Avaya Solution & Interoperability Test Lab

Application Notes for Polycom® SoundStation® IP with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Polycom SoundStation IP phone to interoperate with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0. The Polycom SoundStation IP phones are SIP conference phones that can register with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and conferencing requirements.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Polycom SoundStation IP 5000, IP 6000, IP 7000 and Duo (SoundStation IP) to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. SoundStation IP is a SIP conference phone that registers with Avaya Aura® Session Manager as a SIP endpoint combining the functionality of an IP phone and a conferencing station in support of voice communications and conferencing requirements.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the SoundStation IP and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711, G.722, iLBC and G.729)
- Inbound calls
- Outbound calls
- Hold/Resume
- Call Transfer and Conferencing (Blind and Attended)
- Call termination (origination/destination)
- Avaya Features using FAC
    - Call Park/Unpark
    - Call Pickup
    - Call Forward (Unconditional, Busy/no answer)
    - Find Me
- Voicemail
- Message Waiting Indicator (MWI)
- Serviceability

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of interoperability compliance testing was primarily on verifying call establishment on the SoundStation IP. The SoundStation IP operations such as inbound calls, outbound calls, hold/resume, transfer, conference, Facility Access Codes, and its interactions with Session Manager, Communication Manager, and other Avaya SIP, and H.323 phones were verified. The serviceability testing introduced failure scenarios to see if SoundStation IP can recover from failures.

2.2. Test Results

The test objectives were verified. For serviceability testing, SoundStation IP operated properly after recovering from failures such as network disconnects, and resets of SoundStation IP.

The features mentioned in **Section 2** worked successfully during compliance testing with the following exceptions, as these features are currently not supported by the SoundStation IP:

- Blind Conference Call
- Long Hold Recall Timer
- Find Me
- Blind Conference
- iLBC Codec is supported only between the SoundStation IP endpoints
- At least one hardware-supported codec needs to be listed on SoundStation IP for iLBC or G.722 to work. Additionally, these codecs need to be configured at the top of the list in **Section 6.2**.
- For Facility Access Codes (FAC) to work properly, please refer to **Section 7.4** for proper configuration.

2.3. Support

For technical support on Polycom SoundStation IP, please contact via the following:

- Web: http://support.polycom.com

3. Reference Configuration

Once SoundStation IP registers as a SIP endpoint with Session Manager, it can place and receive voice calls with various supported features as listed above in **Section 2.1**. The reference configuration used for the compliance test is shown in **Figure 1** below.

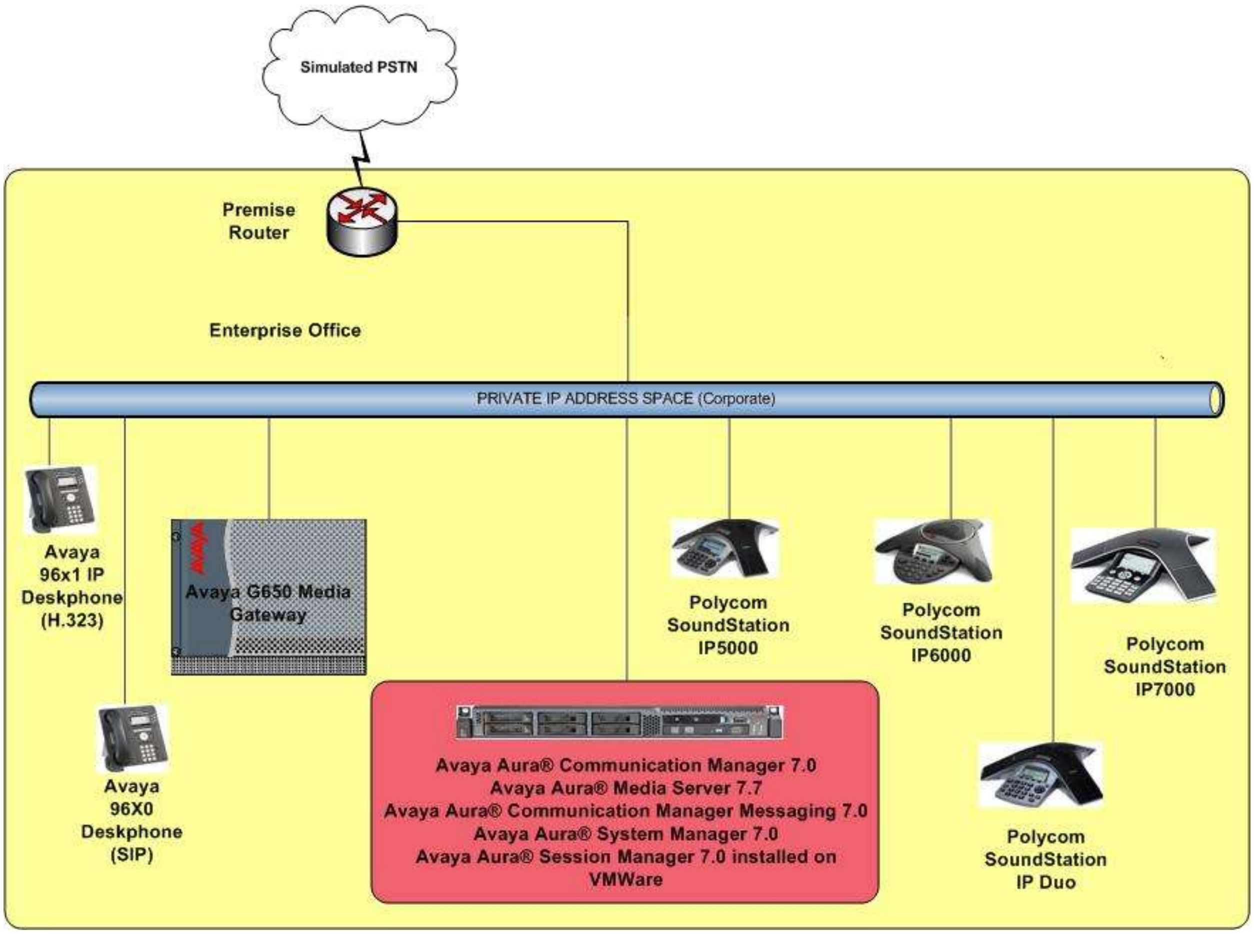


Figure 1: Polycom® SoundStation® IP with Avaya Aura® Session Manager and Avaya Aura® Communication Manager

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya Aura® System Manager installed on VMWare	R7.0 (7.0.0.0.16266)
Avaya Aura® Session Manager installed on VMWare	R7.0 (7.0.0.0.700007)
Avaya Aura® Communication Manager installed on VMWare	R7.0 (vcm-07.00.0.441.0)
Avaya Aura® Media Server installed on VMWare	R7.7 (v.7.7.0.226)
Avaya Aura® Communication Manager Messaging installed on VMWare	R7.0 (vcmm-07.00.0.441.0)
Avaya 96x1 IP Deskphone (H323)	R6.2.2313
Avaya 96x0 IP Deskphone (SIP)	R2.6.9.1
Polycom® SoundStation® IP	UCS 4.0.9.0509

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities and corresponding Entity Links between Session Manager and Communication Manager/Communication Manager Messaging
- Define Communication Manager as Administrable Entity (i.e., Managed Element).
- Application Sequence
- Add SIP Users

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "**https://<*ip-address*>/SMGR**", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice.

Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. To add a location, navigate to **Home→Elements→Routing→Domains** and click the **New** (not shown) button on the right.

The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., ***avaya.com***)
- **Type:** Set to ***sip*** (default)
- **Notes:** Descriptive text (optional)

Click **Commit**.

5.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, navigate to **Home→Elements→Routing→Locations** and click on the **New** (not shown) button on the right. The following screen will then be shown. Fill in the following:

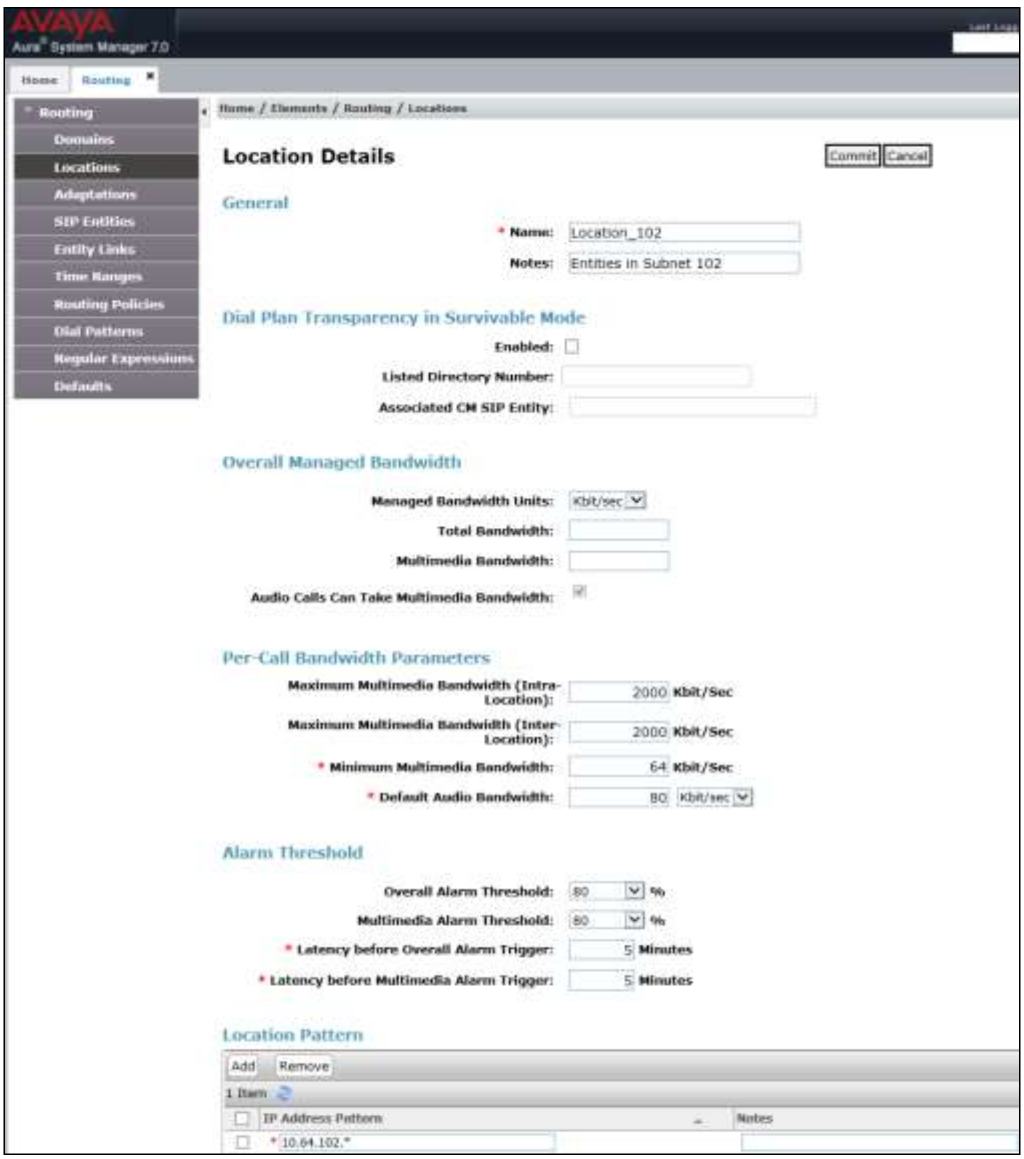
Under ***General***:

- **Name:** A descriptive name
- **Notes:** Descriptive text (optional)

Under ***Location Pattern***:

- **IP Address Pattern:** A pattern used to logically identify the location
- **Notes:** Descriptive text (optional)

The screen below shows addition of the ***Location_102*** location used for Communication Manager and other entities. Similarly a location was defined for Session Manager. Click **Commit** to save the Location definition.



5.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, Communication Manager and Communication Manager Messaging. The screens below also show the corresponding Entity Links.

5.3.1. Session Manager Entity

To add a SIP Entity, navigate to **Home→Elements→Routing→SIP Entities**, and click on **New** (not shown) and configure as follows:

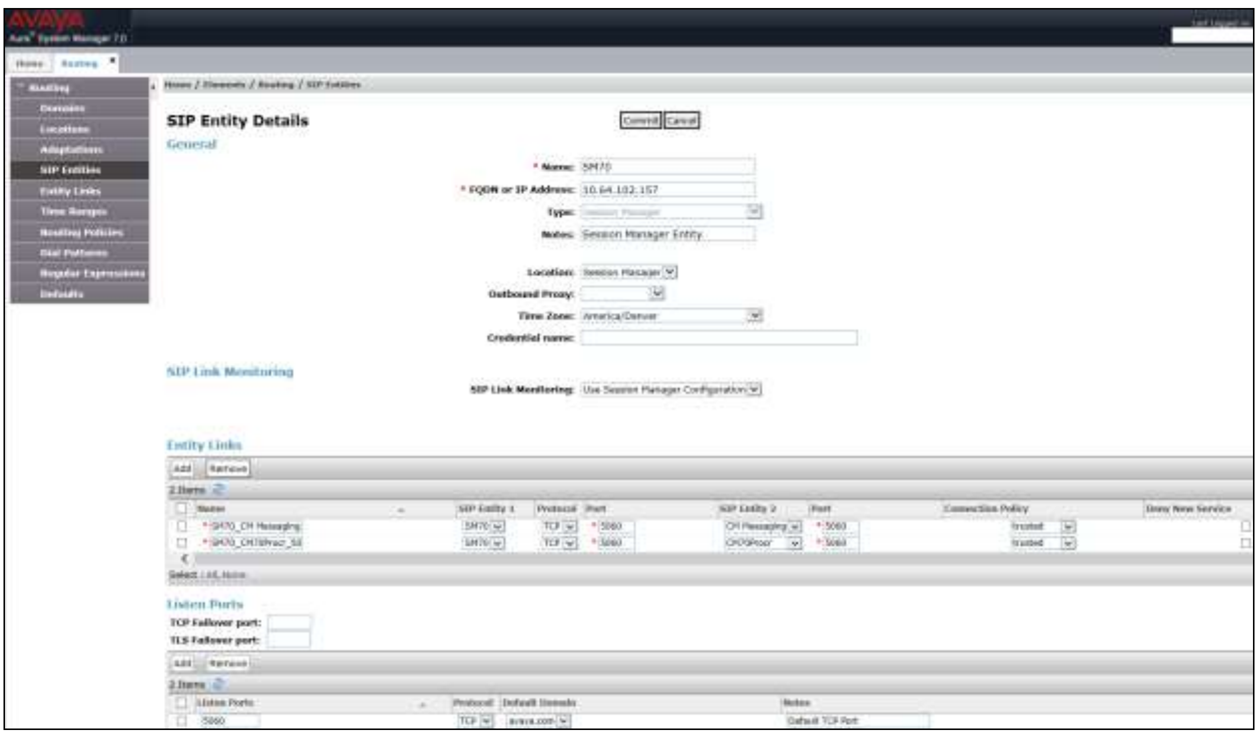
Under ***General***:

- **Name:** Any descriptive name
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager
- **Type:** Select ***Session Manager***
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this location

Under ***Port***, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Listen Ports:** Port number on which the system listens for SIP requests
- **Protocol:** Transport protocol to be used to send SIP requests
- **Default Domain:** The domain used for the enterprise (e.g. ***avaya.com***)

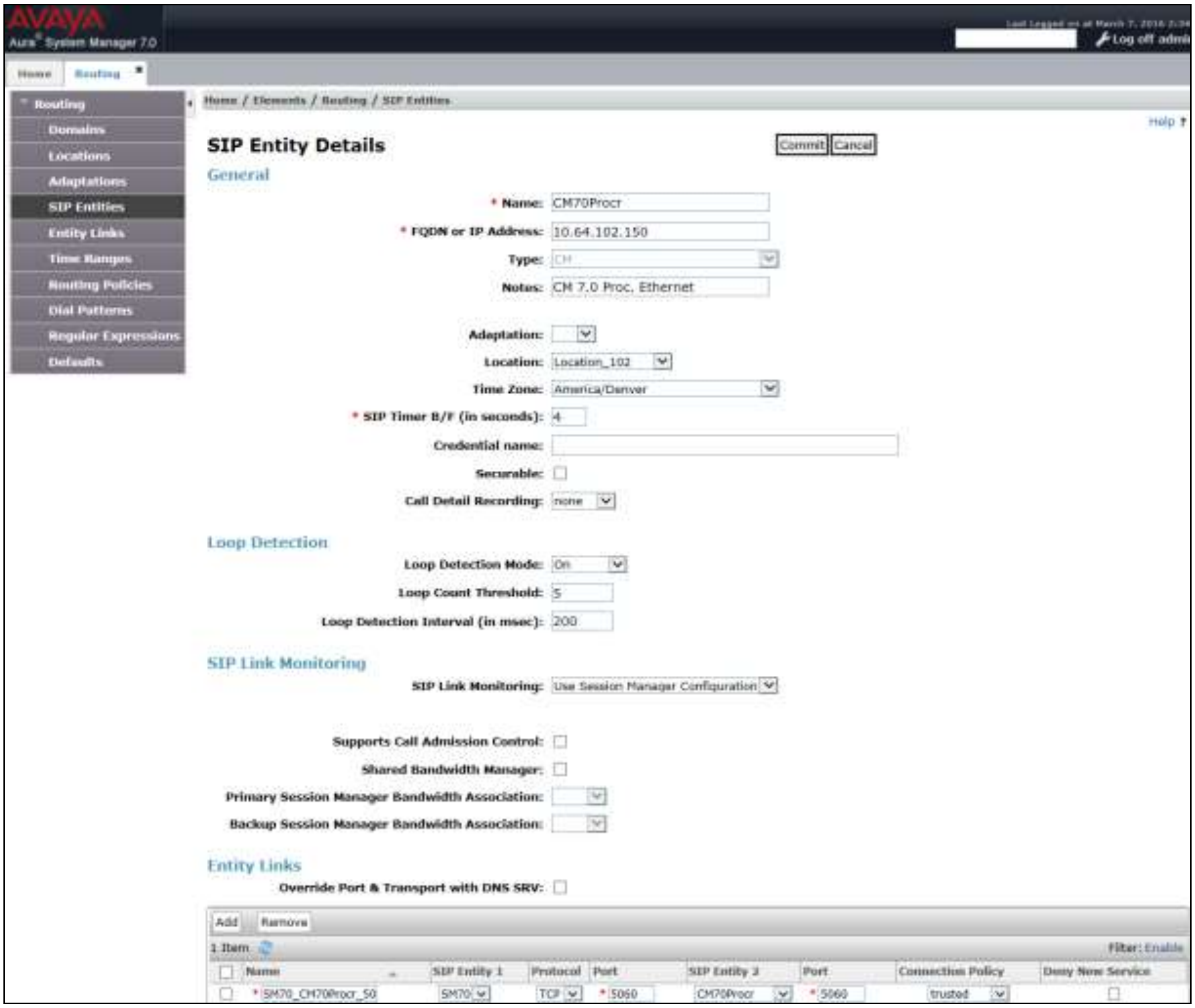
Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



Solution & Interoperability Test Lab Application Notes

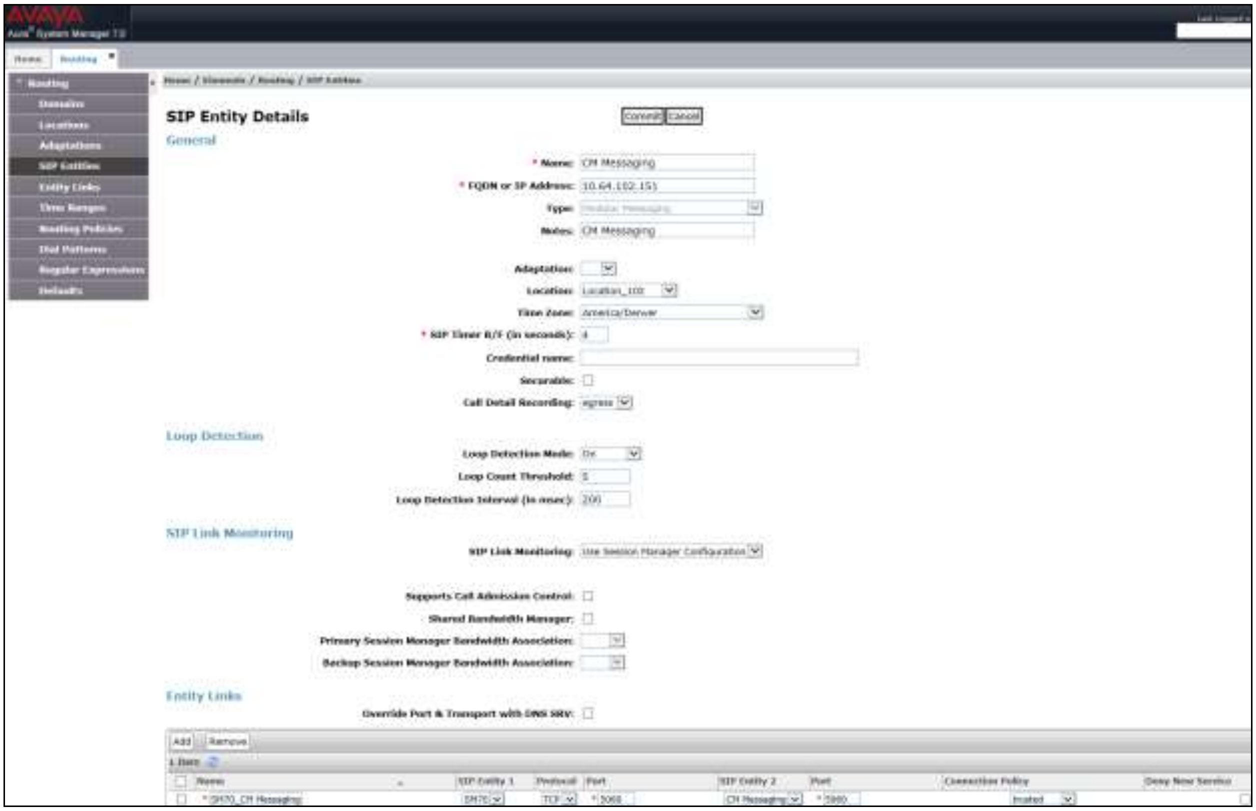
5.3.2. Communication Manager Entity

The following screen displays the Communication Manager entity configured for this reference configuration.



5.3.3. Communication Manager Messaging Entity

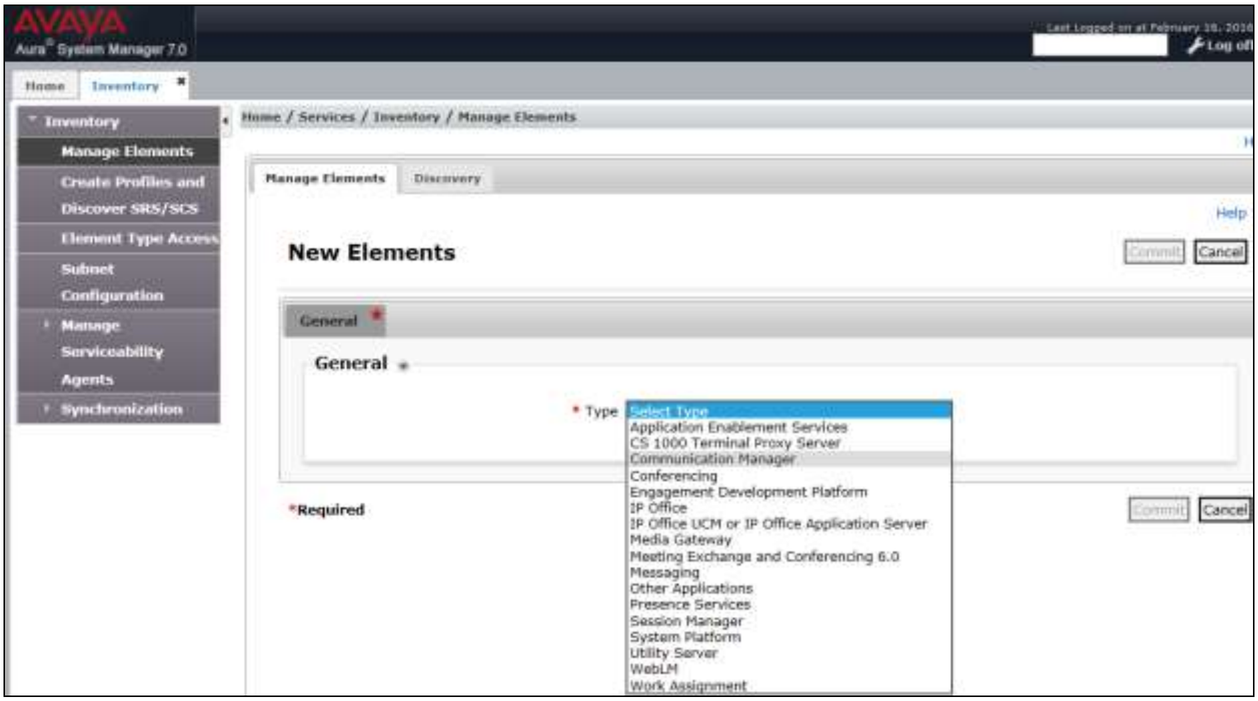
The following screen displays the Communication Manager Messaging entity configured for this reference configuration.



5.4. Define Communication Manager as a Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

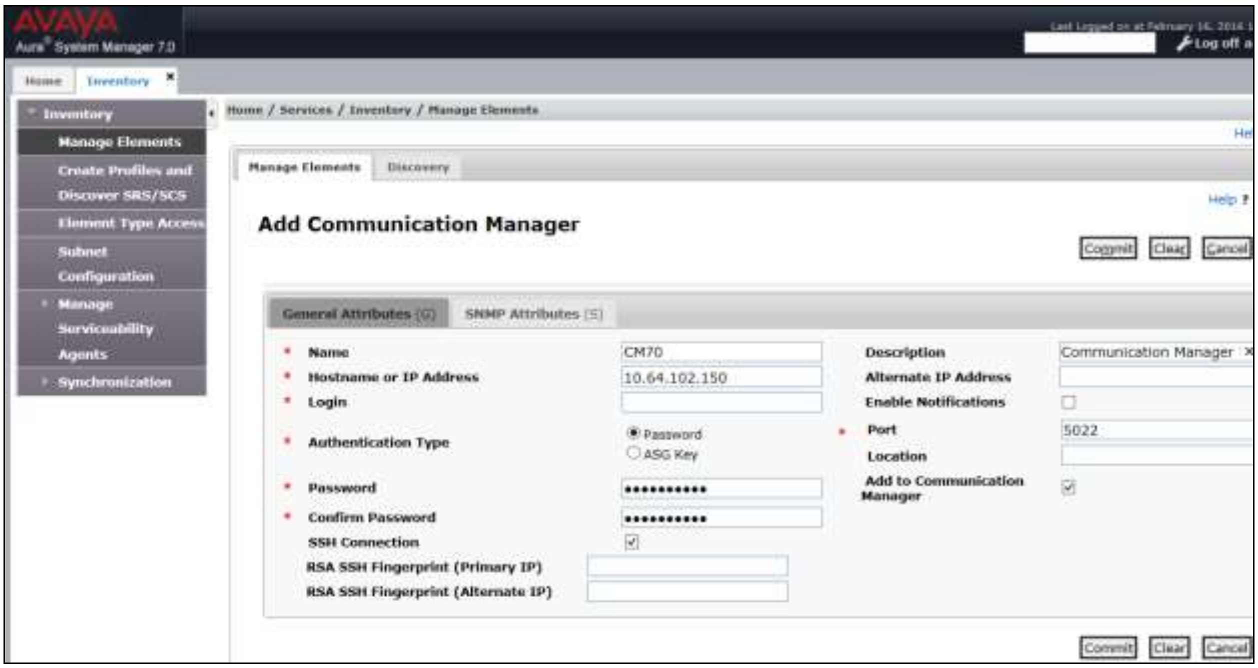
To define Communication Manager as a managed element, navigate to **Home→Services→ Inventory→Manage Elements** on the left and click on the **New** (not shown) button on the right. In the **Type** field that is displayed, select ***Communication Manager***.



In the **Add Communication Manager** screen, fill in the following fields as follows:
Under *General Attributes*:

- **Name:** Enter an identifier for Communication Manager
- **Hostname or IP Address:** Enter the IP address of the administration interface for Communication Manager
- **Login:** Enter the login used for administration access to Communication Manager
- **Authentication Type:** Select the **Password** button
- **Password** Enter a valid password
- **Confirm Password** This should match the password entered in the **Password** field above

Click **Commit** to save.

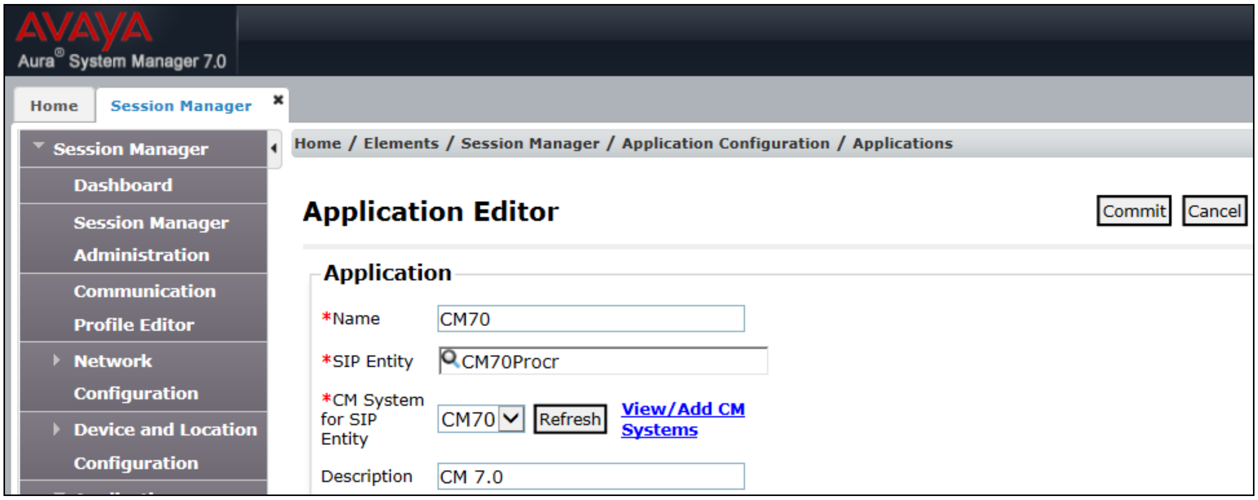


5.5. Add Application Sequence

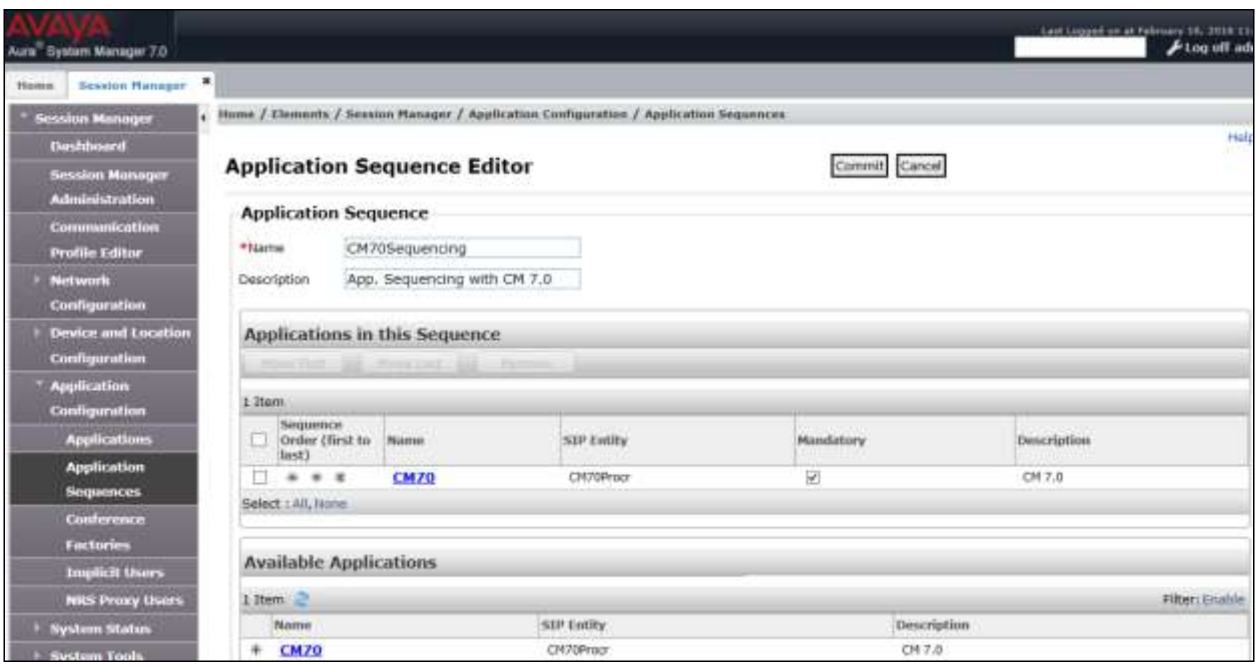
Navigate to **Home→Elements→Session Manager→Application Configuration→Applications** and configure as follows:

- **Name:** Enter any descriptive name
- **SIP Entity:** Select the Communication Manager SIP Entity configured in **Section 5.3.2**
- **CM System for SIP Entity:** Select the system configured in **Section 5.4**

Click **Commit** to save the Application definition.



Next, define the **Application Sequence** for Communication Manager as shown below.

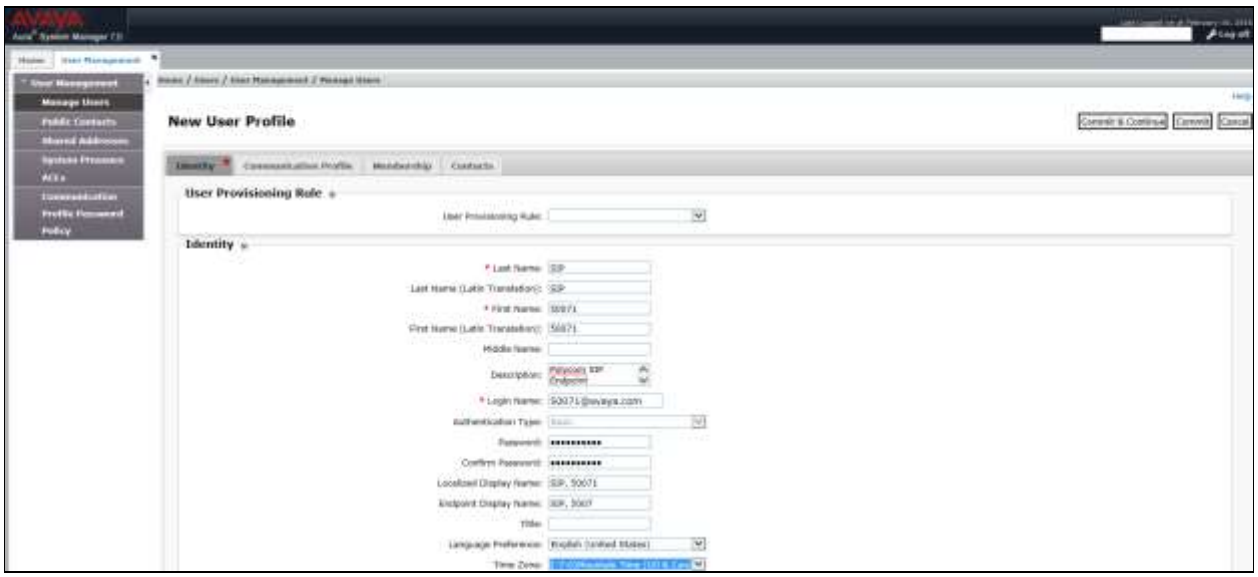


5.6. Add SIP Users

SoundStation IP was entered as a SIP user on Session Manager using the following steps. Navigate to **Home→Users→User Management→Manage Users** and configure as described below. This configuration is automatically synchronized with Communication Manger as verified in **Section 6.3**.

Enter values for the following required attributes for a SIP user in the **New User Profile** form:

- **Last Name:** Enter the last name of the user
- **First Name:** Enter the first name of the user
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., ***50071@avaya.com***)
- **Password:** Enter the password used to register with System Manager
- **Confirm Password:** Re-enter the password from above



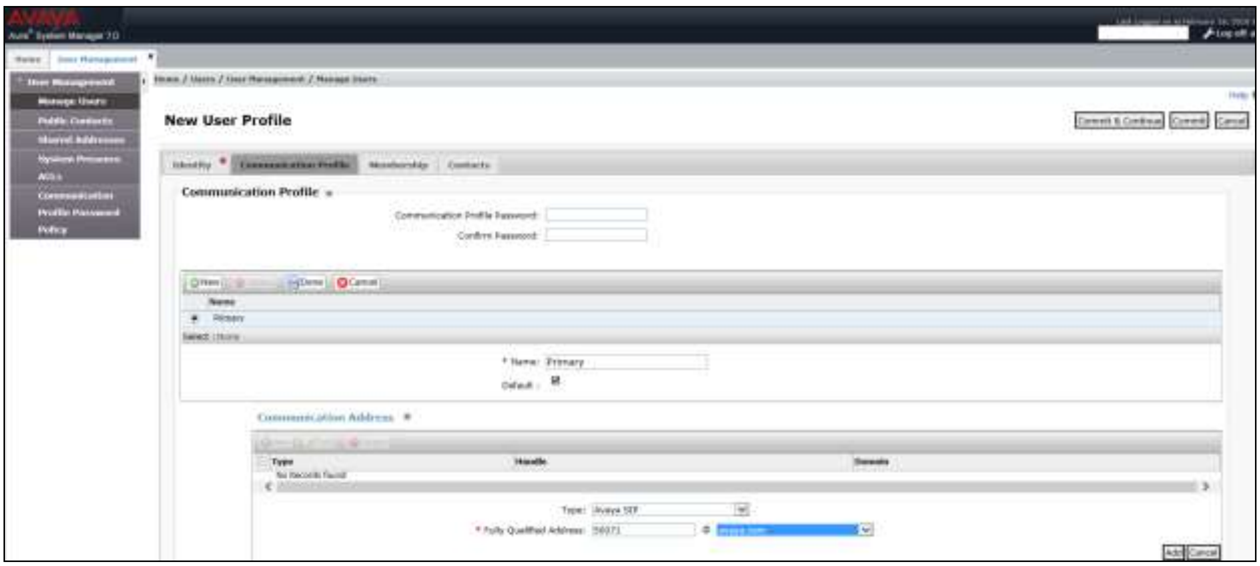
Click the **Communication Profile** tab and select **New** (not shown) to define a **Communication Profile** for a new SIP user. Enter values for the following required fields:

- **Communication Profile Password:** Enter a valid password.
- **Confirm Password:** Make sure that it matches the password entered above.
- **Name:** Enter name of the communication profile.
- **Default:** Check box to indicate that it is the default profile.

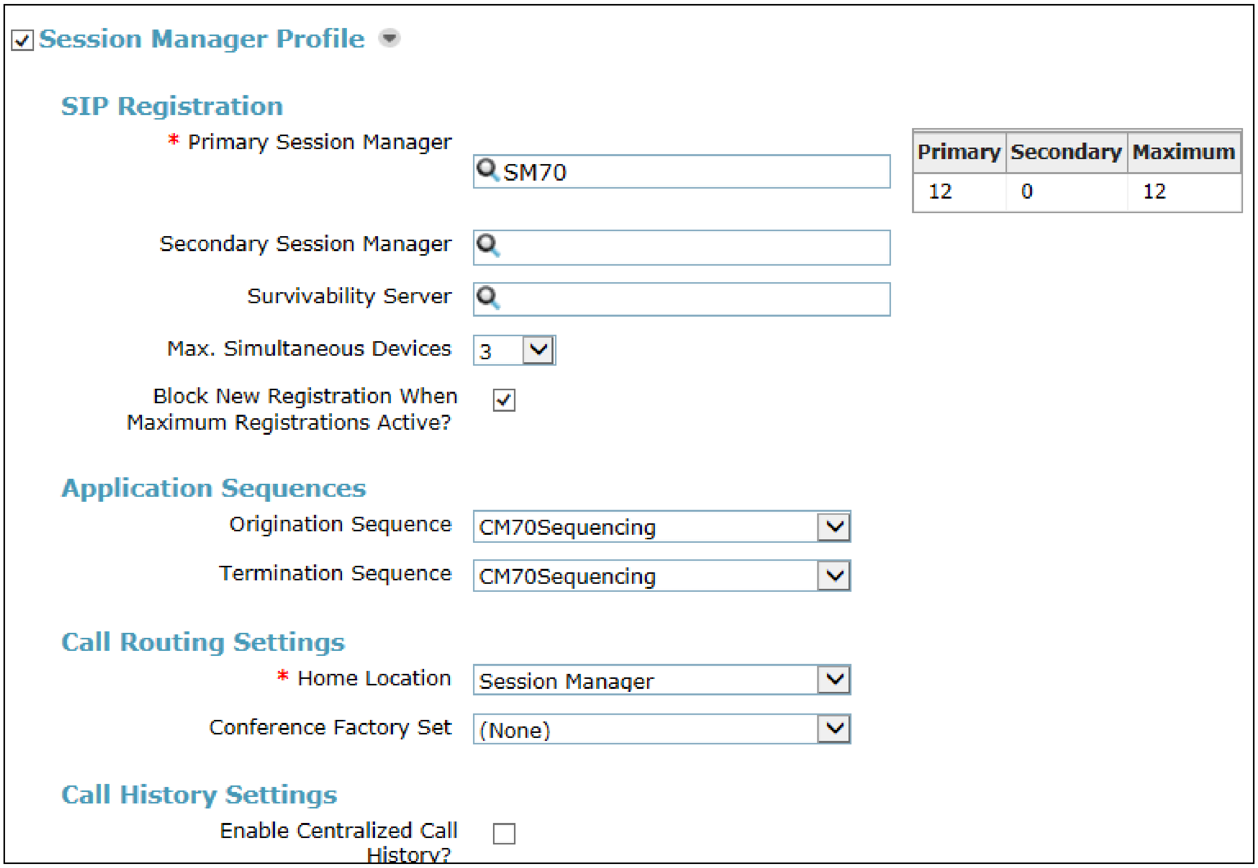
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.

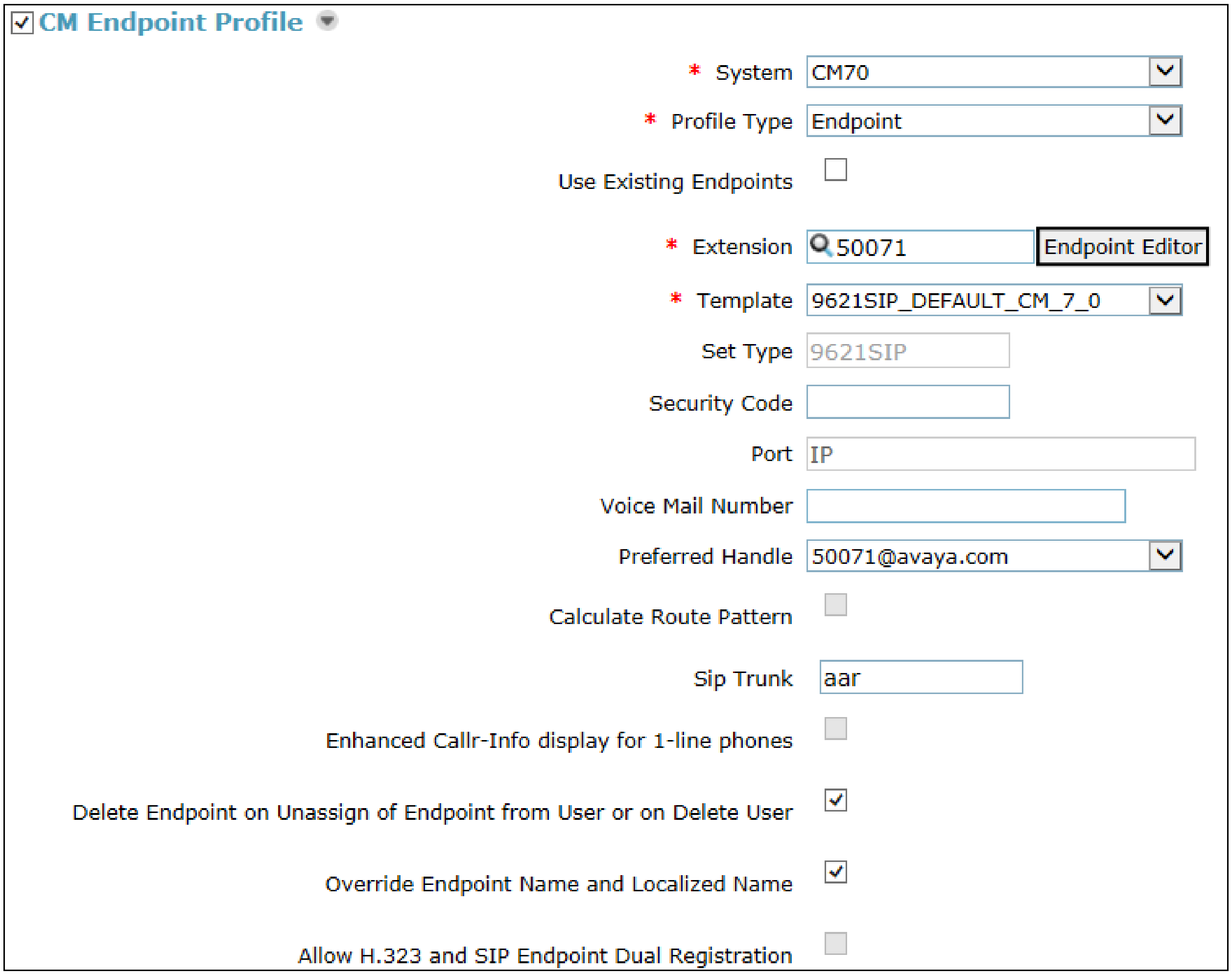


In the **Session Manager Profile** section, specify the Session Manager entity configured in **Section 5.3.1** and assign the **Application Sequence** defined in **Section 5.5** to both the **Originating Sequence** and **Termination Sequence** fields. Additionally, set **Home Location** field to ***Session Manager*** configured in **Section 5.2**.



In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager in **Section 5.4**
- **Profile Type:** Select ***Endpoint***
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager
- **Extension:** Enter extension number of SIP user
- **Template:** Select template for type of SIP phone which is set to **9621SIP_DEFAULT_CM_7_0** for **SoundStation IP**
- **Preferred Handle:** Select the value from the drop-down list



6. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the SoundStation IP as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

6.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 6400  25
                                      Maximum Stations: 2400  10
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  0
                    Maximum Off-PBX Telephones -   OPS: 9600  5
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

```
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  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  160
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

6.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for Session Manager (***ASM70***) and Media Server (***AMS70***). The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
    Name              IP Address
default             0.0.0.0
ASM70               10.64.102.157
CMM70               10.64.102.151
AMS70               10.64.102.158
procr               10.64.102.150
procr6              ::
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is ***avaya.com***. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. The **IP Network Region** form also specifies the **Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region ***2***) is specified in the SIP signaling group.

```
change ip-network-region 2                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: avaya.com
    Name: Main Network Region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
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
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the SoundStation IP. The form is accessed via the **change ip-codec-set *2*** command. Note that IP codec set ***2*** was specified in IP Network Region ***2*** shown above. The following form shows the list of codecs tested. The order of these codecs was changed to support the some of the codecs for reasons listed in **Section 2.2**.

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

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.711A            n         2        20
 3: G.722-64K                   2        20
 4: iLBC                        1       20-30
 5:
 6:
 7:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- **Group Type**: Set to ***sip***
- **Transport Method**: Set to ***tcp***
- **Near-end Node Name**: Set to ***procr*** node configured in this section
- **Far-end Node Name**: Set to ***ASM70*** node configured in this section
- **Far-end Network Region**: Set to network region configured in this section
- **Far-end Domain**: Set to ***avaya.com*** to match the Session Manager domain configured in **Section 5.1**
- Verify **Direct IP-IP Audio Connections** field is set to ***y*** for shuffling
- Verify **DTMF over IP** field is set to the default value of ***rtp-payload*** indicating DTMF transmission using RFC 2833

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

Group Number: 2             Group Type: sip
  IMS Enabled? n       Transport Method: tcp
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                  Far-end Node Name: ASM70
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 2
                                   Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to ***tie***, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

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

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: SIP Endpoints/CM Messaging  COR: 1        TN: 1        TAC: 102
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 2
                                                    Number of Members: 15
```

On **Page 3** of the **Trunk Group** form, set the **Numbering Format** field to ***private***. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 2                                            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

                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
 DSN Term? n
```

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with ***5*** and whose calls are routed over any trunk group, including SIP trunk group ***2***, have the number sent to the far-end for display purposes.

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  33               10                          5     Total Administered: 4
 5  58               10                          5        Maximum Entries: 540
 5  5                2                           5
 5  600              10                          5
```

6.3. Configure Signaling Group For Avaya Aura® Media Server

Another signaling group was created between Communication Manager and Media Server to provide media resources for IP telephony in parallel with Media Gateway G650 resource. Following signaling group was created for this reference configuration:

- **Group Type**: Set to to ***sip***
- **Transport Method**: Set to ***tcp***
- **Peer Detection Enable**: Set to ***n***
- **Peer Server**: Set to ***AMS***
- **Near-end Node Name**: Set to ***procr*** node configured in **Section 6.2**
- **Far-end Node Name**: Set to ***AMS70*** node configured in **Section 6.2**
- **Far-end Network Region**: Set to network region configured in **Section 6.2**

```
add signaling-group 3                                         Page   1 of   1
                                SIGNALING GROUP

Group Number: 3              Group Type: sip
                       Transport Method: tcp

 Peer Detection Enabled? n  Peer Server: AMS



   Near-end Node Name: procr                  Far-end Node Name: AMS70
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                        Far-end Network Region: 2

Far-end Domain: 10.64.102.158
```

6.4. Verify SIP Stations

Use the **display station** command to view each SoundStation IP SIP endpoint configured in **Section 5.6**.

```
display station 50071                                             Page   1 of   6
                                     STATION

Extension: 50071                       Lock Messages? n              BCC: 0
     Type: 9621SIP                     Security Code:                 TN: 1
     Port: S00003                    Coverage Path 1: 1              COR: 1
     Name: 50071 SIP                 Coverage Path 2:                COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 19
                                               Message Lamp Ext: 40012

        Display Language: english

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n
```

Use the **display off-pbx-telephone station-mapping** to verify proper entry of SoundStation IP SIP station in Communication Manager.

```
display off-pbx-telephone station-mapping 50071                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 50071            OPS           -       50071          aar        1
```

On **Page 2**, verify that the **Call Limit** matches the number of *call-appr* entries in the station form.

```
display off-pbx-telephone station-mapping 50071                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 50071          OPS    3           both       all         none
```

7. Configure Polycom® SoundStation® IP SIP Interface

This section describes how to set up the SoundStation IP network interface. Since the configuration for all the SoundStation IP (5000, 6000, 7000 and Duo) conference phone is same, only configuration steps for one model will be listed here. For rest of the phones, these steps can be replicated. Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

7.1. Set the IP address used by SoundStation IP

This section shows how to set the network IP address Soundstation IP.

On the SoundStation IP, push the **Menu** button and navigate to **3. Setting→2. Advance→1. Admin Setting…→Ethernet Menu** and configure as follows:

- **DHCP**: ***Disabled***
- **IP Address**: ***010.080.130.071***
- **Subnet Mask**: ***255.255.255.000***
- **IP Gateway**: ***010.033.005.001***

7.2. Launch Web interface for Polycom SoundStation IP

Open the web browser, and in the address field enter the Duo IP address as format ***http://10.80.130.71*** and the login page will appear as shown below. Select ***Admin***, enter the default password and click **Submit**.

The following home page is displayed.

7.3. Configure the Lines for Polycom® SoundStation® IP

Navigate to **Settings→Lines** and configure **Identification** section as follows:

- **Display Name**: Set to any valid string
- **Address**: Set to the **Login Name** in **Section 5.6**
- **Authentication User ID**: Set to Extension of **Login Name** in **Section 5.6**
- **Authentication Password**: Set to **Communication ProfilePassword** field value configured in **Section 5.6**

Click **Save** (not shown).

7.4. SIP Settings

Navigate to **Settings→SIP** and configure as follows:

Under ***Local Settings*** section,

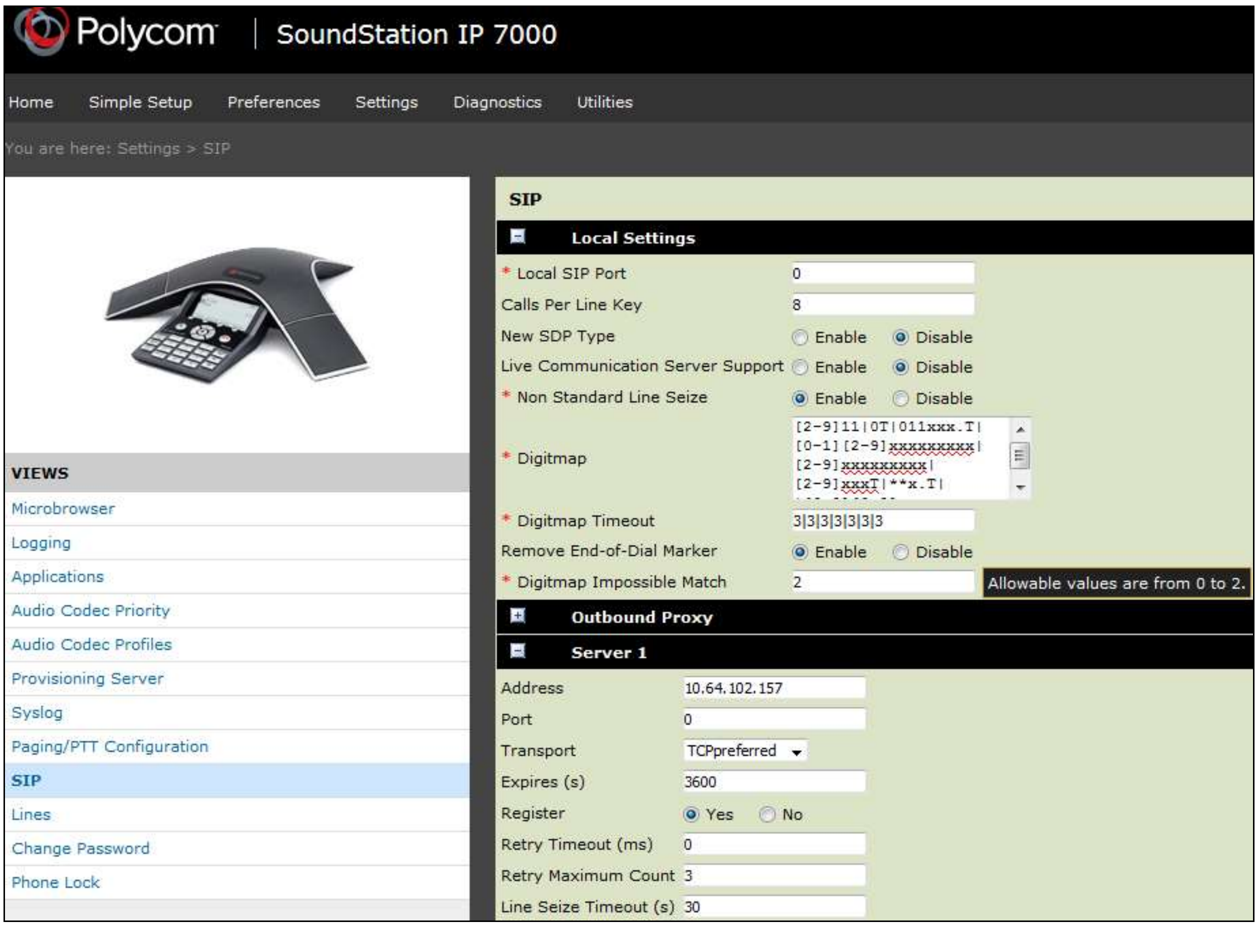
- Set **Digitmap Impossible Match** field to ***2*** to disable the automatic dial if the digits match in **Digitmap** field

Under ***Server1*** section

- **Address**: Set to the IP address of Session Manager signaling interface
- **Port**: Set to ***5060*** for TCP
- **Transport**: Set to ***TCPpreferred***

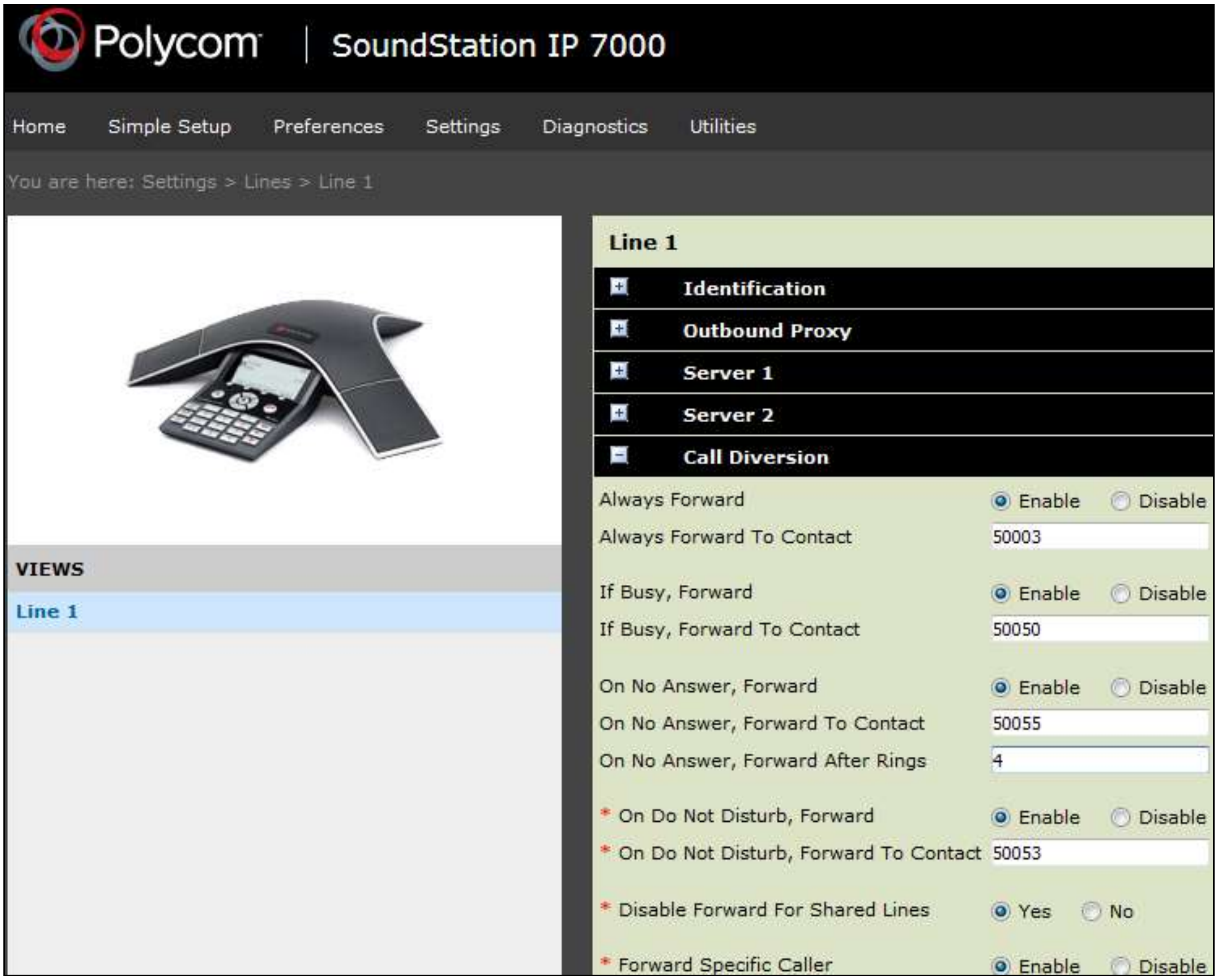
Click **Save** (not shown).

Note: The default local Digitmap configuration may require customization. Refer to **Section 10** [**9**] for further details.



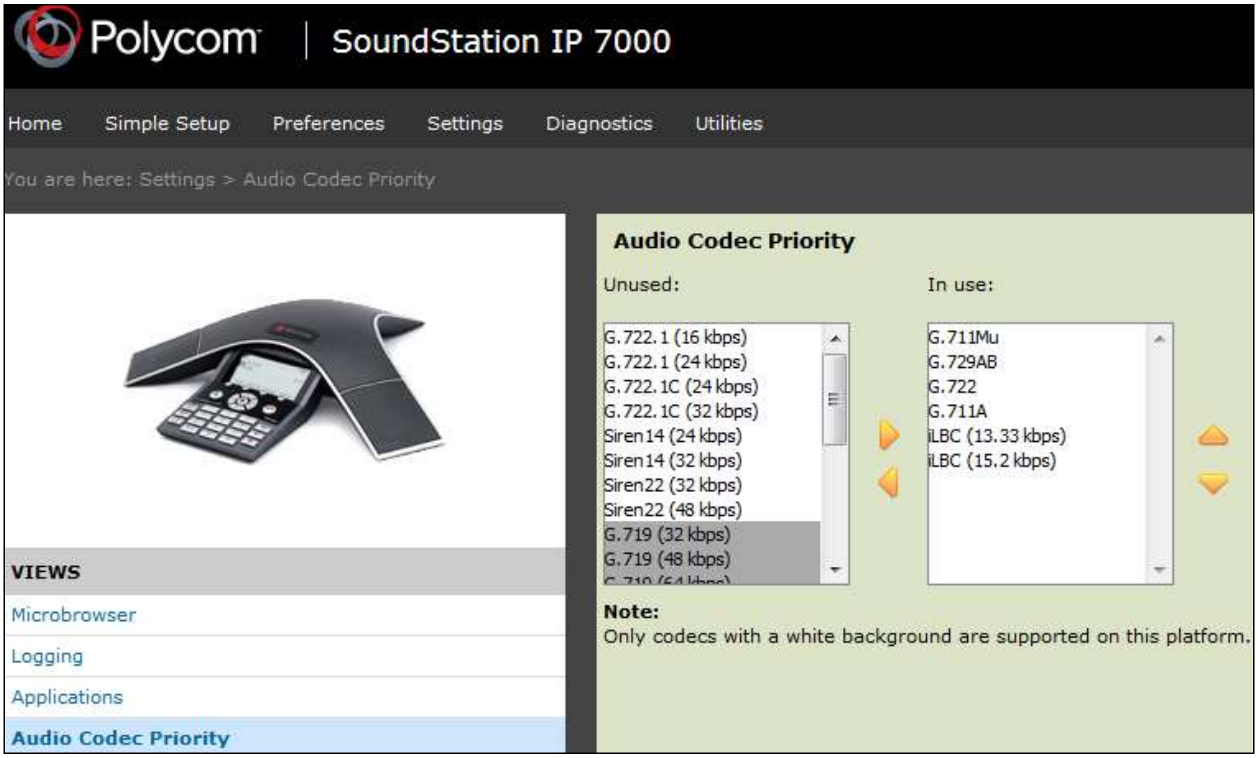
7.5. Local Call Forward Settings

Navigate to **Settings→Lines** and configure **Call Diversion** section as shown screen below. These features can also be enabled directly from the phone too.



7.6. Audio Codec Settings

Navigate to **Settings→Audio Codec Priority** and configure as shown below. The codecs shown in the **In use** column were tested in this reference configuration. The priority can be changed by moving the codecs up or down the order.



7.7. Voice Mail Setting

Navigate to **Settings→Lines** and configure **Message Center** section as follows:

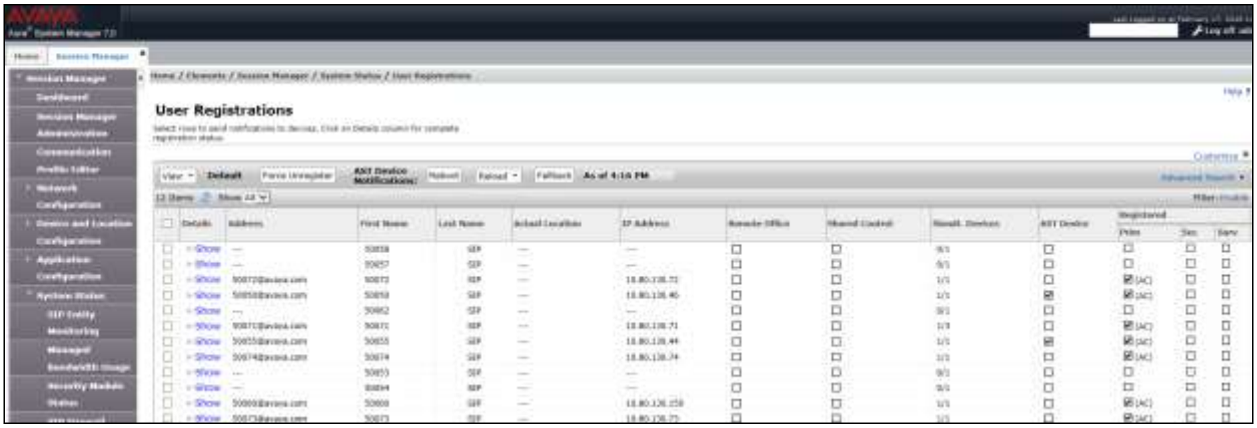
- **Subscription Address**: Set to the Authentication ID field value **Section 7.3**
- **Callback Mode**: Set to the ***Contact***
- **Callback Contact**: Set to voicemail messaging Pilot number

Click **Save** (not shown).

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and Communication Manager with SoundStation IP.

- Verify that SoundStation IP is registered with Session Manager. The following screen shows the registered SIP users with Session Manager:



- Verify that basic calls can be made from and to SoundStation IP and another telephone registered with Communication Manager.

9. Conclusion

These Application Notes describe the configuration steps required for Polycom SoundStation IP conference station to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with the exceptions noted in **Section 2.2**.

10. Additional References

This section references the product documentation available at support.avaya.com relevant to these Application Notes.

[1] Deploying Avaya Aura® System Manager, Release 7.0, November 2015

[2] Administering Avaya Aura® System Manager, Release 7.0, January 2016

[3] Deploying Avaya Aura® Session Manager on VMWare, Release 7.0, August 2015

[4] Administering Avaya Aura® Session Manager, Release 7.0, August 2015

[5] Deploying Avaya Aura® Communication Manager in Virtualized Environment, Release 7.0, August 2015

[6] Deploying and Updating Avaya Aura® Media Server Appliance, Release 7.7, October 2015

[7] Implementing Avaya Aura® Media Server, Release 7.7, January 2016

[8] Deploying Avaya Aura® Communication Manager Messaging, Release 7.0, September 2015

[9] Polycom SoundStation IP (5000, 6000, 7000, and Duo) Conference Phone technical product documentation is available at http://support.polycom.com/PolycomService/support/us/support/voice/soundstation_ip_series/index.html.