**Avaya Solution & Interoperability Test Lab**

# Configuring Avaya 1100-Series and 1200-Series IP Deskphones running R4.0 SIP software with Avaya Aura® Session Manager Release 6.1, Avaya Aura® Communication Manager Release 6.0.1, and Avaya Aura® Messaging Release 6.0.1– Issue 1.0

## Abstract


These Application Notes describe a solution comprised of Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Avaya Aura® Messaging, and Avaya 1100-Series and 1200-Series IP Deskphones with SIP software.

- Avaya Aura® Session Manager provides SIP proxy/routing functionality, routing SIP sessions across a TCP/IP network with centralized routing policies and adaptations to resolve SIP protocol differences across different telephony systems.
- Avaya Aura® Communication Manager serves as an Evolution Server within the Avaya Aura® Session Manager architecture and supports SIP endpoints registered to Avaya Aura® Session Manager and other types of endpoints including Avaya 9600-Series and Avaya 9601-Series IP Deskphones and 2420 Digital Telephones.
- Avaya Aura® Messaging provides a centralized voice mail system for all Communication Manager users.

- During testing, Avaya 1100-Series and 1200-Series SIP Deskphones successfully registered with Session Manager, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as conference, transfer, hold, and transfer to Avaya Aura® Messaging.

These Application Notes provide information for the setup, configuration, and verification of the call flows tested on this solution.





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# 1. Introduction

These Application Notes describe a solution comprised of Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Avaya Aura® Messaging, and Avaya 1100-Series and 1200-Series IP Deskphones with SIP software.

As shown in **Figure 1,** Avaya 1100-Series and 1200-Series IP Deskphones configured as SIP endpoints utilize the Avaya Aura® Session Manager User Registration feature and are supported by Avaya Aura® Communication Manager Evolution Server. Since these telephones were originally developed under the Nortel brand, they do not currently support the Avaya Advanced SIP Telephony (AST) protocol implemented in Avaya 9600-Series or Avaya 9601-Series SIP Deskphones. However, Communication Manager and Session Manager have the capability to extend some advanced telephony features to non-AST telephones. See **References [15]** and **[16]** in **Section 9** for more information on configuring these features on Avaya 1100-Series and 1200-Series IP Deskphones.

**Note:** although Avaya 1100-Series and 1200-Series IP Deskphones support the ability to failover to a secondary SIP Registrar, this functionality was not tested in the sample configuration and will not be described in these Application Notes.

Avaya Aura® Communication Manager Evolution Server supports Avaya 2420 Digital telephones and Avaya 9600-Series IP Deskphones and is connected over a SIP trunk to Avaya Aura® Session Manager Release 6.1, using the SIP Signaling network interface on Session Manager.

Avaya Aura® Messaging consists of an Avaya Aura® Messaging Application Server (MAS) and Avaya Message Storage Server (MSS) running on a single Avaya S8800 server. Avaya Aura® Messaging is also connected over a SIP trunk to Avaya Aura® Session Manager.

All inter-system calls are carried over these SIP trunks.

All users have mailboxes defined on Avaya Aura® Messaging which they access via a dedicated pilot number. Interoperability testing included verifying calls between stations were re-directed to Avaya Aura® Messaging and the calling party was able to leave a voice mail message for the appropriate subscriber.

Avaya Aura® Session Manager is managed by Avaya Aura® System Manager. For the sample configuration, Avaya Aura® System Manager and Avaya Aura® Session Manager each run on an Avaya S8800 Server. Avaya Aura® Communication Manager Evolution Server runs on an Avaya S8800 server with an Avaya G650 Media Gateway.


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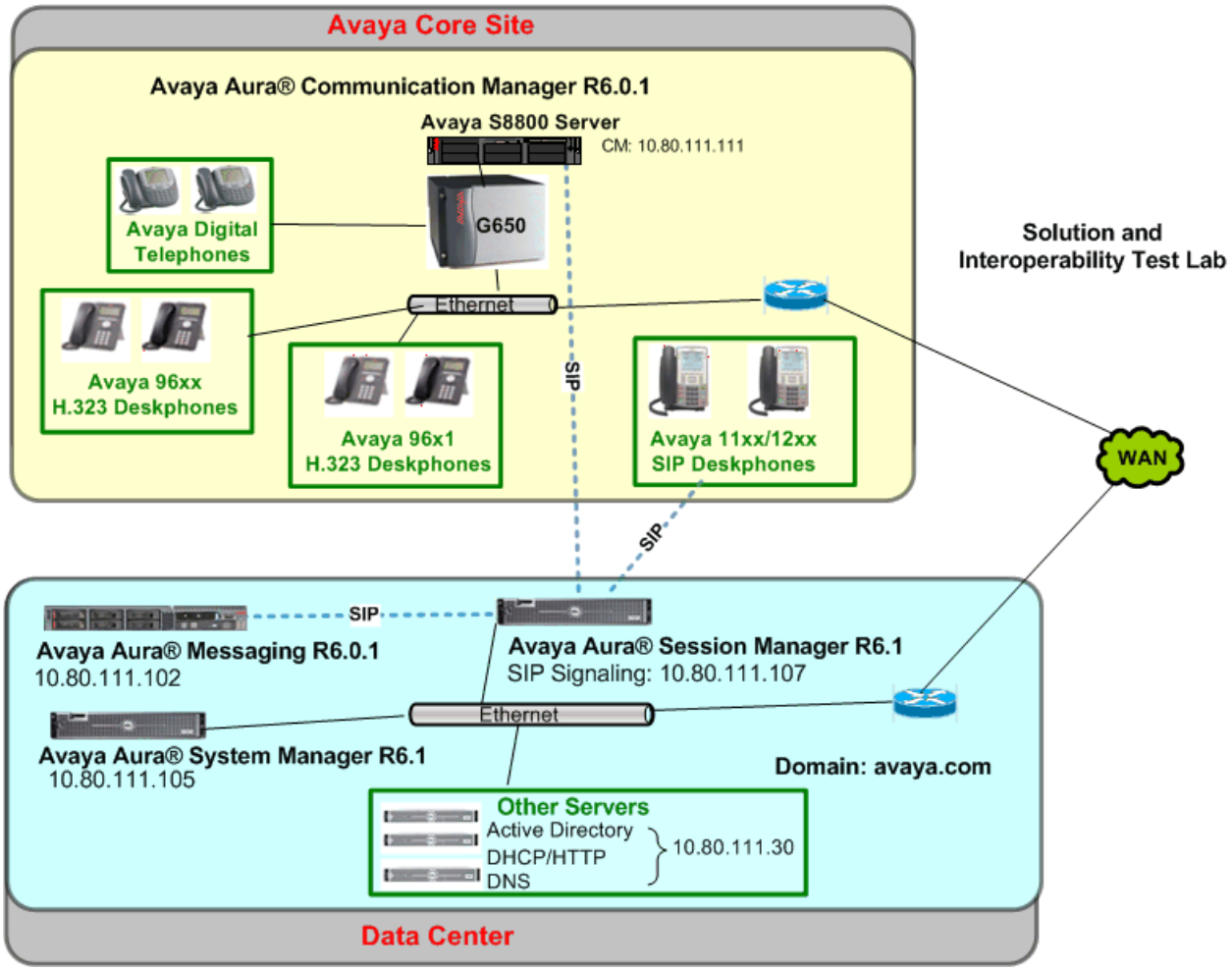



**Figure 1 – Sample Configuration**

In general, a SIP endpoint originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to Communication Manager for origination services. If the call is destined for another SIP endpoint, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP endpoint. If the call is destined for an H.323 or Digital telephone, Communication Manager terminates the call directly.

These Application Notes focus on the configuration of the SIP endpoints, SIP trunks and call routing. These Application Notes assume Avaya Aura® Messaging, Communication Manager and Session Manager are already installed and basic configuration steps have been performed. Only steps relevant to configuration of SIP endpoints will be described in this document. For further details on configuration steps not covered in this document, consult the appropriate document in **Section 9**.


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## 2. Equipment and Software Validated

The following equipment and software were used for the sample configuration.

| Equipment | Software/Firmware |
|---|---|
| Avaya Aura® Session Manager on an Avaya S8800 server | Release 6.1<br>Build 6.1.0.0.610023 |
| Avaya Aura® System Manager on Avaya S8800 server | Release 6.1<br>Version: 6.1.0.4.5072-6.1.4.11 |
| Avaya Aura® Messaging on an Avaya S8800 server | Release 6.0.1<br>Version: 6.0.1-8.0 |
| Avaya Aura® Communication Manager Evolution Server | Release 6.0.1, SP1<br>Version R16x.00.1.510.1-18777 |
| • 1100-Series and 1200-Series IP Deskphone (SIP) | FW: R4.00.04 |
| • Digital Telephones (DCP) | N/A |
| • 9600-Series IP Deskphone (H.323) | FW: R3.1, SP1 |
| • 9601-Series IP Deskphone (H.323) | FW: R6.0, SP1 |

**Table 1: Equipment and Software/Firmware**

**Note:** Avaya 9608 and 9641G IP Deskphones (H.323) were tested in the sample configuration. Avaya 9601 IP Deskphone was not tested since this device does not support H.323 protocol.

**Note:** The following field updates were also installed on Avaya Aura® Messaging. See http://support.avaya.com for more information on installing these field updates.

- W16007rf+ab
- C16007rf+ad
- A14007rf+ac
- M6104rf+ab


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# 3. Configure Avaya Aura® Communication Manager

This section describes the steps needed to configure the SIP trunk between Communication Manager and Session Manager to support calls between SIP telephones and other stations on Communication Manager. These instructions assume the G450 Media Server is already configured on Communication Manager. For information on how to administer these other aspects of Communication Manager, see **References [8]** through **[12]** in **Section 9.**

Avaya and third party SIP telephones are configured as Off-PBX Stations (OPS) in Communication Manager. Communication Manager does not directly control an OPS endpoint, but its features and calling privileges can be applied by associating a local extension with the OPS endpoint. Similarly, a SIP telephone in Session Manager is associated with an extension on Communication Manager. SIP telephones register with Session Manager and use Communication Manager for call origination and termination services.

This section describes the administration of Communication Manager Evolution Server using a System Access Terminal (SAT). Some administration screens have been abbreviated for clarity.

The following administration steps will be described:

- Verify System Capacities and Communication Manager Licensing
- Configure Trunk-to-trunk Transfers
- Configure IP Codec Set
- Configure IP Network Region
- Configure IP Node Names and IP Addresses
- Configure SIP Signaling Groups and Trunk Groups
- Configure Route Pattern
- Administer Numbering Plan
- Administer Locations
- Administer AAR Analysis
- Configure Stations

After completing these steps, the **save translation** command should be performed.

## 3.1. Verify System Capacities and Licensing

This section describes the procedures to verify the correct system capacities and licensing have been configured. If there is insufficient capacity or a required features is not available, contact an authorized Avaya sales representative to make the appropriate changes.


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**Step 1:** Verify Off-PBX Telephone Capacity is sufficient for the expected number of endpoints.

On **Page 1** of the **display system-parameters customer-options** command, verify the limit specified for number of **Maximum Off-PBX Telephones - (OPS)** is sufficient as shown below.

```
display system-parameters customer-options                  Page   1 of  11
                                OPTIONAL FEATURES
G3 Version: V16                            Software Package: Enterprise
       Location: 2                               System ID (SID): 1
                                                             USED
                                Platform Maximum Ports: 6400   45
                                     Maximum Stations: 2400   12
                    Maximum Off-PBX Telephones - EC500: 9600   0
                    Maximum Off-PBX Telephones -   OPS: 9600   8
                    Maximum Off-PBX Telephones - PBFMC: 9600   0
…
```

**Step 2:** Verify SIP Trunk Capacity is sufficient for the expected number of calls.

On **Page 2** of the **display system-parameters customer-options** command, verify the limit specified for number of **Maximum Administered SIP Trunks** is sufficient as shown below.

```
display system-parameters customer-options                  Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                      Maximum Administered H.323 Trunks: 4000  0
            Maximum Concurrently Registered IP Stations: 2400  7
              Maximum Administered Remote Office Trunks: 4000  0
…
                   Maximum Video Capable IP Softphones: 2400  1
                       Maximum Administered SIP Trunks: 4000  10
…
```


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**Step 3:** Verify AAR/ARS Routing features are Enabled on system.

On **Page 3** of **system-parameters customer-options** command, verify the following features are enabled.

- **ARS?** Verify "**y**" is specified.
- **ARS/AAR Partitioning?** Verify "**y**" is specified.
- **ARS/AAR Dialing without FAC?** Verify "**y**" is specified.

```
display system-parameters customer-options                    Page   3 of  11
                                OPTIONAL FEATURES

A/D Grp/Sys List Dialing Start at 01? n                          CAS Main? n
Answer Supervision by Call Classifier? n                 Change COR by FAC? n
                                      ARS? y  Computer Telephony Adjunct Links? y
                      ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
              ARS/AAR Dialing without FAC? y                      DCS (Basic)? y
              ASAI Link Core Capabilities? y                 DCS Call Coverage? n
…
```

**Step 4:** Verify Private Networking feature is Enabled.

On **Page 5** of **display system-parameters customer options** command, verify the **Private Networking** feature is set to "**y**".

```
display system-parameters customer-options                    Page   5 of  11
                                OPTIONAL FEATURES

                     Port Network Support? y              Time of Day Routing? n
                          Posted Messages? n      TN2501 VAL Maximum Capacity? y
                                                      Uniform Dialing Plan? y
                        Private Networking? y    Usage Allocation Enhancements? y
                  Processor and System MSP? y
                        Processor Ethernet? y               Wideband Switching? n
…
```


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## 3.2. Configure Trunk-to-Trunk Transfers

Use the **change system-parameters features** command to enable trunk-to-trunk transfers. This feature is needed when an incoming call to a SIP station is transferred to a different telephony system such as when calls are transferred to Avaya Aura® Messaging. For simplicity, the **Trunk-to-Trunk Transfer** field on **Page 1** was set to "**all**" to enable all trunk-to-trunk transfers on a system wide basis.

**Note**: Enabling this feature poses significant security risk by increasing the risk of toll fraud, and must be used with caution. To minimize the risk, a COS could be defined to allow trunk-to-trunk transfers for specific trunk group(s). For more information regarding how to configure Communication Manager to minimize toll fraud, see **Reference [12]** in **Section 9.**

```
change system-parameters features                                Page   1 of  18
                          FEATURE-RELATED SYSTEM PARAMETERS

                             Self Station Display Enabled? n
                                  Trunk-to-Trunk Transfer: all
              Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
…
```

## 3.3. Configure IP Codec Set

Use the **change ip-codec-set n** command where **n** is the number used to identify the codec set.

Enter the following values:

- **Audio Codec**: Enter "**G.711MU"** and **"G.729**" as supported types.
- **Silence Suppression**: Retain the default value "**n**".
- **Frames Per Pkt**: Enter "**2**".
- **Packet Size (ms):** Enter "**20**".
- **Media Encryption**: Enter the value based on the system requirement. For the sample configuration, "**none**" was used.

```
change 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.711MU           n         2        20
 2: G.729             n         2        20
 3:

     Media Encryption
 1: none
```


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## 3.4. Configure IP Network Region

Use the **change ip-network-region n** command where **n** is an available network region.

Enter the following values and use default values for remaining fields.

- **Authoritative Domain:** Enter the correct SIP domain for the configuration. For the sample configuration, "**avaya.com**" was used.
- **Name:** Enter a descriptive name.
- **Codec Set:** Enter the number of the IP codec set configured in **Section 3.3.**
- **Intra-region IP-IP Direct Audio:** Enter "**yes**".
- **Inter-region IP-IP Direct Audio:** Enter "**yes**".

```
change ip-network-region 1                                   Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: SIP Trunk
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: 16585
```

## 3.5. Add Node Names and IP Addresses

Use the **change node-names ip** command to add the node-name and IP Addresses for the "**procr**" interface on Communication Manager and the SIP signaling interface of Session Manager, if not previously added.

For the sample configuration, the node-name of the SIP Signaling Interface for Session Manager is "**ASM1-6_1**" with an IP address of "**10.80.111.107**".

**Note**: The solution is extensible to configurations using CLAN interface. For these configurations, enter the node-name and IP address of the CLAN interface instead of using the procr interface.

```
change node-names ip                                         Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
ASM1-6_1              10.80.111.107
default               0.0.0.0
procr                 10.80.111.111
```


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## 3.6. Configure SIP Signaling Group and Trunk Group

In the sample configuration, trunk group "**10**" and signaling group "**10**" were used for connecting to Session Manager

**Step 1:** Add Signaling Group for SIP Trunk

Use the **add signaling-group n** command, where **n** is an available signaling group number to create SIP signaling group.

Enter the following values and use default values for remaining fields.

- **Group Type:** Enter "**sip**".
- **IMS Enabled:** Enter "**n**".
- **Transport Method:** Enter "**tcp**".
- **Peer Detection Enabled?** Enter "**y**".
- **Peer Server:** Use default value.
  **Note:** default value is replaced with "**SM**" after SIP trunk to Session Manager is established.
- **Near-end Node Name:** Enter "**procr**" node name from **Section 3.5**.
- **Far-end Node Name:** Enter node name for Session Manager defined in **Section 3.5**
- **Near-end Listen Port:** Verify "**5060**" is used.
- **Far-end Listen Port:** Verify "**5060**" is used.
- **Far-end Network Region**: Enter network region defined in **Section 3.4.**
- **Far-end Domain:** Enter domain name for **Authoritative Domain** field defined in **Section 3.4.**
- **DTMF over IP:** Verify "**rtp-payload**" is used.

**Note:** TCP was used for the sample configuration. However, TLS would typically be used in production environments.

```
add signaling-group 10                                    Page   1 of   1
                              SIGNALING GROUP

Group Number: 10             Group Type:  sip
  IMS Enabled? n             Transport Method: tcp
        Q-SIP? n                                        SIP Enabled LSP? n
     IP Video? n
  Peer Detection Enabled? y  Peer Server: Others

   Near-end Node Name: procr                 Far-end Node Name: ASM1-6_1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
Far-end Domain: avaya.com
                                           Bypass If IP Threshold Exceeded? 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? n                 Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```


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**Step 2:** Add SIP Trunk Group

Add the corresponding trunk group controlled by the signaling group defined **Step 1** using the **add trunk-group n** command where **n** is an available trunk group number.

Fill in the indicated fields as shown below. Default values can be used for the remaining fields.

- **Group Type:** Enter "**sip**".
- **Group Name:** Enter a descriptive name.
- **TAC:** Enter an available trunk access code.
- **Direction:** Enter "**two-way**".
- **Outgoing Display?** Enter "**y**".
- **Service Type:** Enter "**tie**".
- **Signaling Group:** Enter the number of the signaling group added in **Step 1.**
- **Number of Members:** Enter the number of members in the SIP trunk (must be within the limits for number of SIP trunks specified in **Section 3.1**).

**Note**: once the **add trunk-group** command is submitted, trunk members will be automatically generated based on the value in the **Number of Members** field.

```
add trunk-group 10                                          Page   1 of  21
                              TRUNK GROUP

Group Number: 10                    Group Type: sip          CDR Reports: y
  Group Name: SIP trunk to ASM 1          COR: 1        TN: 1       TAC: #10
Direction: two-way        Outgoing Display? y
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 10
                                                    Number of Members: 10
```

On **Page 3**, fill in the indicated fields as shown below. Default values can be used for the remaining fields.

- **Numbering Format** Enter "**private**".
- **Show ANSWERED BY on Display** Enter "**y**".

```
add trunk-group 10                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                                        Maintenance Tests? y

                    Numbering Format: private
                                              UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

 Show ANSWERED BY on Display? y
```


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On **Page 4**, fill in the indicated fields as shown below. Default values can be used for the remaining fields.

- **Support Request History** Enter "**y**".
- **Telephone Event Payload Type** Enter "**120**".

```
add trunk-group 10                                          Page   4 of  21
                                  PROTOCOL VARIATIONS

                     Mark Users as Phone? y
           Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 120
```

## 3.7. Configure Route Pattern

This section provides the configuration of the route pattern used in the sample configuration for routing calls between SIP stations and other stations supported by Communication Manager Evolution Server.

Use **change route-pattern n** command where **n** is an available route pattern.

Fill in the indicated fields as shown below and use default values for remaining fields.

- **Grp No** Enter a row for the trunk group defined in **Section 3.6.**
- **FRL** Enter "**0**".
- **Numbering Format** Enter "**lev0-pvt**".

In the sample configuration, route pattern "**10**" was created as shown below.

```
change route-pattern 10                                      Page   1 of  3
                       Pattern Number: 10  Pattern Name: SIP to ASM
                         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: 10   0                                                     n   user
2:                                                            n   user
3:                                                            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                             lev0-pvt  none
2: y y y y y n  n            rest                             lev0-pvt  none
3: y y y y y n  n            rest                                       none
…
```


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## 3.8. Administer Numbering Plan

Extension numbers used for SIP Users registered to Session Manager must to be added to either the private or public numbering table on Communication Manager. For the sample configuration, private numbering was used and all extension numbers were unique within the private network. However, in many customer networks, it may not be possible to define unique extension numbers for all users within the private network. For these types of networks, additional administration may be required as described in **Reference [9]** in **Section 9**.

**Step 1:** Administer Private Numbering Plan

Use the **change private-numbering n** command, where **n** is the length of the private number.

Fill in the indicated fields as shown below.

- **Ext Len**: Enter length of extension numbers.
  In the sample configuration, "**7**" was used.
- **Ext Code**: Enter leading digit (s) from extension number.
  In the sample configuration, "**444**" was used.
- **Trk Grp(s):** Enter row for trunk group defined in **Section 3.6.**
- **Private Prefix:** Leave blank unless an enterprise canonical numbering scheme is defined in Session Manager. If so, enter the appropriate prefix.
- **Total Length:** Enter "**7**" since a private prefix was not defined.

```
change private-numbering 7                                    Page   1 of  2
                            NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 7  444            10                           7    Total Administered: 1
                                                     Maximum Entries: 540
…
```


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**Step 2:** Administer Uniform Dialplan

Use the **change uniform-dialplan n** command, where **n** is the first digit of the extension numbers used for SIP stations in the system.

In the sample configuration, 7-digit extension numbers starting with "**444-3xxx**" are used for extensions associated with Avaya 1100-Series or 1200-Series SIP Deskphones.

Fill in the indicated fields as shown below and use default values for remaining fields.

- **Matching Pattern** Enter digit pattern of extensions associated with SIP stations.
- **Len** Enter extension length.
- **Net** Enter "**aar**".

```
change uniform-dialplan 6                                      Page   1 of  2
                          UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0
  Matching                  Insert              Node
  Pattern       Len Del     Digits     Net Conv Num
 4443            7   0                 aar  n
 4445            7   0                 aar  n
 777             7   0                 aar  n
 778             7   0                 aar  n
                                       n
```

## 3.9. Administer Locations

This section provides the configuration of the Locations screen. Configuring a default route is necessary to enable stations on Communication Manager to use Avaya Aura® Messaging features such as Call Sender or Auto-Attendant.

Use the **change locations** command to identify a default proxy route. Set the **Proxy Rte** field to use the Route Pattern defined in **Section 3.7.**

```
change locations                                        Page   1 of  16
                          LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc  Name       Timezone Rule  NPA  ARS  Atd      Disp   Prefix   Proxy Sel
No               Offset            FAC  FAC      Parm             Rte Pat
1:  Main        + 00:00   0                        1                10
2:                      :
3:                      :
```


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## 3.10. Administer AAR Digit Analysis

This section provides the configuration of the AAR (Automatic Alternate Routing) pattern used in the sample configuration for routing calls between SIP users and other stations on Communication Manager Evolution Server.

In the sample configuration, extension numbers starting with digits "**444-3xxx**" are assigned to SIP stations supported by Communication Manager Evolution Server.

**Note:** Other methods of routing may be used.

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

Fill in the indicated fields as shown below and use default values for remaining fields.

- **Dialed String** Enter leading digit (s) of extension numbers assigned to SIP Stations.
- **Min** Enter minimum number of digits that must be dialed.
- **Max** Enter maximum number of digits that may be dialed.
- **Route Pattern** Enter Route Pattern defined in **Section 3.7.**
- **Call Type** Enter "**unku**".

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    443                   7    7     10       unku         n
    4443                  7    7     10       unku         n
    4445                  7    7     10       unku         n
    778                   7    7     10       unku         n
```

## 3.11. Configure Stations

For each SIP user defined in Session Manager, add a corresponding station on Communication Manager. The extension number defined for the SIP station will be the login ID the user enters to register to Session Manager. The configuration is the same for all of the 1100-Series or 1200-Series IP Deskphones except for the desired number of call appearances.

**Note:** Instead of manually defining each station using the Communication Manager SAT interface, an alternative option is to automatically generate the SIP station when adding a new SIP user using System Manager. See **Section 4.7** for more information on adding SIP users.


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Use the **add station n** command where **n** is a valid extension number defined in the system.

On **Page 1,** enter the following values and use defaults for remaining fields.

- **Phone Type:** Enter "**9630SIP**".
- **Port:** Leave blank. Once the command is submitted, a virtual port will be assigned (e.g. "**S0000**").
- **Name:** Enter a display name for user.
- **Security Code:** Enter the number used to log in station.
  **Note:** this number should match the "**Communication Profile Password**" field defined when adding this user in System Manager. See **Section 4.7** for more information.
- **Coverage Path 1:** Enter the coverage path number previously defined for coverage to Avaya Aura® Messaging.

```
add station 4443120                                    Page   1 of  6

                              STATION

Extension: 444-3120                  Lock Messages? n            BCC: 0
     Type: 9630SIP                 Security Code: 123456         TN: 1
     Port:                         Coverage Path 1: 1           COR: 1
     Name: SIP Station User        Coverage Path 2:             COS: 1
                                   Hunt-to Station:

STATION OPTIONS
                                       Time of Day Lock Table:
             Loss Group: 19
                                             Message Lamp Ext: 666-4029
        Display Language: english              Button Modules: 0
          Survivable COR: internal
   Survivable Trunk Dest? y                      IP SoftPhone? n

                                                 IP Video? n
```

On **Page 2,** enter the following values and use defaults for remaining fields.

- **MWI Served User Type:** Enter "**sip-adjunct**".

```
add station 4443120                                    Page   2 of  6

                              STATION
FEATURE OPTIONS
         …
        H.320 Conversion? n      Per Station CPN - Send Calling Number? y
                                          EC500 State: enabled

    MWI Served User Type: sip-adjunct
```


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On **Page 4,** add the desired number of **call-appr** entries in the **BUTTON ASSIGNMENTS** section. This governs how many concurrent calls can be supported. Avaya 1100-Series IP Deskphones have the capability of handling 11 call appearances, while 1200-Series can handle 10 call appearances. In the sample configuration, three call appearances were configured here to support conferencing scenarios.

**Note:** Avaya 1100-Series IP Deskphones display only one local call appearance button when idle. So the number of entries shown below is not required to match the number of appearances displayed on the telephone.

```
add station 4443120                                    Page   4 of  6

                              STATION
SITE DATA
…

BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:
```

On **Page 6,** enter the following values and use defaults for remaining fields.

- **SIP Trunk:** Enter "**aar**" to use Route Pattern defined in **Section 3.7**

```
add station 4443120                                    Page   6 of  6

                              STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                     SIP Trunk: aar
```


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## 3.12. Verify Off-PBX-Telephone Station-Mapping

Use the **change off-pbx-telephone station-mapping xxx** command where **xxx** is an extension assigned to an 1100-Series or 1200-Series SIP telephone to verify an Off-PBX station mapping was automatically created for the SIP station.

On **Page 1**, verify the following fields were correctly populated.

- **Application** Verify "**OPS**" is assigned.
- **Trunk Selection** Verify "**aar**" is assigned.

```
change off-pbx-telephone station-mapping 4443120              Page   1 of  3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk      Config  Dual
 Extension                 Prefix                    Selection  Set     Mode
 444-3120         OPS          -      4443120          aar        1
                               -
                               -
```

On **Page 2**, verify the following fields were correctly populated.

- **Call Limit** Verify "**3**" is assigned.
  **Note:** if more than 3 call appearances were assigned to the station as described in **Section 3.11,** modify this field to match the number of call appearances.
- **Mapping Mode**: Verify "**both**" is assigned.
- **Calls Allowed:** Verify "**all"** is assigned.

```
change off-pbx-telephone station-mapping 4443120              Page   2 of  3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION


 Station       Appl    Call        Mapping    Calls      Bridged     Location
 Extension     Name    Limit       Mode       Allowed    Calls
 444-3120       OPS      3          both       all        none
```

Configuration of Communication Manager is complete. Use the **save translation** command to save these changes.

**Note:** After making a change on Communication Manager which alters the dial plan or numbering plan, synchronization between Communication Manager and System Manager needs to be completed and SIP telephones must be re-registered.

See **Section 4.8** for more information on how to perform an on-demand synchronization.


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# 4. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® Session Manager to support registrations of SIP endpoints.

These instructions assume other administration activities have already been completed such as defining the SIP entities for Avaya Aura® Messaging, Avaya Aura® Communication Manager, and Session Manager, defining the network connection between System Manager and Session Manager, defining Communication Manager as a Managed Element and defining the Entity Links for the SIP trunks between each SIP entity and Session Manager.

For more information on configuring SIP Trunks, see **Reference [17]** in **Section 9**. For more information on other aspects of administering Session Manager, see **References [2]** through **[5]** in **Section 9.**

The following administration activities will be described:

- Define SIP Domain and Locations.
- Define Routing Policies and Dial Patterns which control routing between SIP Entities.
- Define Applications and Application Sequences supporting SIP Users.
- Add new SIP Users.
- Synchronize changes with Avaya Aura® Communication Manager.

**Note:** Some administration screens have been abbreviated for clarity.

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "**http://<ip-address>/SMGR**", where "**<ip-address>"** is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials.


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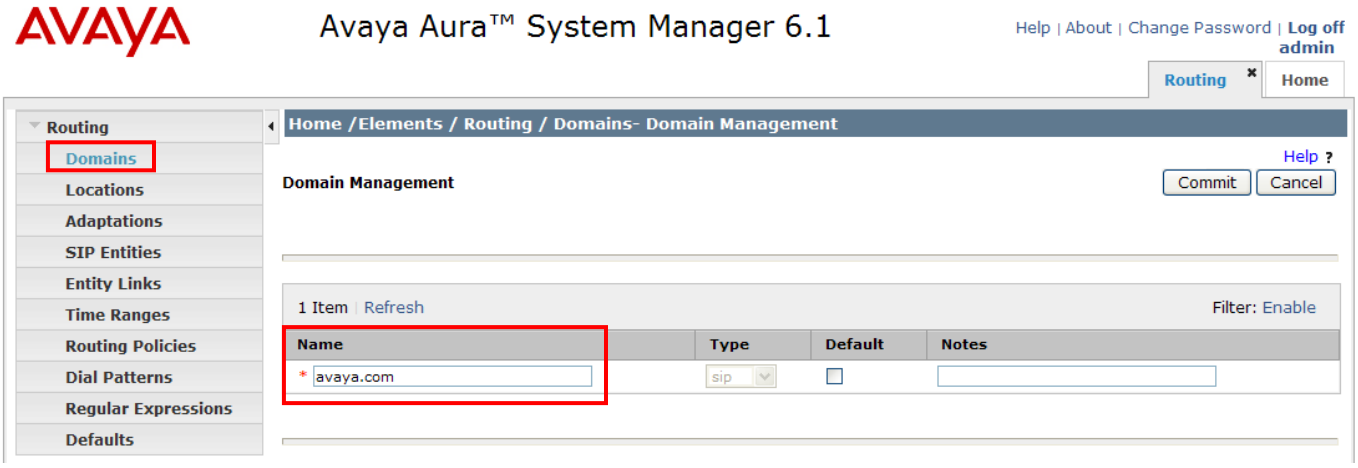
## 4.1. Define SIP Domain

Expand **Elements → Routing** and select **Domains** from the left navigation menu.

Click **New** (not shown)**.** Enter the following values and use default values for remaining fields**.**

- **Name** Enter the Authoritative Domain Name specified in **Section 3.4.** In the sample configuration, "**avaya.com**" was used.
- **Type** Verify "**SIP**" is selected.
- **Notes** Add a brief description. [Optional]

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.





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## 4.2. Define Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing.

Expand **Elements → Routing** and select **Locations** from the left navigation menu.

Click **New** (not shown)**.** In the **General** section**,** enter the following values and use default values for remaining fields**.**

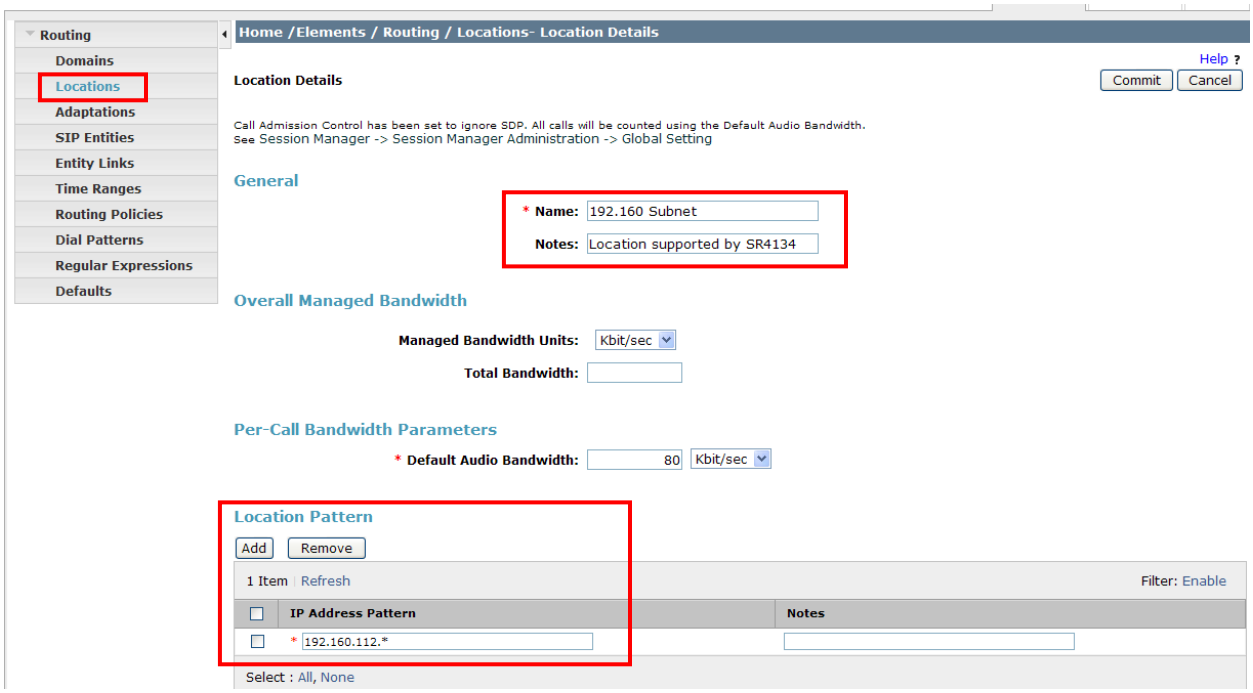
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional].

In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the logical pattern used to identify the location. For the sample configuration, "**192.160.112.***" was used.
- **Notes** Add a brief description. [Optional].

Click **Commit** to save.

The screen below shows the Location defined for the Avaya 1100-Series and 1200-Series SIP Deskphones used in the sample configuration.





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## 4.3. Define Routing Policy

Routing policies describe the conditions under which calls will be routed to non-SIP stations on Communication Manager Evolution Server or to Avaya Aura® Messaging.

**Note:** Since the SIP stations are registered on Session Manager, a routing policy does not need to be defined for calls to SIP stations.

To add a routing policy, expand **Elements → Routing** and select **Routing Policies.**

Click **New** (not shown). In the **General** section, enter the following values.

- **Name:** Enter an identifier to define the routing policy for making calls to non-SIP stations on Communication Manager Evolution Server.
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional].

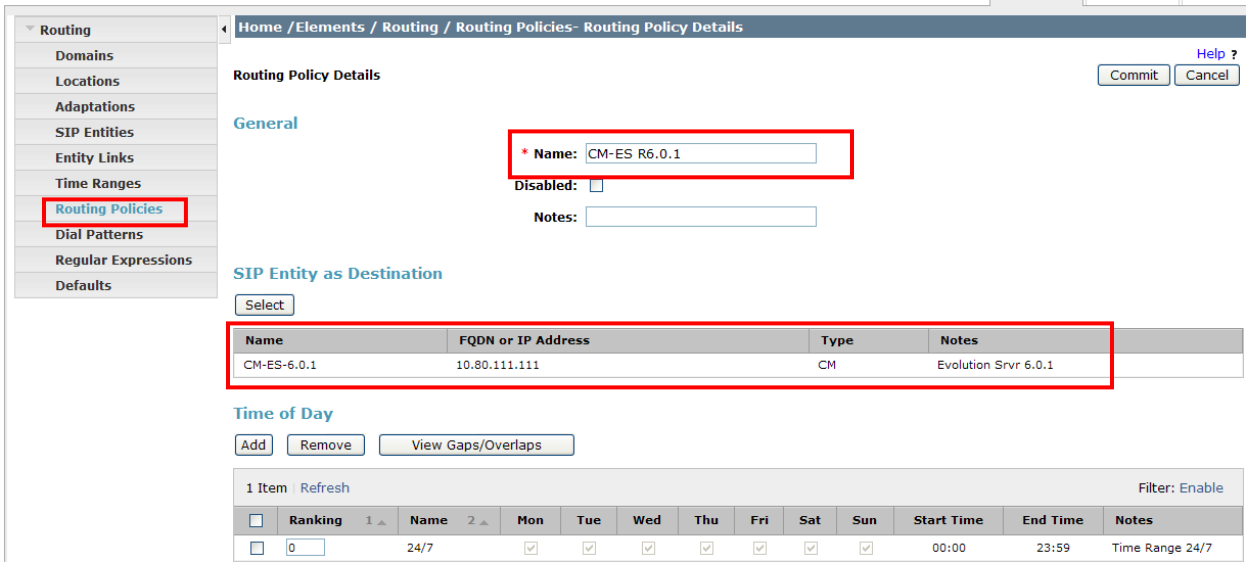
In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the SIP Entity associated with Communication Manager Evolution Server and click **Select.**
- The selected SIP Entity displays on the **Routing Policy Details** page.

Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

**Note**: the routing policies defined in this section are examples and were used in the sample configuration. Other routing policies may be appropriate for different customer networks.

The following screen shows the Routing Policy for Communication Manager Evolution Server.





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## 4.4. Define Dial Pattern

This section describes the steps to define a dial pattern to route calls to non-SIP stations on Communication Manager Evolution Server. In the sample configuration, 7-digit extensions beginning with "**4441**" are assigned to non-SIP stations on Communication Manager.

**Note:** Since the SIP stations are registered on Session Manager, a dial pattern does not need to be defined for SIP stations supported by Communication Manager Evolution Server.

To define a dial pattern, expand **Elements → Routing** and select **Dial Patterns.**

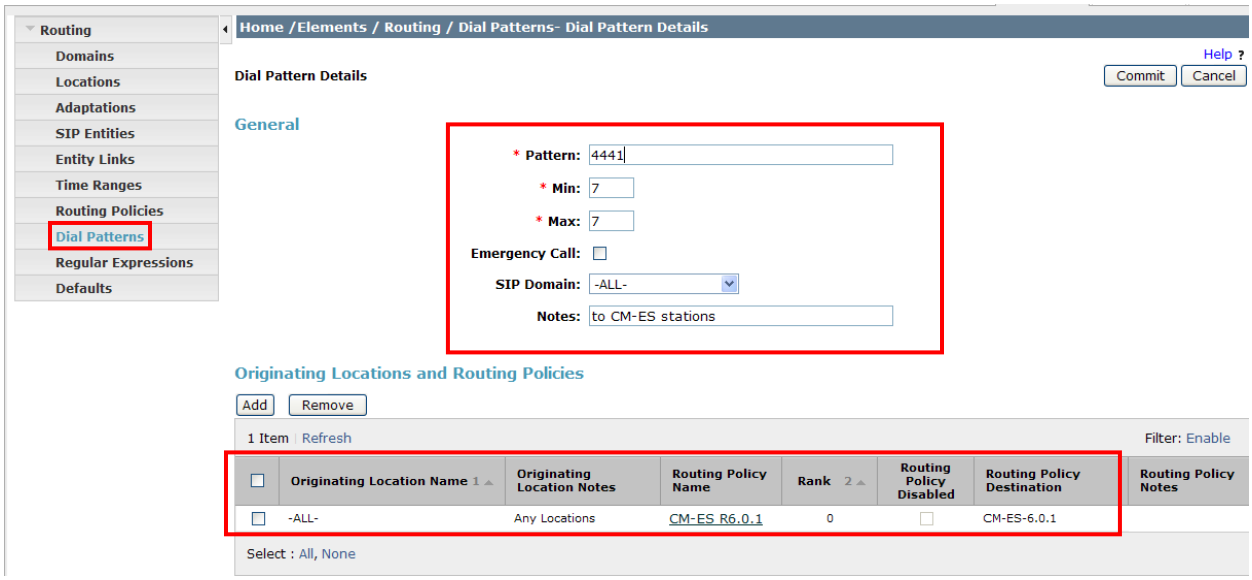
Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern:** Add dial pattern associated with non-SIP stations.
- **Min:** Enter the minimum number digits that must to be dialed.
- **Max:** Enter the maximum number digits that may be dialed.
- **SIP Domain:** Select the SIP Domain defined in **Section 4.1**.
- **Notes:** Enter a brief description. [Optional].

In the **Originating Locations and Routing Policies** section, click **Add.**
The **Originating Locations and Routing Policy List** page opens (not shown).

- In **Originating Locations** table, select "**ALL**".
- In **Routing Policies** table, select the appropriate Routing Policy defined for Communication Manager Evolution Server in **Section 4.3.**
- Click **Select** to save these changes and return to **Dial Pattern Details** page.

Click **Commit** to save the new definition. The following screen shows the Dial Pattern defined for routing calls to non-SIP stations on Communication Manager Evolution Server.





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## 4.5. Define Application

To support SIP stations registered to Session Manager, an Application must be defined for Communication Manager Evolution Server.

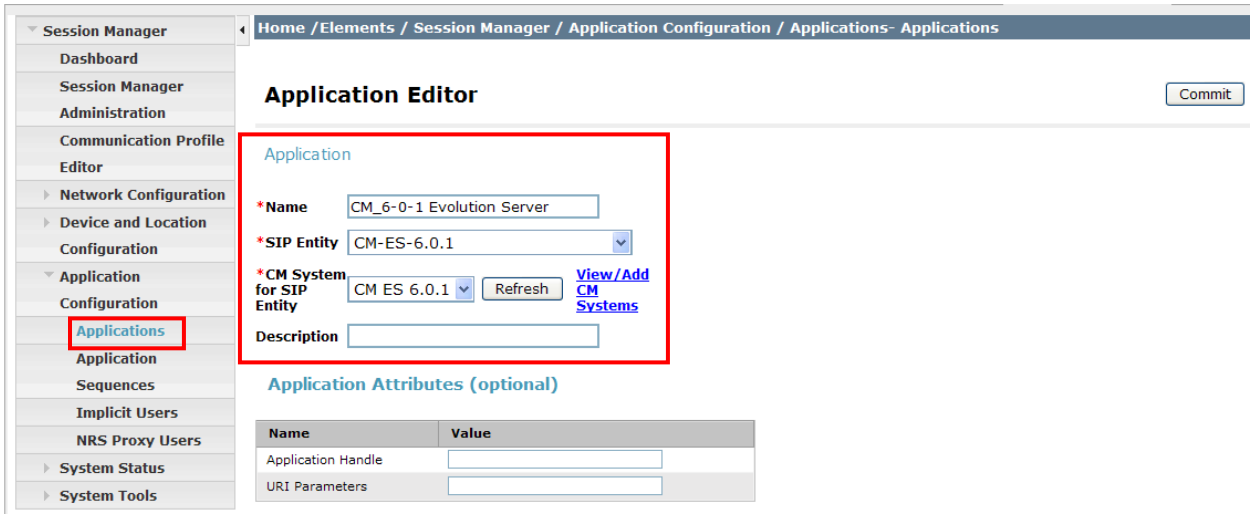
To define a new Application, expand **Elements → Session Manager → Application Configuration** and select **Applications** from the left navigational menu.

Click **New** (not shown). In the **Application Editor** section, enter the following values.

- **Name** Enter name for the application.
- **SIP Entity** Select SIP Entity associated with Communication Manager Evolution Server.
- **CM System for SIP Entity:** Select name of Managed Element associated with Communication Manager.
  In the sample configuration, **"CM ES 6.0.1"** was used.
- **Description:** Enter description [Optional].

Leave fields in the **Application Attributes (optional)** section blank.

Click **Commit** to save application. The screen below shows the Application defined for Communication Manager Evolution Server in the sample configuration.





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## 4.6. Define Application Sequence

The second step in defining an Application to support SIP stations registered to Session Manager is to define an Application Sequence.

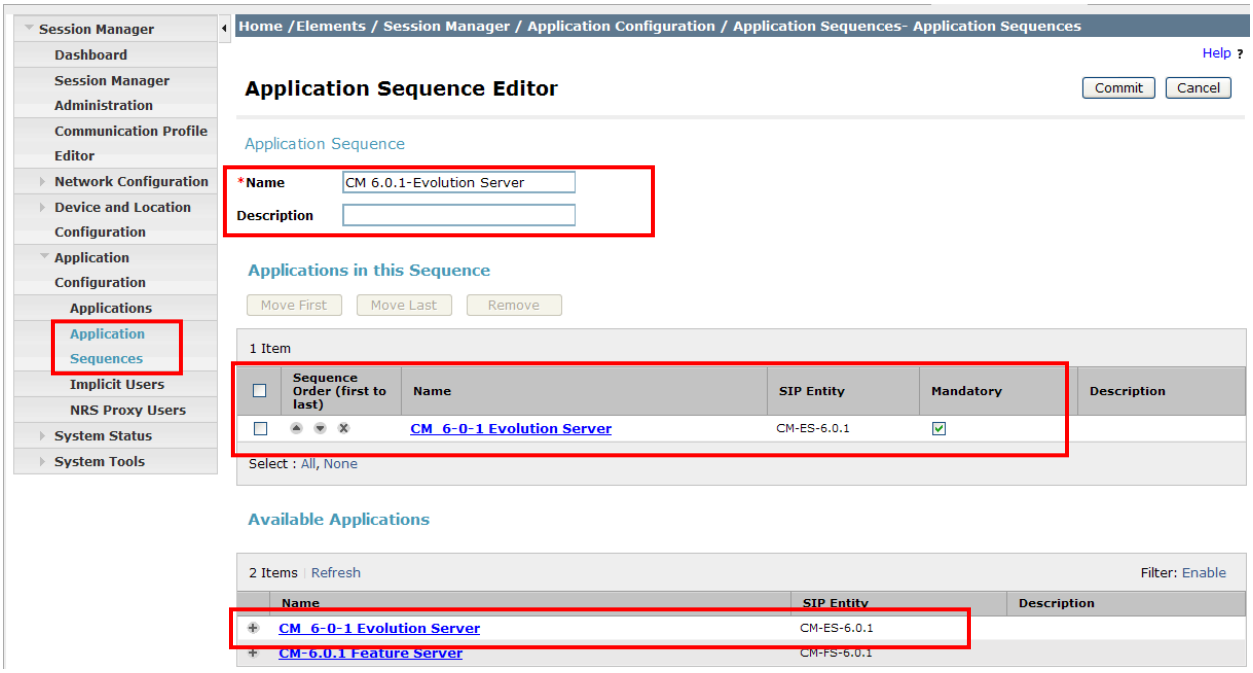
Expand **Elements → Session Manager → Application Configuration** and select **Application Sequences** from the left navigation menu.

Click **New** (not shown). In the **Application Sequence** section, enter the following values.

- **Name** Enter name for the application sequence.
- **Description** Enter description [Optional].

In the **Available Applications** table, click + icon associated with the Application for Communication Manager Evolution Server defined in **Section 4.5** to select this application.

Verify a new entry is added to the **Applications in this Sequence** table and the **Mandatory** column is ☑ as shown below.




**Note:** The Application Sequence defined for Communication Manager Evolution Server can only contain a single Application.

Click **Commit** to save the new Application Sequence.


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## 4.7. Add SIP Users

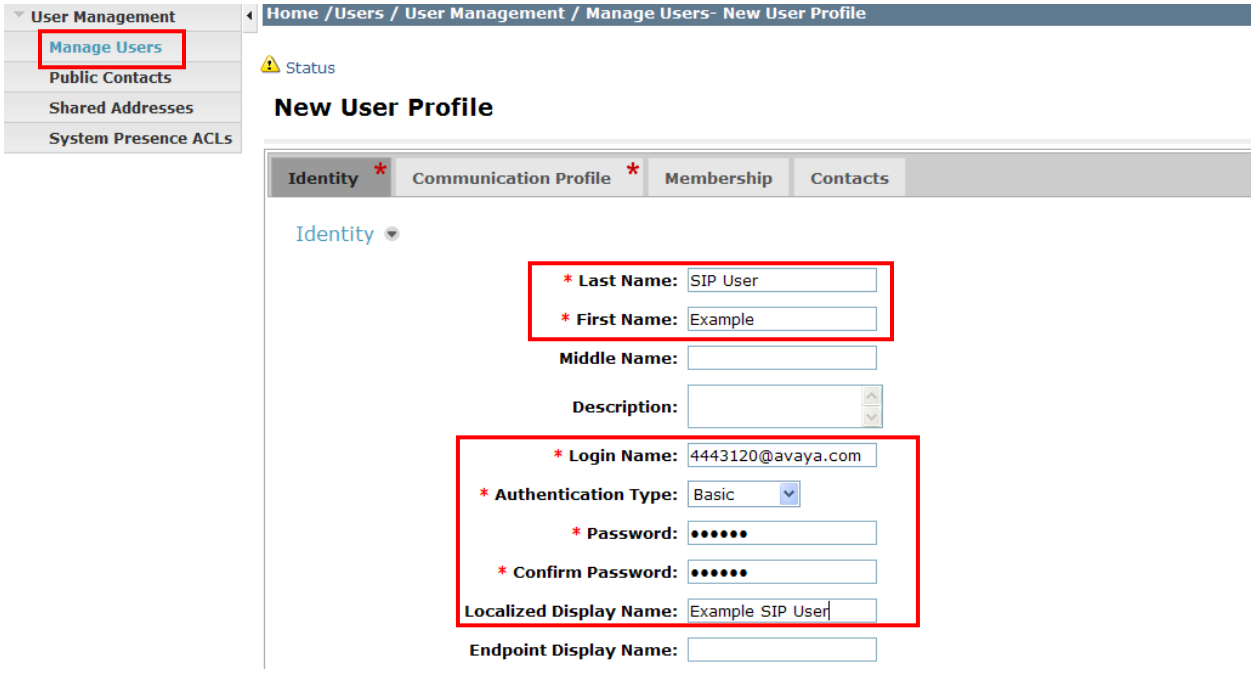
Add new SIP users for each Avaya 1100-Series or 1200-Series SIP station defined in **Section 3.11.** Alternatively, use the option to automatically generate the SIP station after adding a new SIP user.

To add new SIP users, expand **Users → User Management** and select **Manage Users.**

**Step 1**: Click **New** (not shown). Enter values for the following required attributes for a new SIP user in the **Identity** section and use default values for remaining fields.

- **Last Name:** Enter last name of user
- **First Name:** Enter first name of user
- **Login Name:** Enter "**extension number@<domain>**" where **"<domain>"** matches domain defined in **Section 4.1**.
- **Authentication Type:** Verify **"Basic"** is selected.
- **Password:** Enter password to be used to log into System Manager.
- **Confirm Password:** Repeat value entered above.
- **Localized Display Name:** Enter display name for user.

The screen below shows results from **Step 1** for a new SIP user.



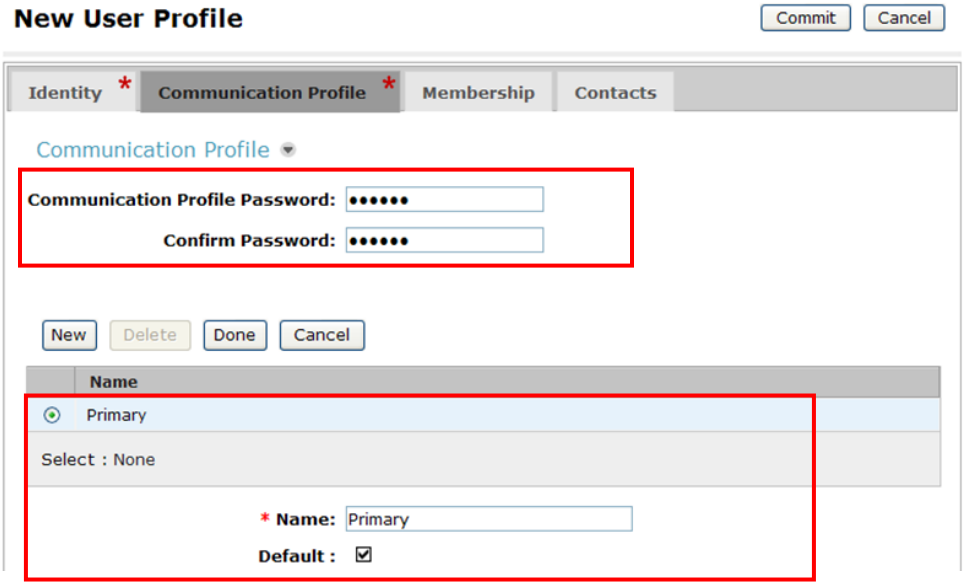


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**Step 2**: Select **Communication Profile** tab on **New User Profile** page and enter numeric value used to logon in the **Communication Profile Password** and **Confirm Password** fields.

**Note: Password** should match the **Security Code** field defined in **Section 3.11.**

Verify there is a default entry identified as the **Primary** profile as shown below:



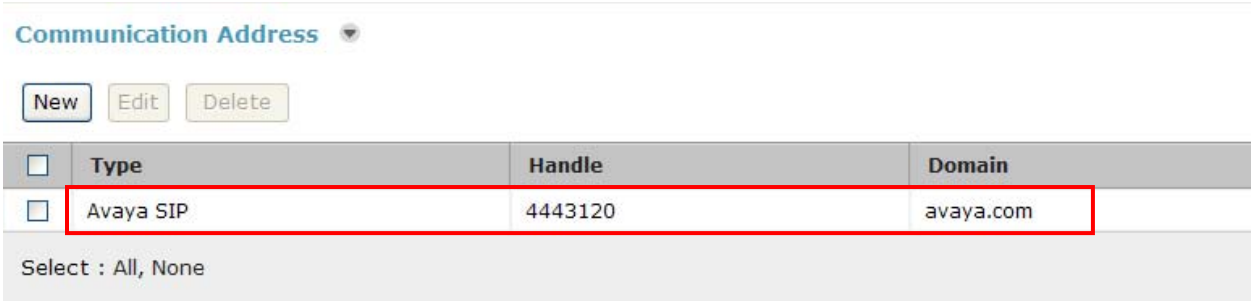

If an entry does not exist, select **New** and enter values for the following required attributes:

- **Name:** Enter "**Primary**".
- **Default:** Verify ☑.

**Step 3**: In the **Communication Address** sub-section, select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required attributes:

- **Type:** Select "**Avaya SIP**" from drop-down menu.
- **Fully Qualified Address:** Enter same extension number as used for **Login Name** in **Step 1**. **Note:** value is shown in **Handle** field after address is added.
- **Domain:** Verify Domain matches Domain name defined in **Section 4.1**.

Click **Add** (not shown) to save the Communication Address for the new SIP user. The screen below shows results from **Step 3:**





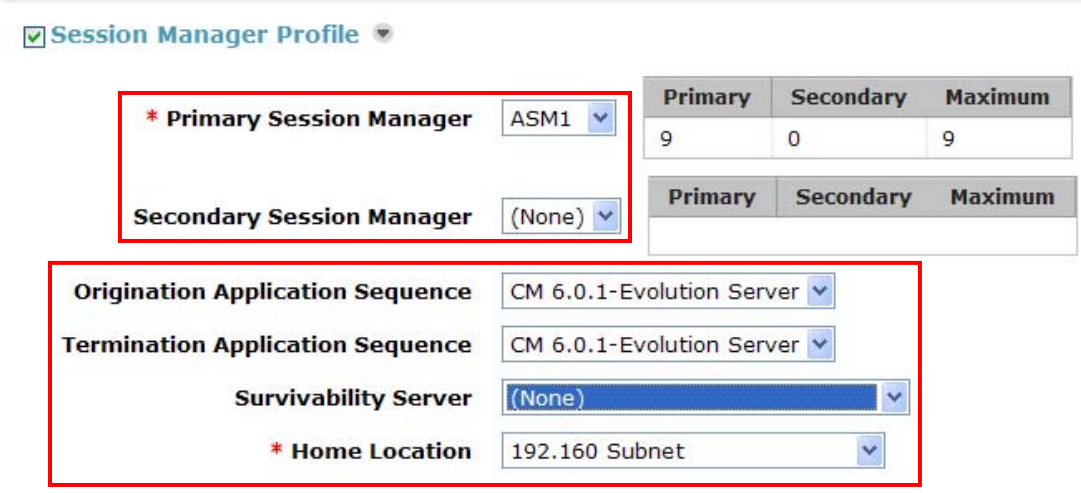
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**Step 4**: In the **Session Manager Profile** section, enter ☑ to expand section.

Enter values for the following fields**.**

- **Primary Session Manager** — Select **Session Manager**.
- **Secondary Session Manager:** — Select "**(None)**" from drop-down menu.
- **Origination Application Sequence** — Select **Application Sequence** defined in **Section 4.6** for Communication Manager.
- **Termination Application Sequence** — Select **Application Sequence** defined in **Section 4.6** for Communication Manager.
- **Survivability Server** — Select "**(None)**" from drop-down menu.
- **Home Location** — Select **Location** defined in **Section 4.2**.

The screen below shows results from **Step 4**.





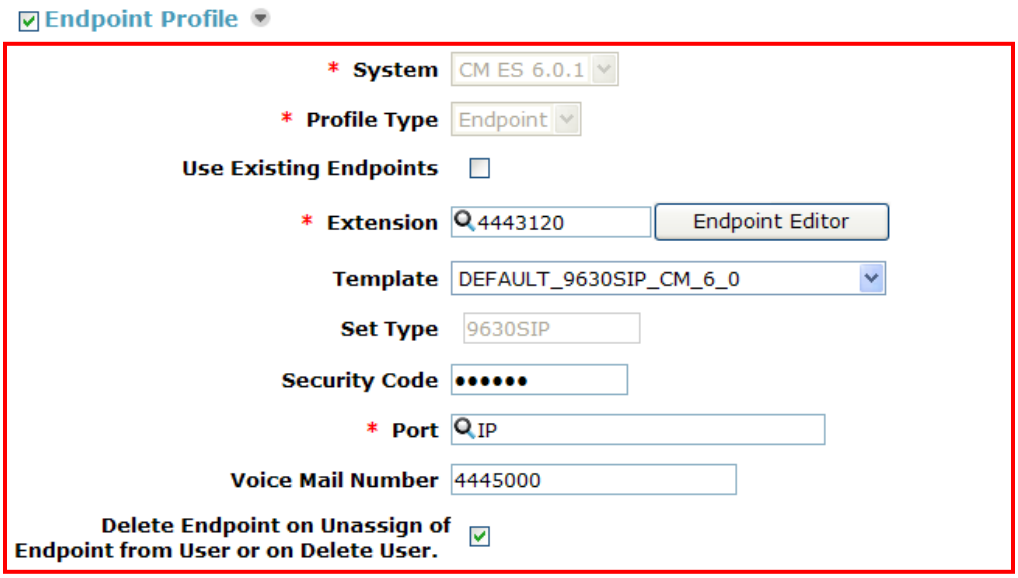
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**Step 5**: In the **Endpoint Profile** section, enter ☑ to expand section.

Enter values for the following fields.

- **System** Select Managed Element associated with Communication Manager Evolution Server.
- **Use Existing Endpoints** Leave unchecked to automatically create new endpoint when new user is created.
  Else, enter ☑ if endpoint was already defined in **Section 3.11**.
- **Extension** Enter same extension number used for **Login Name** in **Step 1**.
- **Template** Select "**DEFAULT_9630SIP_CM_6_0**".
- **Security Code** Enter numeric value used to log on to SIP telephone
  **Note:** this field must match the value entered for the **Communication Profile Password** field in **Step 2**.
- **Port** Select "**IP**" from drop down menu.
- **Voice Mail Number** Enter **Pilot Number** for Avaya Aura® Messaging.
- **Delete Station on Unassign of Endpoint** Enter ☑ to automatically delete station when **Endpoint Profile** is un-assigned from user.

The screen below shows the results from **Step 5** when adding a new SIP user in the sample configuration.



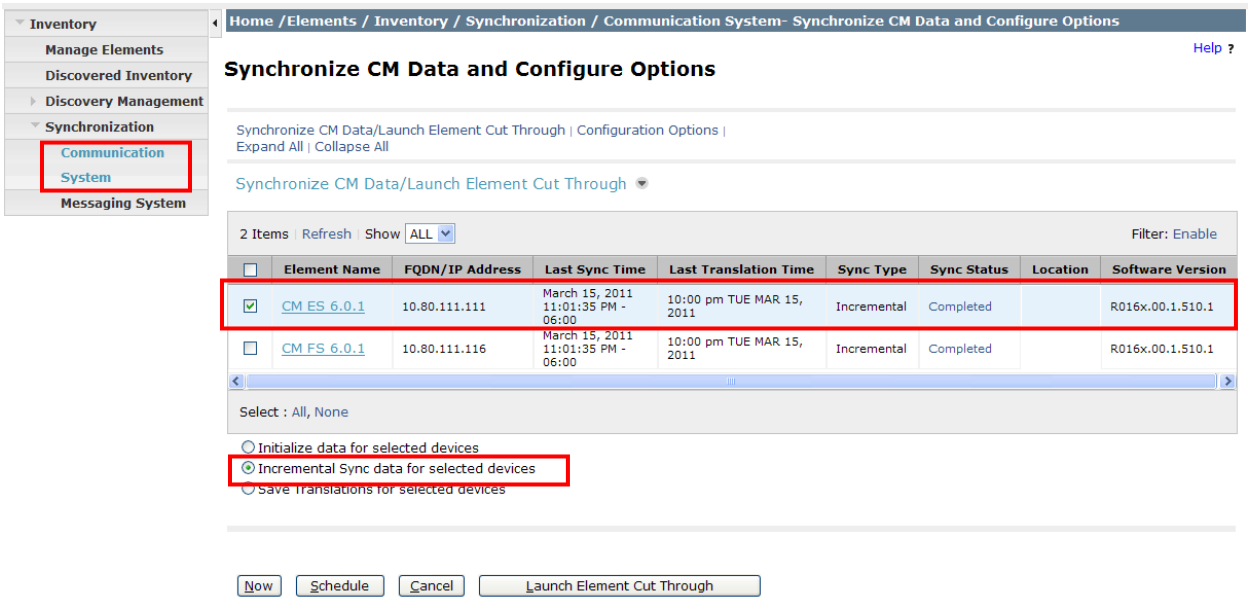

Click **Commit** (not shown) to save definition of new user.


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## 4.8. Synchronize Changes with Avaya Aura® Communication Manager

After completing these changes in System Manager, perform an on demand synchronization. Expand **Elements → Inventory → Synchronization** and select **Communication System.**

On the **Synchronize CM Data and Configure Options** page, expand the **Synchronize CM Data/Launch Element Cut Through** table and select the row associated with Communication Manager Evolution Server as shown below.




Click ◉ to select **Incremental Sync data for selected devices** option. Click **Now** to start the synchronization.

Use the **Refresh** button in the table header to verify status of the synchronization.

Verify synchronization successfully completes by verifying the status in the **Sync. Status** column is "**Completed**".

**Note:** Depending on the number of administration changes made, synchronization might take several minutes to complete.


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# 5. Configure Avaya 1100-Series and 1200-Series IP Deskphones

This section describes the basic configuration of the Avaya 1100-Series and 1200-Series IP Deskphones used in the sample configuration. For additional information on configuring these types of endpoints, see **References [8, 9]** in **Section 9**.

## 5.1. Configure Initial Network Parameters

Network configuration of the telephone can be accomplished either manually or via DHCP. Once network configuration is finished, configuration files are used to configure other settings.

To manually configure the telephone, access the **Device Settings** screen on the telephone and enter the appropriate password. Enter the appropriate values for **IP address, mask, default gateway, file server address**, and **file server access type** fields. For the sample configuration, "**HTTP**" was selected as the type of file server.

When the telephone boots, it locates the **"<ModelNumber>SIP.cfg**" file from the root directory of the HTTP server, where **<ModelNumber>** is the model number for the specific telephone. For example, for the 1165E Deskphone, the file name would be "**1165eSIP.cfg**". This configuration file contains the following three sections:

- **[DEVICE_CONFIG]** Main device configuration file for configuring local features
- **[FW]** Firmware Release
- **[DIALING_PLAN]** Local dial plan

Each section specifies the **FILENAME** to be accessed and the **PROTOCOL** to be used for downloading the file from the file server. One of the configuration files used in sample configuration for configuring 1165E Deskphone is shown below.

**Note:** A value of "**FORCED**" for the **DOWNLOAD_MODE** for each section ensures explicit control for when files will be downloaded.

```
[DEVICE_CONFIG]
DOWNLOAD_MODE   FORCED
VERSION         000100
PROTOCOL        HTTP
FILENAME        1165DeviceConfig.dat

[FW]
DOWNLOAD_MODE   FORCED
VERSION         SIP1165e04.00.04.00
PROTOCOL        HTTP
FILENAME        SIP1165e04.00.04.00.bin

[DIALING_PLAN]
DOWNLOAD_MODE   FORCED
VERSION         000020
PROTOCOL        HTTP
FILENAME        dialplan.txt
```


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## 5.2. Configure Local Telephone Features

After the configuration file in the previous section has been downloaded, the telephone will download the files referenced and will automatically upgrade to the firmware version specified. After upgrading the firmware, the telephone reboots and downloads the specified main device configuration and local dial plan files.

An annotated copy of the main device configuration file used in the sample configuration is shown below. The important parameters, their use, and the values used for the sample configuration are shown in **bold**.

**Note:** the file shown below has been abbreviated for clarity and does not contain many of the default settings.

```
# ----------------------------------------------------
# SIP Proxy Server Domain information
# Note: Multiple domains can be defined. The first domain corresponds to the
# domain used in the sample configuration and should match the domain
# configured in Communication Manager and Session Manager
# ----------------------------------------------------
SIP_DOMAIN1 avaya.com
SIP_DOMAIN3 abc.com
SIP_DOMAIN4 xyz.com
SIP_DOMAIN5 test.com

#------DNS domain. Should match domain specified in Section 3.4
DNS_DOMAIN avaya.com

# ----------------------------------------------------
# Specifies Session Manager as the SIP registrar for domain avaya.com
# A second address parameter could be specified as a backup registrar for
# failover (not tested).
# ----------------------------------------------------
SERVER_IP1_1 10.80.111.107
SERVER_IP1_2 10.80.111.107

SERVER_IP2_1 0.0.0.0
SERVER_IP2_2 0.0.0.0

#------UDP Port numbers
# Note: UDP was not used in the sample configuration
SERVER_PORT1_1 5060
SERVER_PORT1_2 5060
SERVER_PORT2_1 5060
SERVER_PORT2_2 5060

#------TCP Port numbers, enter 0 to disable
# TCP is used in the sample configuration
SERVER_TCP_PORT1_1 5060
SERVER_TCP_PORT1_2 5060
SERVER_TCP_PORT2_1 0
SERVER_TCP_PORT2_2 0
```


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```
#------TLS Port numbers, 0 to disable. If enabled, 5061 is typically used.
# Note: TLS was not used in the sample configuration.
SERVER_TLS_PORT1_1 0
SERVER_TLS_PORT1_2 0
SERVER_TLS_PORT2_1 0
SERVER_TLS_PORT2_2 0

#------Listening ports
SIP_UDP_PORT 5060
SIP_TCP_PORT 5060
SIP_TLS_PORT 0

#------Server retries
SERVER_RETRIES1 3
SERVER_RETRIES2 3
SERVER_RETRIES3 3

#------Recovery & Log levels
RECOVERY_LEVEL 2
LOG_LEVEL 255

#------Firmware update
AUTO_UPDATE YES
AUTO_UPDATE_TIME 0

#------Service Package
# Not supported in this configuration
ENABLE_SERVICE_PACKAGE NO

#------Service Package http or https
#SERVICE_PACKAGE_PROTOCOL HTTP

# ------------------------------------------------------
# Banner
# ------------------------------------------------------
FORCE_BANNER YES
BANNER       Avaya

#------Autologin
AUTOLOGIN_ENABLE  YES

#------Enable/Disable SIP ping
SIP_PING YES

#-------------------------------------------------------
# VMAIL Settings
# Voice mail extension dialed when “messages” button is pressed.
# Enter Pilot Number for Avaya Aura® Messaging
# ------------------------------------------------------
VMAIL 4445000
VMAIL_DELAY 600

#------Specify Transfer, Hold, and Conference settings.
TRANSFER_TYPE    STANDARD
HOLD_TYPE        RFC3261
ENABLE_3WAY_CALL YES
REDIRECT_TYPE    RFC3261
```


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```
#------Maximum number of Multi user logins
MAX_LOGINS 5


#------Enable UPDATE method
ENABLE_UPDATE    YES
ENABLE_PRACK     YES

#------PROXY Checking
PROXY_CHECKING YES

#------File Manager
FM_CONFIG_ENABLE YES
FM_CERTS_ENABLE YES
FM_CONFIG_ENABLE YES

# Local Storage Limits
# ----------------------------------------------------
MAX_INBOX_ENTRIES        100
MAX_OUTBOX_ENTRIES       100
MAX_REJECTREASONS        20
MAX_CALLSUBJECT          20
MAX_PRESENCENOTE         20
MAX_IM_ENTRIES           999
MAX_ADDR_BOOK_ENTRIES    100

#------Session Timer Setttings
SESSION_TIMER_ENABLE      NO
RECOVERY_LEVEL 2

#------End
```

For more information describing other configuration settings for Avaya 1100-Series and 1200-Series SIP Deskphones, see **References [13]** and **[14]** in **Section 9**.


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## 5.3. Configure Local Dial Plan

The telephone will use a local dial plan configuration file to determine when enough digits have been pressed to complete dialing, so that the user need not press an additional key to launch the call. The **DIALING_PLAN** file is downloaded from the file server at boot time as specified in the Configuration file described in **Section 5.1**. An annotated copy of the local dial plan file used in the sample configuration is shown below.

In the sample configuration, since users dial "**444xxxx**" to call other stations or the Pilot Number for Avaya Aura® Messaging, an entry was added to local dial plan file. This entry corresponds to the dial plan configuration in Communication Manager. There is also an entry for long distance dialing using the FAC "9" for ARS routing.

**Note**: each entry also allows for the telephone user to press the "#" key to indicate that dialing is complete.

```
/* ------------------------------------------------------------------ */
/*                                                                    */
/* Avaya 1100-Series and 1200-Series IP Deskphone Dial Plan           */
/*                                                                    */
/* ------------------------------------------------------------------ */

/* Domain used in the dialed URL of the SIP INVITE message */
$n="avaya.com"
$t=300

%%

/* DIGITMAP: 12 digits starting with 9 followed by an initial 1 */
(9[^1]x{10})|(9[^1]x{10})#     && sip:$$@$n;user=phone          && t=300

/* DIGITMAP: 7 Digit Extensions beginning with 444 */
(444x{4})|(444x{4})#     && sip:$$@$n;user=phone          && t=300

/* End of Dial Plan */
```


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# 6. Verification Steps

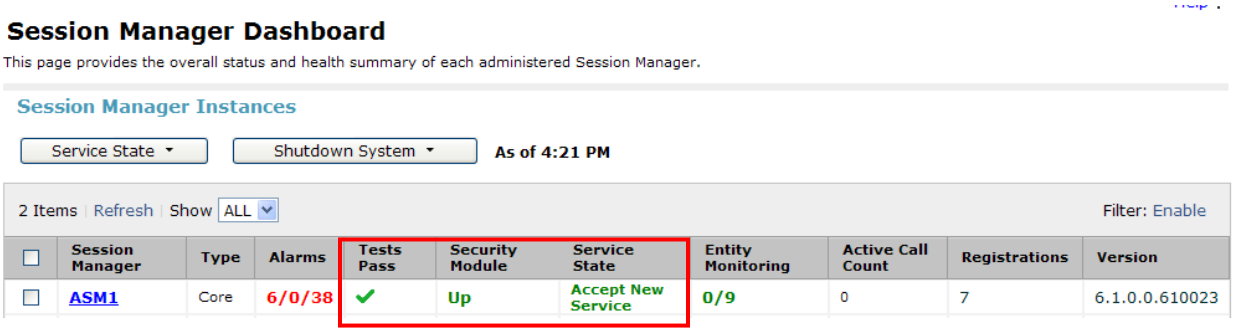
## 6.1. Verify Avaya Aura® Session Manager Operational Status

**Step 1:** Verify overall system status of Session Manager.

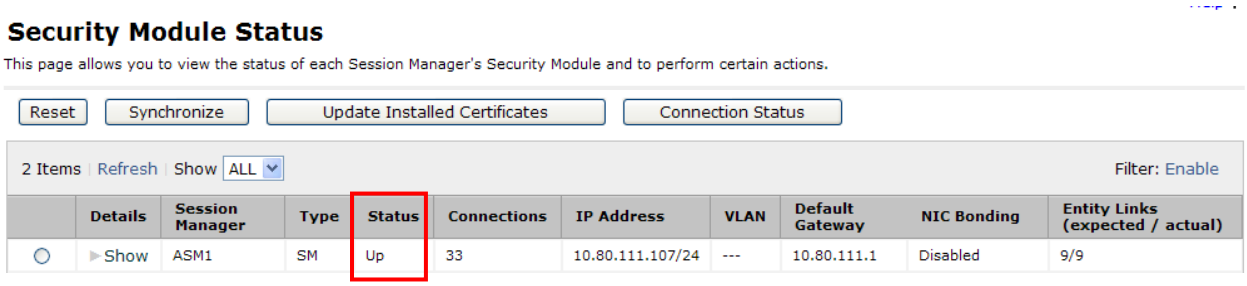
Expand **Elements → Session Manager** and select **Dashboard** to verify the overall system status for both Session Managers.

Specifically, verify the status of the following fields as shown below:

- **Tests Pass** ✔
- **Security Module** Up
- **Service State** Accept New Service




Expand **Elements → Session Manager → System Status** and select **Security Module Status** to view more detailed status information on the status of Security Module for the Session Manager. Verify the **Status** column displays "**Up**" as shown below.





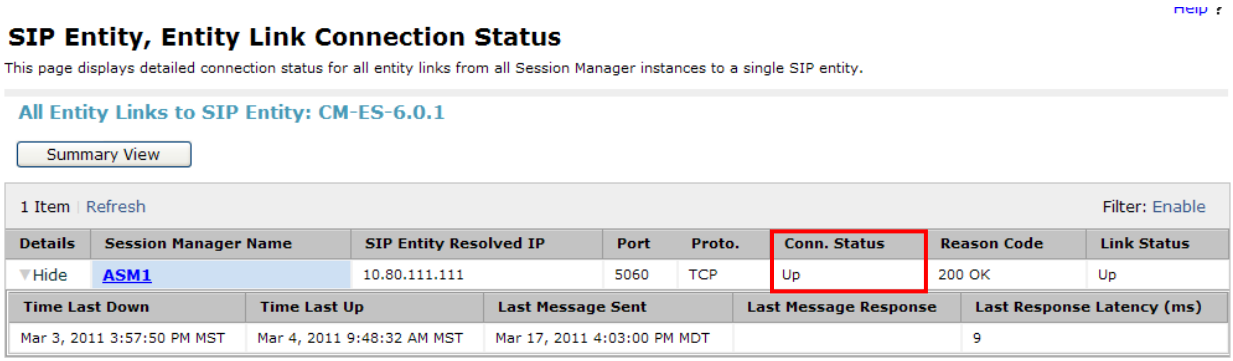
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**Step 2**: Verify status of the SIP Trunks between Session Manager and either Communication Manager or Avaya Aura® Messaging.

Expand **Elements → Session Manager → System Status** and select **SIP Entity Monitoring** to view more detailed status information for one of the SIP Entity Links.

Select the SIP Entity for Communication Manager Evolution Server from the **All Monitored SIP Entities** table to open the **SIP Entity, Entity Link Connection Status** page.

In the **All Entity Links to SIP Entity: CM-ES-6.0.1** table, verify the **Conn. Status** for link is "**Up**". Click **Show** in the **Details** column to view additional status information for the selected link as shown below:



**SIP Entity, Entity Link Connection Status**

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

**All Entity Links to SIP Entity: CM-ES-6.0.1**

Summary View

1 Item | Refresh — Filter: Enable

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| Hide | ASM1 | 10.80.111.111 | 5060 | TCP | Up | 200 OK | Up |

| Time Last Down | Time Last Up | Last Message Sent | Last Message Response | Last Response Latency (ms) |
|---|---|---|---|---|
| Mar 3, 2011 3:57:50 PM MST | Mar 4, 2011 9:48:32 AM MST | Mar 17, 2011 4:03:00 PM MDT | | 9 |

Repeat the steps to verify the Entity Link status for SIP Trunk to Avaya Aura® Messaging.

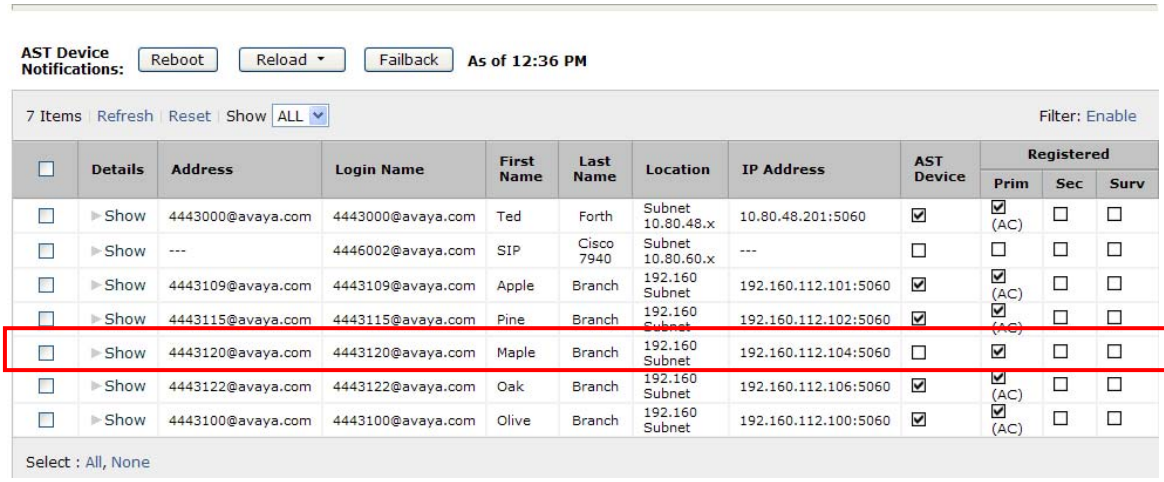

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**Step 3**: Verify Registrations of SIP Endpoints

Expand **Elements → Session Manager →System Status** and select **User Registrations** to verify the SIP endpoints have successfully registered with Session Manager.

For example, the screen below highlights an Avaya 1100-Series SIP Deskphone successfully registered to Session Manager.

**Note:** As previously mentioned, Avaya 1100-Series and 1200-Series SIP Deskphones do not currently support the Avaya Advanced SIP Telephony (AST) protocol. However, Communication Manager and Session Manager have the capability to extend some advanced telephony features to non-AST telephones. See **References [15]** and **[16]** in **Section 9** for more information on configuring these features on Avaya 1100-Series and 1200-Series IP Deskphones.



AST Device Notifications: Reboot | Reload | Failback | As of 12:36 PM

7 Items | Refresh | Reset | Show ALL | Filter: Enable

| | Details | Address | Login Name | First Name | Last Name | Location | IP Address | AST Device | Registered | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| | | | | | | | | | Prim | Sec | Surv |
| ☐ | Show | 4443000@avaya.com | 4443000@avaya.com | Ted | Forth | Subnet 10.80.48.x | 10.80.48.201:5060 | ☑ | ☑ (AC) | ☐ | ☐ |
| ☐ | Show | --- | 4446002@avaya.com | SIP | Cisco 7940 | Subnet 10.80.60.x | --- | ☐ | ☐ | ☐ | ☐ |
| ☐ | Show | 4443109@avaya.com | 4443109@avaya.com | Apple | Branch | 192.160 Subnet | 192.160.112.101:5060 | ☑ | ☑ (AC) | ☐ | ☐ |
| ☐ | Show | 4443115@avaya.com | 4443115@avaya.com | Pine | Branch | 192.160 Subnet | 192.160.112.102:5060 | ☑ | ☑ (AC) | ☐ | ☐ |
| ☐ | Show | 4443120@avaya.com | 4443120@avaya.com | Maple | Branch | 192.160 Subnet | 192.160.112.104:5060 | ☐ | ☑ | ☐ | ☐ |
| ☐ | Show | 4443122@avaya.com | 4443122@avaya.com | Oak | Branch | 192.160 Subnet | 192.160.112.106:5060 | ☑ | ☑ (AC) | ☐ | ☐ |
| ☐ | Show | 4443100@avaya.com | 4443100@avaya.com | Olive | Branch | 192.160 Subnet | 192.160.112.100:5060 | ☑ | ☑ (AC) | ☐ | ☐ |

Select : All, None


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## 6.2. Verify Avaya Aura® Communication Manager Operational Status

Verify the status the SIP trunk group on Communication Manager Evolution Server by using the **status trunk n** command, where **n** is the trunk group administered in **Section 3.6.**

Verify that all trunks in the trunk group are in the "**in-service/idle**" state as shown below:

```
status trunk 10
                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy
0010/001 T00006   in-service/idle    no
0010/002 T00007   in-service/idle    no
0010/003 T00008   in-service/idle    no
0010/004 T00009   in-service/idle    no
0010/005 T00014   in-service/idle    no
0010/006 T00015   in-service/idle    no
0010/007 T00043   in-service/idle    no
0010/008 T00044   in-service/idle    no
0010/009 T00045   in-service/idle    no
0010/010 T00046   in-service/idle    no
```

Verify the status the SIP signaling group by using the **status signaling-group** command, where **n** is the signaling group number administered in **Section 3.6.**

Verify the signaling group is "**in-service**" as indicated in the **Group State** field shown below:

```
status signaling-group 10
                         STATUS SIGNALING GROUP

       Group ID: 10                               Active NCA-TSC Count: 0
     Group Type: sip                               Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
    Group State: in-service
```


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Use the SAT command, **list trace tac #**, where **#** is the trunk access code the trunk group defined in **Section 3.6** to trace trunk group activity for the SIP trunk between Session Manager and Communication Manager. For example, the trace below illustrates a call from an Avaya 1100-Series SIP Deskphone using Extension "**444-3120**" to an IP (H.323) station using Extension "**444-1000**".

```
list trace tac #10                                          Page   1
                                LIST TRACE
time            data

16:04:08 TRACE STARTED 03/17/2011 CM Release String cold-00.1.510.1-18777
16:04:17 SIP<INVITE sip:4441000@avaya.com;user=phone SIP/2.0
16:04:17 SIP>SIP/2.0 180 Ringing
16:04:17     dial 4441000
16:04:17     ring station    4441000 cid 0x92
16:04:17     G729A ss:off ps:20
             rgn:1 [10.80.48.194]:2704
             rgn:1 [10.80.111.108]:2052
16:04:17     xoip options: fax:Relay modem:off tty:US  uid:0x50001
             xoip ip: [10.80.111.108]:2054
16:04:17 SIP<PRACK sip:4441000@10.80.111.111;transport=tcp SIP/2
```

On Communication Manager, use the SAT command, **list trace station xxx**, where **xxx** is a valid extension number for a SIP telephone. For example, the trace below illustrates a second call from the same SIP telephone used in the previous trace to the same IP station.

```
list trace station 4443120                                  Page   1
                                LIST TRACE
time            data

16:06:39 TRACE STARTED 03/17/2011 CM Release String cold-00.1.510.1-18777
16:06:44     active station    4443120 cid 0x93
16:06:44     dial 4441000
16:06:44     ring station    4441000 cid 0x93
16:06:44     G729A ss:off ps:20
             rgn:1 [10.80.48.194]:2704
             rgn:1 [10.80.111.108]:2050
16:06:44     xoip options: fax:Relay modem:off tty:US  uid:0x50001
             xoip ip: [10.80.111.108]:2072
16:06:46     active station    4441000 cid 0x93
```


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## 6.3. Call Scenarios Verified

Verification scenarios for the configuration described in these Application Notes included the following call scenarios:

**Basic Calls:**

- Place calls from Avaya 1100-Series or 1200-Series SIP Deskphones registered to Session Manager to either digital or IP (H.323) stations. Answer the call and verify talkpath.
- Place calls from Avaya 1100-Series or 1200-Series SIP Deskphones registered to Session Manager to either digital or IP (H.323) stations. Answer the call and place the call on Hold. Return to the held call and verify talkpath.
- Verify calls can be transferred from Avaya 1100-Series or 1200-Series SIP Deskphones registered to Session Manager to other stations on Communication Manager.
- Verify calls can be forwarded from Avaya 1100-Series or 1200-Series SIP Deskphones registered to Session Manager to other stations on Communication Manager.
- Verify Avaya 1100-Series or 1200-Series SIP Deskphones registered to Session Manager can create conferences with other SIP Deskphones and non-SIP stations on Communication Manager Evolution Server.
- Repeat the above scenarios with calls originating from non-SIP stations on Communication Manager Evolution Server to Avaya 1100-Series or 1200-Series SIP Deskphones registered to Session Manager.

**Basic Messaging Features:**

- Use Pilot Number to access Avaya Aura® Messaging and verify Avaya 1100-Series or 1200-Series SIP subscribers are properly recognized and can login without entering their mailbox number.
- Verify calls between Avaya 1100-Series or 1200-Series SIP subscribers are forwarded to the correct Avaya Aura® Messaging mailbox in both No Answer and Busy conditions.
- Verify calls between Avaya 1100-Series or 1200-Series SIP subscribers are successfully forwarded to Avaya Aura® Messaging and the correct Personal Greetings are played in both No Answer and Busy conditions.
- Verify Avaya 1100-Series or 1200-Series SIP subscribers can leave voice mail messages for other subscribers.
- Verify Avaya Aura® Messaging sends appropriate Message Waiting Notification messages when Avaya 1100-Series or 1200-Series SIP subscribers leave or retrieve messages.

**Supplemental Features:**

- Use Auto Attendant Number to access Avaya Aura® Messaging and verify Avaya Aura® Messaging can successfully transfer calling party to correct Avaya 1100-Series or 1200-Series SIP subscriber
- When Reach-Me is activated for Avaya 1100-Series or 1200-Series SIP subscribers, verify Avaya Aura® Messaging can successfully call the Reach-Me destination. After subscriber accepts call, verify calling party is connected to subscriber.


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- Verify Avaya 1100-Series or 1200-Series SIP subscribers could use Reply, Forward and Call Sender features with other subscribers.
- Verify Avaya Aura® Messaging sends appropriate Message Waiting Notification messages when Avaya 1100-Series or 1200-Series SIP subscribers use Reply or Forward features.
- Verify Avaya 1100-Series or 1200-Series SIP subscribers were able to create 3-party conferences when call was forwarded or re-directed to Avaya Aura® Messaging.

**Long Duration Scenarios**

- Verify Avaya 1100-Series or 1200-Series SIP subscribers can remain on active call with other stations for at least 30 minutes.
- Verify Avaya 1100-Series or 1200-Series SIP subscribers can place a call on hold to other stations for at least 30 minutes.
- Verify Avaya 1100-Series or 1200-Series SIP subscribers can leave long voice mail messages for other subscribers.

## 6.4. Known Limitations

Since Avaya 1100-Series and 1200-Series IP Deskphones with SIP software have not implemented Presence features, testing with Avaya Presence Services has been deferred until a future time.


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# 7. Acronyms

| | |
|---|---|
| AAR | Automatic Alternate Routing (Routing on Communication Manager) |
| ARS | Automatic Route Selection (Routing on Communication Manager) |
| CLAN | Control LAN (Control Card in Communication Manager) |
| DCP | Digital Communications Protocol |
| DTMF | Dual Tone Multi Frequency |
| IMS | IP Multimedia Subsystem |
| IP | Internet Protocol |
| SAT | System Access Terminal (Communication Administration Interface) |
| SIL | Solution Interoperability and Test Lab |
| SIP | Session Initiation Protocol |
| SM | Avaya Aura® Session Manager |
| SMGR | System Manager (used to configure Session Manager) |
| SSH | Secure Shell |
| SSL | Secure Socket Layer |
| TAC | Trunk Access Code (Communication Manager Trunk Access) |
| TCP | Transmission Control Protocol |
| TCP/IP | Transmission Control Protocol/Internet Protocol |
| TLS | Transport Layer Security |
| URL | Uniform Resource Locator |


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# 8. Conclusion

These Application Notes describe how to configure Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server to support Avaya 1100-Series or 1200-Series SIP Deskphones.

Interoperability testing included making bi-directional calls between SIP telephones and other types of stations on Communication Manager Evolution Server. In addition, various features including hold, transfer, and conference were tested. Interoperability testing also included verification that calls from Avaya 1100-Series or 1200-Series SIP subscribers were successfully forwarded to Avaya Aura® Messaging in both busy and no-answer scenarios and Avaya 1100-Series or 1200-Series SIP subscribers could use supplemental Avaya Aura® Messaging features such as Auto Attendant and Reach-Me .


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# 9. Additional References

Avaya Product documentation relevant to these Application Notes is available at http://support.avaya.com

**Avaya Aura® Session Manager**

1) Avaya Aura® Session Manager Overview, Doc ID 100068105.
2) Installing and Configuring Avaya Aura® Session Manager, Doc ID 03-603473.
3) Avaya Aura® Session Manager Case Studies, Doc ID 03-603478.
4) Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325.
5) Administering Avaya Aura® Session Manager, Doc ID 03-603324.

**Avaya Aura® Messaging**

6) Administering Avaya Aura® Messaging, document dated February 2011.
7) Implementing Avaya Aura® Messaging, document dated December 2010.

**Avaya Aura® Communication Manager**

8) SIP Support in Avaya Aura® Communication Manager Running on Avaya S8xxx Servers, Doc ID 555-245-206
9) Administering Avaya Aura® Communication Manager, Doc ID 03-300509
10) Administering Avaya Aura® Communication Manager Server Options, Doc ID 03-603479
11) Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide, Doc ID 210-100-500.
12) Avaya Toll Fraud Security Guide, Doc ID 555-025-600

**Avaya IP Deskphones (SIP)**

13) SIP Software for Avaya 1100 Series IP Deskphones -Administration, Release 4.0 NN43170-600.
14) SIP Software for Avaya 1200 Series IP Deskphones - Administration, Release 4.0 NN43170-601.

**Avaya Application Notes**

15) Advanced Feature Support for Avaya 1100 and 1200 Series IP Deskphones R3.2 with Avaya Aura® Communication Manager 6.0 and Avaya Aura® Session Manager 6.0
16) Application Notes for Avaya 1100- and 1200-Series IP Deskphones R3.2 with Avaya Aura® Communication Manager R6, Avaya Aura® Session Manager R6, and Avaya Modular Messaging R5.2.
17) Configuring SIP Trunks among Avaya Aura® Session Manager R6.1, Avaya Communication Server 1000E R7.5, and Avaya Aura® Messaging R6.0.1


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