



# **CCS-UC-1**

## SIP Endpoint with Avaya Aura<sup>®</sup> 6.3 System

Configuration Guide

Crestron Electronics, Inc.

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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.

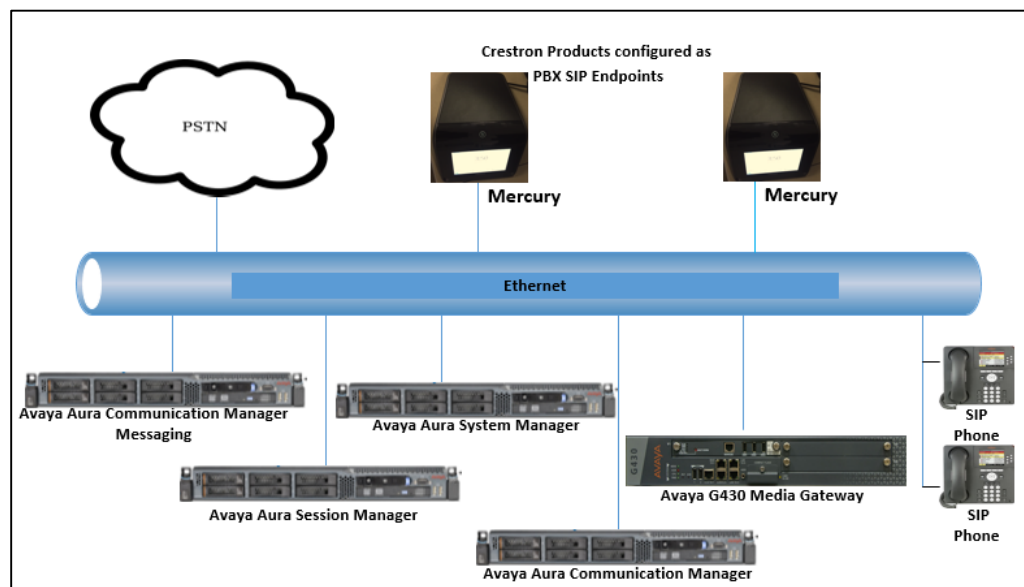
## Audience

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 as 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.

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## 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.

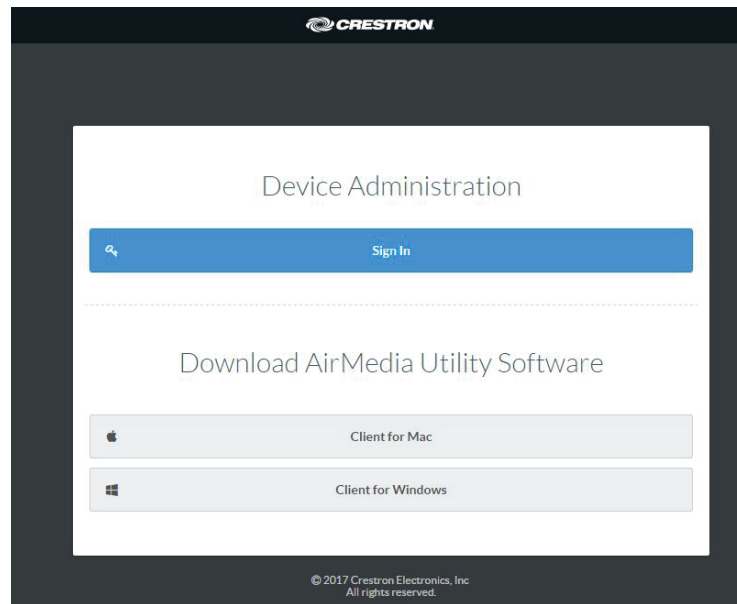
### Configure the device

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.



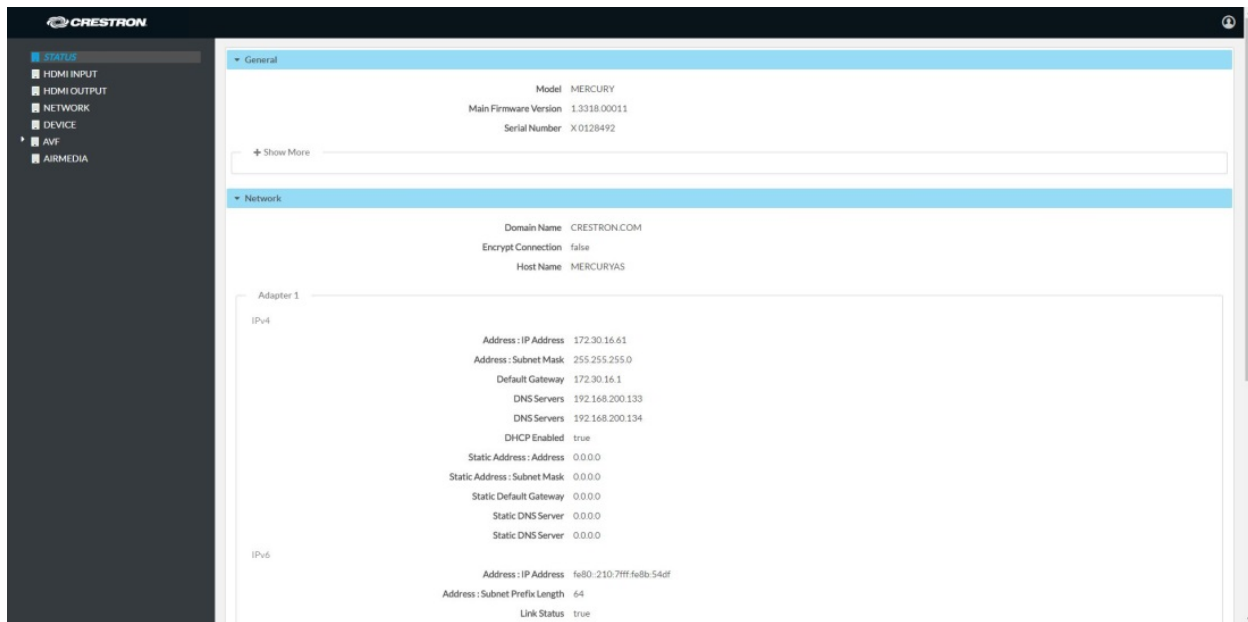
### Crestron Mercury: Login to Web GUI



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](http://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

CRESTRON

STATUS  
HDMI INPUT  
HDMI OUTPUT  
**NETWORK**  
DEVICE  
AVF  
AIRMEDIA

Network Setting

Revert Save Changes

Host Name MERCURY-00107F0522

Domain Name lab.tekvizion.com

Adapter 1

DHCP Enabled  Off (DHCP settings will apply to all adapters)

IP Address 10.70.4.50

Subnet Mask 255.255.255.0

Default Gateway 10.70.4.1

DNS Server 1 10.64.1.3

DNS Server 2 10.64.1.3

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*

The screenshot shows the Crestron web GUI interface for configuring SIP parameters. The left sidebar contains a navigation menu with options: STATUS, HDMI INPUT, HDMI OUTPUT, NETWORK, **DEVICE**, AVF, and AIRMEDIA. The main content area is titled 'SIP Calling' and includes a 'Revert' button and a 'Save Changes' button. The configuration fields are as follows:

Enable SIP	On
Transport Type	UDP
Server IP Address	10.70.4.7
Port	5060
Server Username	2101
Server Password	••••
Server Realm	*
Local Extension	2101
Proxy Server	NONE
SIP Server Status	Online

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 F1d 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**.

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

```
IP CODEC SET
```

```
Codec Set: 1
```

Audio Codec	Silence Suppression	Frames Per Pkt	Packet 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.

```

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

```

## 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:** *sjp* 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:** *rtsp-payload* was used in this example.
- **Direct IP-IP Audio Connections?** *y* was used in this example.

```

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
Incoming Dialog Loopbacks: eliminate                   Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                             RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3                    Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y                               IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n                Initial IP-IP Direct Media? n
Alternate Route Timer(sec): 6
F1=Cancel F2=Refresh F3=Submit F4=Clr F1d 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.

```

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

Incoming Dialog Loopbacks: eliminate          Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                    RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3           Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y                       IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n       Initial IP-IP Direct Media? n
Alternate Route Timer(sec): 6

F1=Cancel F2=Refresh F3=Submit F4=Clr F1d 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
Signal 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
                                                         UII 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:** *sjp* 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
Signalng 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
```

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

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

                                                         Replace Restricted Numbers? n
                                                         Replace Unavailable Numbers? n

    Modify Tandem Calling Number: no

    Show ANSWERED BY on Display? y
```

```
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
2:
3:
4:
5:
6:

BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM  No.  Numbering  LAR
0 1 2 M 4 W      Request      rest
                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

```

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
                                           Intw
1: 3    0
2:
3:
4:
5:
6:
                                           n   user
                                           n   user
                                           n   user
                                           n   user
                                           n   user
                                           n   user

BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM  No. Numbering LAR
0 1 2 M 4 W      Request      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
public-ntwrk
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 String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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.

```

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 String	Total		Route	Call	Node	ANI
	Min	Max	Pattern	Type	Num	Reqd
214242	10	10	2	nat1		n
						n
						n
						n
						n
						n
						n
						n
						n
						n
						n
						n
						n
972265	10	10	2	nat1		n
972598	10	10	2	nat1		n
						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 String	Total		Route	Call	Node	ANI
	Min	Max	Pattern	Type	Num	Reqd
180	11	11	deny	fnpa		n
1800	11	11	2	nat1		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	nat1		n
187	11	11	2	nat1		n
1877	11	11	2	nat1		n
188	11	11	2	nat1		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

```
display ars analysis 12
```

Page 1 of 2

ARS DIGIT ANALYSIS TABLE							
Location: all							
Percent Full: 0							
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd	
120	11	11	deny	fnpa		n	
1200	11	11	deny	fnpa		n	
121	11	11	2	nat1		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 String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI 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
                                Next Path Number:
                                Hunt after Coverage? n
                                Linkage

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

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

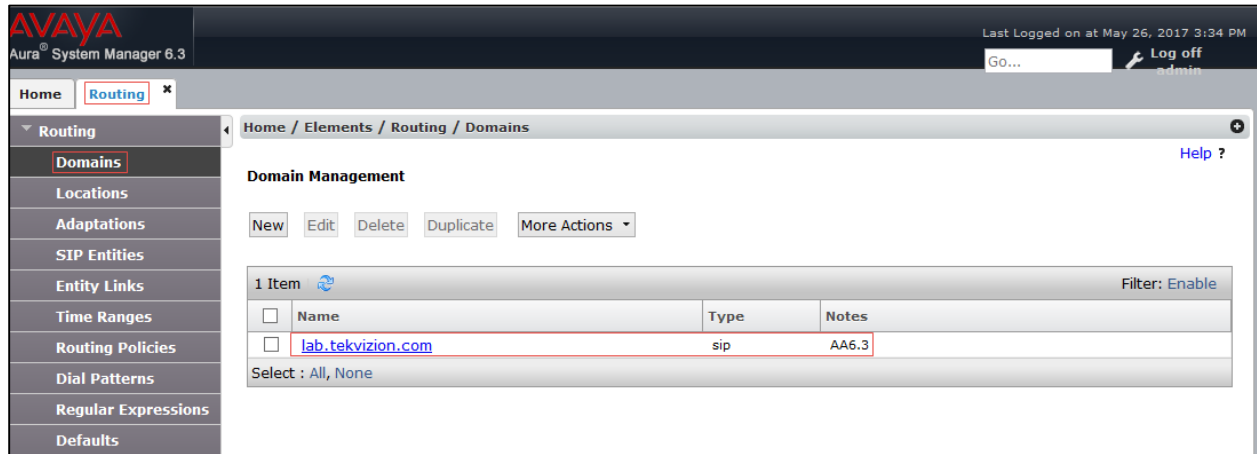
## Domain

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

AVAYA  
Aura System Manager 6.3

Last Logged on at May 26, 2017 3:34 PM

Home Routing

Home / Elements / Routing / Locations

**Location Details** [Commit] [Cancel] Help ?

**General**

\* Name: Plano

Notes: Plano

**Dial Plan Transparency in Survivable Mode**

Enabled:

Listed Directory Number: [ ]

Associated CM SIP Entity: [ ]

Latency: [ ]

Alarm Trigger: 5 Minutes

**Location Pattern**

[Add] [Remove]

4 Items [Filter: Enable]

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* [ ]	[ ]
<input type="checkbox"/>	* [ ]	[ ]
<input type="checkbox"/>	* 10.70.4.0/24	AA 6.3
<input type="checkbox"/>	* 10.89.17.0/24	AA 7.0

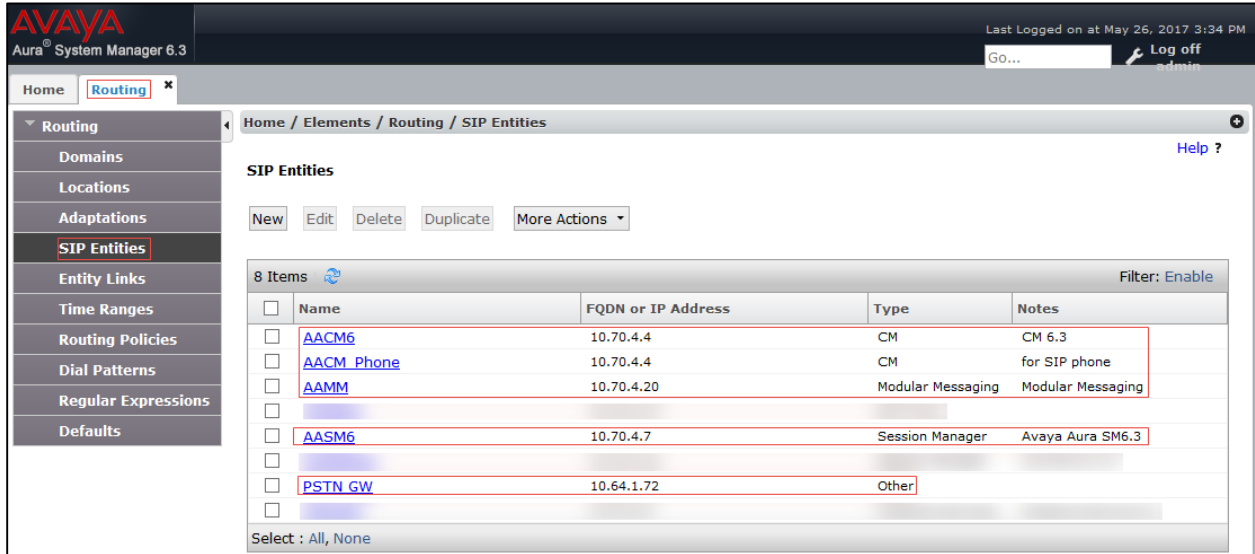
Select : All, None

[Commit] [Cancel]

## 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



AVAYA  
Aura System Manager 6.3

Home / Elements / Routing / SIP Entities

SIP Entities

New Edit Delete Duplicate More Actions

8 Items Filter: Enable

<input type="checkbox"/>	Name	FQDN or IP Address	Type	Notes
<input type="checkbox"/>	AACM6	10.70.4.4	CM	CM 6.3
<input type="checkbox"/>	AACM Phone	10.70.4.4	CM	for SIP phone
<input type="checkbox"/>	AAMM	10.70.4.20	Modular Messaging	Modular Messaging
<input type="checkbox"/>	AASM6	10.70.4.7	Session Manager	Avaya Aura SM6.3
<input type="checkbox"/>	PSTN GW	10.64.1.72	Other	

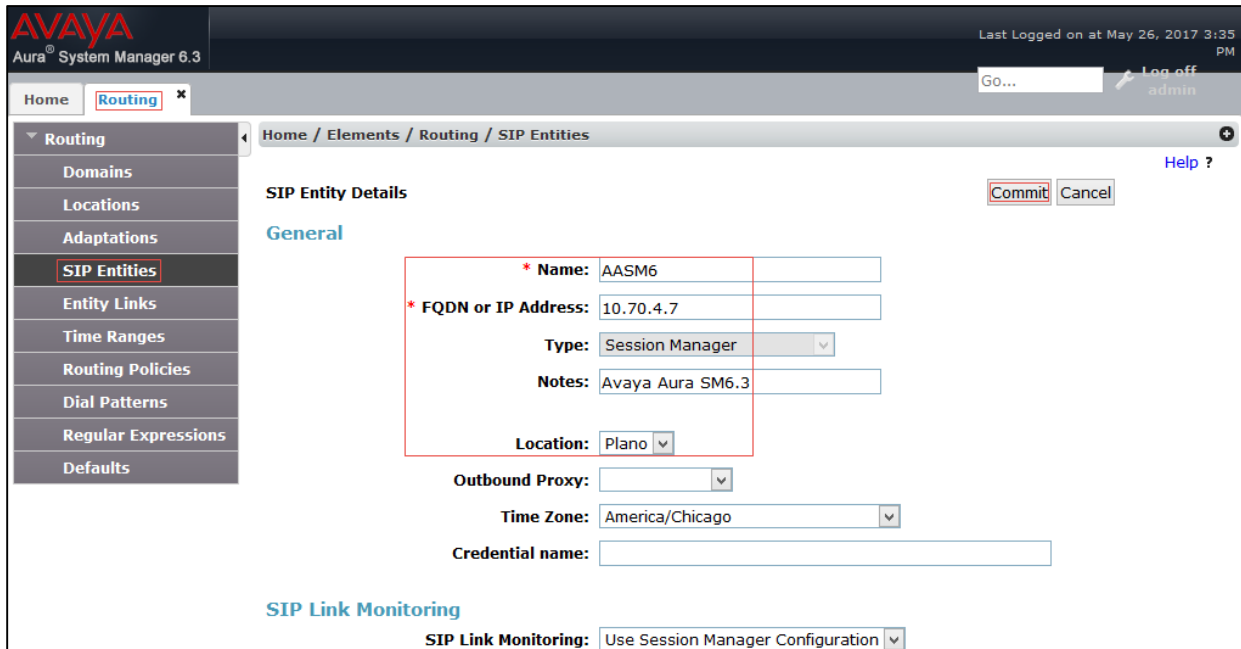
Select : All, None

### 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



AVAYA  
Aura System Manager 6.3

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

General

\* Name: AASM6

\* FQDN or IP Address: 10.70.4.7

Type: Session Manager

Notes: Avaya Aura SM6.3

Location: Plano

Outbound Proxy:

Time Zone: America/Chicago

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager 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*

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'General' section. A red box highlights the following fields in the 'General' section:

- \* Name: AACM6
- \* FQDN or IP Address: 10.70.4.4
- Type: CM
- Notes: CM 6.3
- Adaptation: [dropdown]
- Location: Plano
- Time Zone: America/Chicago
- \* SIP Timer B/F (in seconds): 4
- Credential name: [text field]
- Call Detail Recording: none

Buttons for 'Commit' and 'Cancel' are visible in the top right corner of the configuration area.

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

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with 'SIP Entities' highlighted. The main content area displays the 'SIP Entity Details' form. The 'General' tab is active, and a red box highlights the following fields:

- \* Name:** AAMM
- \* FQDN or IP Address:** 10.70.4.20
- Type:** Modular Messaging
- Notes:** Modular Messaging
- Adaptation:** (empty)
- Location:** Plano

Other visible fields include:

- Time Zone:** America/Chicago
- \* SIP Timer B/F (in seconds):** 4
- Credential name:** (empty)
- Call Detail Recording:** none

A **Commit** button is located in the top right corner of the form area.

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

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* AASM6_AACM6_5060_T1	AASM6	TCP	* 5060	AACM6	* 5060	trusted	<input type="checkbox"/>
<input type="checkbox"/>	* AASM6_AACM_Phone_5060	AASM6	TCP	* 5062	AACM_Phone	* 5060	trusted	<input type="checkbox"/>
<input type="checkbox"/>	* AASM6_PSTN GW_5060_	AASM6	UDP	* 5060	PSTN GW	* 5060	trusted	<input type="checkbox"/>

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

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 6.3', and a 'Log off admin' button. The breadcrumb trail is 'Home / Elements / Routing / Routing Policies'. The left sidebar lists various configuration options, with 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and contains the following sections:

- General:**
  - Name: to\_AACM
  - Disabled:
  - \* Retries: 0
  - Notes: (empty field)
- SIP Entity as Destination:**
  - Select button
  - Table with columns: Name, FQDN or IP Address, Type, Notes. One row is visible: AACM6, 10.70.4.4, CM, CM 6.3.
- Time of Day:**
  - Buttons: Add, Remove, View Gaps/Overlaps
  - 1 Item table with columns: Ranking, Name, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time, End Time, Notes. One row is visible: 0, 24/7, all days checked, 00:00, 23:59, Time Range 24/7.
- Dial Patterns:**
  - Buttons: Add, Remove
  - 5 Items table with columns: Pattern, Min, Max, Emergency Call, SIP Domain, Originating Location, Notes. One row is visible: 1, 11, 11, unchecked, -ALL-, Plano, PSTN Calling.
- Regular Expressions:**
  - Buttons: Add, Remove
  - 0 Items table with columns: Pattern, Rank Order, Deny, Notes.

At the bottom right of the configuration area, there are 'Commit' and 'Cancel' buttons.

*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

**AVAYA**  
Aura® System Manager 6.3

Last Logged on at May 30, 2017 10:46 AM

Home Routing

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

**General**

\* Name: to AAMM

Disabled:

\* Retries: 0

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
AAMM	10.70.4.20	Modular Messaging	Modular Messaging

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

**Dial Patterns**

Add Remove

1 Item Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
2301	4	4	<input type="checkbox"/>	lab.tekvizion.com	Plano	

Select : All, None

**Regular Expressions**

Add Remove

0 Items Filter: Enable

Pattern	Rank Order	Deny	Notes
---------	------------	------	-------

Commit Cancel

*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)*

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and includes a 'Commit' button. The 'General' section contains the following fields:

- Name:** toPSTN
- Disabled:**
- Retries:** 0
- Notes:** (empty text box)

The 'SIP Entity as Destination' section features a 'Select' button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
PSTN GW	10.64.1.72	Other	


The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below is a table with one item:

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

At the bottom of the Time of Day section, there is a 'Select' dropdown menu set to 'All, None'.

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

**Dial Patterns**

8 Items  Filter: [Enable](#)

<input type="checkbox"/>	Pattern ▲	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	1	11	11	<input type="checkbox"/>	-ALL-	Plano	PSTN Calling
<input type="checkbox"/>	1800	11	36	<input type="checkbox"/>	-ALL-	Plano	
<input type="checkbox"/>	1866	11	36	<input type="checkbox"/>	-ALL-	Plano	
<input type="checkbox"/>	1877	11	11	<input type="checkbox"/>	-ALL-	Plano	
<input type="checkbox"/>	1888	11	11	<input type="checkbox"/>	-ALL-	Plano	
<input type="checkbox"/>	214	10	10	<input type="checkbox"/>	lab.tekvizion.com	Plano	
<input type="checkbox"/>	972352	6	36	<input type="checkbox"/>	lab.tekvizion.com	Plano	
<input type="checkbox"/>							

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)

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes 'Home' and 'User Management'. The left sidebar lists 'User Management' with sub-items: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is titled 'User Profile Edit: 2101@lab.tekvizion.com' and includes buttons for 'Commit & Continue', 'Commit', and 'Cancel'. The 'Identity' tab is active, showing a 'User Provisioning Rule' dropdown and an 'Identity' section with the following fields:

- \* Last Name: Mercury1
- Last Name (Latin Translation): Mercury1
- \* First Name: 2101
- First Name (Latin Translation): 2101
- Middle Name: (empty)
- Description: (empty)
- Update Time: May 22, 2017 12:17:24 F
- \* Login Name: 2101@lab.tekvizion.com
- \* User Type: Basic

Below the 'Identity' section is a 'Change Password' link and a 'Source' dropdown set to 'local'. Other fields include 'Localized Display Name: Mercury1, 2101', 'Endpoint Display Name: Mercury1, 2101', 'Title: (empty)', 'Language Preference: English (United States)', 'Time Zone: (empty)', 'Employee ID: (empty)', 'Department: (empty)', and 'Company: (empty)'. At the bottom, there are sections for 'Address' and 'Localized Names', and a '\*Required' legend. The bottom right corner has 'Commit & Continue', 'Commit', and 'Cancel' buttons.

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)*

**User Profile Edit: 2101@lab.tekvizion.com** Commit & Continue Commit Cancel

Identity \* **Communication Profile** Membership Contacts

**Communication Profile**

Communication Profile Password: ..... [Edit](#)

**Name**

Primary

Select : None

\* Name:

Default :

**Communication Address**

<input type="checkbox"/>	Type	Handle	Domain
<input type="checkbox"/>	Avaya SIP	2101	lab.tekvizion.com

Select : All, None

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.

**Session Manager Profile** ▼

**SIP Registration**

\* Primary Session Manager  ▼

Secondary Session Manager  ▼

Survivability Server  ▼

Max. Simultaneous Devices  ▼

Block New Registration When Maximum Registrations Active?

Primary	Secondary	Maximum
6	0	6

**Application Sequences**

Origination Sequence  ▼

Termination Sequence  ▼

**Call Routing Settings**

\* Home Location  ▼

Conference Factory Set  ▼

**Call History Settings**

Enable Centralized Call History?

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**.

**CM Endpoint Profile** ▼

\* System AACM ▼

\* Profile Type Endpoint ▼

Use Existing Endpoints

\* Extension 2101 Endpoint Editor

Template Select/Reset ▼

Set Type 9608SIP

Security Code

Port S00000

Voice Mail Number

Preferred Handle (None) ▼

Enhanced Callr-Info display for 1-line phones

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

Override Endpoint Name and Localized Name

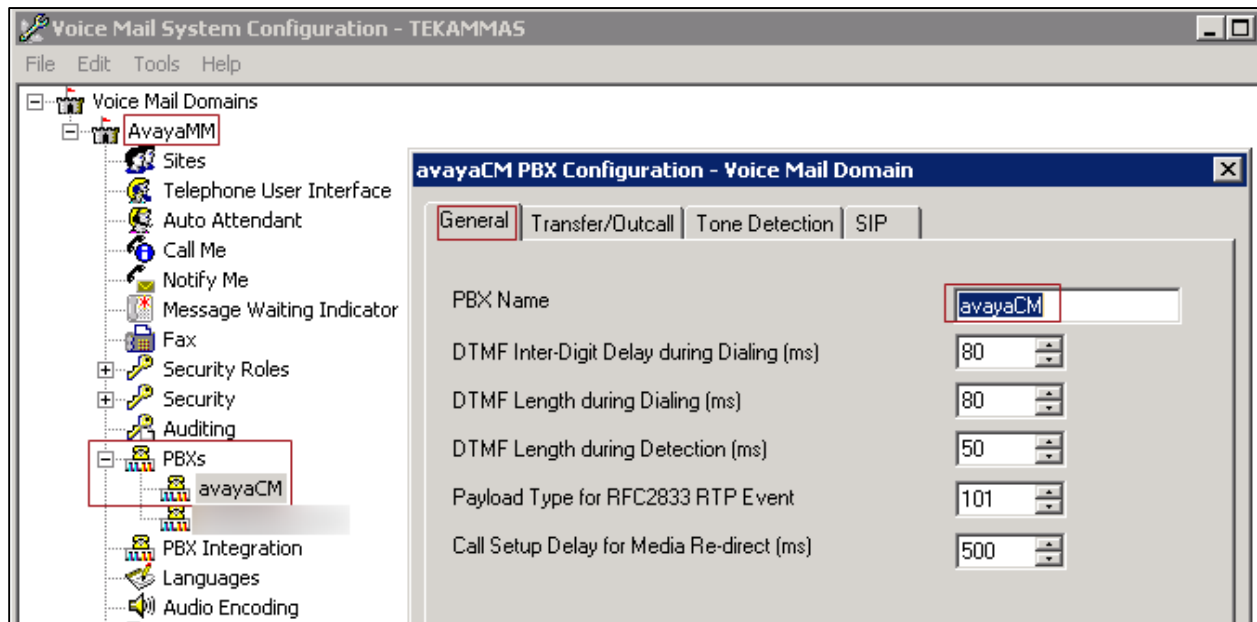
# Avaya Modular Messaging

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

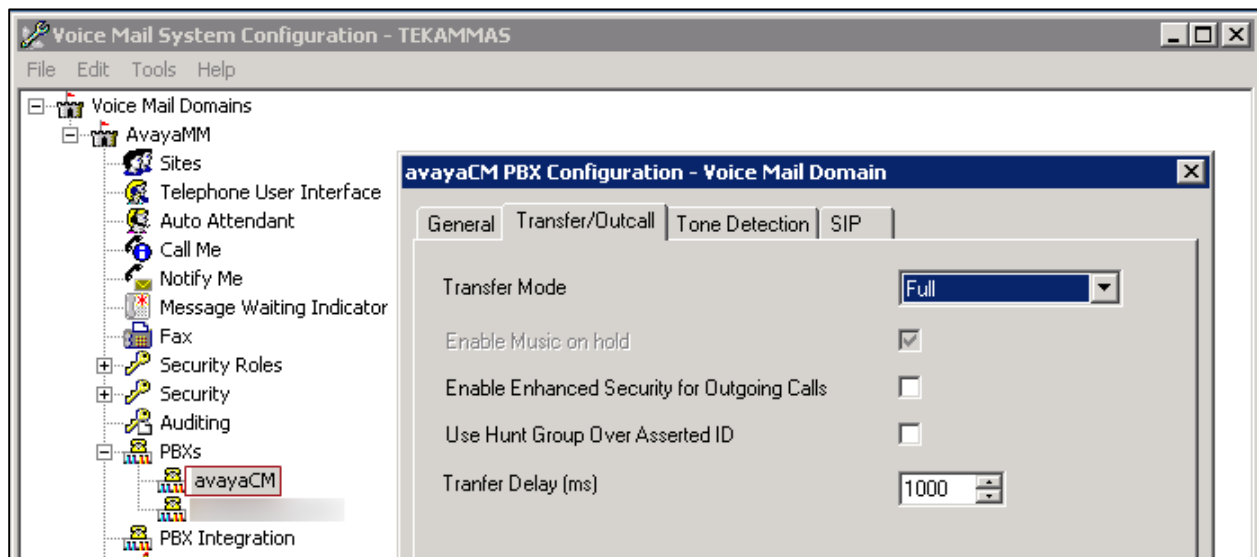
## Integration with the Avaya Aura System

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:

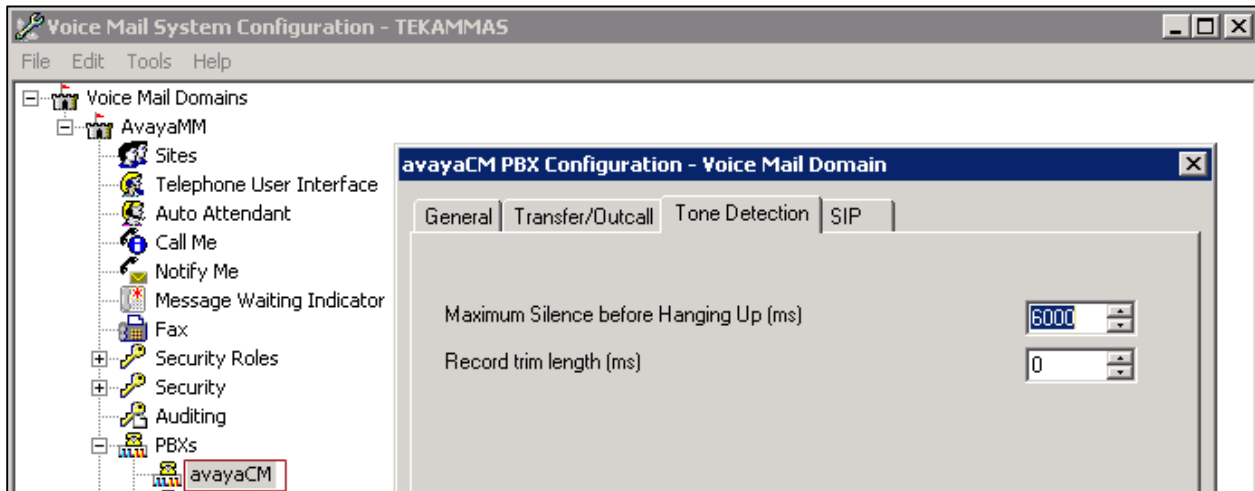
*Avaya Aura Modular Messaging: PBX Configuration: Voice Mail Domain: General Tab*



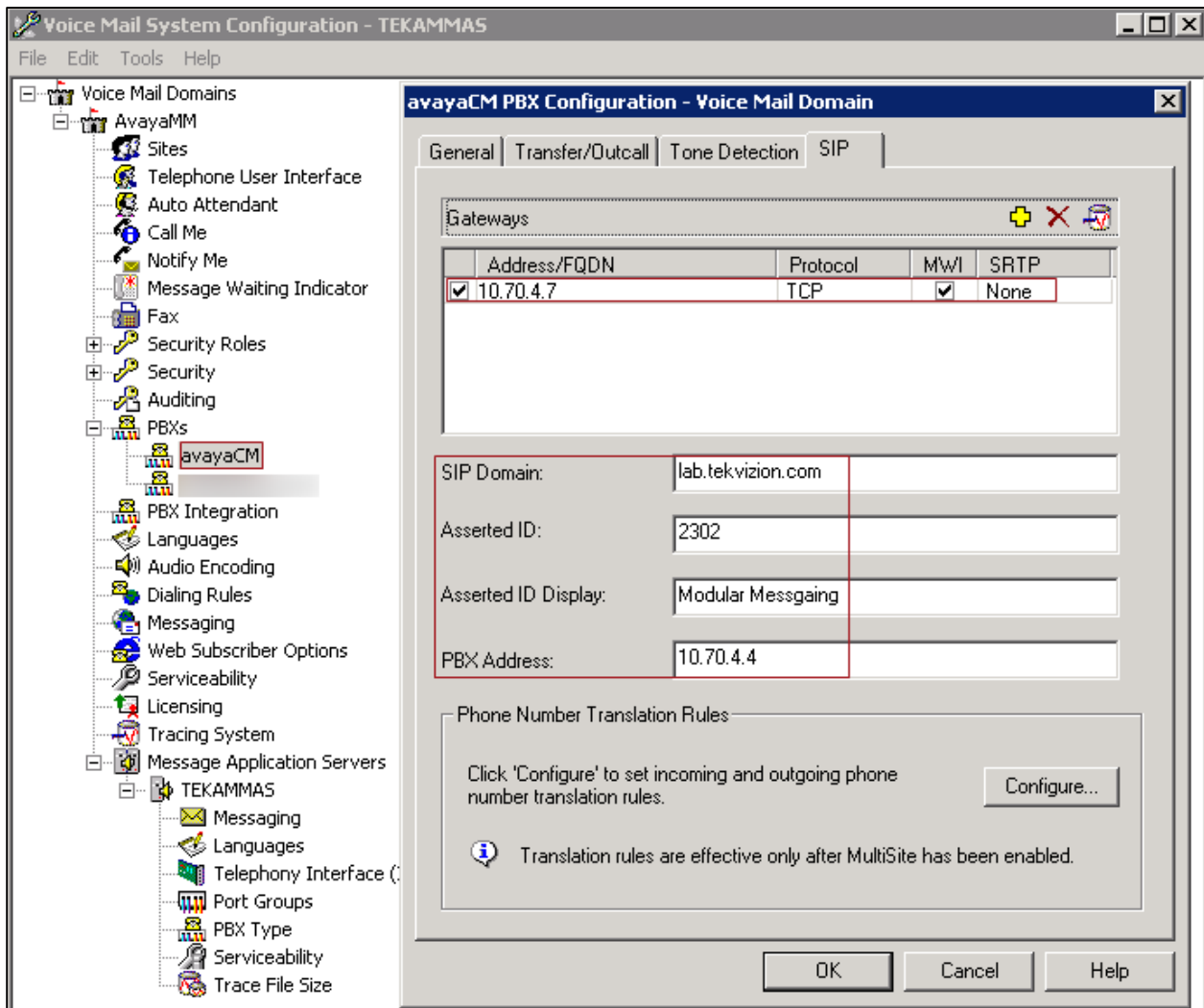
*Avaya Aura Modular Messaging: PBX Configuration: Voice Mail Domain" Transfer/Outcall Tab*



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)*

The screenshot shows the Avaya Modular Messaging Administration GUI. The header includes the Avaya logo and the text 'Modular Messaging Messaging Administration'. Below the header, there is a navigation menu on the left with categories like 'Messaging Administration', 'Server Administration', and 'IMAP/SMTP Administration'. The main content area is titled 'Manage Subscribers' and contains a form for adding a local subscriber. The form has a field for 'Local Subscriber Mailbox Number' with the value '2621' and an 'Add or Edit' button. Below the form is a table of subscribers with columns for 'Machine Name', 'Local Subscriber Mailboxes', 'Total Subscribers', and 'Filtered Subscribers'. The table shows two rows: 'Local Subscribers' for the machine 'tekammss' and 'Remote Subscribers' for the machine 'internet'.

	<u>Machine Name</u>	<u>Local Subscriber Mailboxes</u>	<u>Total Subscribers</u>	<u>Filtered Subscribers</u>
• <b>Local Subscribers</b>	tekammss	3	7	7
• <b>Remote Subscribers</b>	internet		0	0

## Edit Local Subscriber

BASIC INFORMATION * (Required Fields)			
<b>*Last Name</b>	Mercury2	<b>First Name</b>	SIP
<b>*Password</b>		<b>*Mailbox Number</b>	2621
<b>*Numeric Address</b>	2621	<b>PBX Extension</b>	2621
<b>*Class Of Service</b>	0 - test	<b>*Community ID</b>	1

SUBSCRIBER DIRECTORY			
<b>Email Handle</b>	2621 @tekammss.tekvizion.com	<b>Telephone Number</b>	
<b>Common Name</b>	Mercury DUT2	<b>ASCII Version of Name</b>	DUT2, Mercury

SUBSCRIBER SECURITY			
<b>Immediately Expire Password?</b>	no	<b>Is Mailbox Locked?</b>	no

MAILBOX FEATURES			
<b>Personal Operator Mailbox</b>		<b>Personal Operator Schedule</b>	Always Active
<b>VoiceMail Enabled</b>	yes	<b>Intercom Paging</b>	paging is off

TUI MESSAGE ORDER			
<b>TUI New Message Order</b>	urgent first then newest	<b>TUI Saved Message Order</b>	urgent first then newest
<b>TUI Deleted Message Order</b>	urgent first then newest	<b>TUI Admin Message Order</b>	urgent first then newest

SECONDARY EXTENSIONS	
No Secondary Extensions ^ v	<--Add--
	Delete
<a href="#">Secondary Extension</a>	<input type="text"/>
<a href="#">Caller Application</a>	(none) v

MISCELLANEOUS	
<a href="#">Miscellaneous1</a>	<input type="text"/>
<a href="#">Miscellaneous2</a>	<input type="text"/>
<a href="#">Miscellaneous3</a>	<input type="text"/>
<a href="#">Miscellaneous4</a>	<input type="text"/>

Save	Delete	Launch Subscriber Options
Back	Help	

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**Configuration Guide – DOC. 7881A**  
**(2049217)**  
**07.17**  
Specifications subject to  
change without notice.