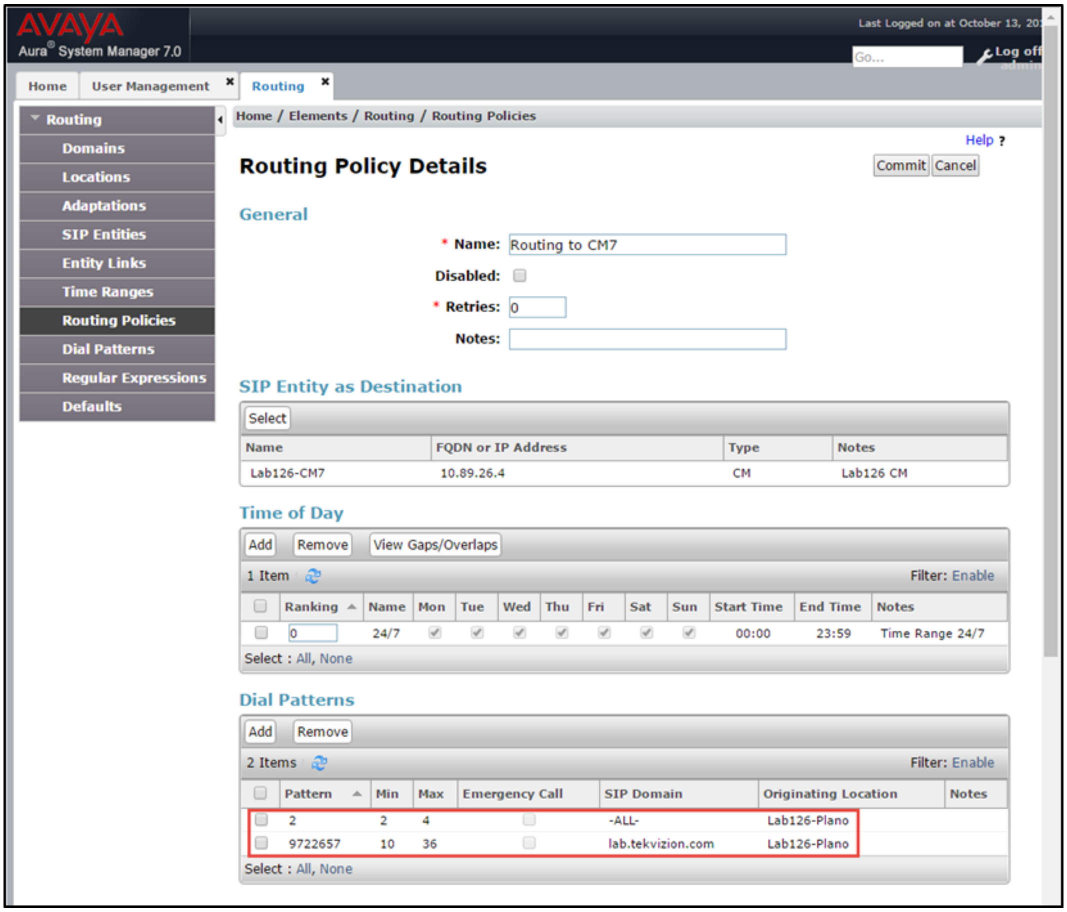
Routing policies describe the conditions under which the SIP entities receive routed calls. This example creates a routing policy for each of the following:

- Communication Manager
- PSTN gateway
- Voice mail

To add a routing policy for Avaya CM:

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

Avaya Aura SM: Routing Policy Configuration (1/3)

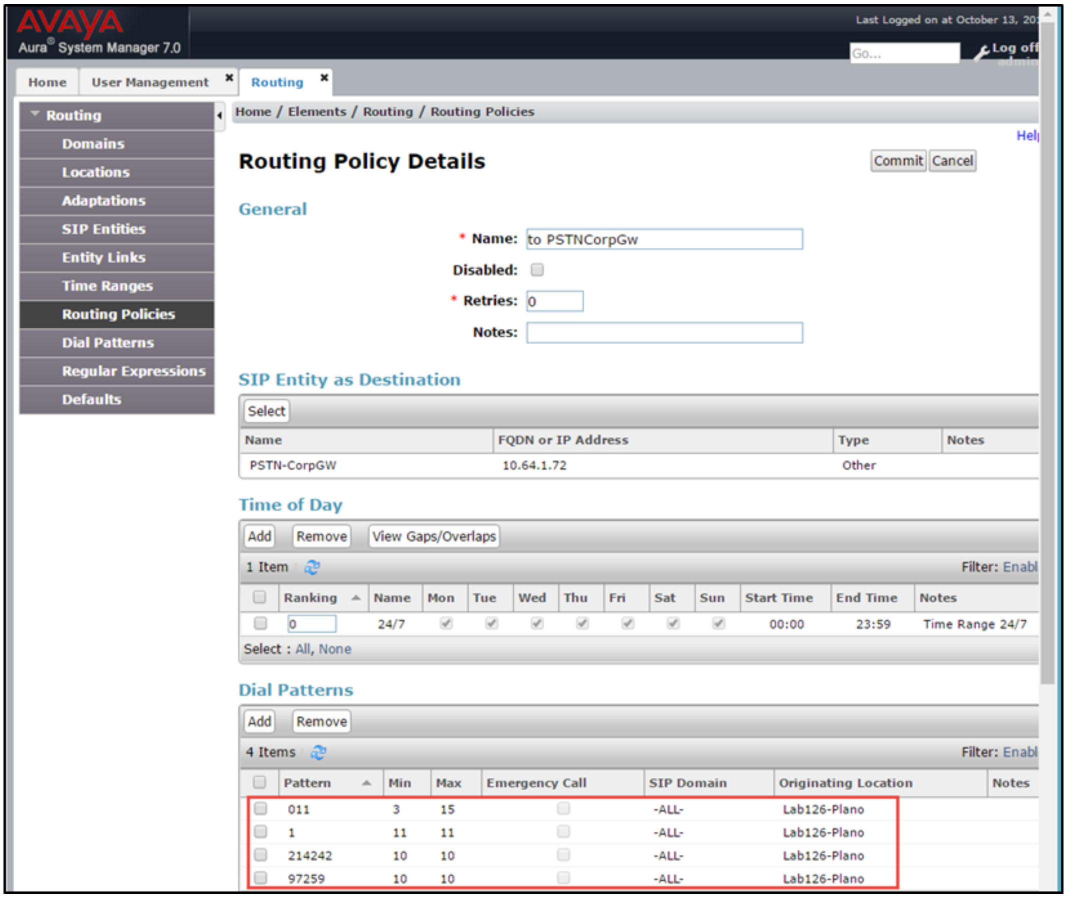


3. In the **General** section, enter **Routing to CM7** for the **Name** (for this example). The remaining fields use default values.
4. In the **SIP Entity as Destination** section, select the Avaya CM. This example uses **Lab126-CM7**.
5. In the **Dial Patterns** section, add the patterns to a call per the routing policy.
   a. Add the **2** pattern. The Avaya and Crestron endpoints have their first 4-digit extensions starting with **2**.
   b. Add the **9722657** pattern. The 10-digit Avaya and Crestron endpoints DID start with **9722657**.

To add a routing policy for the PSTN GW:

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

Avaya Aura SM: Routing Policy Configuration (2/3)

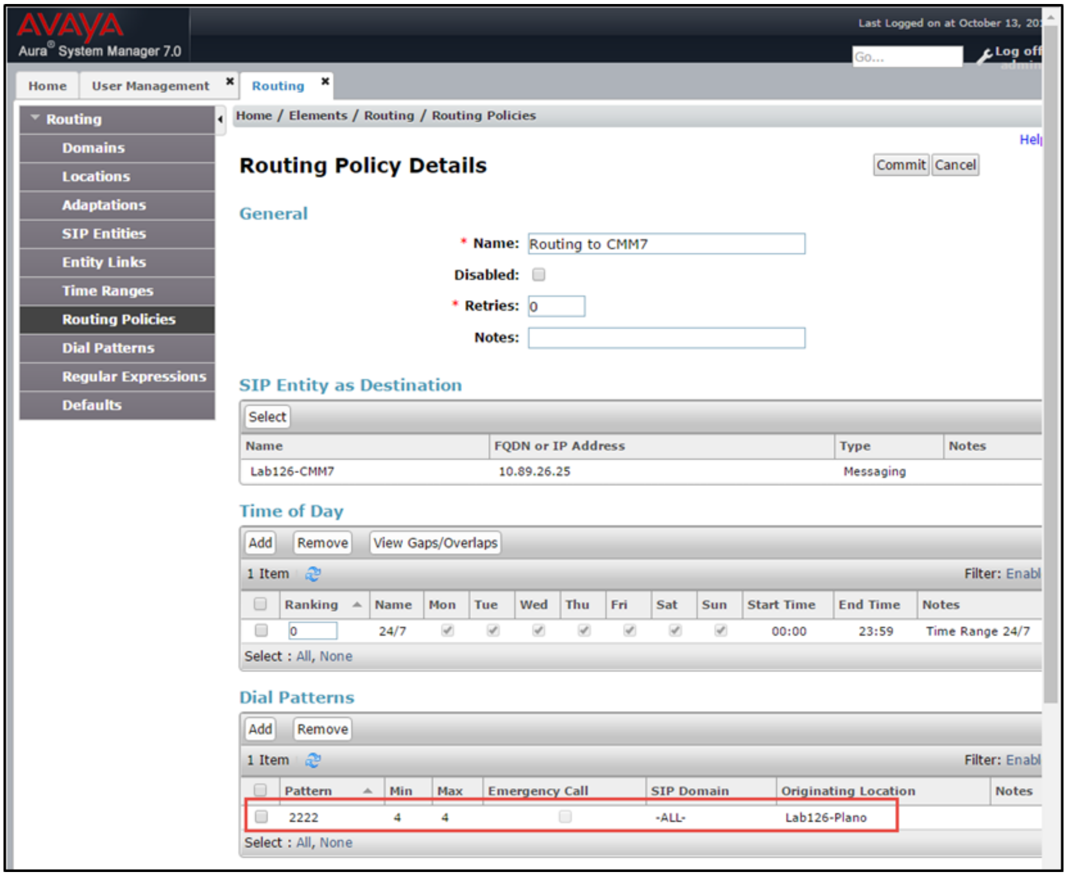


3. In the **General** section, enter **to PSTNCorpGW** for the **Name** (for this example). The remaining fields use default values.
4. In the **SIP Entity as Destination** section, select **PSTNCorpGW**.
5. In the **Dial Patterns** section, add the patterns to a call per the routing policy.
   a. Add the **011** pattern, an 11-digit international dialing pattern starting with **1**.
   b. Add the **1** pattern, an 11-digit national dialing pattern starting with **1**.
   c. Add the **214242** pattern, a 10-digit PSTN dialing pattern starting with **214242**.

To add a routing policy for the Avaya Communication Manager Messaging - Voice Mail System:

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

Avaya Aura SM: Routing Policy Configuration (3/3)



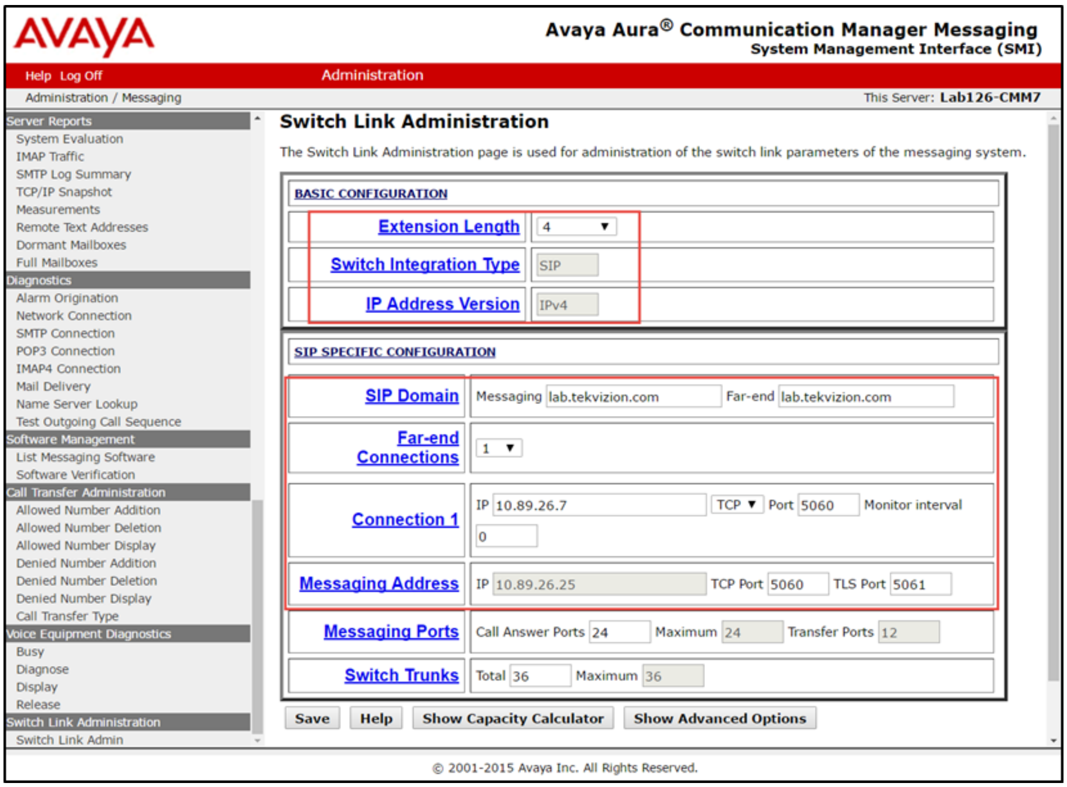
3. In the **General** section, enter **Routing to CMM7** for the **Name** to reach PSTN. The remaining fields use default values.
4. In the **SIP Entity as Destination** section, select **Lab126-CMM7**.
5. In the **Dial Patterns** section, add the **2222** pattern, used as the voice mail pilot in this example.

Avaya Communication Manager Messaging

This section describes the steps for configuring the Avaya Communication Manager Messaging to work with Avaya Aura Session Manager via SIP trunking.

Switch Link Administration

To administer the switch link:

1. Click **Administration** > **Messaging** > **Switch Link Administration** > **Switch Link Admin**.
   Avaya Communication Manager Messaging: Switch Link Administration
   

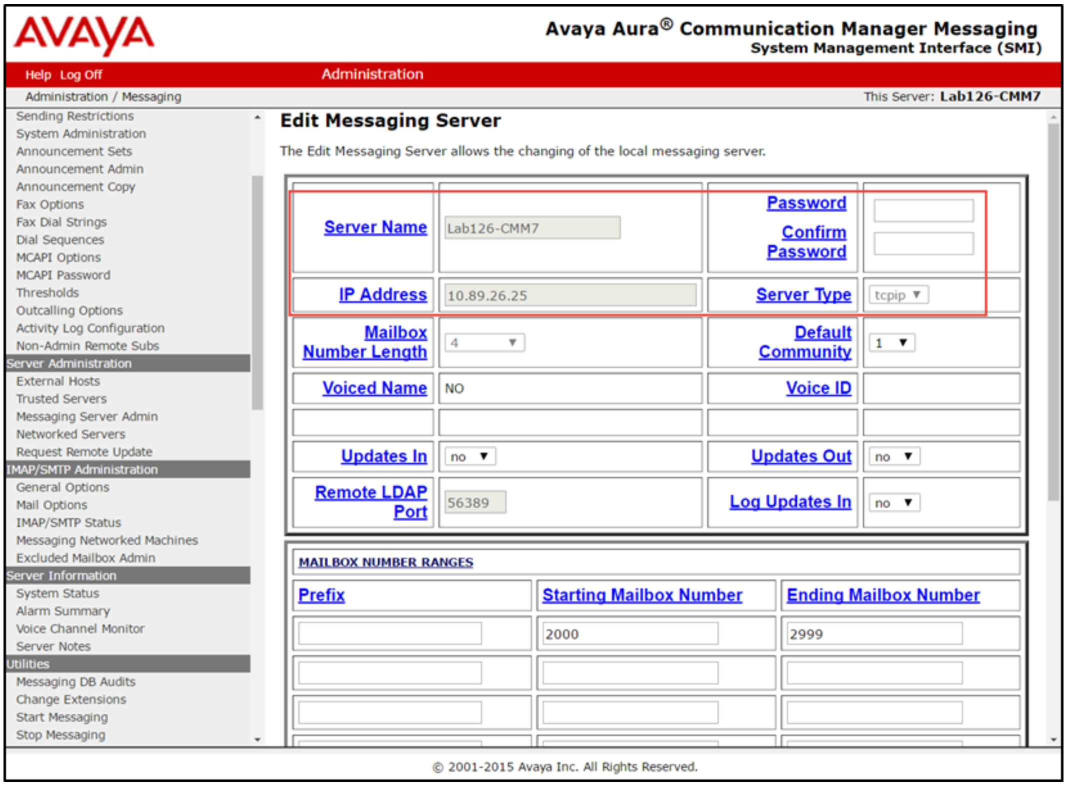
2. Under **BASIC CONFIGURATION**, do the following (for this example):
   a. Select **4** for the **Extension Length**.
   b. Enter **SIP** for the **Switch Integration Type**.
   c. Enter **IPV4** for the **IP Address Version**.
3. Under **SIP SPECIFIC CONFIGURATION**, do the following (for this example):
   a. Enter **lab.tekvizion.com for** the **SIP Domain**.
   b. Enter **10.89.26.7** (the Avaya Session Manager IP) for **Connection 1**.
   c. Enter **10.89.26.25** for the **Messaging Address**.

Messaging Server

For this example, configure the parameters for the Communication Manager Messaging Server:

1. Click **Administration** > **Messaging** > **Server Administration** > **Messaging Server Admin**.

   Avaya Communication Manager Messaging: Messaging Server Configuration

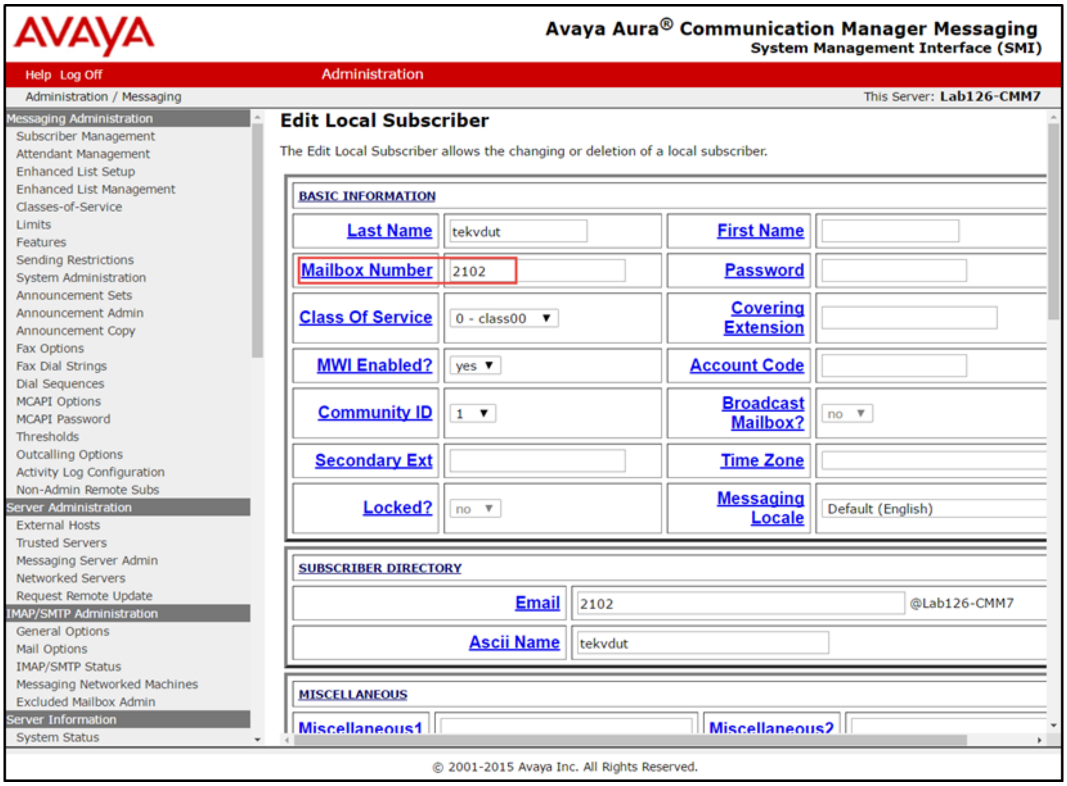
2. Enter **Lab126-CMM7** for the **Server Name**.
3. Enter **10.89.26.25** for the **IP Address**.
4. Enter **2000** for the **Starting Mailbox Number**.
5. Enter **2999** for the **Ending Mailbox Number**.

Subscriber

To create a subscriber for the messaging server:

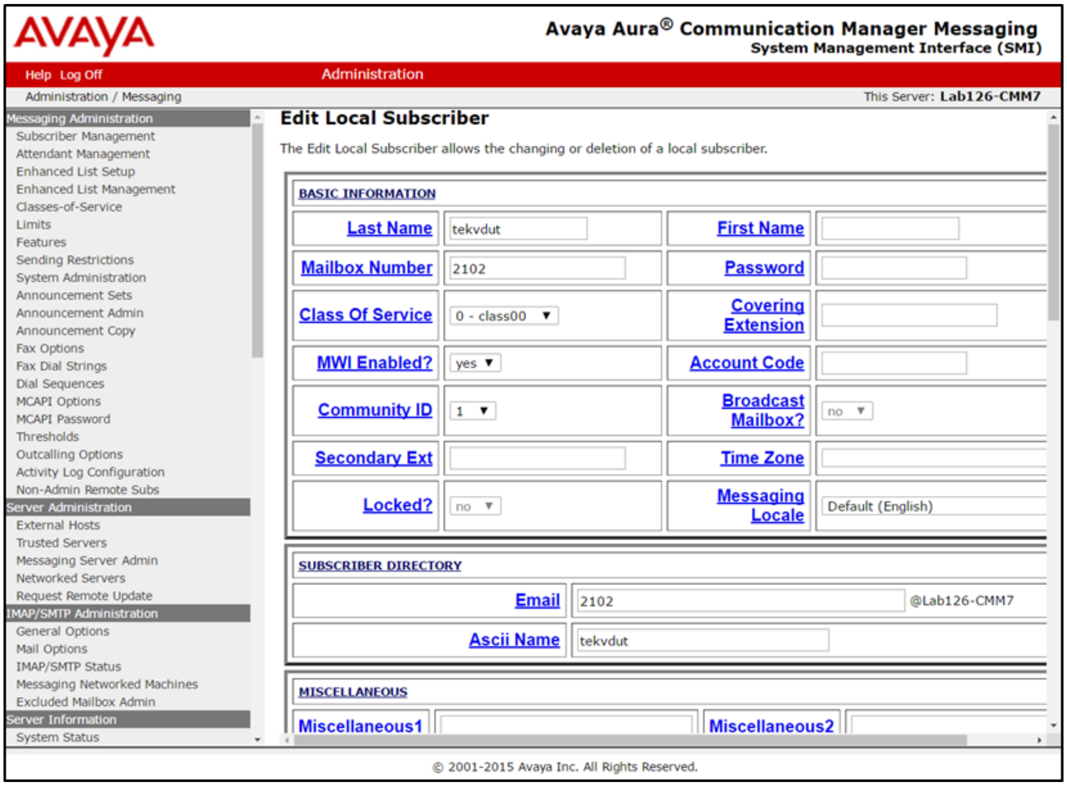
1. Click **Administration** > **Messaging** > **Messaging Administration** > **Subscriber Management**.
2. Click **Add**.

   Avaya Communication Manager Messaging: Subscriber Configuration (1/3)

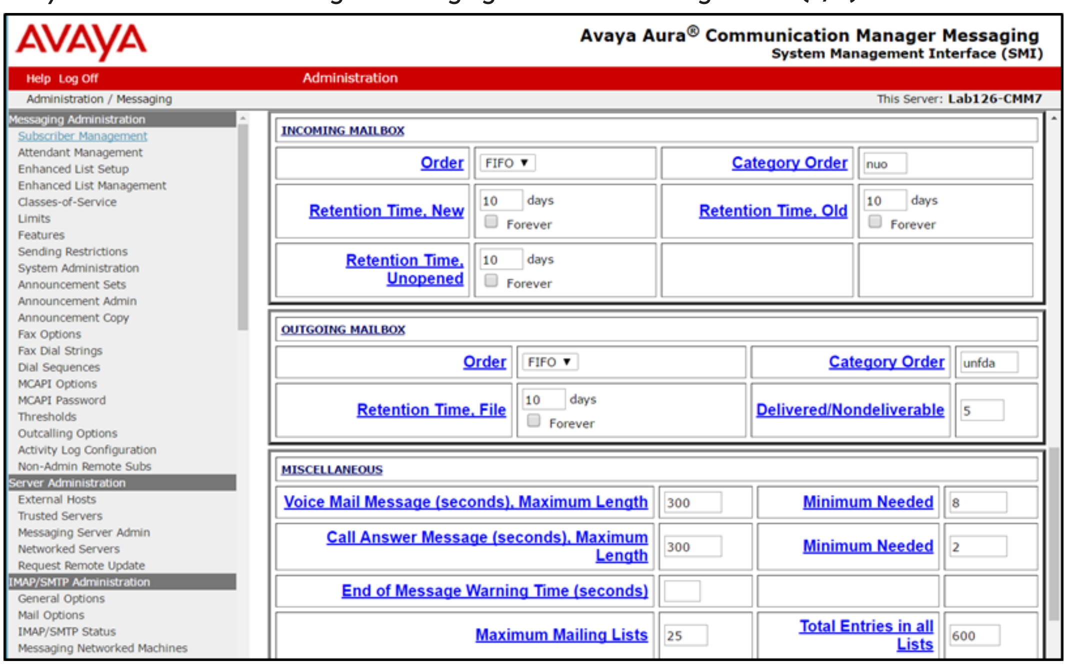


3. Enter **tekvdut** for the **Last Name** (for this example).
4. Enter **2102** for the **Mailbox Number** (for this example).
5. Select **yes** for **MWI Enabled**.
6. Leave all other fields at the default values.

Avaya Communication Manager Messaging: Subscriber Configuration (2/3)



Avaya Communication Manager Messaging: Subscriber Configuration (3/3)



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