CRESTRON®

CCS-UC-1

SIP Endpoint with Avaya Aura® 6.3 System

Configuration Guide

Crestron Electronics, Inc.

This document was written by the Technical Publications department at Crestron.

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CCS-UC-1: SIP Endpoint with Avaya Aura 6.3

Introduction

This configuration guide describes the necessary procedure to configure a Crestron Mercury™ device to register to the Avaya® Aura Communication Manager as a basic SIP endpoint.

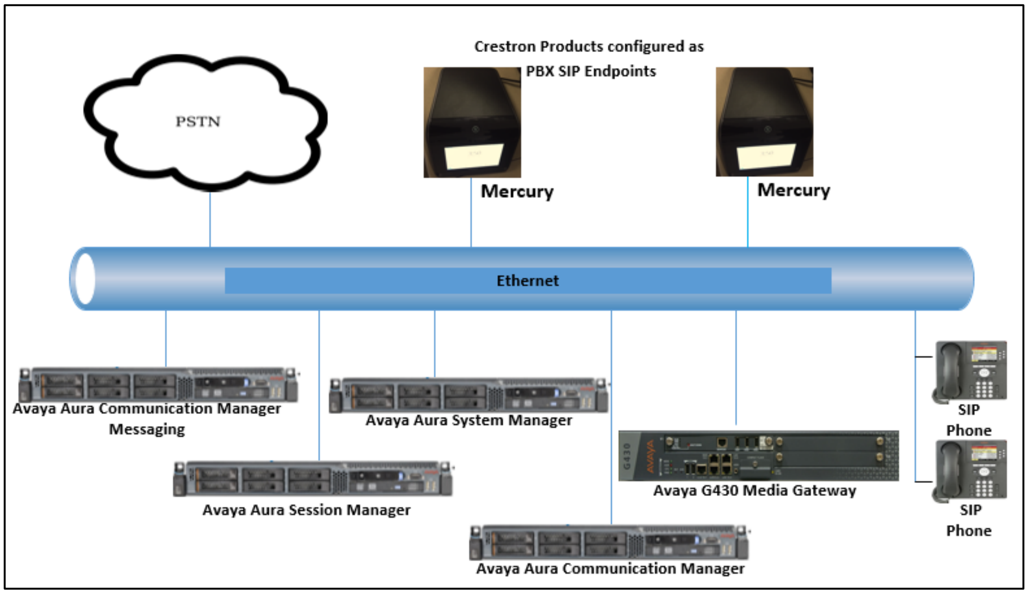
Audience

This document is intended for users attempting to configure and use the Crestron Mercury devices as SIP Endpoints registering to the Avaya Aura Session Manager 6.3.

Topology

The network topology for the Crestron Mercury Endpoint to interop with the Avaya Aura 6.3 is as shown below.

Crestron Mercury: SIP Endpoint Integration with Avaya: 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 Mercury device as the SIP Endpoints

Software Requirements

- Avaya Aura Communication Manager v 6.3
- Avaya Aura System Manager v 6.3
- Avaya Aura Session Manager v 6.3
- Avaya g430 Media Gateway v 36.18.30/1
- Crestron Mercury devices v 1.3390.0034

Hardware Requirements

- Avaya components either in a virtual environment or separate hardware servers.
    - Avaya Aura Communication Manager
    - Avaya Aura Session Manager
    - Avaya G430 Media Gateway
    - Avaya Aura Session Manager
    - Avaya Aura Modular Messaging
- PSTN Gateway for PSTN Calling
- Avaya phones (2) in SIP mode
- Crestron Mercury devices (2)

Product Description

The Crestron Mercury device is a complete solution for conference rooms. It acts an all-in-one touch screen, speakerphone, and AirMedia® product for conference rooms that integrates microphones and speakers into the user interface at the table.

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

The Crestron Mercury web interface is used to control all Crestron Mercury devices on the network.

Summary

The Crestron Mercury devices, in secure mode, are configured on the Avaya Aura as SIP endpoints. The devices successfully register to the Avaya Aura Session Manager with digest authentication.

Features Supported

- Registration with Digest Authentication
- Basic Calls with G729, G722, G711u, and G711a codecs
- Caller ID (limited to only calling number)
- DTMF support
- Early media support
- Retrieval of a parked call
- Transferee in a call transfer
- Conference participant
- Member of hunt group
- Voice mail access and interaction

Features Not Supported

- Caller ID presentation in the form of name and number
- Call hold and resume
- Call forwarding on the device (forwarding can be configured on the PBX for the 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)
- Call park (initiating call park)
- Message waiting indicator

Known Issues and Limitations

- The device fails to maintain an active call during a PBX network outage. As soon as the Crestron Mercury device loses connectivity with the PBX, it drops the currently active call. This issue is tracked via Crestron's Bugzilla™ software Defect: 128016.
- When the device's power is cycled during a call and the device recovers and reregisters to the Avaya PBX, incoming calls to the device fail until the other party in the previously active call disconnects.
- When a device is in an active call and there is a network outage on the PBX, only outbound calls can be made from the device once the PBX recovers and the device reregisters. The device is unable to receive calls until the Avaya Communication Manager updates the status of the device extension from "active" to "idle." Any call to the device receives a busy treatment as long as the status of the device extension on the Avaya CM is "active."

- A call made by the device to certain models of Avaya phones puts those phones in an auto-answer mode. One such model is the Avaya 9640G.
- Caller ID is not supported on the device. Currently only the calling party number is displayed as the caller ID. This issue is tracked via Crestron's Bugzilla software Defect: 119006.
- When a call is rejected on the Avaya Aura 6.3, the calling party receives a delayed response to the *603 Decline* SIP message. Therefore, if the device places a call that is rejected by the called party, the user receives an appropriate error treatment in the form of a reorder tone, but after a delay.
- The active call timer on the device unit does not reflect the correct call duration. The active call duration includes the time for which the unit was being alerted also. This issue is tracked via Crestron's Bugzilla software Defect: 124001.
- The first ringback heard on the device is stuttered. It resembles a mix of local and remote ringback. This issue is tracked via Crestron's Bugzilla software Defect: 122421.
- On the device's web user interface, there is currently no notification provided to the user when certain configurations are missing. This issue is tracked via Crestron's Bugzilla software Defect: 125193.
- On the device's web user interface, a configuration of DHCP OFF on the Network configuration page mandates configuration of both adapters. The user is unable to save changes unless both adapters are configured and is notified of an invalid IP against the default of 0.0.0.0 for an unused adapter. This issue is tracked via Crestron's Bugzilla software Defect: 126236.
- On the Crestron Mercury device, for certain called numbers that cannot be reached or are invalid, the user hears only a reorder tone and does not have the option to disconnect the call except by pressing the call button again. This issue is tracked via Crestron's Bugzilla software Defect: 122633.

Crestron Mercury Configuration

Setup

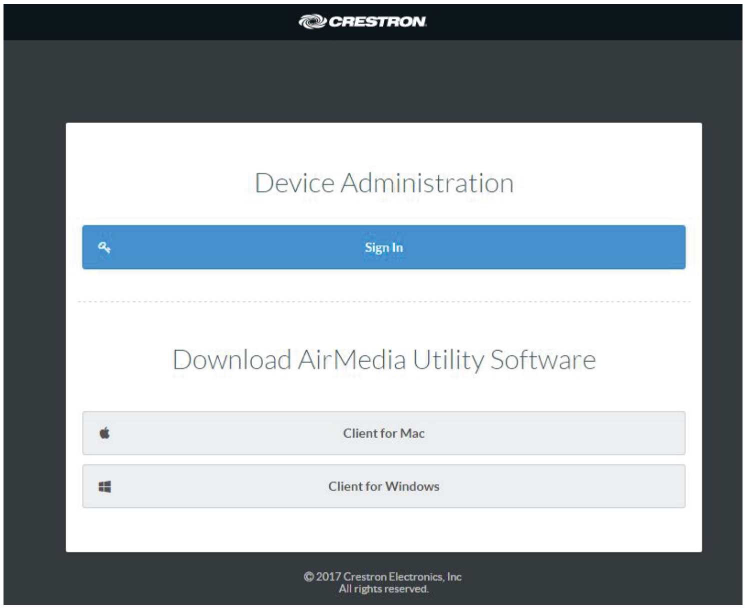
The LAN port of the Crestron Mercury device needs to be connected to one PoE+ port to power it up for network connectivity with the Avaya Aura. The PoE switch that is used should have the LLDP functionality enabled for the device to power up and be completely functional. By default the "poeplus" configuration is set to Off on the device.

Configure the device

To configure the device, follow this procedure:

1. Access the web GUI for the device by using an http session with the device's IP address. The device IP address used in this test was 10.70.4.50.

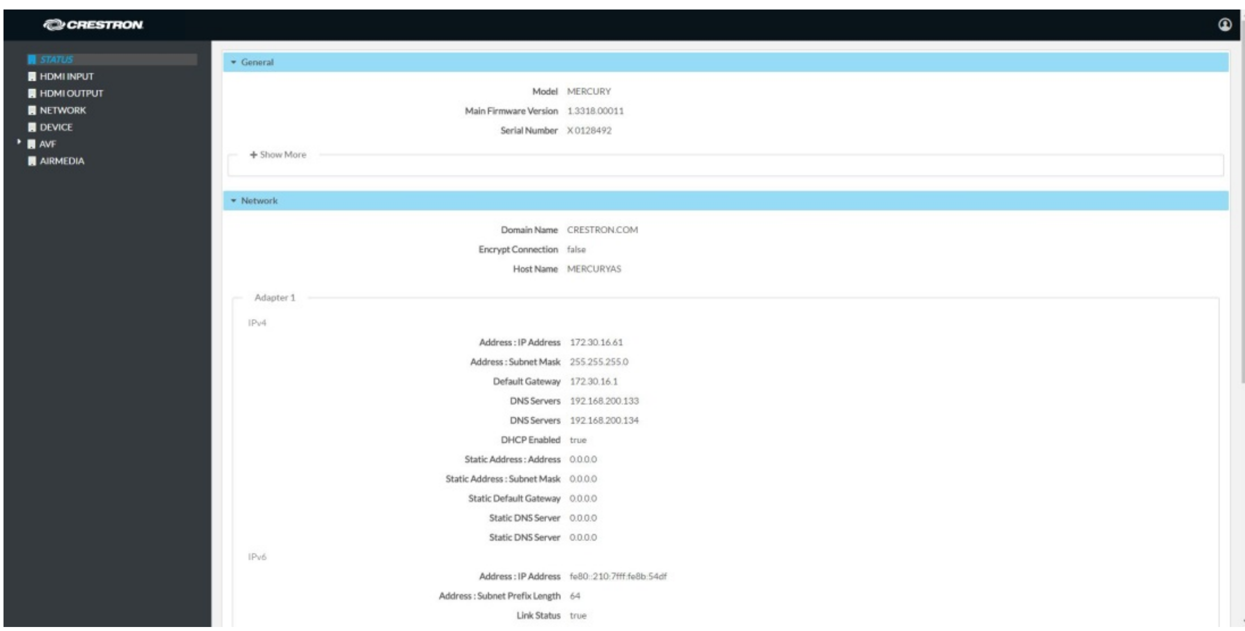
Crestron Mercury: Login to Web GUI



2. Click **Sign In** and log in to the device. For information on device administration, refer to the CCS-UC-1 Supplemental Guide (Doc. 7844) at www.crestron.com/manuals.

   The Status screen that appears displays basic information on the device.

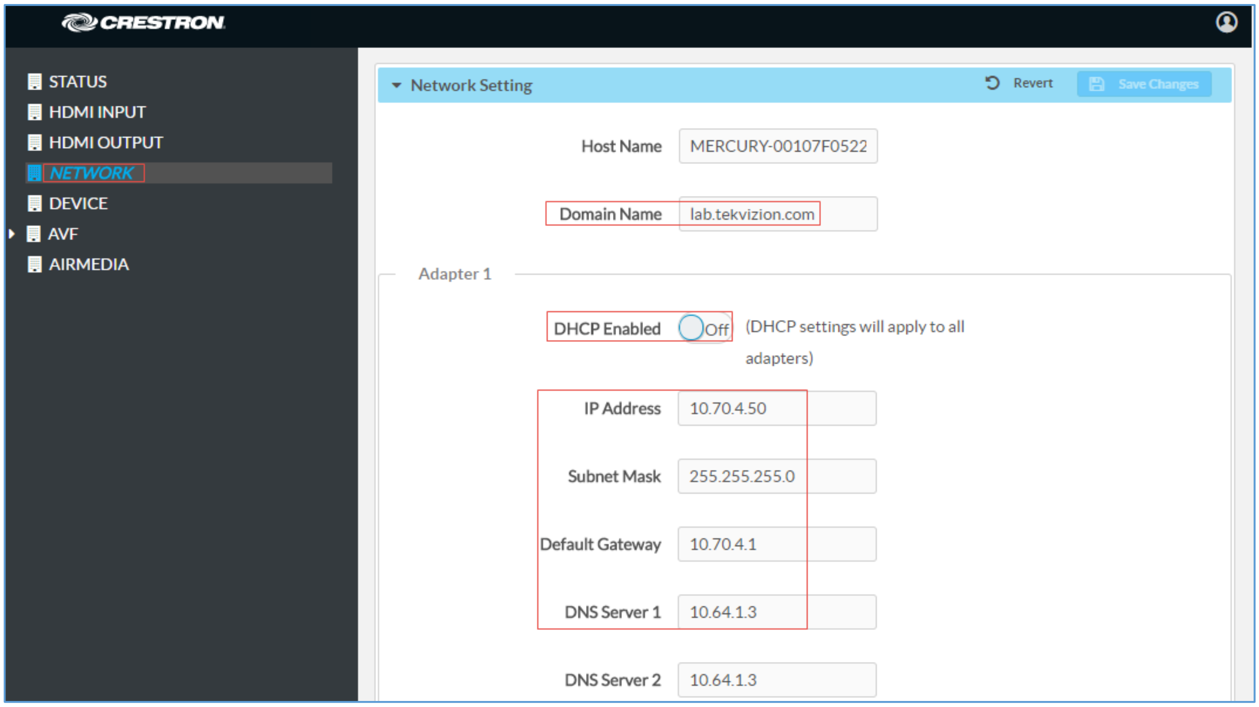
Crestron Mercury: Status



   The device can be configured from the **Network** page.

3. On the web GUI, navigate to **Network**.

Crestron: Mercury Configuration: Network Screen



4. Enter the following parameters to configure the Crestron Mercury device.
   - Domain Name: *lab.tekvizion.com*, used in this example (mostly auto-detected by device when in DHCP mode).
   - DHCP: Choose either of the following:
     - Obtain an IP address automatically.
     - Use the following IP address.

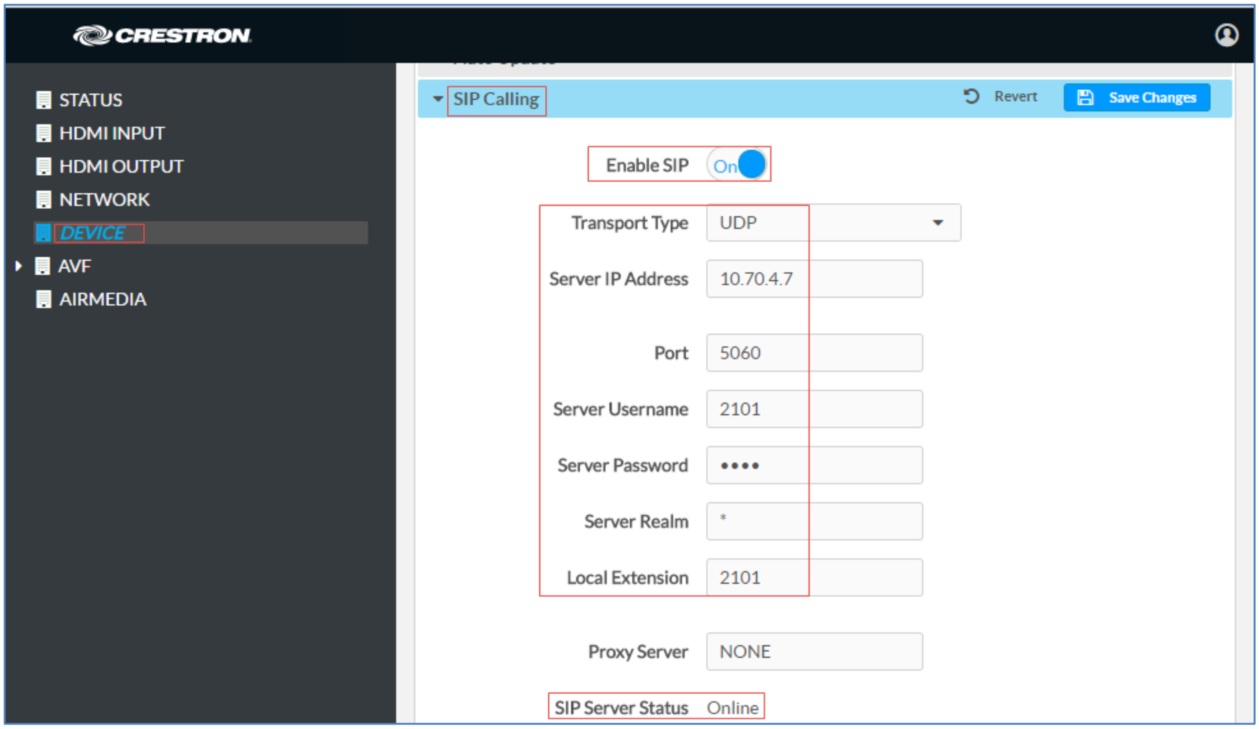
     For the test, a static IP was configured.
     - IP address: *10.70.4.50* was used in this example.
     - Subnet Mask: *255.255.255.0* was used in this example.
     - Default Gateway: *10.70.4.1* was used in this example.
     - DNS Servers: *10.64.1.3* was used in this example.
5. Click Save Changes.

Configure SIP Parameters

To configure the SIP parameters, follow this procedure:

1. On the web GUI, navigate to Device > SIP Calling.

Crestron: Mercury: Device Configuration: TLS SIP Parameters



2. Enable the check box for Enable SIP.
3. Configure the Server IP Address: Enter the IP address of the Avaya Aura Session Manager node: *10.70.4.7* was used in this example.
4. Configure the Port: *5060* was used in this example.
5. Configure the Server Username: Enter the end user configured on Avaya Aura Communication Manager for this device. *2101* was used in this example (the other end user configured was *2621*).
6. Configure the Server Password: Enter the password as configured on Avaya Aura Communication Manager for this end user.
7. Configure the Local Extension: Enter the directory number that was configured for this device on Avaya Aura Communication Manager. *2101* was used in this example (the other extension configured for the second Crestron Mercury device was *2621*).
8. Leave all other fields at their default values.
9. Click Save Changes.

Once the device successfully registers with the Avaya Aura Session Manager, the SIP Server Status updates its status to show *Online.*

Avaya Aura Communication Manager Configuration

This section describes the Avaya Aura Communication Manager (Avaya CM) configuration necessary to support the registration of the devices using digest authentication and connectivity to PSTN.

NOTE: It is assumed that the general installation and basic Avaya Aura configuration have already been administered.

Node Names

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

Use the **change node-names ip** command to add the node name. In this example, *procr* and *AASM* were added with their respective IPs.

- *AASM* is an Avaya Aura Session Manager used in this example and is used to register the SIP phones and third-party SIP devices.
- *procr* is used to register the SIP trunk.

Avaya Aura CM: Node Configuration

```
change node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
AASM                    10.70.4.7
AASMHA                  10.70.4.24
default                 0.0.0.0
procr                   10.70.4.4
procr6                  ::

( 5  of 5    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

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.

For the test, **ip-codec-set 1** was configured with the following codecs supported by Crestron Mercury device: **G.729**, **G.711MU**, **G.722**, and **G.711A**.

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.729              n          2        20
 2: G.711MU            n          2        20
 3: G.722-64K                     2        20
 4: G.711A             n          2        20
 5:
 6:
```

Network Region

Configure an IP Network region 1 using the change ip-network-region 1 command.

To configure an IP Network region, issue the above command and do the following:

- Set Authoritative Domain: *lab.tekvizion.com* was used in this example.
- Set Name: provide any relevant name.
- Codec Set: *1*, which is programmed in the previous step.
- Set Intra-region IP-IP Direct Audio: *Yes*
- Set Inter-region IP-IP Direct Audio: *Yes*
- Retain all other default configurations.

Avaya Aura CM: Network Region Configuration

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: lab.tekvizion.com
    Name: [redacted]                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
```

Signaling Group

For this test, two signaling groups are configured:

- *signaling-group 2* for SIP Trunk to PSTN gateway for PSTN calls
- *signaling-group 3* for the SIP devices: Avaya SIP Phones and Crestron Mercury devices

Using the command ***add signaling-group 2***, add and configure the Signaling Group 2 as follows:

- Group Number: *2* was used in this example.
- Group Type: *sip* was used in this example.
- Transport Method: *tcp* was used in this example.
- Near-end Node Name: *procr* was used in this example.
- Near-end Listen Port: *5060* was used in this example.
- Far-end Node Name: *AASM* was used in this example.
- Far-end Listen Port: *5060* was used in this example.
- Far-end Network Region: *1* was used in this example.
- Far-end Domain: *lab.tekvizion.com was* used in this example.
- DTMF over IP: *rtp-payload* was used in this example.
- Direct IP-IP Audio Connections? *y* was used in this example.

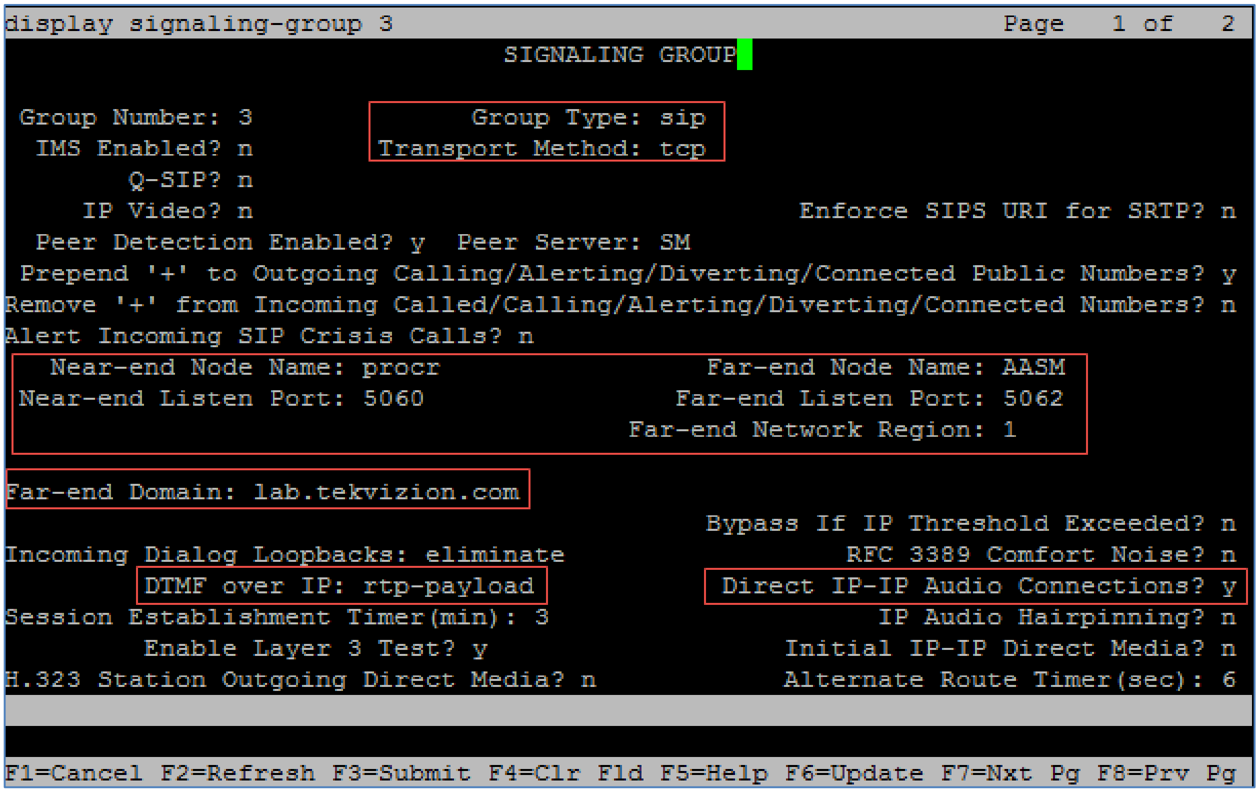
Avaya Aura CM: Signaling Group Configuration: For PSTN Calls

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

 Group Number: 2            Group Type: sip
   IMS Enabled? n       Transport Method: tcp
         Q-SIP? n
      IP Video? y          Priority 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: AASM
 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
```

Using the command *add signaling-group 3*, add and configure the Signaling Group 3 as follows:

- Group Number: *3* was used in this example.
- Group Type: *sip* was used in this example.
- Transport Method: *tcp* was used in this example.
- Near-end Node Name: *procr* was used in this example.
- Near-end Listen Port: *5060* was used in this example.
- Far-end Node Name: *AASM* was used in this example.
- Far-end Listen Port: *5062* was used in this example.
- Far-end Network Region: *1* was used in this example.
- Far-end Domain: *lab.tekvizion.com was* used in this example.
- DTMF over IP: *rtp-payload* was used in this example.
- Direct IP-IP Audio Connections? *y* was used in this example.

Avaya Aura CM: Signaling Group Configuration: For the SIP Devices



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

 Group Number: 3              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        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: AASM
 Near-end Listen Port: 5060                       Far-end Listen Port: 5062
                                               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
```

Trunk Groups

Similar to the signaling groups, two trunk groups were configured for this test:

- Trunk Group 2, which utilized a public numbering plan to place PSTN calls via a PSTN GW
- Trunk Group 3, which utilized a private numbering plan to access the stations registered to the Avaya Session Manager

Use the add trunk-group n command to add a new trunk group where *n* is the trunk group number.

Configure Trunk Group 2:

- Group Number: *2* was used in this example.
- Group Name: *Trunk to PSTN* was used in this example.
- Group Type: *sip* was used in this example.
- TAC: *#002* was used in this example.
- Signaling Group: *2* was used in this example.
- Number of Members: *6* was used in this example.
- Preferred Minimum Session Refresh Interval (sec): *900*
- Numbering Format: *public*
- Send Diversion Header?: *y* was used in this example.

- Telephone Event Payload Type: *101* was used in this example.
- Identity for Calling Party Display: *From* was used in this example.

Avaya Aura CM: Trunk Group Configuration to PSTN (1/4)

```
display trunk-group 2                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip            CDR Reports: y
  Group Name: Trunk to PSTN               COR: 1          TN: 1         TAC: #002
   Direction: two-way         Outgoing Display? n
 Dial Access? n                                         Night Service:
Queue Length: 0
Service Type: public-ntwrk           Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 2
                                                    Number of Members: 6
```

Avaya Aura CM: Trunk Group Configuration for PSTN (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): 900

 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
```

Avaya Aura CM: Trunk Group Configuration for PSTN (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: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no





 Show ANSWERED BY on Display? y
```

Avaya Aura CM: Trunk Group Configuration for PSTN (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? y
                                   Support Request History? y
                              Telephone Event Payload Type: 101


                        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
```

Configure Trunk Group 3:

- Group Number: *3* was used in this example.
- Group Name: *SIP PHONE* was used in this example.
- Group Type: *sip was* used in this example.
- Service Type: *tie* was used in this example.

- TAC: *#003* was used in this example.
- Signaling Group: *3* was used in this example.
- Number of Members: *10* was used in this test
- Preferred Minimum Session Refresh Interval (sec): *900*
- Numbering Format: *private*

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

```
display trunk-group 3                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: SIP PHONE                   COR: 1        TN: 1        TAC: #003
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                      Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 3
                                                     Number of Members: 10
```

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

```
display 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): 900

 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
```

Avaya Aura CM: Trunk Configuration to Session Manager (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


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

Avaya Aura CM: Trunk Configuration to Session Manager (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:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        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
```

Route Pattern

Two route patterns were configured for this test:

- Route pattern 2 for the SIP trunk to PSTN
- Route pattern 3 for the SIP devices/phones

Use **change route-pattern x** command to specify the routing preference.

Configure Route Pattern 2:

- **Pattern Name**: *SIP TRUNK* was used in this example.
- **Grp No**: *2* was used in this example.
- **FRL**: *0* is given as it has the least restriction.
- **Numbering Format**: *unk-unk* (Avaya uses unknown-unknown to address international numbers).
- Retain all other default configurations.

Avaya Aura CM: Route Pattern 2 Configuration

```
display route-pattern 2                                         Page   1 of   3
                    Pattern Number: 2       Pattern Name: SIP TRUNK
                             SCCAN? n      Secure SIP? 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  No. Numbering LAR
   0 1 2 M 4 W     Request                                 Dgts Format
                                                     Subaddress
 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
```

Configure Route Pattern 3:

- **Pattern Name:** *SIP Phone* was used in this example.
- **Grp No:** 3 was used in this example.
- **FRL:** *0* is given as it has the least restriction.
- Retain all other default configurations.

Avaya Aura CM: Route Pattern 3 Configuration

```
display route-pattern 3                                        Page   1 of   3
                    Pattern Number: 3      Pattern Name: SIP Phone
                             SCCAN? n     Secure SIP? n
    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  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                      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
```

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.

For the test a DID starting with 972265727x was used. The **inc-call-handling-trmt** on the trunk-group 2 (used to route the internal calls from PSTN) was configured to delete the first 10 digits and insert the 4 digit extensions.

Avaya Aura CM: Inbound Routing Configuration

```
change inc-call-handling-trmt trunk-group 2                    Page   1 of  30
                         INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number       Del Insert
 Feature        Len      Digits
 public-ntwrk  [redacted]
 public-ntwrk  [redacted]
 public-ntwrk   10 9722657277        10  2620
 public-ntwrk   10 9722657278        10  2621
 public-ntwrk   10 9722657279        10  2101
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
```

Outbound Routing

Automatic Route Selection (ARS)

The Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the PSTN. In the sample configuration, the single digit *9* is used as the ARS access code. PBX users dial 9 to initiate a call to PSTN. This common configuration is illustrated below with little elaboration.

Use the change dialplan analysis command to define a dialed string beginning with the following parameters:

- *2* of length *4* to dial extensions (*ext*)
- *9* of length *1* as a feature access code (*fac*)
- * of length *1* as a feature access code (*fac*)

Avaya Aura CM: Outbound Routing Configuration: Dial Plan Analysis Table

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

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

The following feature access codes were configured for this test:

- Call Park Access Code*: *30* was used in this example.
- Answer Back Access Code*: *31* was used in this example to retrieve a parked call.
- Auto Route Selection (ARS): *9* was used in this example.

Avaya Aura CM: Outbound Routing Configuration: 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: *31

      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:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: *30
                       Call Pickup Access Code:
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.

For this example, the following entries were configured:

- *214, 214242*, and *972*: to accommodate the lab and generic PSTN test numbers used during the test
- *1800,186,187, and 188*: to accommodate all 18xx numbers
- *121*: to accommodate calling the lab PSTN prefixed by a 1

Avaya Aura CM: Outbound Routing Configuration: Auto Route Selection (1/3)

```
display ars analysis 2                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                  Location: all            Percent Full: 0

        Dialed             Total     Route     Call    Node   ANI
        String            Min  Max   Pattern   Type    Num    Reqd
   214242                 10   10    2         natl           n
   [redacted]                                                 n
                                                              n
                                                              n
                                                              n
                                                              n
                                                              n
                                                              n
                                                              n
                                                              n
                                                              n
                                                              n
   972265                 10   10    2         natl           n
   972598                 10   10    2         natl           n
   [redacted]                                                 n


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

Avaya Aura CM: Outbound Routing Configuration: Auto Route Selection (2/3)

```
display ars analysis 18                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                  Location: all            Percent Full: 0

        Dialed             Total     Route     Call    Node   ANI
        String            Min  Max   Pattern   Type    Num    Reqd
   180                    11   11    deny      fnpa           n
   1800                   11   11    2         natl           n
   1800555                11   11    deny      fnpa           n
   1809                   11   11    deny      fnpa           n
   181                    11   11    deny      fnpa           n
   182                    11   11    deny      fnpa           n
   183                    11   11    deny      fnpa           n
   184                    11   11    deny      fnpa           n
   185                    11   11    deny      fnpa           n
   186                    11   11    2         natl           n
   187                    11   11    2         natl           n
   1877                   11   11    2         natl           n
   188                    11   11    2         natl           n
   189                    11   11    deny      fnpa           n
   190                    11   11    deny      fnpa           n


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

Avaya Aura CM: Outbound Routing Configuration: Auto Route Selection (3/3)

```
display ars analysis 12                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route     Call    Node   ANI
          String          Min  Max   Pattern   Type    Num    Reqd
    120                   11   11    deny      fnpa           n
    1200                  11   11    deny      fnpa           n
    121                   11   11    2         natl           n
    122                   11   11    deny      fnpa           n
    123                   3    3     1         svcl           n
```

Auto Alternative Routing

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

The following entries were configured for this test:

- **Dialed Number**: *2*, utilizing route pattern 3 for Avaya SIP phones and Crestron Mercury SIP devices
- **Dialed Number**: *2301*, utilizing route pattern 3 to access voice mail
- **Dialed number**: *9*, utilizing route pattern 2 for PSTN numbers

Avaya Aura CM: AAR Digit Analysis Table Configuration

```
display aar analysis 1                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route     Call    Node   ANI
          String          Min  Max   Pattern   Type    Num    Reqd
    2                     4    4     3         aar            n
    2301                  4    4     3         aar            n
    3                     7    7     2000      aar            n
    4                     4    4     1         aar            n
    6                     7    7     2000      aar            n
    7                     7    7     2000      aar            n
    8                     7    7     2000      aar            n
    9                     10   10    2         aar            n
```

Hunt Group

One hunt group was configured for this test:

Extension *2100* was used in this example for the **Hunt Group** feature.

Use the ***add hunt-group n*** command to add a new hunt group where *n* is the available hunt group number.

Configure the Hunt Group:

- **Group Number:** *2* was used in this example.

- Group Name: *AvayaHG* was used in this example.
- Group Extension: *2100* was used in this example.
- Group Type: *circ* was used in this example to enable sequential ringing on the hunt-group members.
- Coverage Path: *2* was used in this example, which includes hunt-group members that will be alerted sequentially.

Avaya Aura CM: Hunt Group Configuration

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

            Group Number: 2
              Group Name: AvayHG
         Group Extension: 2100
              Group Type: circ                        Coverage Path: 2
                      TN: 1                  Night Service Destination:
                     COR: 1                             MM Early Answer? n
           Security Code:                        Local Agent Preference? n
 ISDN/SIP Caller Display:
```

Use the ***add coverage path n*** command (where *n* is the available coverage path number) to add the coverage path that includes members of the hunt group.

Coverage path 2 was used in this example. This is invoked by Hunt Group 2.

The following coverage points were configured:

- Point1: *2621*, Rng: 2 is used in this example.
- Point2: *2105*, Rng: 2 is used in this example.
- Point3: *2101,* Rng: 2 is used in this example.

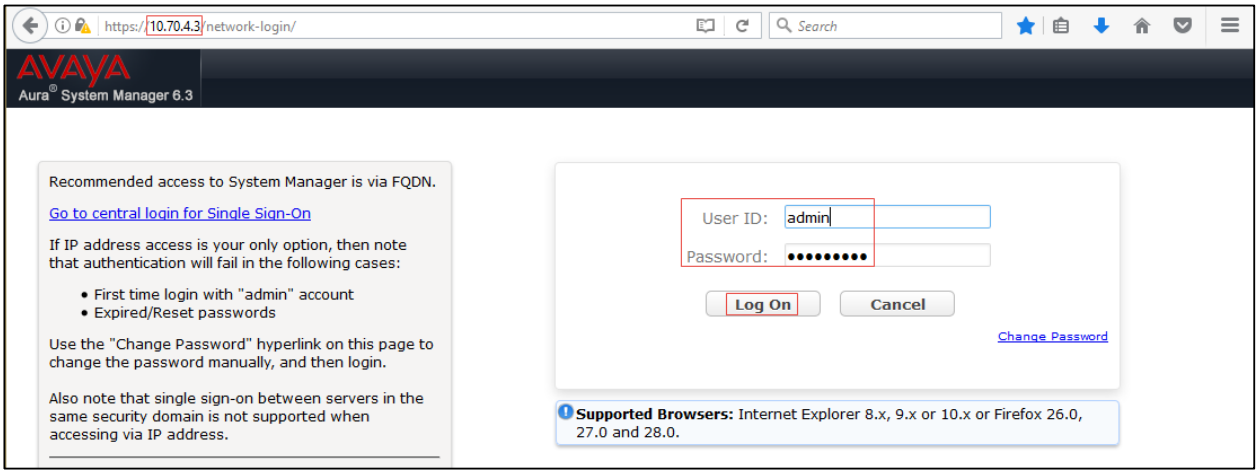
Avaya Aura CM: Hunt Group Coverage Path Configuration

```
display coverage path 2
                                 COVERAGE PATH

                   Coverage Path Number: 2
     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: 2621           Rng: 2  Point2: 2105           Rng: 2
  Point3: 2101           Rng: 2  Point4:
  Point5:                        Point6:
```

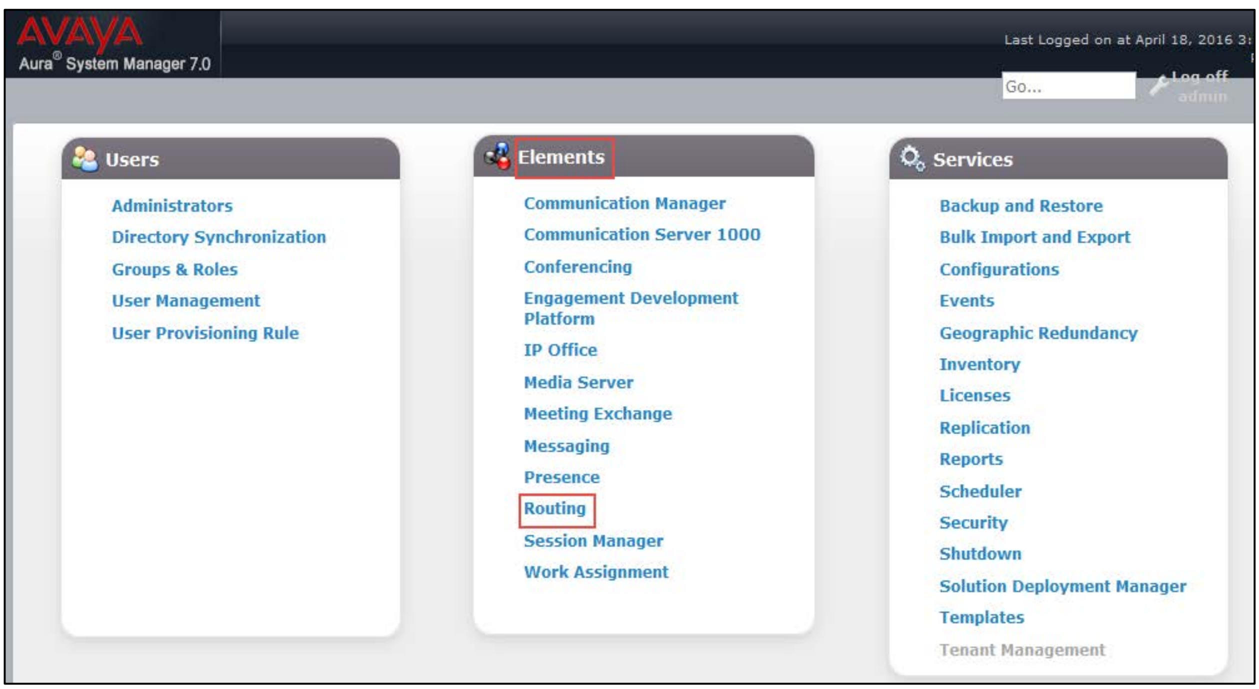
Avaya Aura Session Manager Configuration

1. Access Avaya Aura System Manager Web login screen via https://<IP Address/FQDN>. IP address *10.70.4.3* was used in this example.
2. Log in with the User Id as **admin** and associated password, and then click **Log on**.

Avaya Aura SM: Login Screen



Avaya Aura SM: Navigation Menu



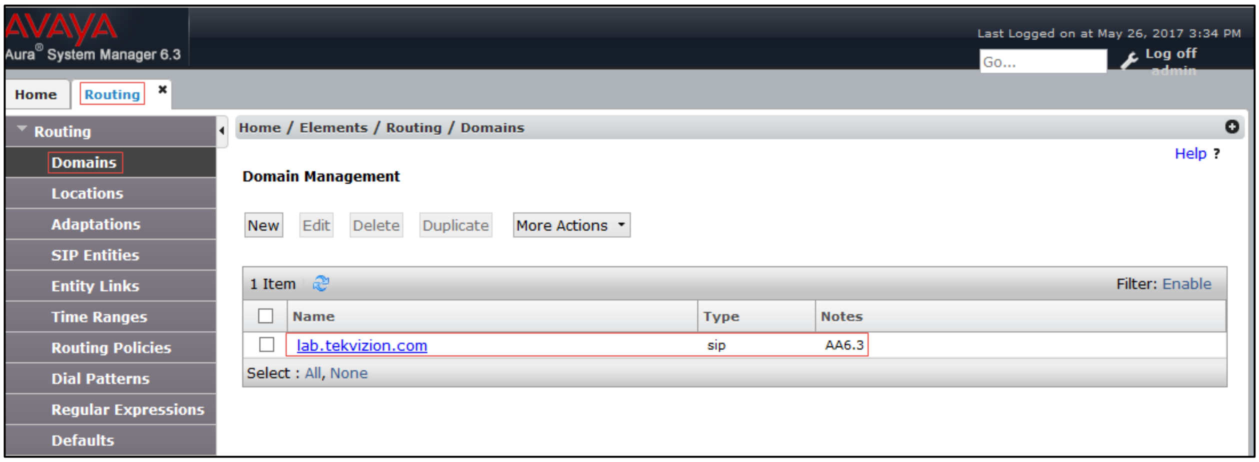
Domain

Create a SIP domain for each domain that Session Manager will need to be aware of, in order to route calls.To configure a domain, perform the following procedure:

1. Navigate to: **Home** > **Routing** > **Domains**.

2. Click **New**.
3. Enter the following information:
    - **Name:** Enter the domain name: *lab.tekvizion.com* is used in this example.
    - **Type:** Select *sip* from the pull-down menu.
    - **Notes:** Add a brief description (optional).
4. Click **Commit** to save.

Avaya Aura SM: Domain Configuration



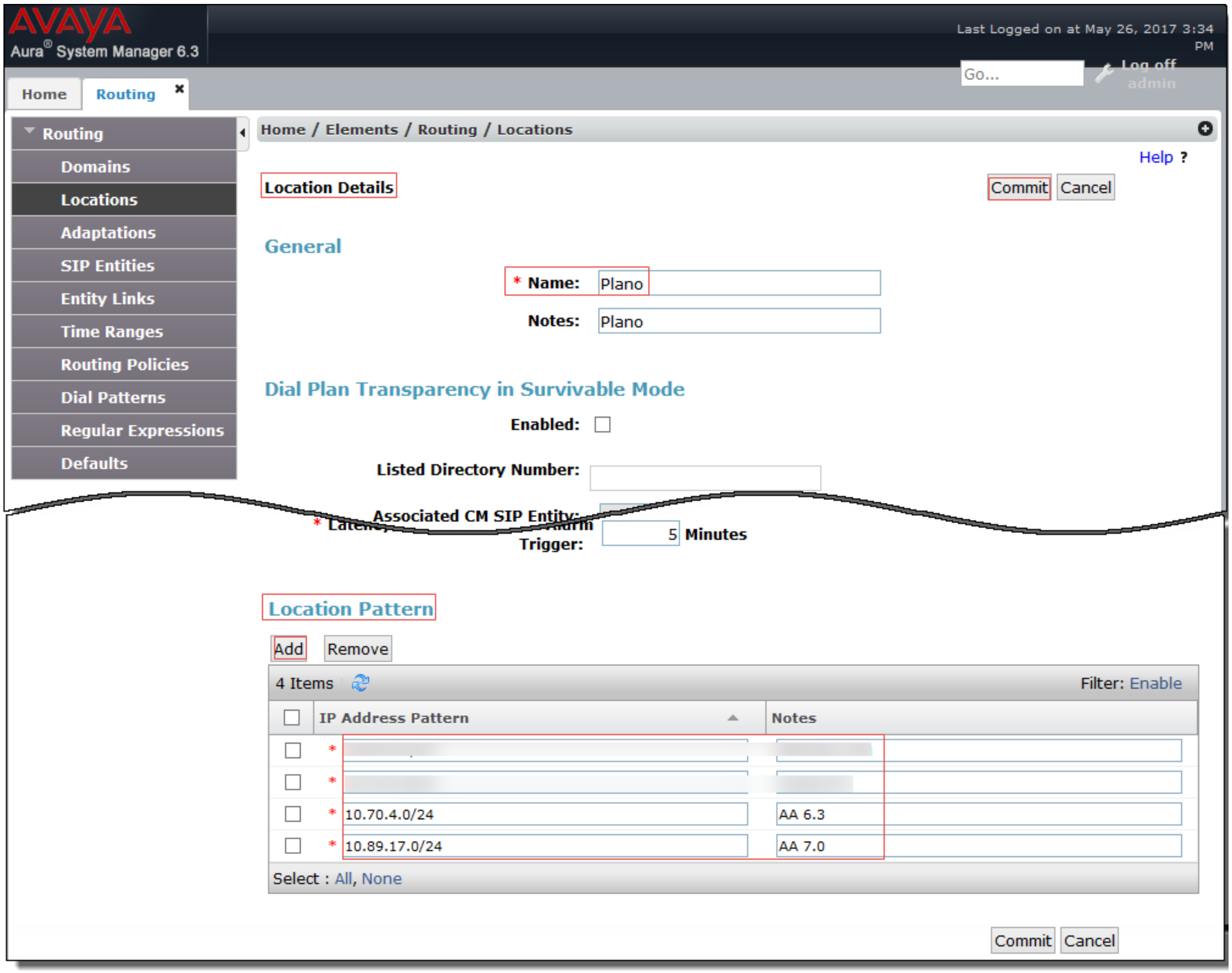
Location

Locations can be used 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, perform the following procedure:

1. Navigate to **Routing** > **Locations**.
2. Click **New**.
3. In the **General** section, enter the following values:
    - **Name:** Enter a descriptive name for the location: *Plano* was used in this example.
    - **Notes:** Add a brief description (optional).
4. Retain all other default configurations.
5. Click **Commit** to save.
6. Under **Location Pattern**, Select **Add** to add **IP Address Patterns** for different networks that are part of the topology:
    - **10.64.0.0/16:** tekVizion
    - **10.70.4.0**: Avaya 6.3
7. Retain all other default configurations.
8. Click **Commit** to save.

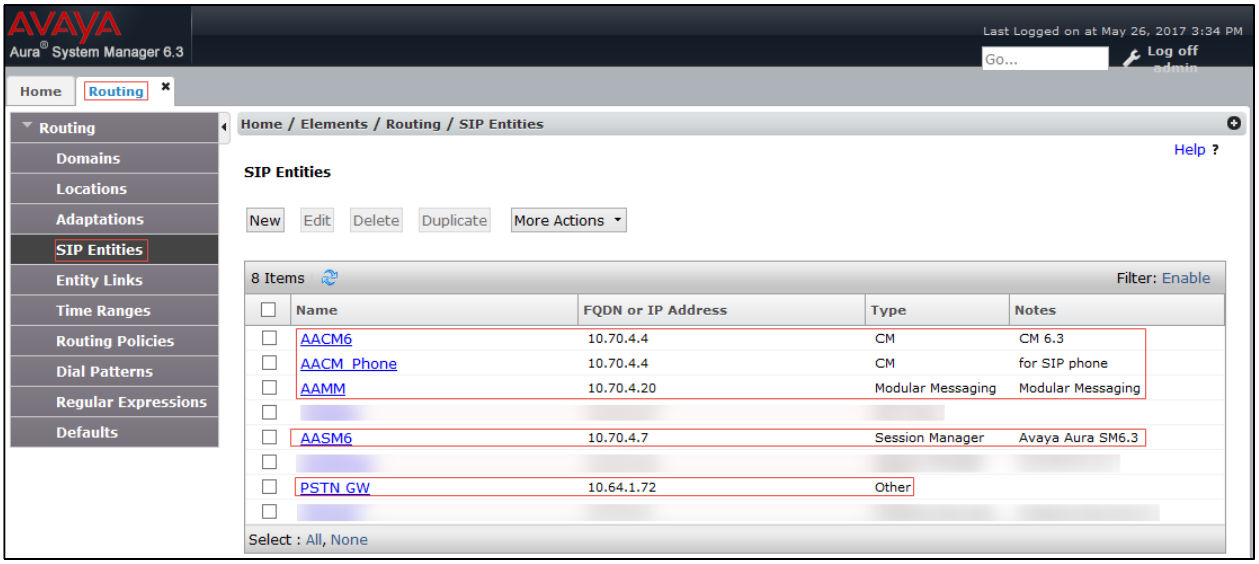
Avaya Aura SM: Location Configuration



SIP Entity and Entity links

A SIP Entity must be added for each network element that is part of the topology and that will participate in the test validation. This includes the Session Manager, Communication Manager, Modular Messaging, and the PSTN gateway.

Avaya Aura SM: SIP Entity Configuration

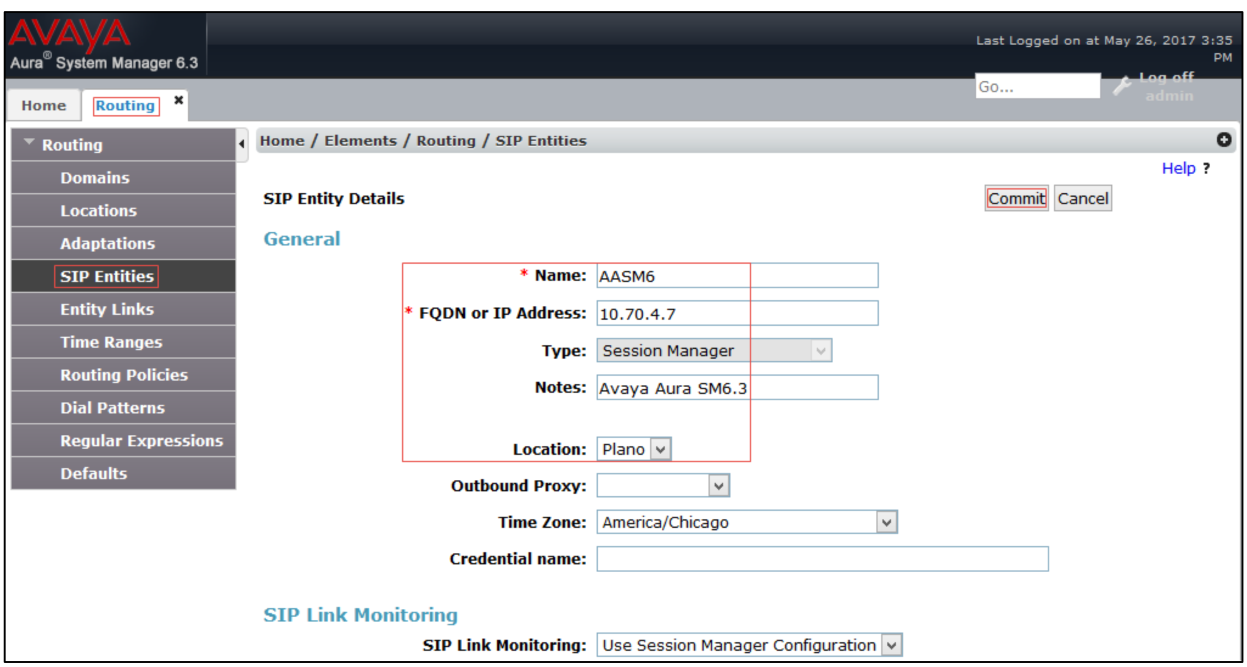


Add SIP Entity for Session Manager

To add a SIP entity, perform the following procedure:

1. Navigate to **Routing** > **SIP Entities**.
2. Click on the **New** button.

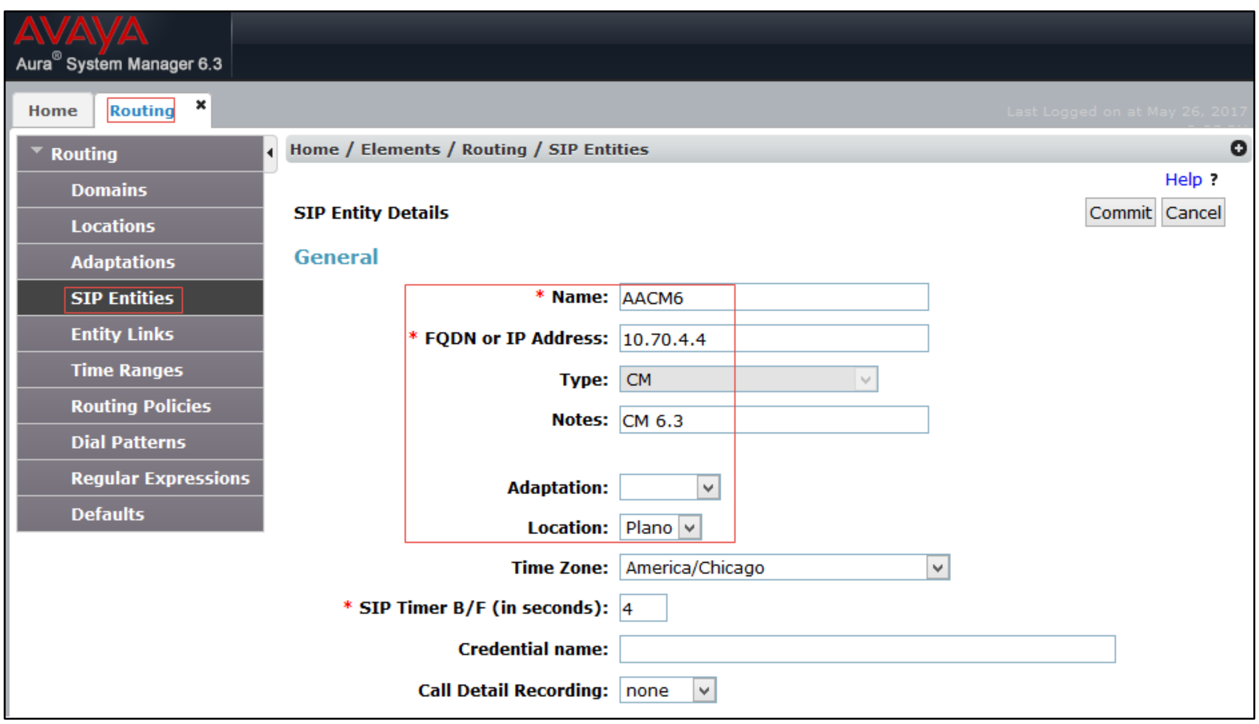
Avaya Aura CM: SIP Entity for SM Configuration



3. In the **General** section, enter the following values
   - **Name:** Enter a descriptive name: *AASM6* was used for the Avaya SM in this example.
   - **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity interface that is used for SIP signaling: *10.70.4.7* was used in this example.
   - **Type:** Enter *Session Manager* for Session Manager.
   - **Location:** Select one of the locations defined previously: *Plano* was used in this example.
   - **Time Zone:** Select the time zone for the location above.
   - To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen.
   - In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:
     - **Port:** Port number on which the CM will listen for SIP requests: *5060* was used in this example.
     - **Protocol:** Transport protocol to be used to send SIP requests: *TCP* was used in this example.

Add SIP Entity for Communication Manager

Avaya Aura SM: SIP Entity for CM Configuration



To add a SIP entity for the Avaya CM, follow this procedure:

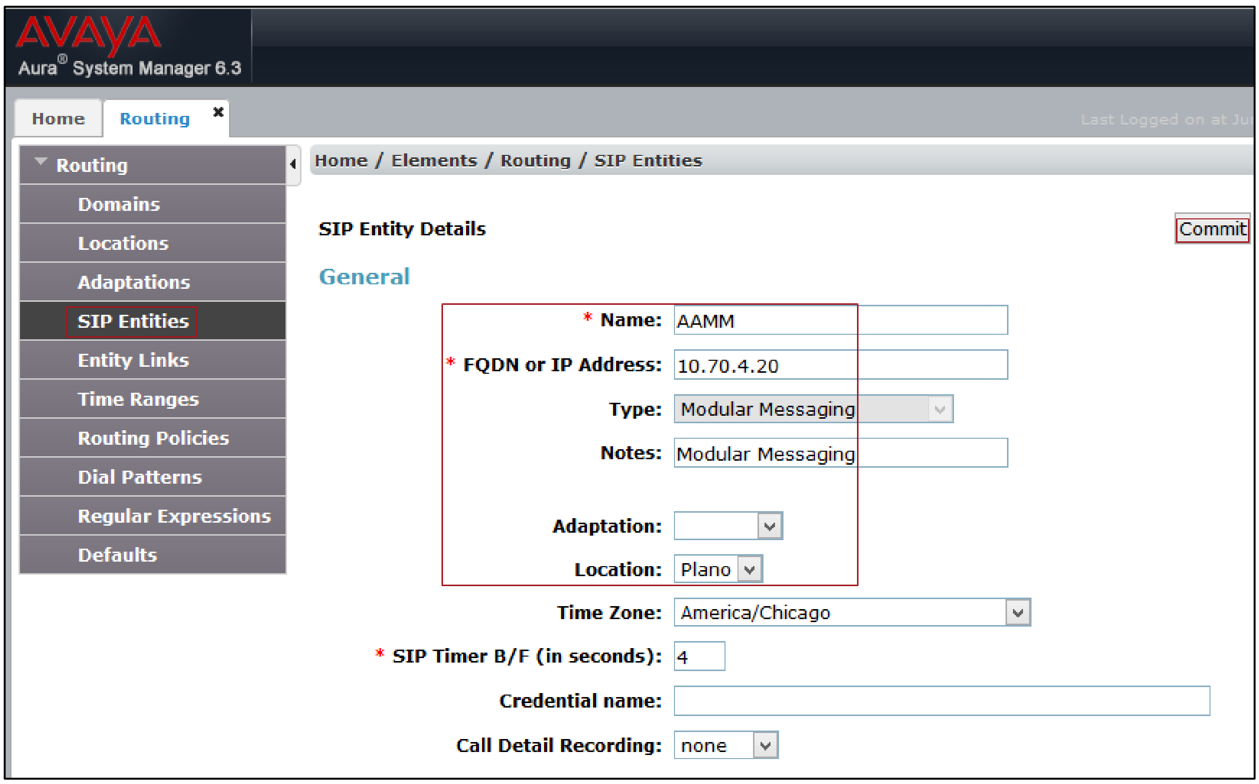
1. Add a SIP entity for the Avaya CM:
   - ***Name:** *AACM6* was used in this example for an SIP entity of Avaya CM.

- * **FQDN or IP address:** *10.70.4.4* was used in this example.
- **Type:** *CM* was used in this example.
- **Notes:** Add a description.
- **Adaptation:** None was used in this example.
- **Location:** Select one of the locations defined previously: *Plano* was used in this example.
- **Time Zone:** Select the time zone for the location above.

2. Click **Commit.**

Add SIP Entity for Avaya Modular Messaging System

Avaya Aura SM: SIP Entity for Modular Messaging Configuration



To add a SIP entity for the Avaya Modular Message System, follow this procedure:

1. Add a SIP entity for the Avaya Modular Messaging:
    - ***Name:** *AAMM* was used in this example for an SIP entity of Avaya Modular Messaging.
    - ***FQDN or IP address:** *10.70.4.20* was used in this example.
    - **Type:** *Modular Messaging* was used in this example.
    - **Notes:** Add a description.
    - **Adaptation:** None was used in this example.

- **Location:** Select one of the locations defined previously. *Plano* was used in this example.
- **Time Zone:** Select the time zone for the location above.

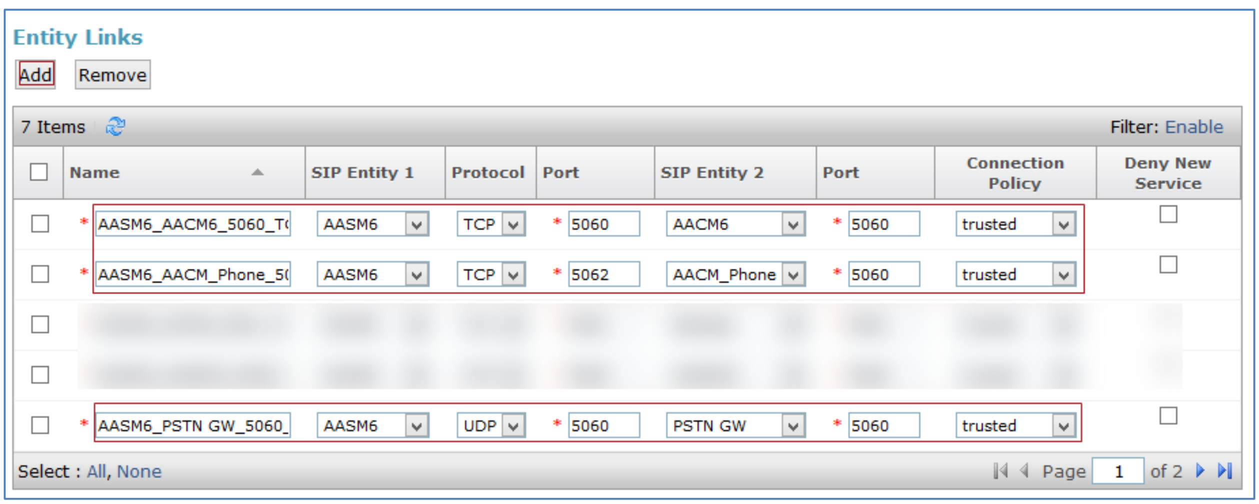
2. Click **Commit**.

Add Entity Links

To configure the SIP entity link for the SM, perform the following procedure:

1. Under **Entity Links**, click **Add**.
   - **SIP Entity 1**: Select *AASM6,* which is configured in the previous step from the drop-down menu.
   - **SIP Entity 2**: Leave the default value *AACM6*.
   - **Protocol**: *TCP* was used in this example.
   - **Ports**: Set both Ports to *5060*.
   - **Connection Policy**: *trusted* was used in this example.
   - Retain all other default configurations.
2. Click **Commit.**

Avaya Aura CM: SIP Entity Link for SM Configuration

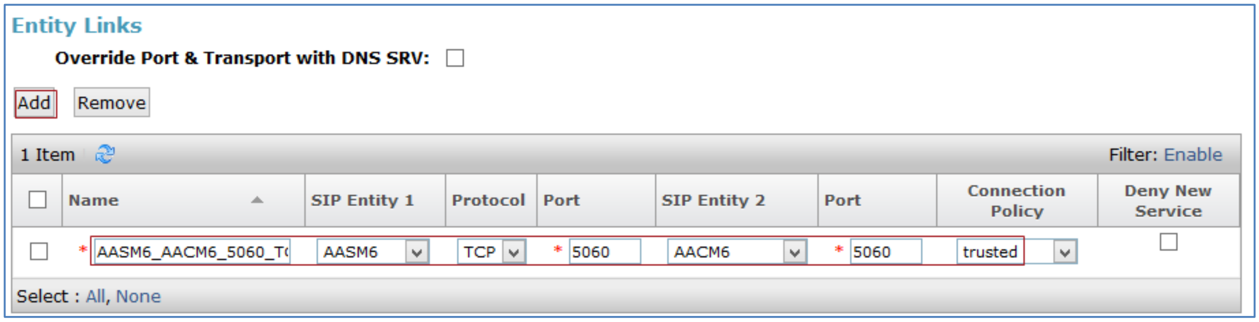


To configure the entity link for the CM, perform the following procedure:

1. Under **Entity Links**, click **Add**.
   - **SIP Entity 1**: Select *AASM6*, which is configured in the previous step from the drop-down menu.
   - **SIP Entity 2**: Leave the default value *AACM6*.
   - **Protocol**: *TCP* was used in this example.
   - **Ports**: Set both Ports to *5060*.
   - **Connection Policy**: *trusted* was used in this example.
   - Retain all other default configurations.

2. Click **Commit**.

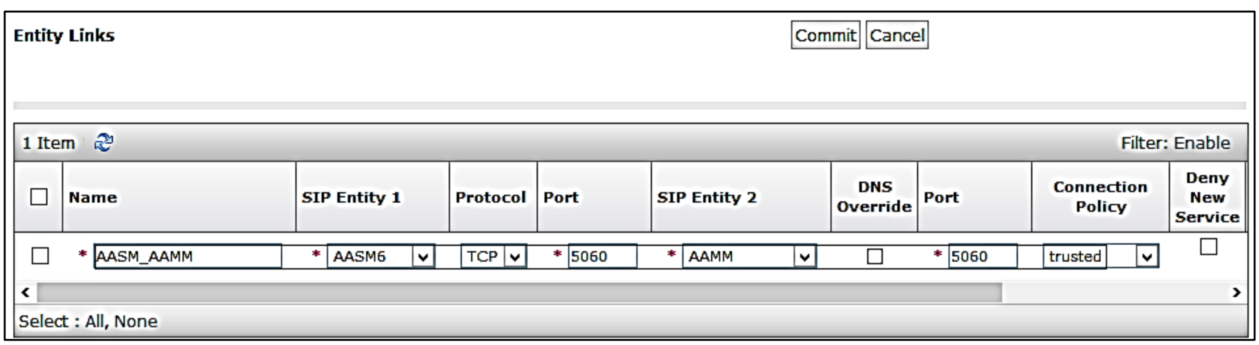
Avaya Aura CM: SIP Entity Link for CM Configuration



To configure the entity link for the Modular Messaging, perform the following procedure:

1. Under **Entity Links**, click **Add**.
    - **SIP Entity 1**: Select *AASM6*, which is configured in the previous step from the drop-down menu.
    - Set **SIP Entity 2**: Leave the default Value *AAMM*.
    - Set **Protocol**: *TCP* was used in this example.
    - Set **Ports**: Set both Ports to *5060*.
    - Set **Connection Policy**: *trusted* was used in this example.
    - Retain all other default configurations.
2. Click **Commit**.

Avaya Aura CM: SIP Entity Link for Modular Messaging Configuration



Routing Policy

Routing Policies describe the conditions under which calls are routed to the SIP entities. Three routing policies were added for this test: one for Communication Manager, one for voice mail, and one to the PSTN GW.

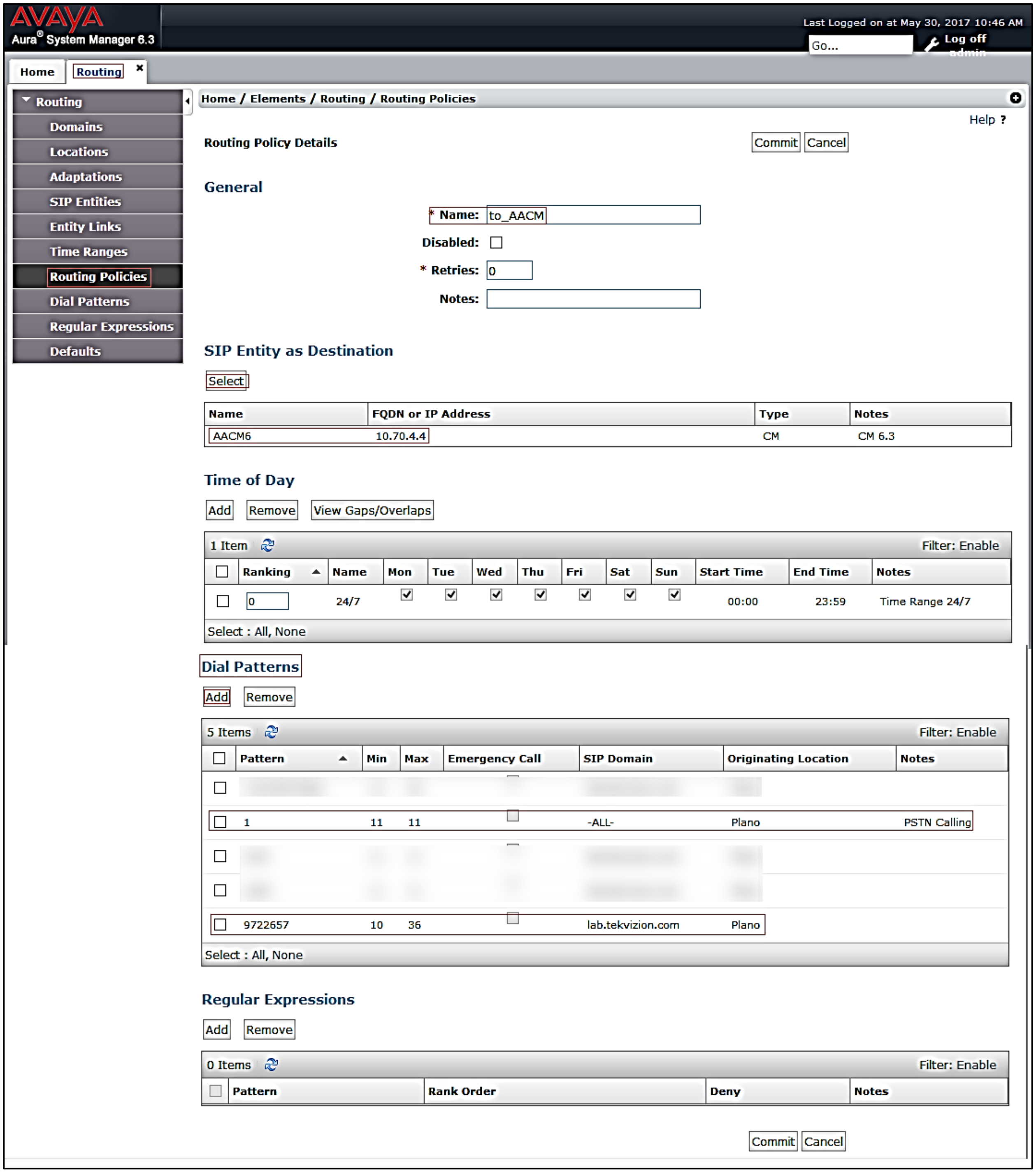
Routing Policy to Communication Manager

To add a routing policy for Avaya CM, perform the following procedure:

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

3. In the **General** section, enter the following values:
    - **Name:** *to_AACM* was used in this example.
    - **SIP Entity as Destination:** Select the Avaya CM. *AACM6* was used in this example.
    - Retain all other default configurations.
4. Add the following dial patterns that can be routed using this policy:
    - For PSTN calling:
        - Select **Pattern:** *1*
        - Select **Min:** *11*
        - Select **Max:** *11*
    - For calling the 10 digit DID of Avaya and Crestron Mercury devices:
        - Select **Pattern:** *9722657*
        - Select **Min:** *10*
        - Select **Max:** *36*
5. Click **Commit**.

Avaya Aura SM: Routing Policy to Communication Manager Configuration



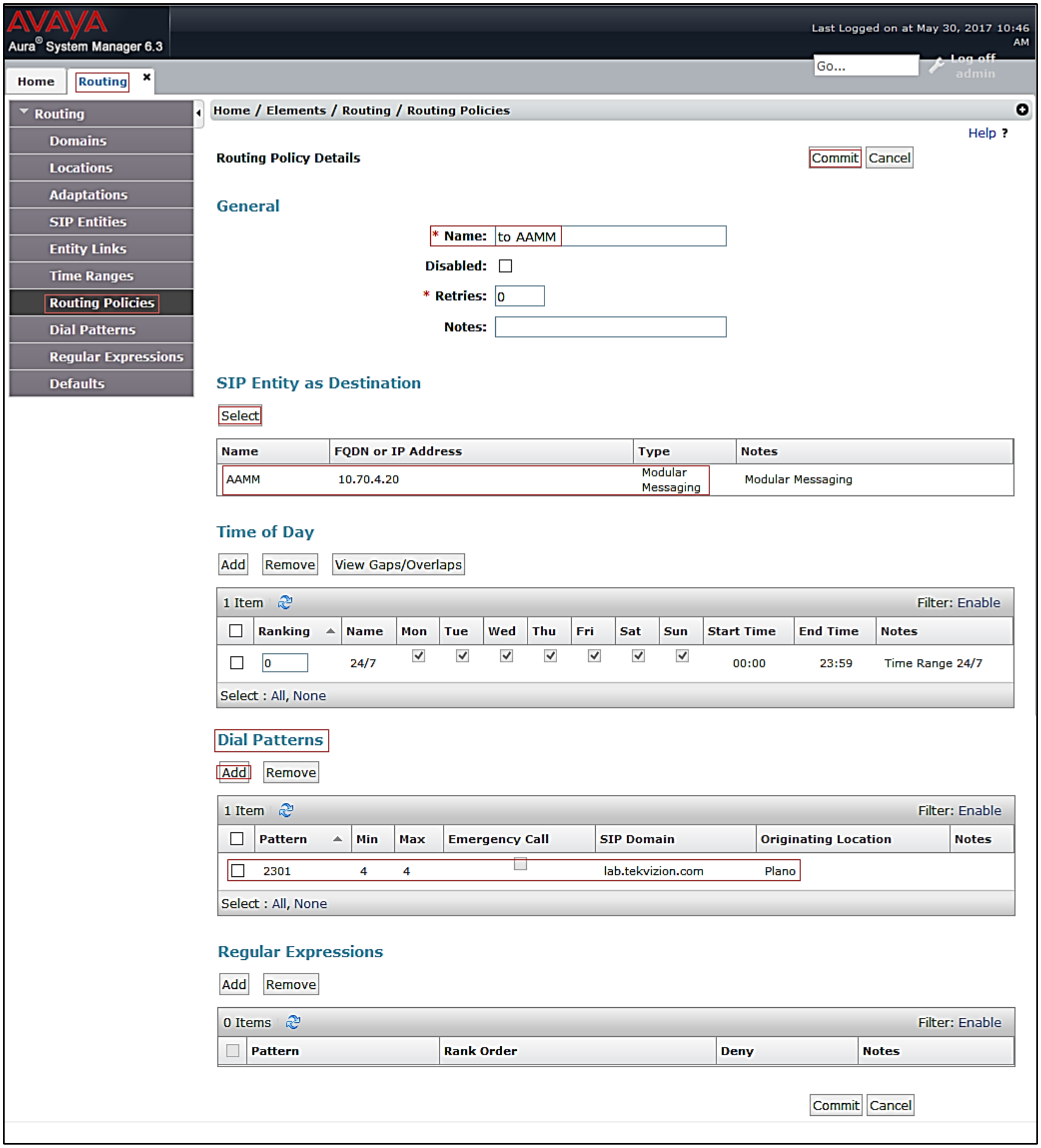
Routing Policy to Avaya Modular Messaging

To add a routing policy for Avaya Modular Messaging, perform the following procedure:

1. Navigate to **Routing** > **Routing Policies**
2. Click on the **New** button

3. In the **General** section, enter the following values:
    - **Name:** *AAMM* is used in this example.
    - **SIP Entity as Destination:** Select the Avaya MM: *AAMM* was used in this example.
    - Retain all other default configurations.
4. Add the dial pattern that can be routed using this policy for voice mail:
    - Select **Pattern:** *2301*
    - Select **Min:** *4*
    - Select **Max:** *4*
5. Click **Commit**.

Avaya Aura SM: Routing Policy to Avaya Modular Messaging Configuration



Routing Policy to PSTN GW

To add a routing policy for the PSTN GW, follow this procedure:

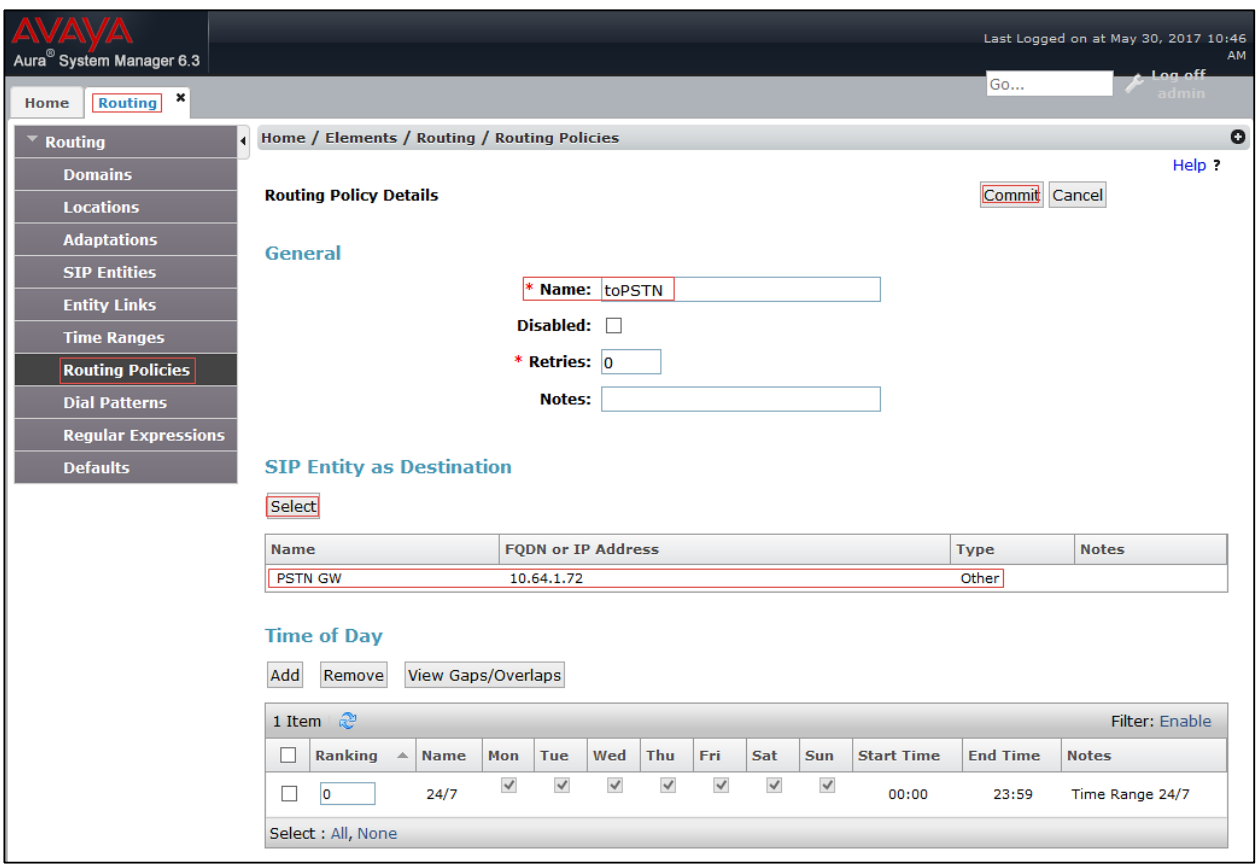
1. Navigate to **Routing** > **Routing Policies**.
2. Click **New**.

3. In the **General** section, enter the following values:
   - **Name:** *toPSTN* is used in this example.
   - **SIP Entity as Destination:** Select the PSTN GW: *PSTN GW* used in this example.
   - Retain all other default configurations.
4. Add the following Dial patterns that can be routed using this policy:
   - *1*
   - *1800*
   - *1866*
   - *1877*
   - *188*
   - *214*
   - *972352*

NOTE: These are the starting digits of all PSTN numbers used in this example.

5. Click **Commit**.

Avaya Aura SM: Routing Policy to PSTN GW Configuration (1/2)



Avaya Aura SM: Routing Policy to PSTN GW Configuration (2/2)

Dial Patterns

Add Remove

8 Items   Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
1	11	11		-ALL-	Plano	PSTN Calling
1800	11	36		-ALL-	Plano	
1866	11	36		-ALL-	Plano	
1877	11	11		-ALL-	Plano	
1888	11	11		-ALL-	Plano	
214	10	10		lab.tekvizion.com	Plano	
972352	6	36		lab.tekvizion.com	Plano	
[illegible]	[illegible]	[illegible]		[illegible]	[illegible]	[illegible]

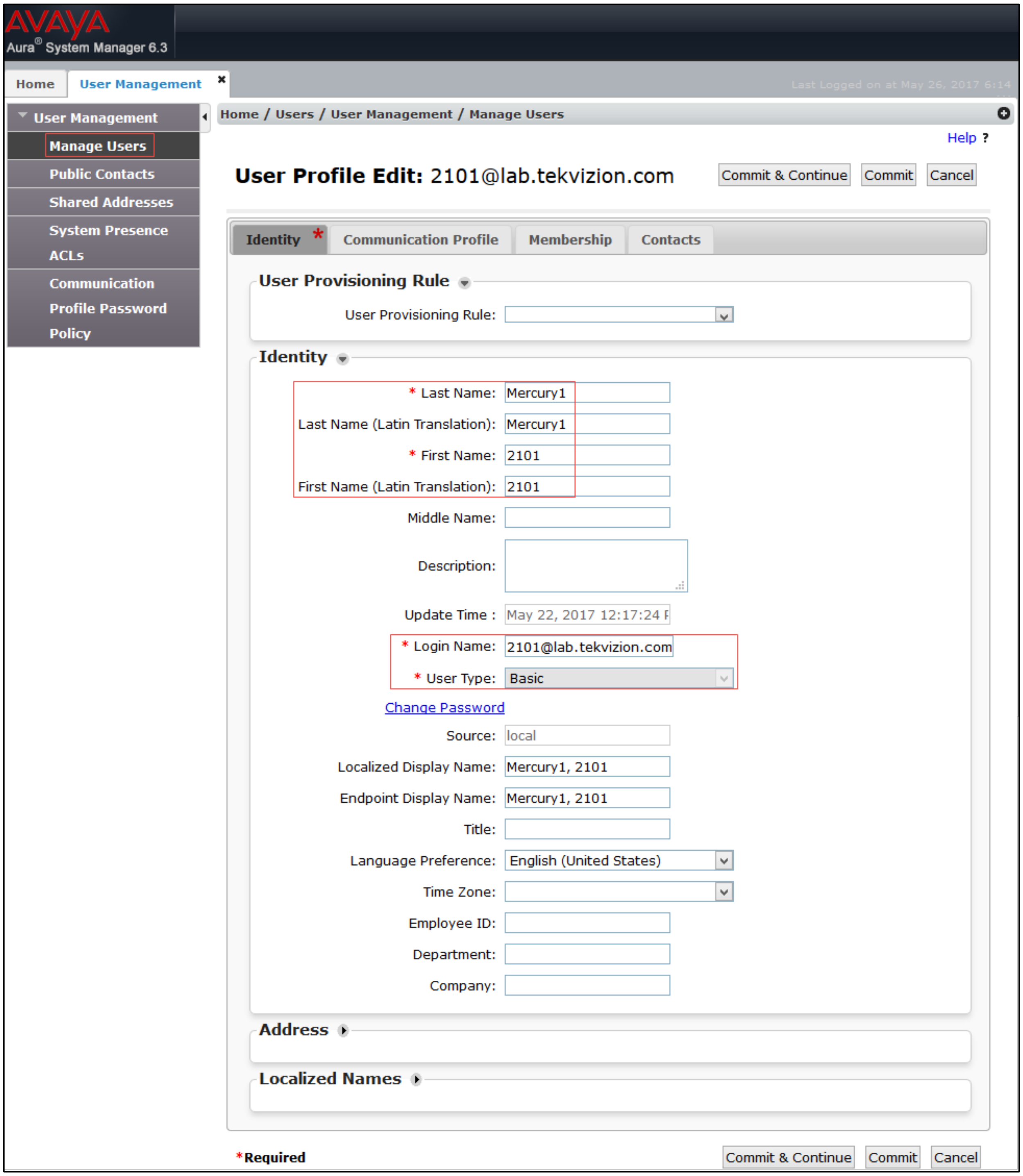
Select : All, None

Configure User for Each Device/Phone

A user was configured for each phone or Crestron device used in the example. To configure a user for each device/phone, follow this process:

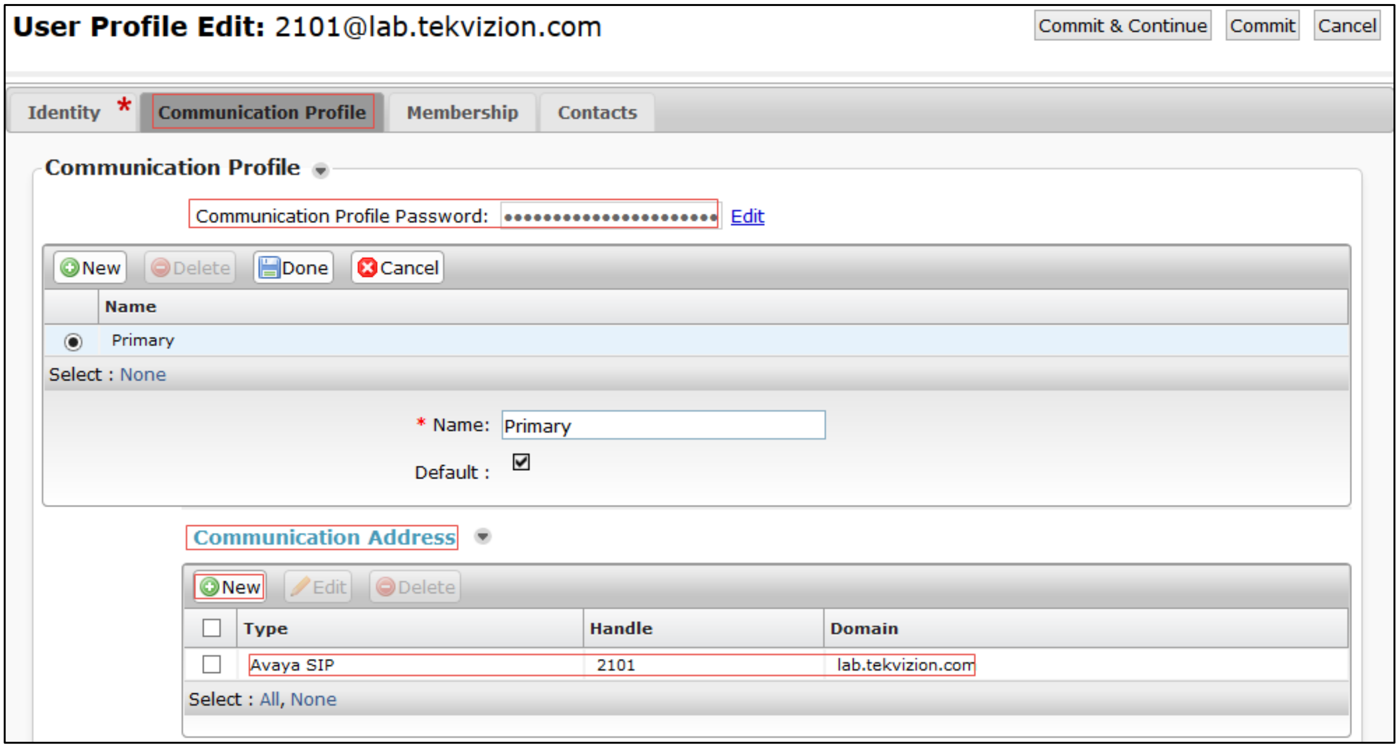
1. Navigate to **Home** > **User Management** > **Manage Users**.
2. Click **Add New**. The User Profile configuration window appears.

Avaya Aura CM: Phone Configuration (1/4)



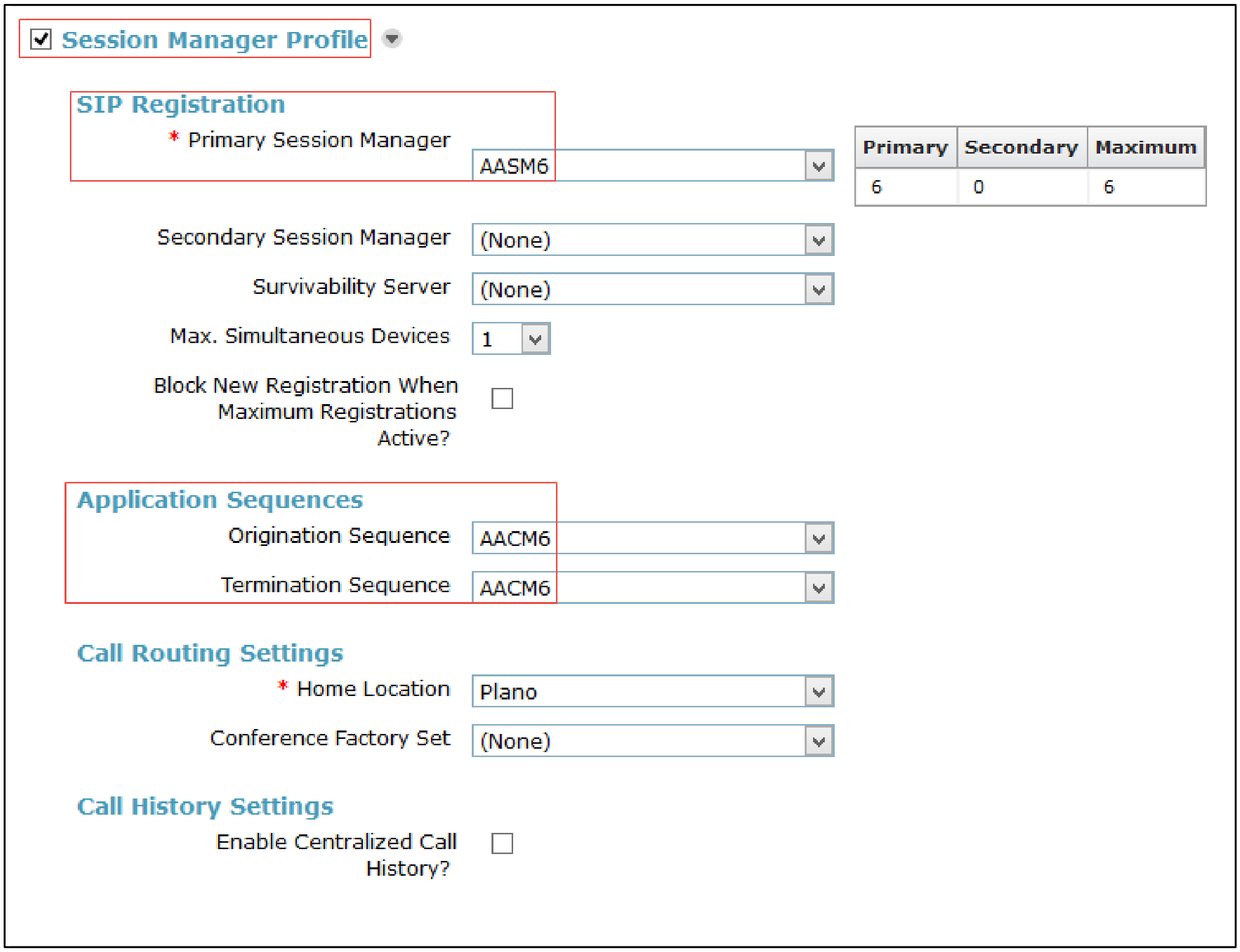
3. Configure **Last Name** and **First Name:** *Mercury1 2101* was used in this example.
4. Configure **Login Name:** *2101@lab.tekvizion.com* was used in this example.
5. Select the **Communication Profile** tab.

Avaya Aura CM: Phone Configuration (2/4)



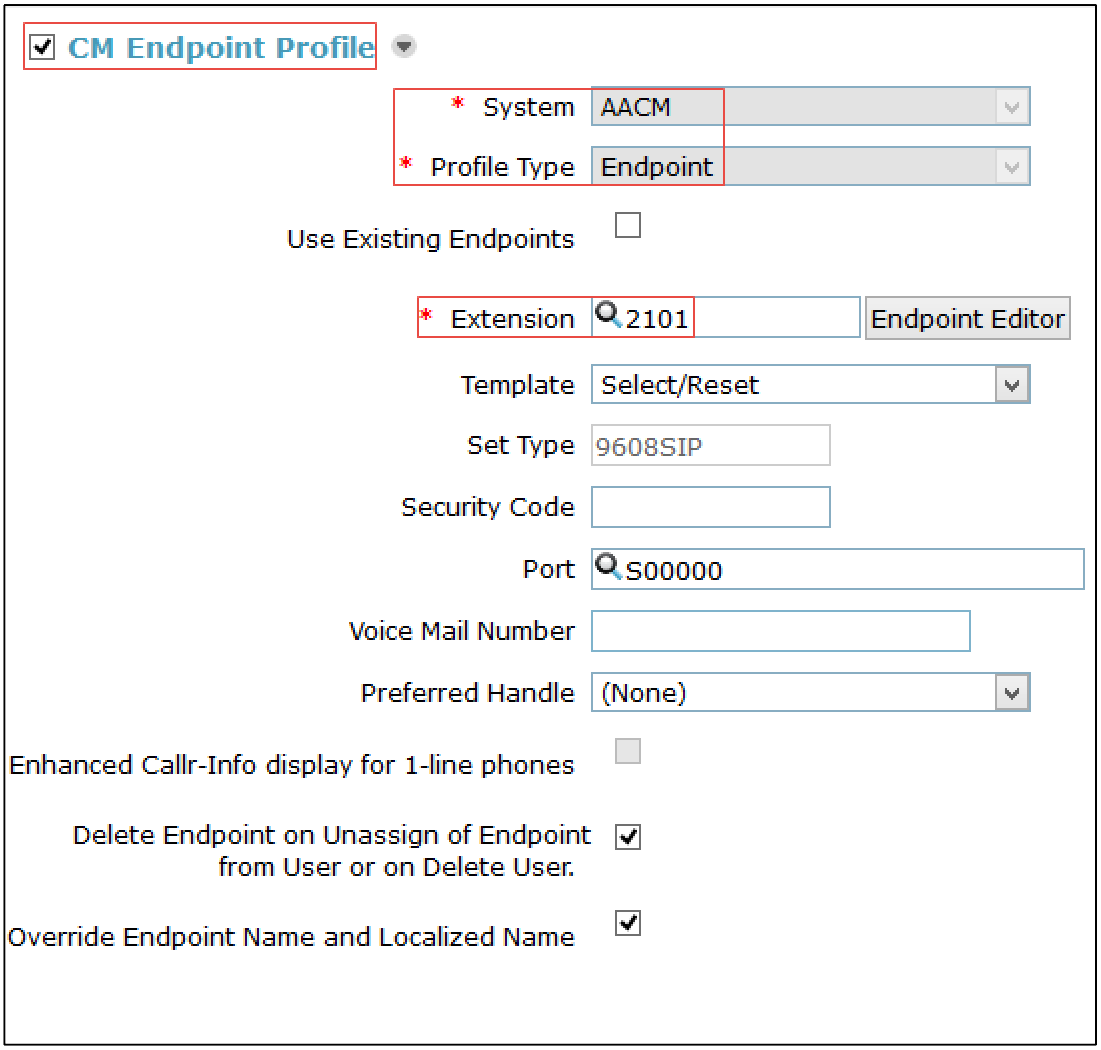
6. Configure **Communication Profile Password:** Enter the desired password for the SIP user to use for registration.
7. Confirm the password.
8. Scroll down to the Communication Address subsection, and click **New** to add a new address.
   - **Type**: *Avaya SIP*
   - **Handle** and **Domain** Address: *2101@lab.tekvizion.com*
9. Check the **Session Manager Profile** check box and configure as follows:
   - **SIP registration:** In **Primary Session Manager**: *AASM6* was used in this example.
   - **Application Sequences:**
     - **Origination Sequence:** *AACM6* was used in this example.
     - **Termination Sequence:** *AACM6* was used in this example.

Avaya Aura CM: Phone Configuration (3/4)



10. Check the **CM Endpoint Profile** check box and configure as follows:
    - Configure **System:** *AACM* was used in this example.
    - Configure **Profile Type**: *Endpoint* was used in this example.
    - Configure **Extension:** *2101* was used in this example.
11. Click **Commit**.

Avaya Aura CM: Phone Configuration (4/4)



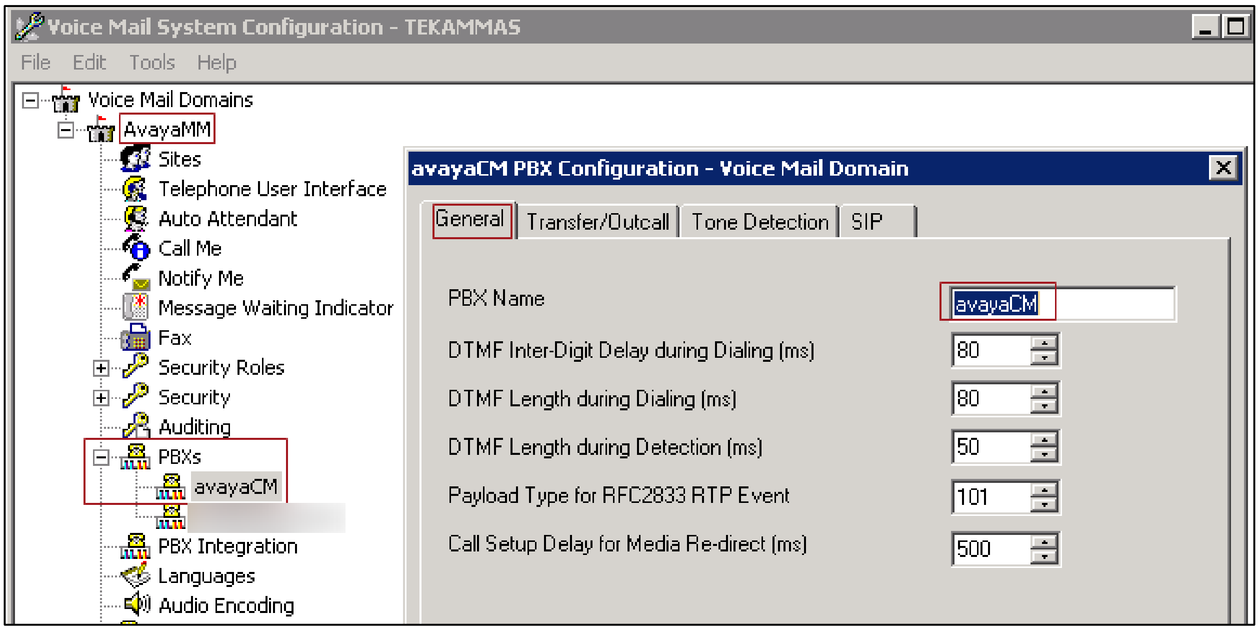
Avaya Modular Messaging

This section describes the configuration related to enabling voice mail for the user.

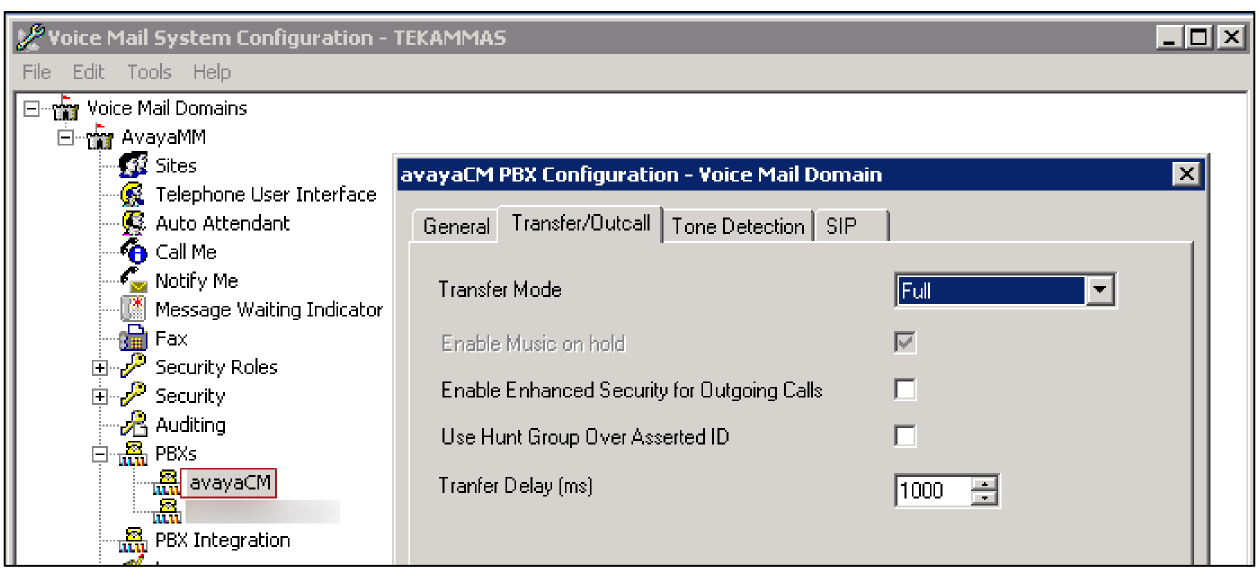
Integration with the Avaya Aura System

It is assumed that the basic configuration and integration of the Messaging Application Server with the Avaya Aura System is complete. The following screen shots outline the important configurations with respect to this example:

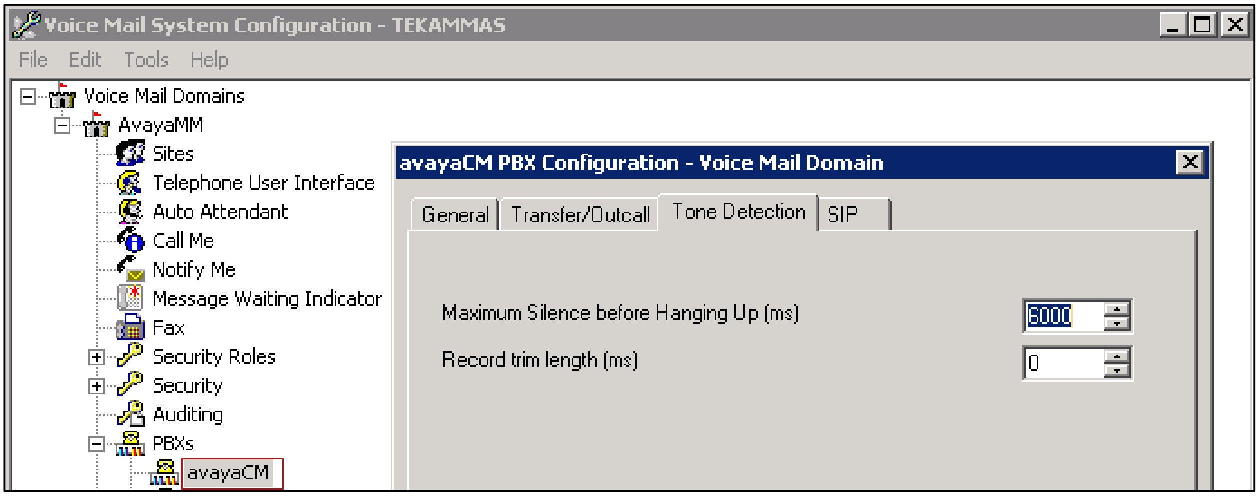
Avaya Aura Modular Messaging: PBX Configuration: Voice Mail Domain: General Tab



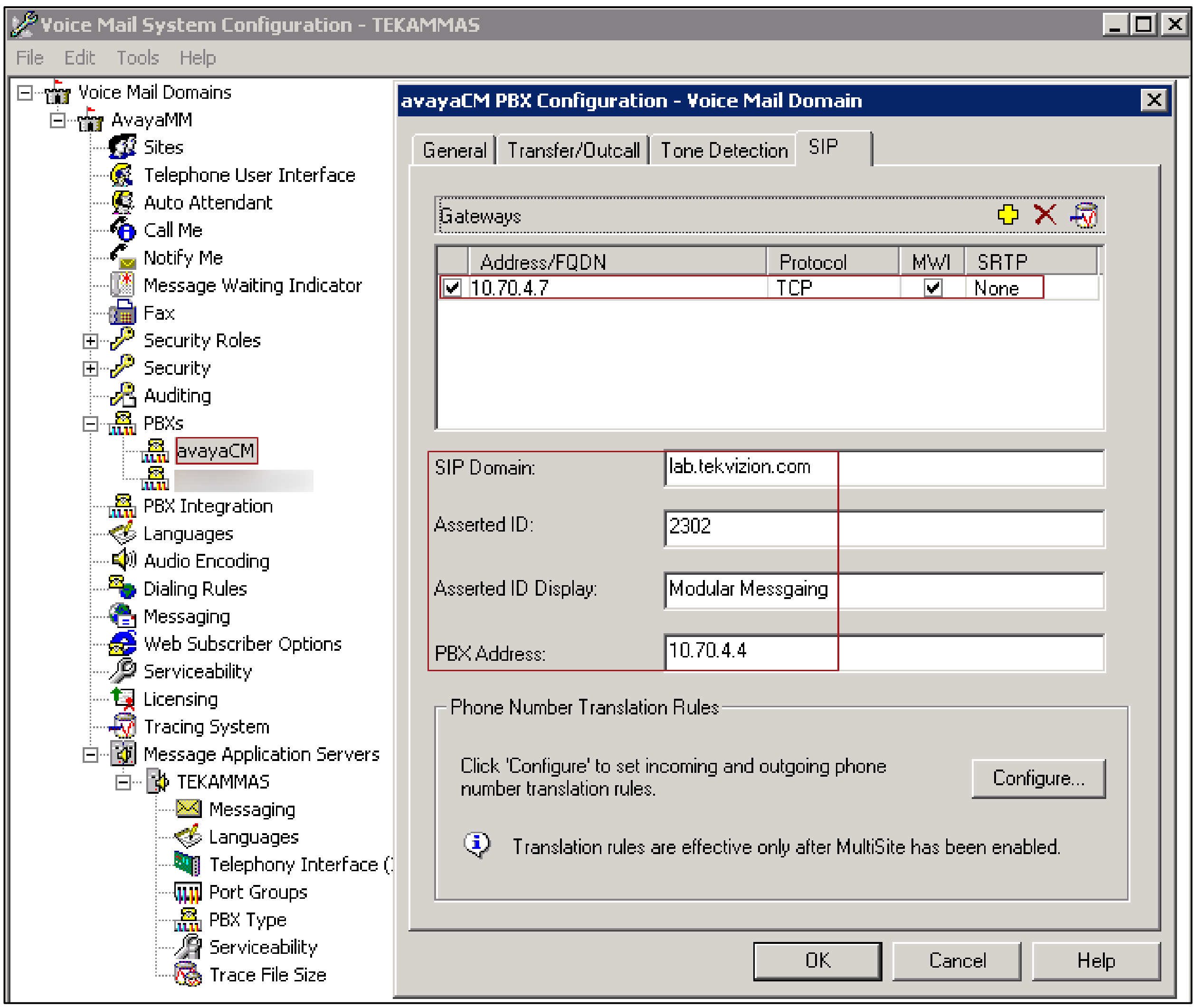
Avaya Aura Modular Messaging: PBX Configuration: Voice Mail Domain" Transfer/Outcall Tab



Avaya Aura Modular Messaging: PBX Configuration: Voice Mail Domain: Tone Detection Tab



Avaya Aura Modular Messaging: PBX Configuration: Voice Mail Domain: SIP Tab



Add a User on the Avaya Modular Messaging System

Access the Aura Modular Messaging Administration GUI via a web browser using its IP address. Log in using the appropriate credentials.

To add a user to this voice mail system, perform the following procedure:

1. Navigate to Messaging Administration > Subscriber Management.
2. Enter the extension of the Crestron Mercury device against Local Subscriber Mailbox Number: *2621* was used in this example.
3. Click Add or Edit.
4. Fill out details as shown below and click Save.

Avaya Aura Modular Messaging: Messaging Administration: Subscriber Management: Add User (1/4)

Avaya Aura Modular Messaging: Messaging Administration: Subscriber Management: Add User (2/4)

Edit Local Subscriber

BASIC INFORMATION
*** (Required Fields)**

*Last Name	Mercury2	First Name	SIP
*Password		*Mailbox Number	2621
*Numeric Address	2621	PBX Extension	2621
*Class Of Service	0 - test	*Community ID	1

SUBSCRIBER DIRECTORY

Email Handle	2621 @tekammss.tekvizion.com	Telephone Number	
Common Name	Mercury DUT2	ASCII Version of Name	DUT2, Mercury

Avaya Aura Modular Messaging: Messaging Administration: Subscriber Management: Add User (3/4)

SUBSCRIBER SECURITY

Immediately Expire Password?	no	Is Mailbox Locked?	no

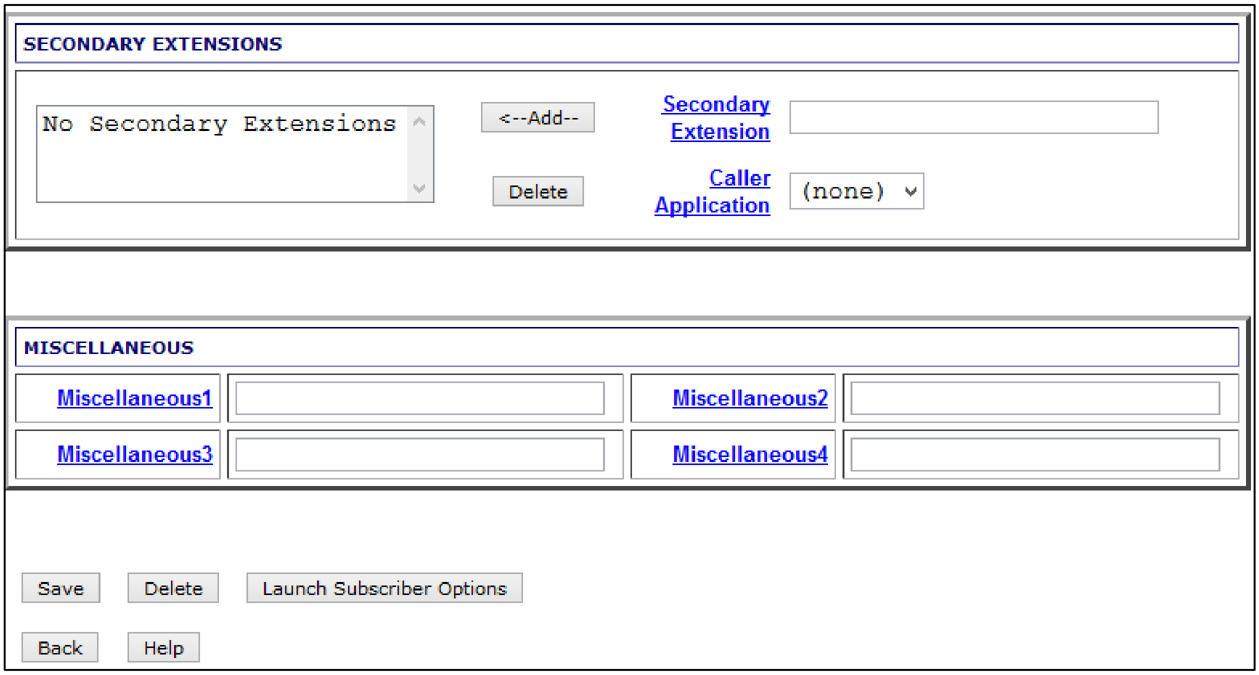
MAILBOX FEATURES

Personal Operator Mailbox		Personal Operator Schedule	Always Active
VoiceMail Enabled	yes	Intercom Paging	paging is off

TUI MESSAGE ORDER

TUI New Message Order	urgent first then newest	TUI Saved Message Order	urgent first then newest
TUI Deleted Message Order	urgent first then newest	TUI Admin Message Order	urgent first then newest

Avaya Aura Modular Messaging: Messaging Administration: Subscriber Management: Add User (4/4)



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