Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Polycom® SoundStation IP running UC Software release 4.0.2 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager Release 6.2 - Issue 1.0

Abstract

These Application Notes describe a solution for supporting interoperability between Polycom SoundStation IP conference telephones running UC software release 4.0.2 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager release 6.2. Emphasis of the testing was to verify voice calls of SoundStation IP as SIP endpoints registered to Avaya Aura® Session Manager.

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 provide detail configurations of the Polycom SoundStation IP (including 5000, 6000, 7000 and Duo models) with a SIP infrastructure consisting of Avaya Aura® Session Manager (SM) and Avaya Aura® Communication Manager (CM). During compliance testing, SoundStation IP SIP Conference Phones successfully registered with Session Manager, established calls with other Avaya telephones, and executed telephony features such as Hold, Transfer, and Conference.

2. General Test Approach and Test Results

The general test approach was to have the SoundStation IP to register to Session Manager. Calls were then placed from Avaya telephone clients/users to and from the SoundStation IP. Other telephony features such as busy, hold, DTMF, transfer, conference and codec negotiation were also verified.

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

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Registration of the SoundStation IP to SM.
- Call establishment of SoundStation IP with Avaya telephones.
- Telephony features: Basic calls, conference, blind and consultative transfer, DTMF (dual tone multi frequency), leaving and retrieving voicemail message, busy, hold, call forward busy, call forward unconditional, call forward no answer, MWI (Message Waiting Indicator), Reject and Do not Disturb (DND).
- Codec negotiation – G.711, G.729 and G722.
- SoundStation IP calls PSTN telephone via SIP trunk.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. The SoundStation IP was registered to SM successfully. Calls have been made between Avaya telephones and SoundStation IP with clear voice path.

2.3. Support

Technical support for the Polycom SoundStation IP conference phone can be obtained through Polycom global technical support:

- Phone: 1-888-248-4143 or 1-408-474-2067
- Web: http://support.polycom.com

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway.
- Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- System Manager used to configure Session Manager.
- Avaya Modular Messaging providing voice mail service for the SIP endpoints.

In this test configuration, the 2 Polycom SoundStation Duo (hereafter referred to as Duo) were used as representative of the Polycom SoundStation IP phones and they were registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

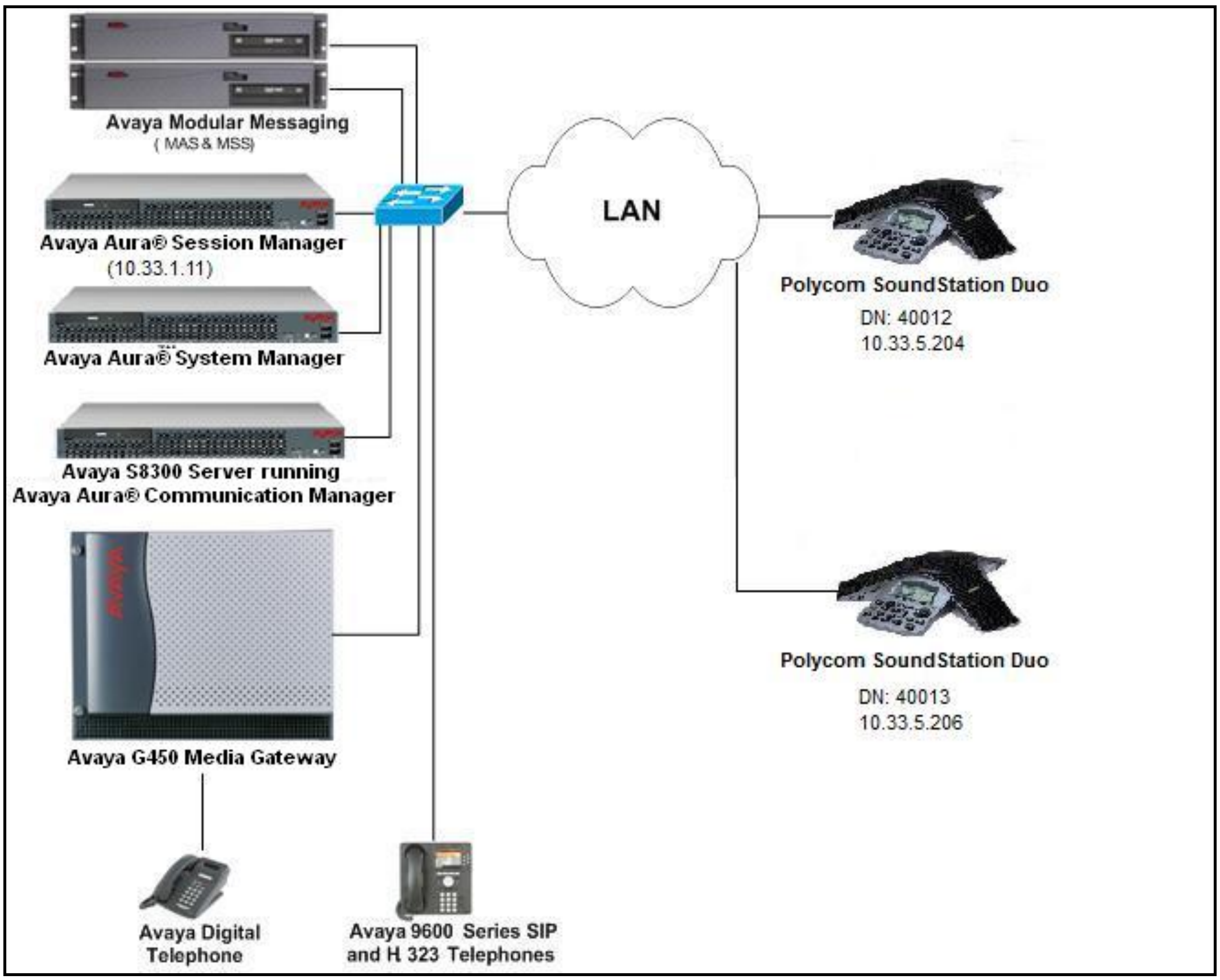


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

The following equipment and software/firmware were used for the reference configuration:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8300 Servers and G450 Media Gateway	6.2 (Avaya CM/R016x.02.0.823.0) (System Platform 6.2.1.0.9) 31.22.0
Avaya Aura® System Manager running on an Avaya S8800 Server	6.2.12.0 (Patch 6.2.12.202 Build Number 6.2.14.1.1925)
Avaya Aura® Session Manager running on S8800 Server.	6.2 (6.2.2.0.622005)
Avaya Modular Messaging	5.2
Avaya 9641G SIP Telephone	6.2.0.69 (SIP)
Avaya 9611G H.323 Telephone	S6.2209
Avaya Digital Telephones	N/A
Polycom SoundStation Duo (SIP)	UC software 4.0.2.8017

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the Duo as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. **Section** Error! Reference source not found. covers the station configuration for the Duo. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

5.1. 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  15
  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.)
```

5.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8300 Server processor, the C-LAN board in the G450 Media Gateway, and Session Manager. 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
interopsm           10.33.1.11
procr               10.33.1.22
procr6              ::

( 4 of 4   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

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 *bvwdev.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 G450 Media Gateway. The **IP Network Region** form also specifies the **IP 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 '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: bvwdev.com
    Name: Main Network Region
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: 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 Duo. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. However, the **IP Codec Set** form may specify multiple codecs, including G.711, G.729A, and G.722, which are supported by the Duo conference SIP phones.

```
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:
 3:
 4:
 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:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the recommended TLS port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *bvwdev.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
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
     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: interopsm
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1
                                   Far-end Secondary Node Name:
Far-end Domain: bvwdev.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 10                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: SIP Trunk to Interop SM     COR: 1        TN: 1        TAC: #10
   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: 10
                                                    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 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

                                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 '400' and whose calls are routed over any trunk group, including SIP trunk group "10", 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  400              10                          5
 5  600              10                          5
```

5.3. Configure Stations

Use the **add station** command to add a station for each Duo phone to be supported. Use *9620SIP* for the **Station Type** and include the **Coverage Path** for voice mail, if applicable. The **Name** field is optional. Use the default values for the other fields on Page 1. The SIP station can also be configured automatically by Session Manager as described in **Section 6.7**.

```
add station 40012                                             Page   1 of   6
                                     STATION

Extension: 40012                         Lock Messages? n               BCC: 0
     Type: 9620SIP                       Security Code:                  TN: 1
     Port: S00003                      Coverage Path 1: 1               COR: 1
     Name: SIP, 40012                  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
```

On Page 2, set the **MWI Served User Type** field to the appropriate value to allow MWI notifications to be sent to the Duo.

```
add station 40012                                             Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                      Coverage Msg Retrieval? y
                                                            Auto Answer: none
             CDR Privacy? n                            Data Restriction? n
                                             Idle Appearance Preference? n
 Per Button Ring Control? n                Bridged Idle Line Preference? n
   Bridged Call Alerting? n                    Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n         Per Station CPN - Send Calling Number?
                                                        EC500 State: enabled

    MWI Served User Type: qsig-mwi
              AUDIX Name:
                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 40012        Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

Use the **change off-pbx-telephone station-mapping** command to map Communication Manager extensions (e.g., 40012) to the same extension configured in Session Manager. Enter the field values shown. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not show in these Application Notes. The **Configuration Set** value can reference a set that has the default settings.

```
change off-pbx-telephone station-mapping 40012                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 40012            OPS           -     40012            aar        1
```

On Page 2, change the **Call Limit** to match the number of *call-appr* entries in the station form. Also, verify that **Mapping Mode** is set to *both* (the default value for a newly added station).

```
change off-pbx-telephone station-mapping 78500                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 40012         OPS      3        both         all         none
```

6. 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 corresponding to Session Manager and Communication Manager.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Application Sequence.
- Define Communication Manager as Administrable Entity (i.e., Managed Element).
- Session Manager, to be managed by System Manager.
- 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.

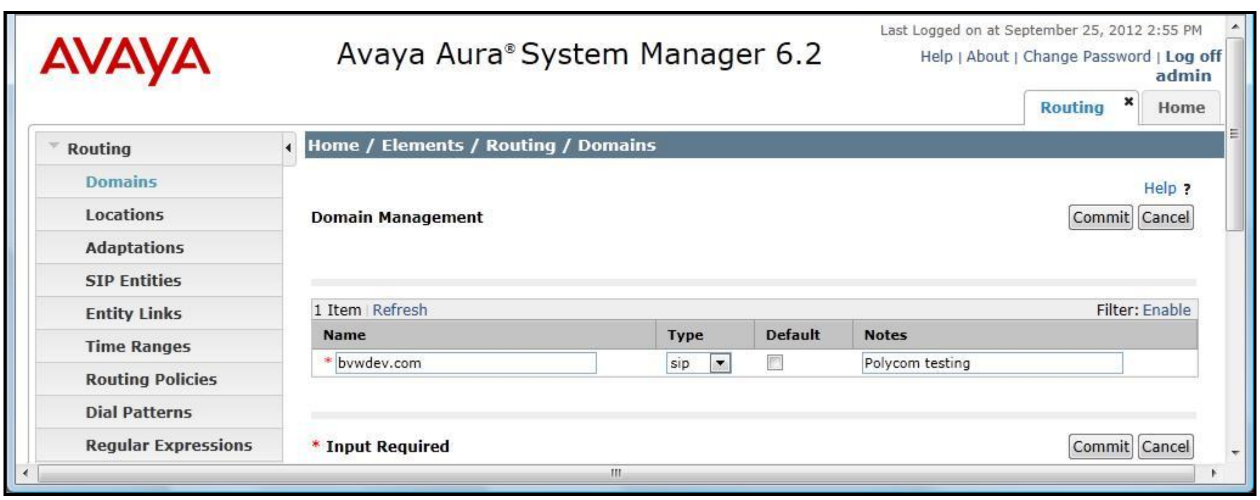
6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. It can be done by selecting **SIP Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *bvwdev.com*).
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.



6.2. 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, select **Locations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

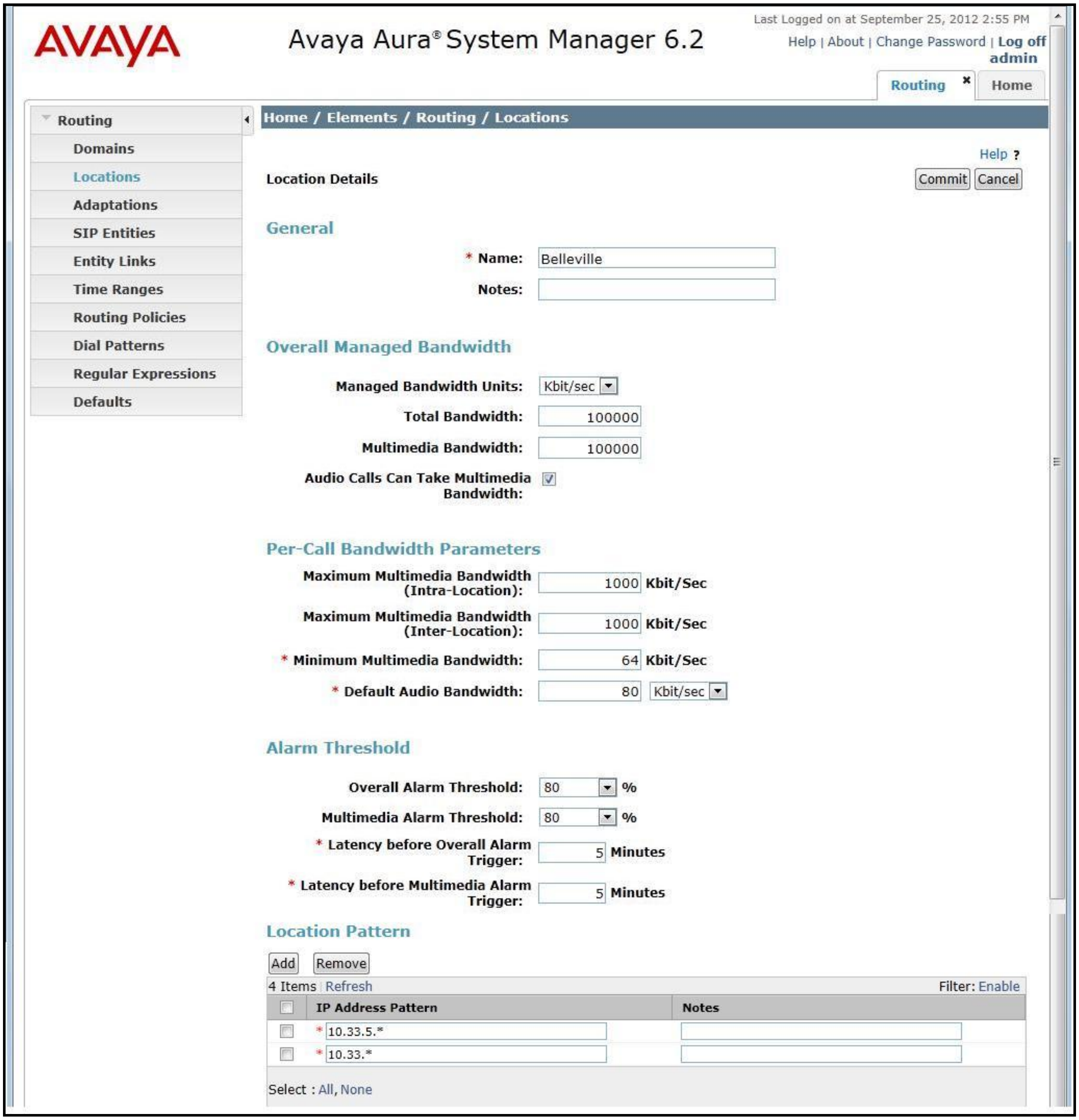
Under *General*:

- **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 *BR-DevConnect* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition.


AVAYA
Avaya Aura® System Manager 6.2
Last Logged on at September 25, 2012 2:55 PM
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / Locations
Help ?
Location Details
Commit
Cancel
General
* Name: Belleville
Notes:
Overall Managed Bandwidth
Managed Bandwidth Units: Kbit/sec
Total Bandwidth: 100000
Multimedia Bandwidth: 100000
Audio Calls Can Take Multimedia Bandwidth:
Per-Call Bandwidth Parameters
Maximum Multimedia Bandwidth (Intra-Location): 1000 Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec
* Minimum Multimedia Bandwidth: 64 Kbit/Sec
* Default Audio Bandwidth: 80 Kbit/sec
Alarm Threshold
Overall Alarm Threshold: 80 %
Multimedia Alarm Threshold: 80 %
* Latency before Overall Alarm Trigger: 5 Minutes
* Latency before Multimedia Alarm Trigger: 5 Minutes
Location Pattern
Add
Remove
4 Items | Refresh
Filter: Enable
IP Address Pattern
Notes
* 10.33.5.*
* 10.33.*
Select : All, None

6.3. Add SIP Entities

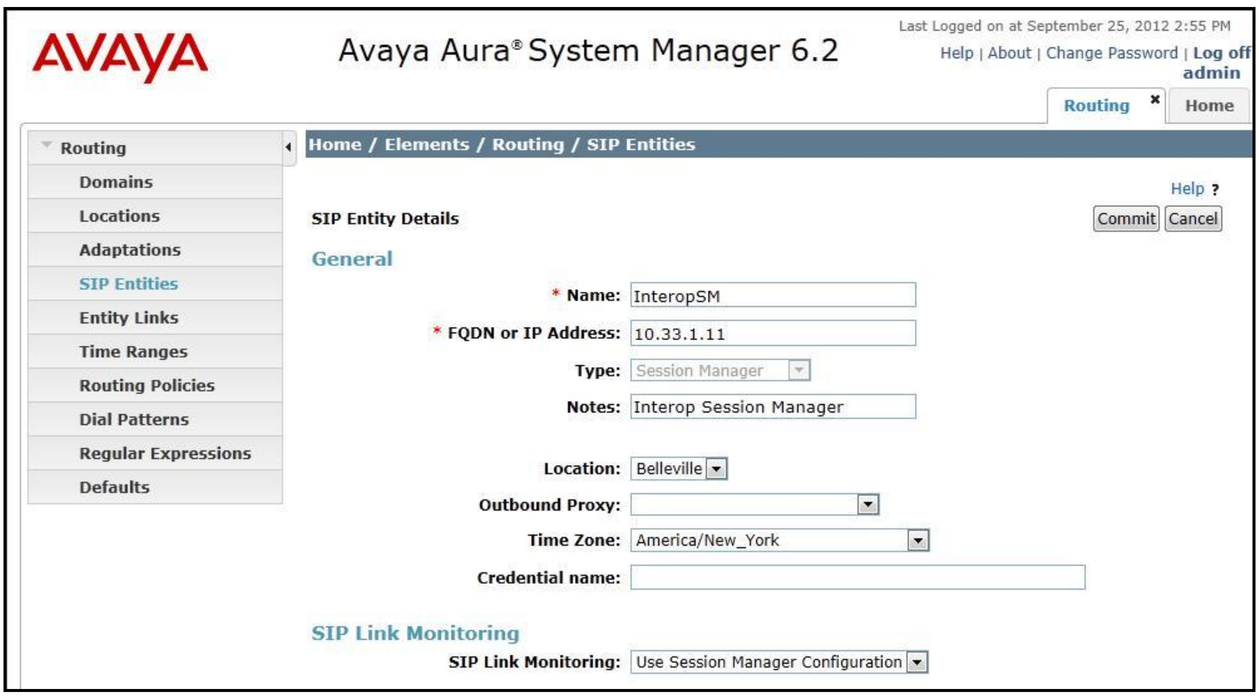
In the sample configuration, a SIP Entity is added for Session Manager and the C-LAN in the G650 Media Gateway.

6.3.1. Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

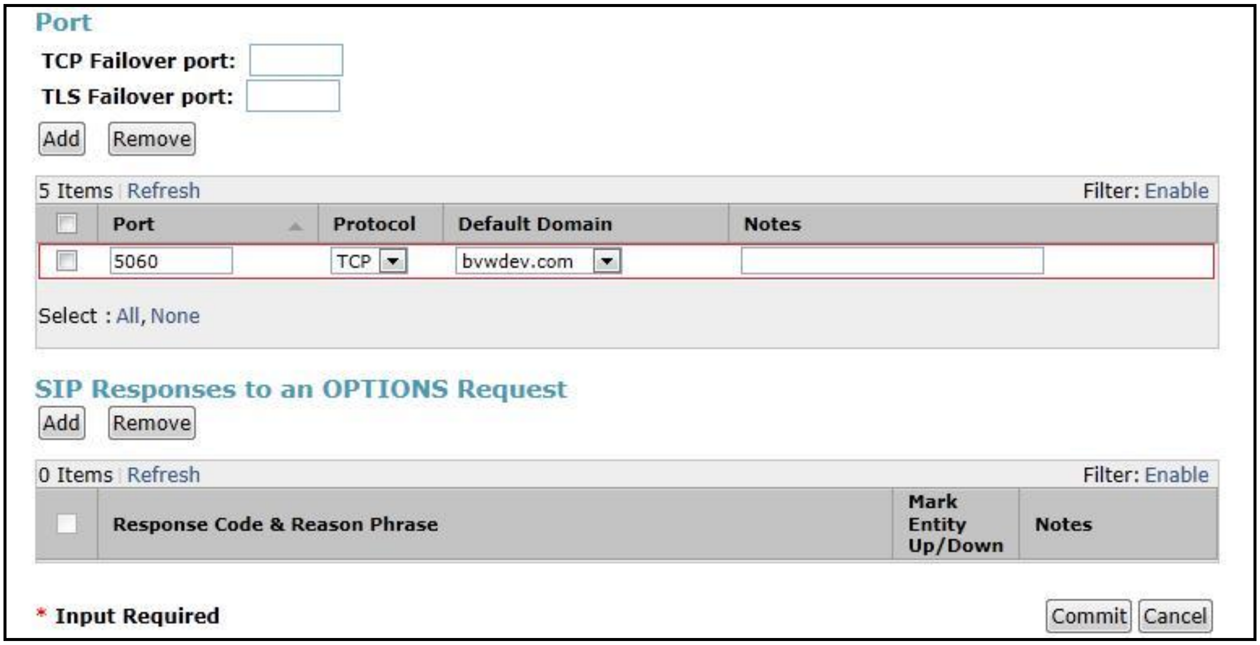
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Specify *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.



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

- **Port:** 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. *bvwdev.com*).

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



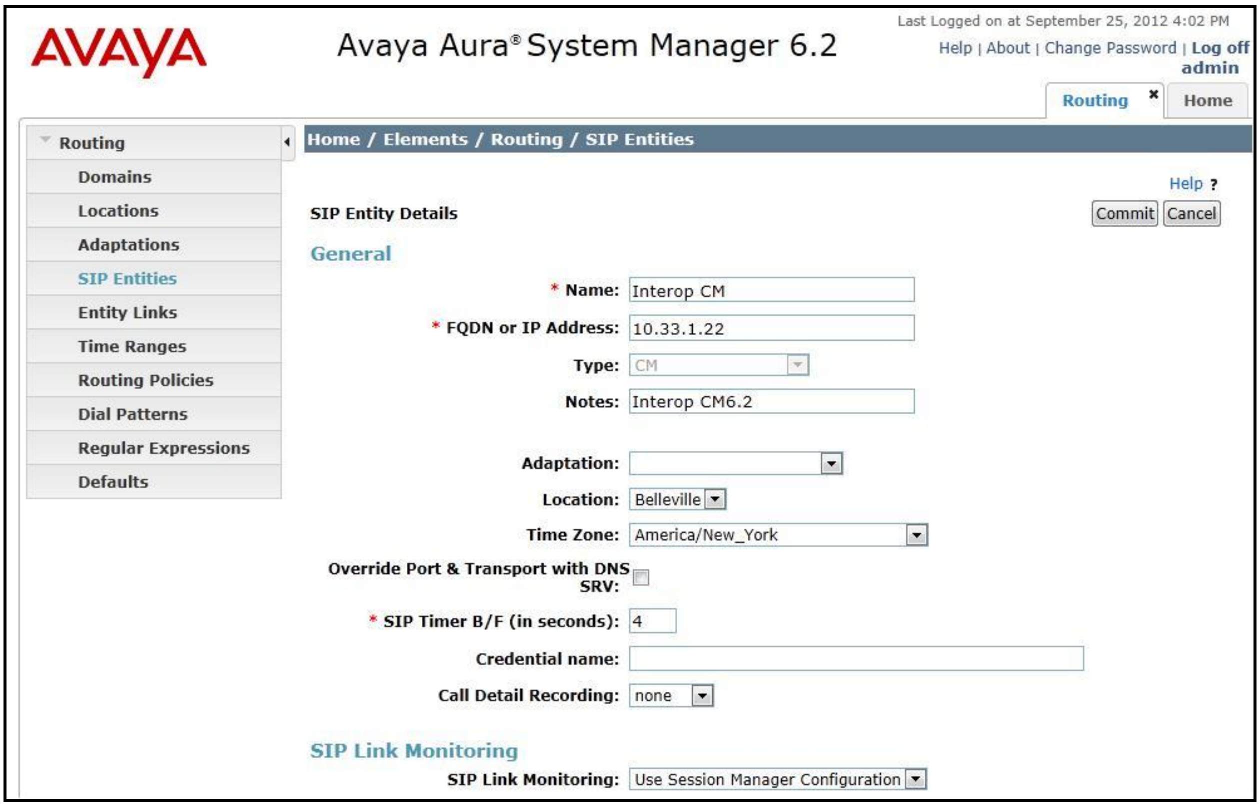
6.3.2. Avaya Aura® Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Specify *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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

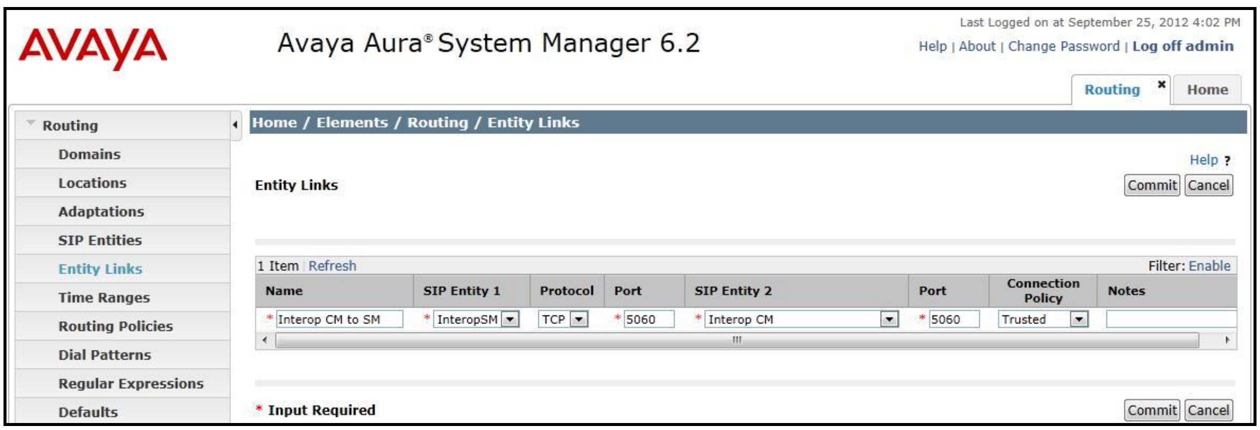


6.4. Add Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g. *Interop CM to SM*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in **Section** Error! Reference source not found. will be denied.*

Click **Commit** to save the Entity Link definition.



6.5. Define Communication Manager as 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.

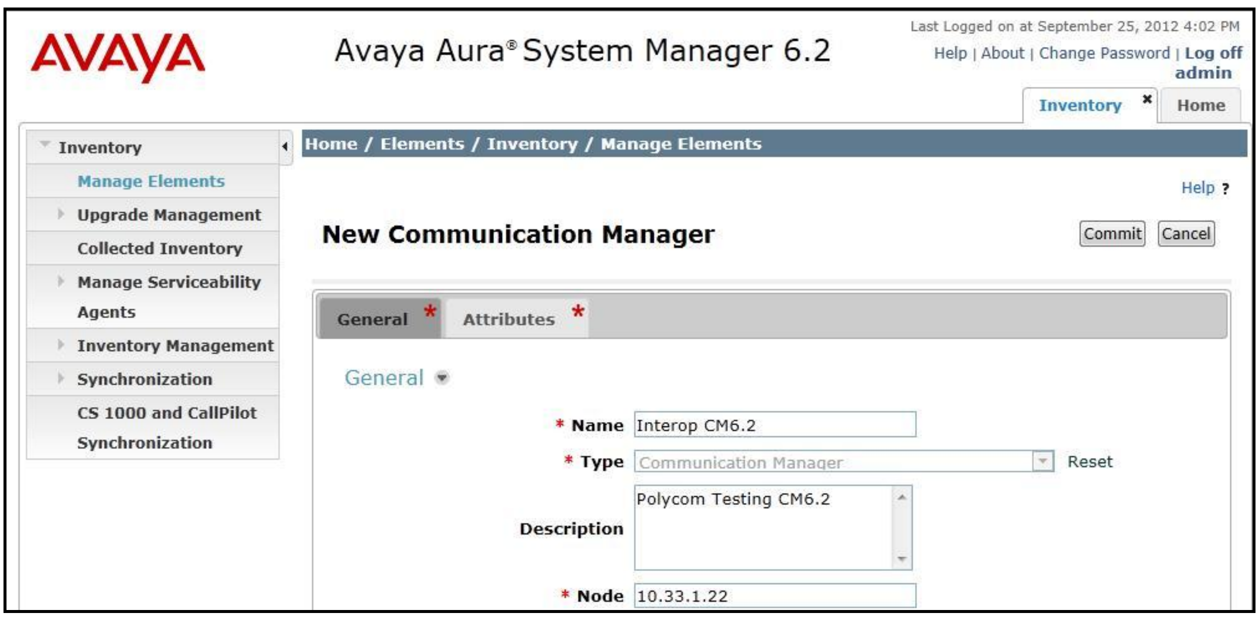
To define Communication Manager as a managed element, select **Elements→Inventory→Manage Elements** on the left and click on the **New** button on the right. In the **Application Type** field that is displayed, select *CM*.

In the **New CM Instance** screen, fill in the following fields as follows:

Under *Application*:

- **Name:** Enter an identifier for Communication Manager.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for Communication Manager.

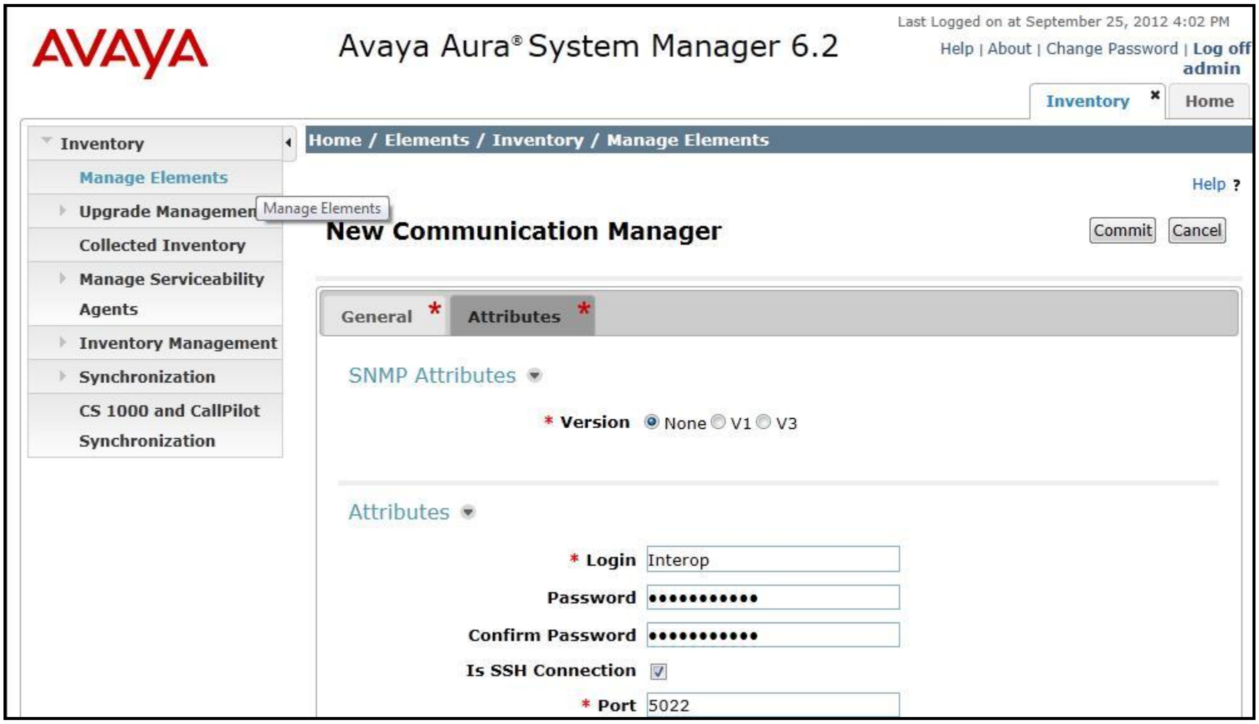
Click **Commit** to save the settings.



Under *Attributes*:

- **Login / Password:** Enter the login and password used for administration access.
- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Click **Commit** to save the settings.

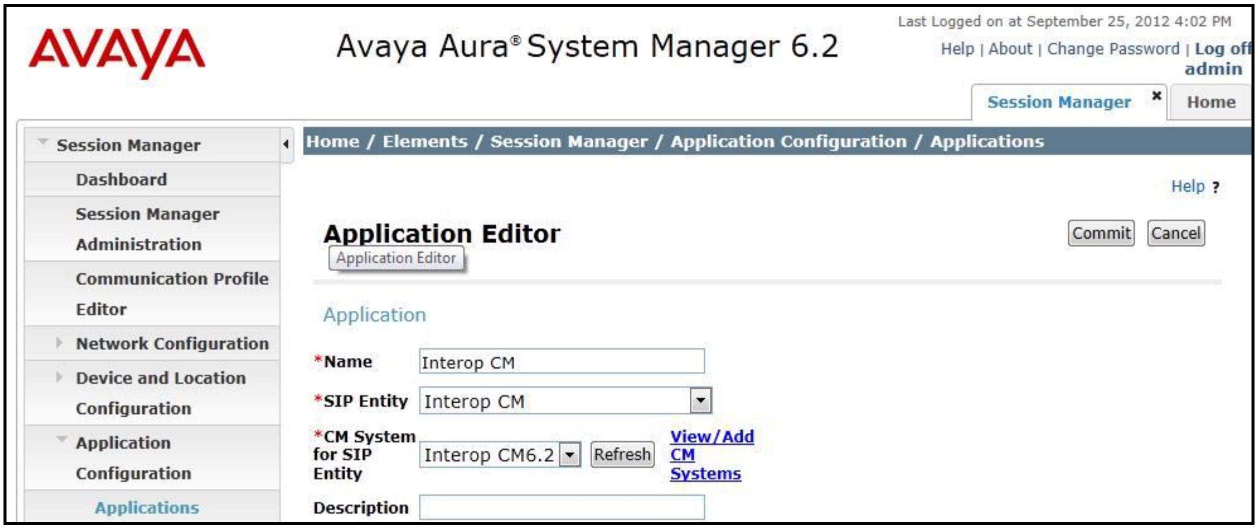


6.6. Add Application Sequence

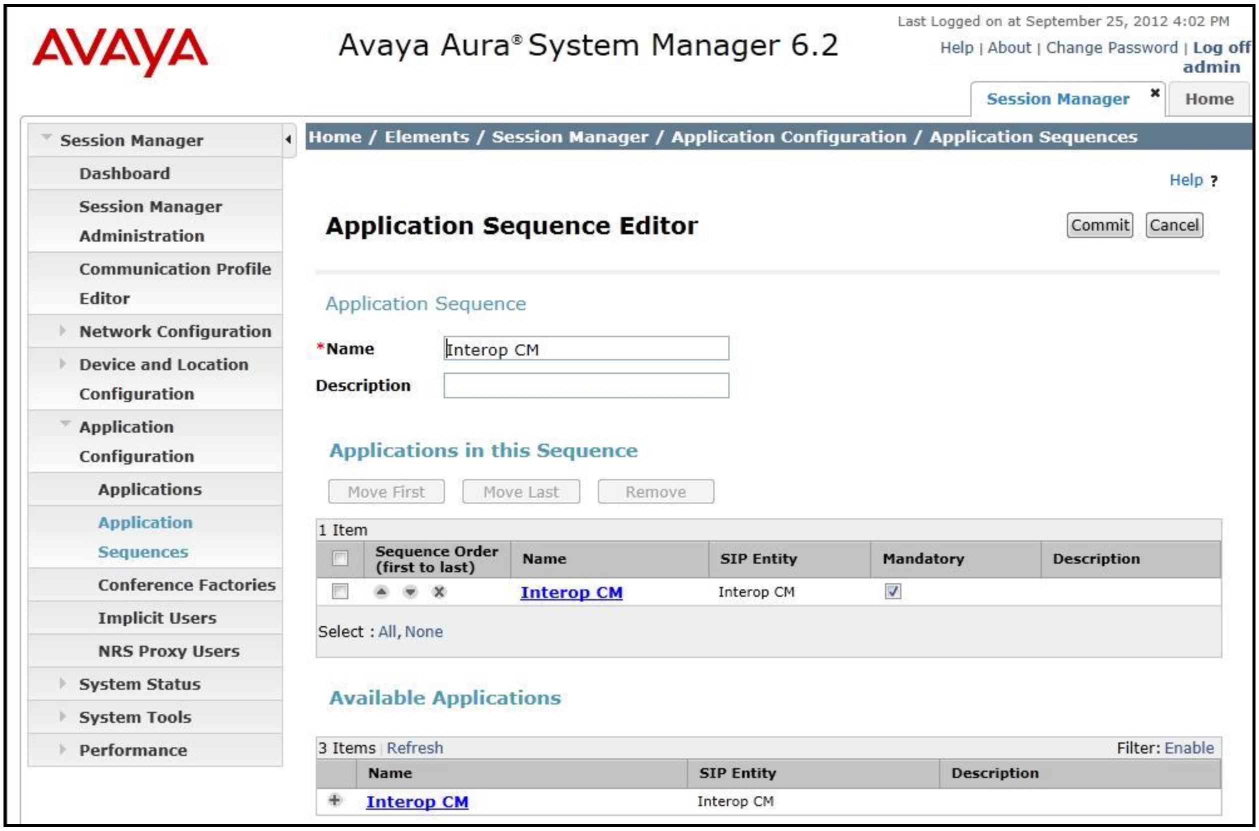
Define an application for Communication Manager. Fill in the following fields:

- **SIP Entity:** Select the Communication Manager SIP entity.
- **CM System for SIP Entity** Select the Communication Manager managed element.

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



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



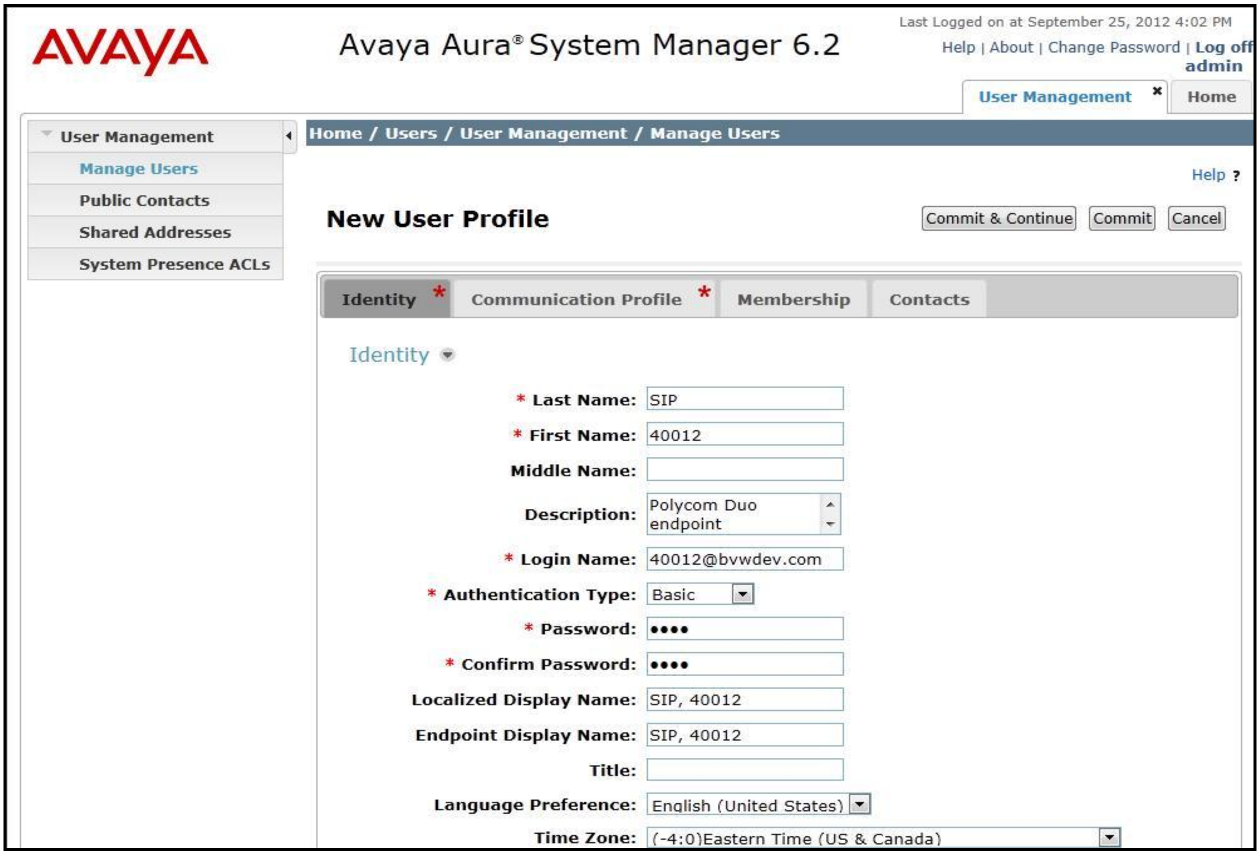
6.7. Add SIP Users

Add SIP users corresponding to the SoundStation Duo defined in **Section** Error! Reference source not found.**.** Alternatively, use the option to automatically generate the SIP stations on Communication Manager Feature Server when adding a new SIP user.

Enter values for the following required attributes for a new SIP user in the new user 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., 40012@bvwdev.com).
- **Authentication Type:** Select *Basic***.**
- **SMGR Login Password:** Enter the password which will be used to log into System Manager.
- **Confirm Password:** Re-enter the password from above.
- **Shared Communication Profile Password:** Enter the password which will by the SIP phone to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.

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



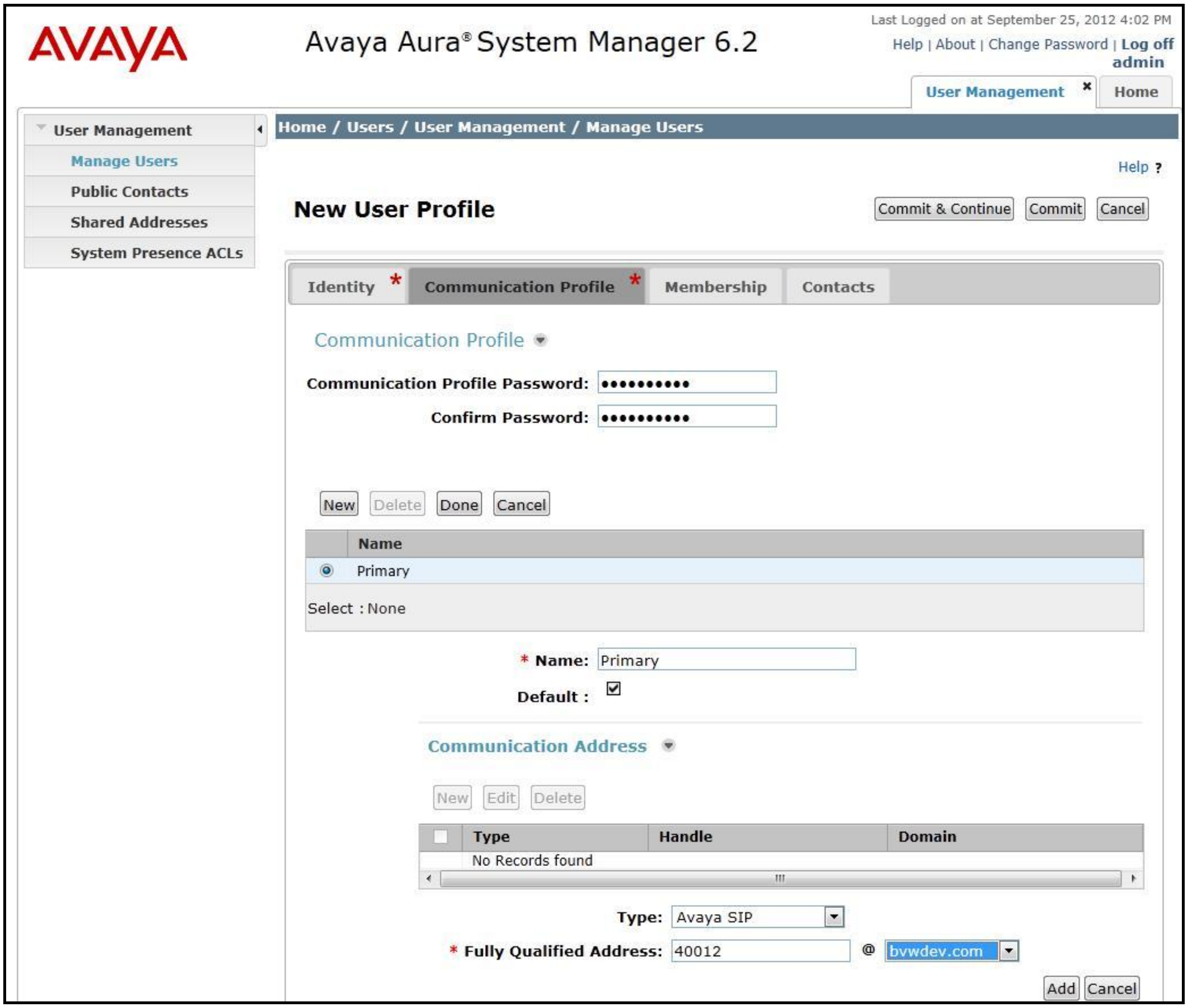
Click the *Communication Profile* tab and select **New** to define a **Communication Profile** for the new SIP user. Enter values for the following required fields:

- **Name:** Enter name of communication profile.
- **Default:** Select field to indicate that this is the default profile.

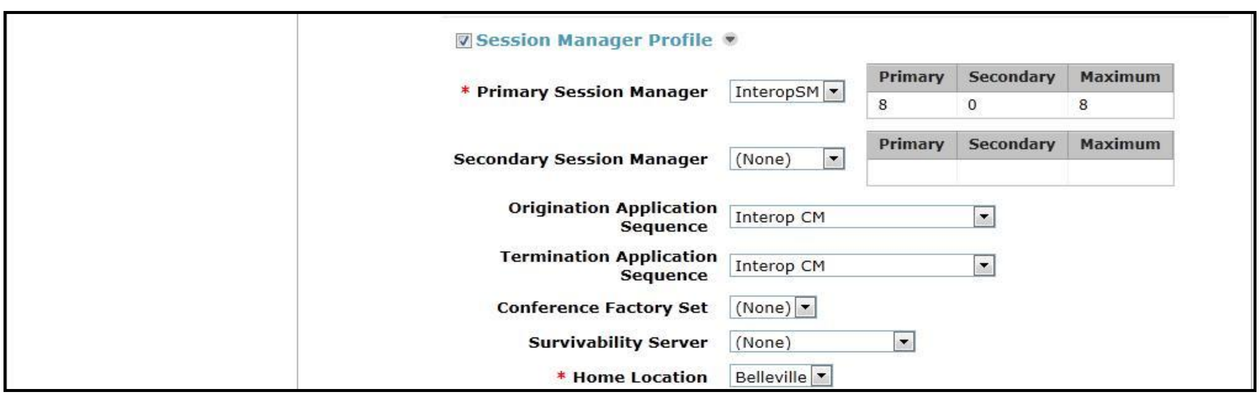
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *sip*.
- **SubType:** Select *username*.
- **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 and assign the **Application Sequence** defined in **Section** Error! Reference source not found. to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence.

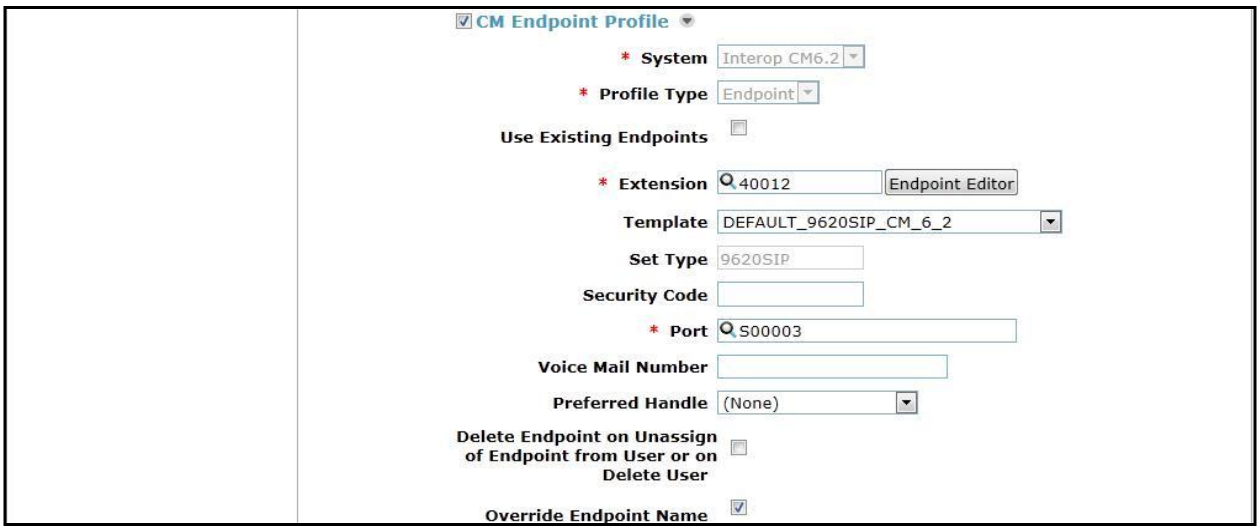


In the *CM Endpoint Profile* section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- 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.
- **Port:** Enter *S00003*.
- **Override Endpoint Name:** Enable the field.

.

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



6.8. Add Session Manager

To complete the configuration, adding Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add**, and fill in the fields as described below and shown in the following screen:

Under *Identity*:

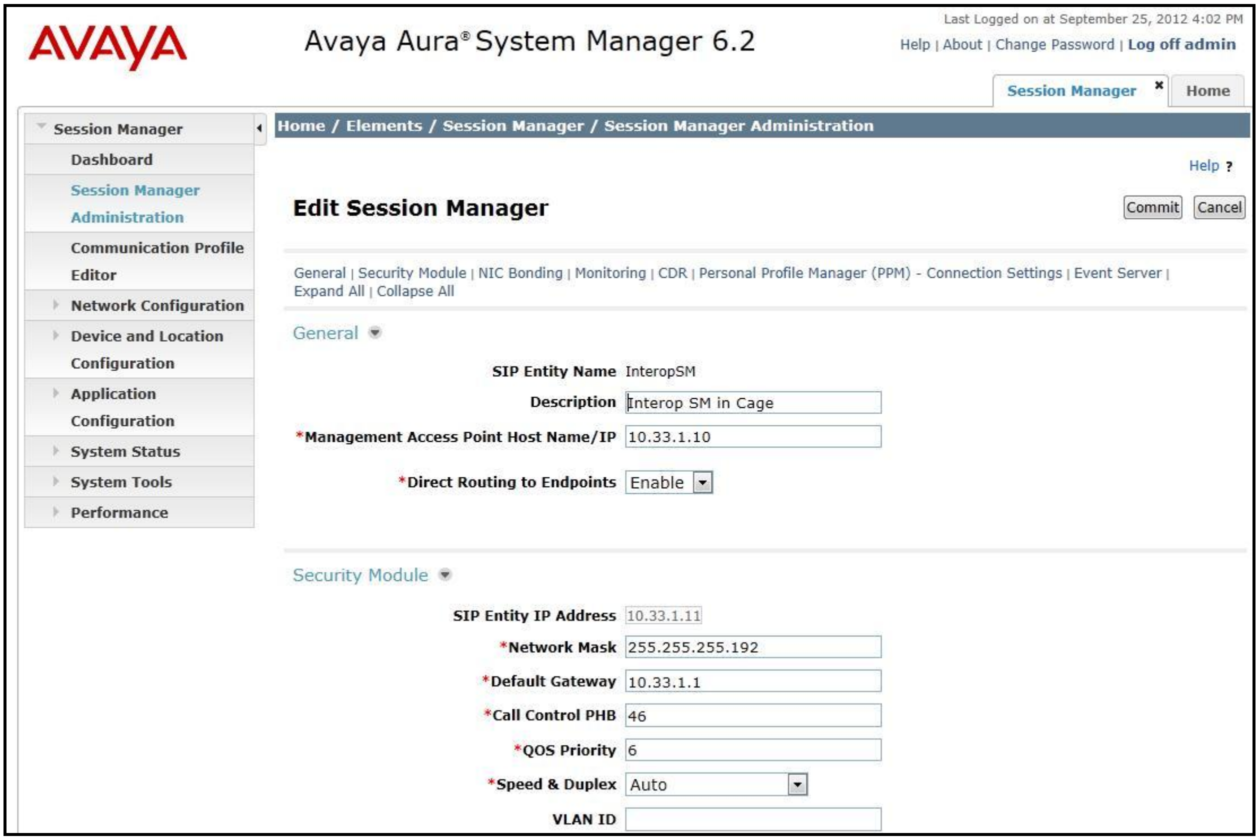
- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager.
- **Description:** Descriptive comment (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP

- **Default Gateway**: address of Session Manager. Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Save** to add this Session Manager.



7. Configure Polycom SoundStation Duo SIP interface

This section describes how to set up the Duo network interface, to access the Duo SIP endpoint web interface and to configure the Duo for testing.

7.1. Determine the IP address used by the Duo

This section shows how to determine the network IP address used by the Duo.

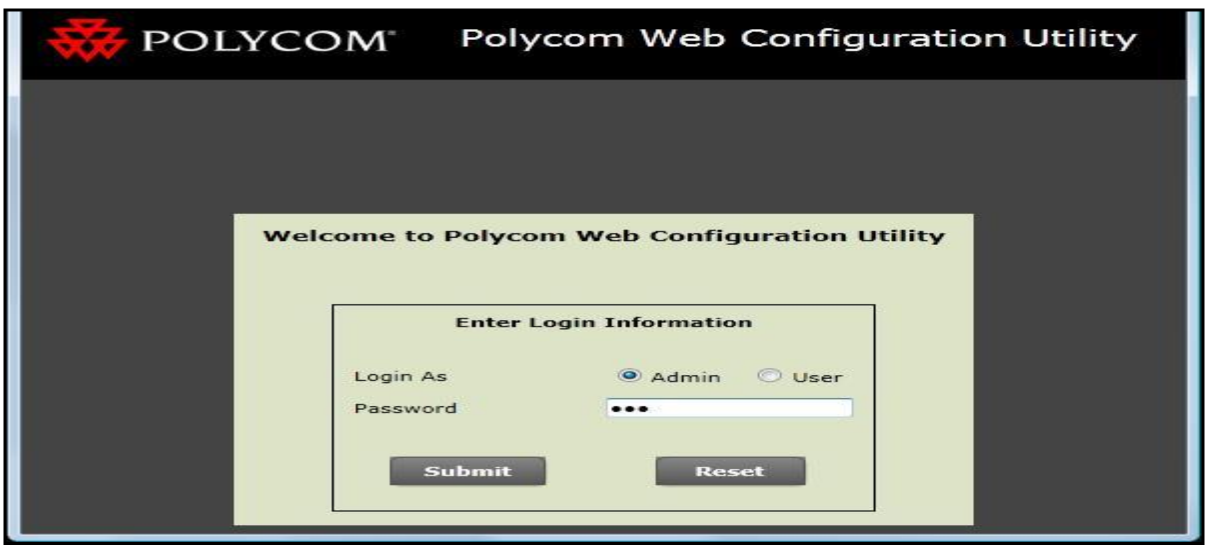
On the Duo (not shown), push the '**Menu**' button and navigate to **2. Status ➔ 2. Network ➔ 1. TCP/IP Parameters.** In this example configuration, the following parameters are used as bellow. Others are left at default.

- **DHCP**: Enabled
- **IP Address**: 010.033.005.204
- **Subnet Mask**: 255.255.255.000
- **IP Gateway**: 010.033.005.001

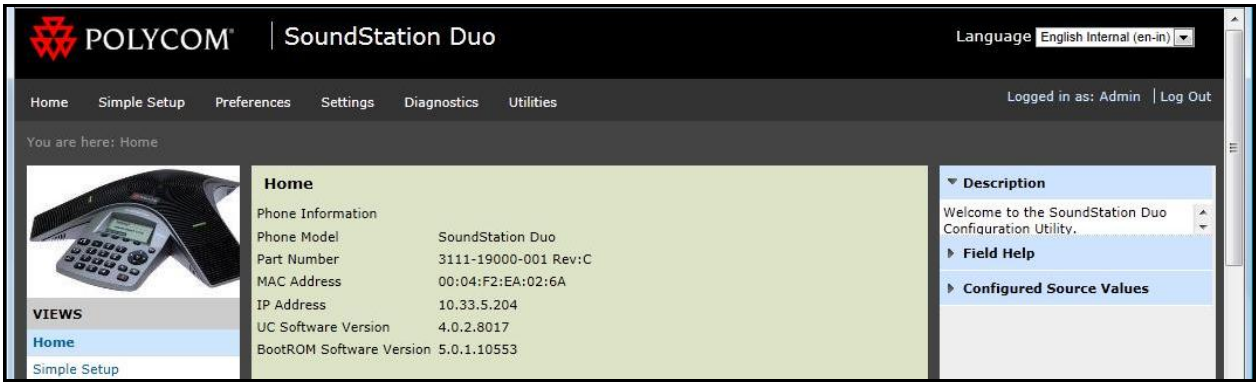
7.2. Polycom SoundStation Duo Web Configuration Utility

This section shows how to log in to the home page of Duo Web Configuration Utility to manage and configure the Duo phone.

Open the web browser, and in the address field enter the Duo IP address as format **http://10.33.5.204** and the Duo login page will appear as shown bellow. Select 'Admin' and enter the default password, **456**.

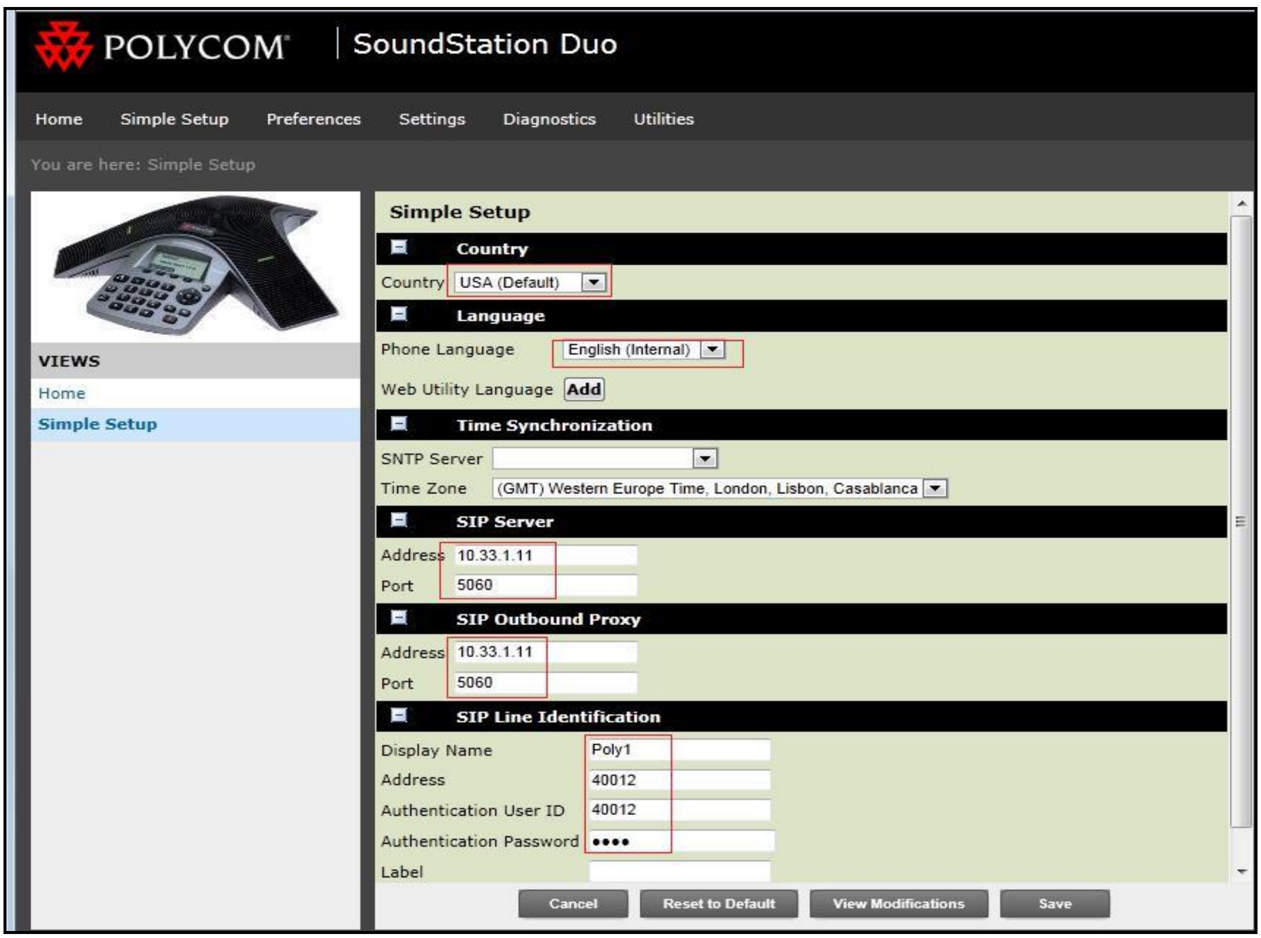


Click **Submit**, the homepage of Duo appears.



7.3. Configure the Lines for Polycom SoundStation Duo

This section shows how to configure the Duo to register with Session Manager. On the homepage of configuration screen, click on the **Simple Setup** menu, the **Simple Setup** page appears as shown below. Enter values as highlighted in areas marked with red boxes in the screenshot below and leave other areas at default. Click **Save**.

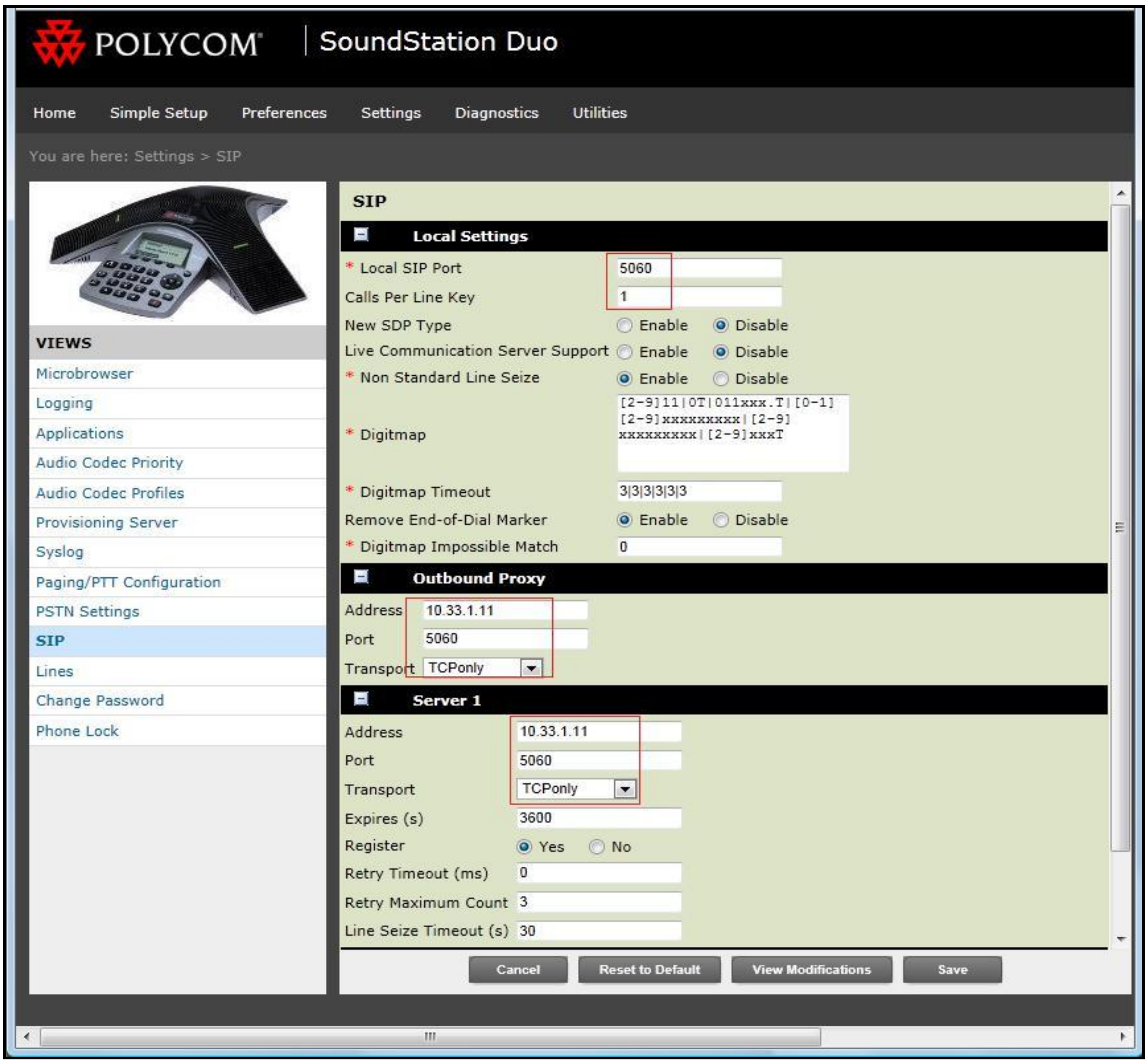


7.4. SIP Settings

This section shows how to set SIP parameters for Duo.

On the homepage of the Duo Web Configuration Utility, navigate to menu **Settings → SIP**, **SIP** page appears. Enter values as highlighted in the areas marked with red-boxes in the screenshot below and leave other areas at default. Click **Save**.

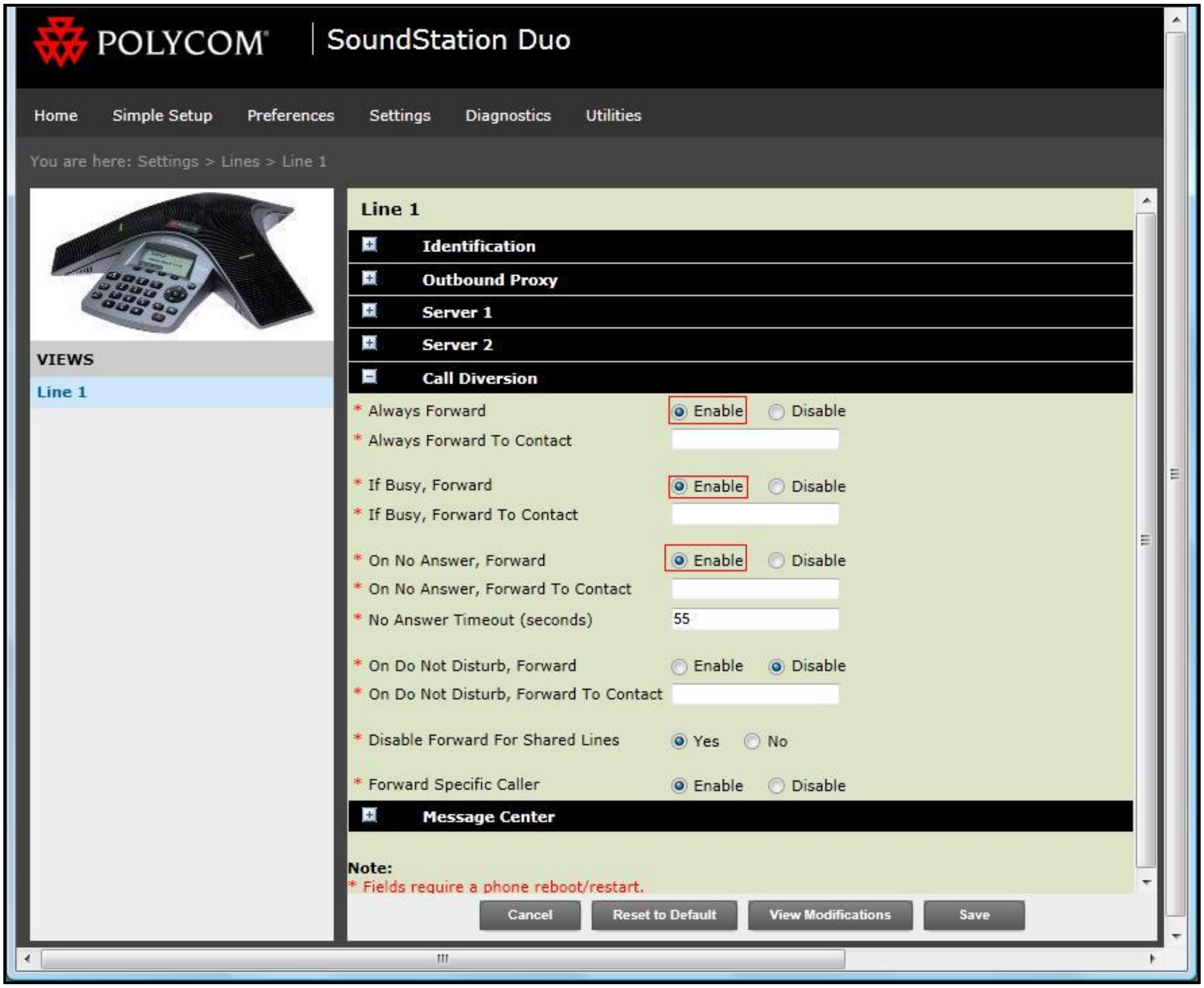
Note: The default local Digitmap configuration used by Duo may require customization. More detailed information about local Digitmap configuration is available in the Administrator's Guide for Polycom UC Software [3] and Polycom Technical Bulletin 11572 [4] – see **Section 10** for additional reference.



7.5. Local Call Forward Settings

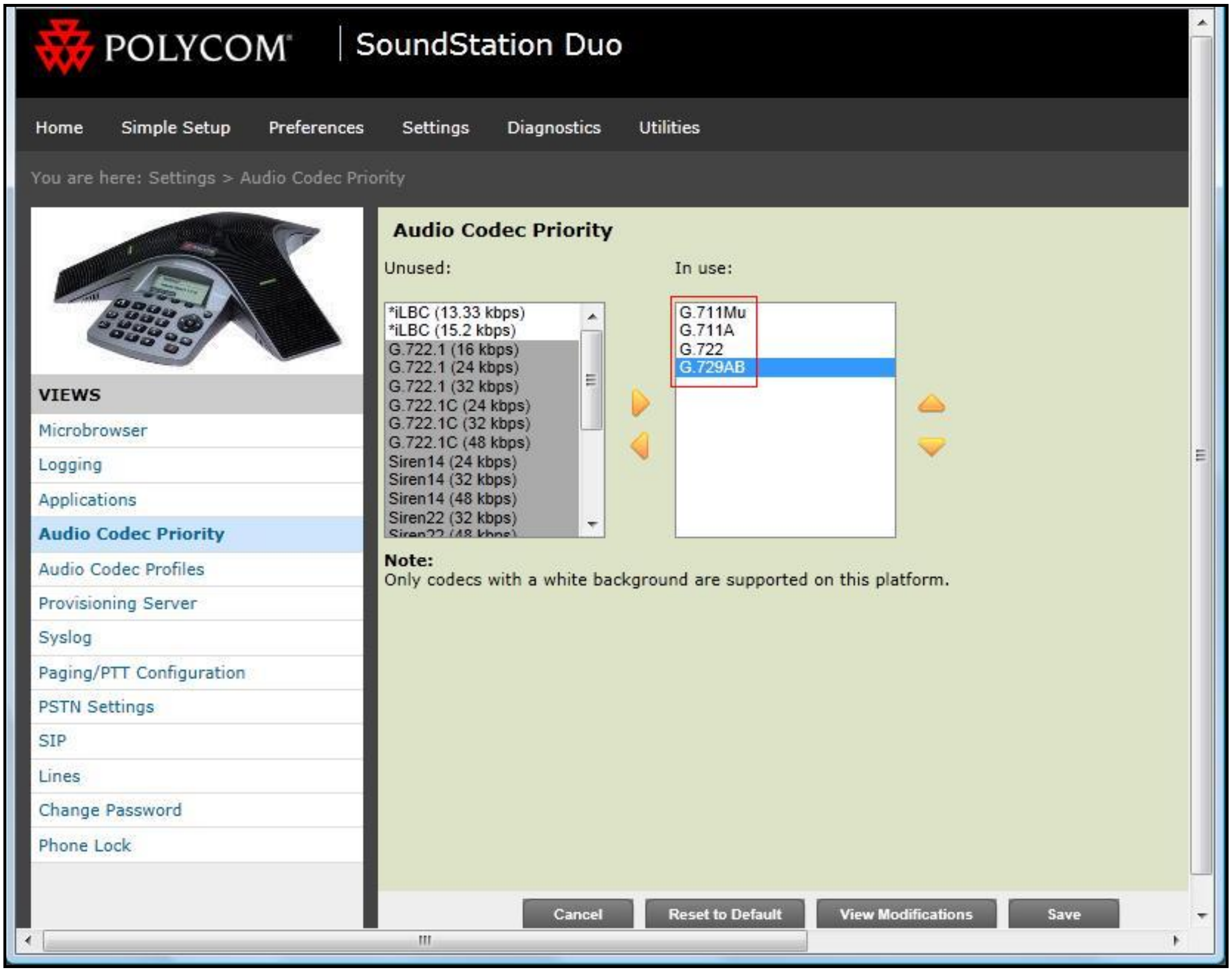
This section shows how to set up call forward settings for Duo.

On the homepage of the Duo Web Configuration Utility, navigate to menu **Settings → Lines, Line 1** page appears. Click on 'Call Diversion' to expand the 'Call Diversion' section. Enable values as highlighted in the areas marked with red boxes and leave other areas at default. Click **Save**.



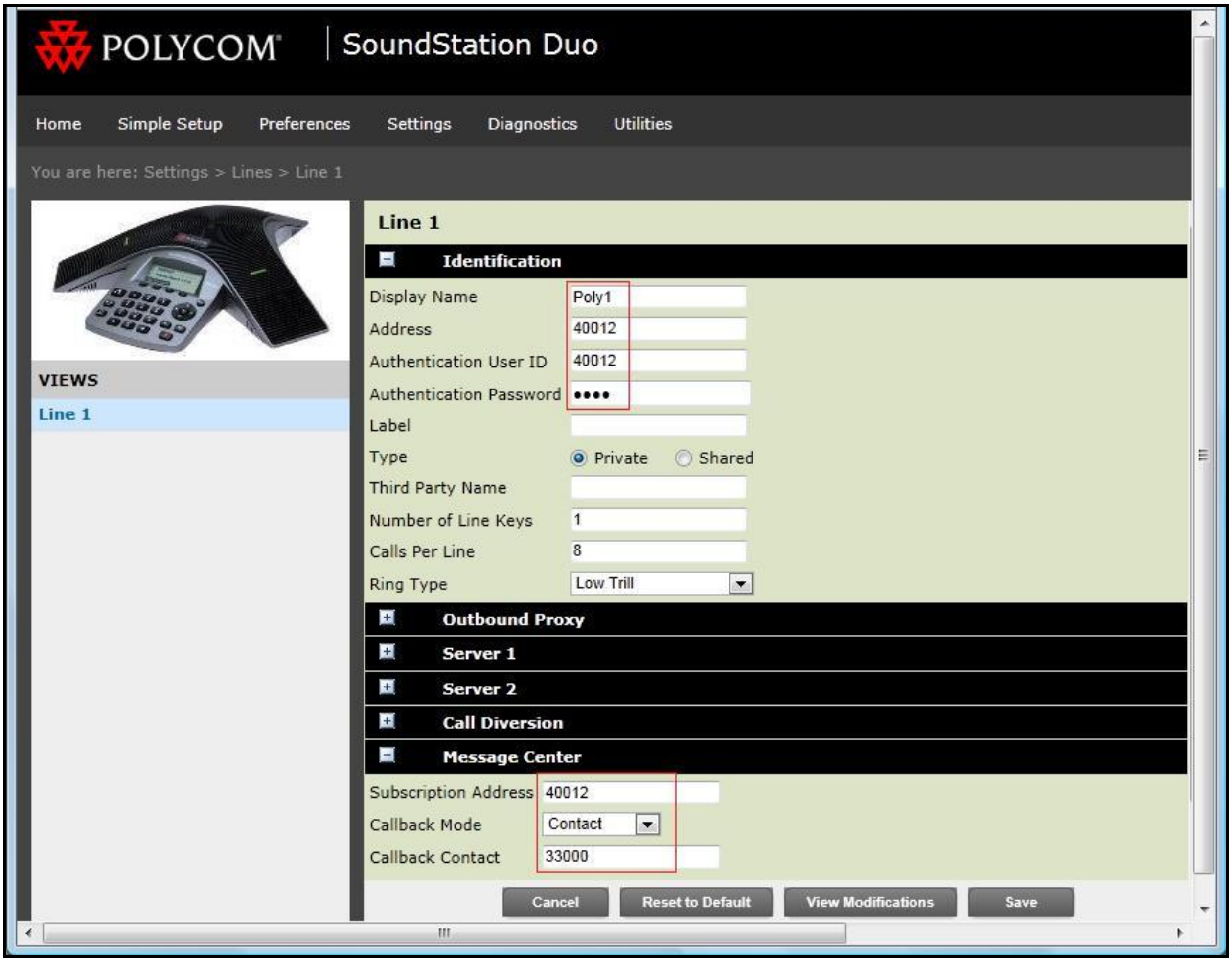
7.6. Audio Codec Settings

On the homepage of Duo Web Configuration Utility, navigate to menu **Settings → Audio Codec Priority**. Select the codec list in the order of priority as shown in the areas marked with a red box in the screenshot below. Click **Save**.



7.7. Voice Mail Setting

On the homepage of the Duo Web Configuration Utility, navigate to menu **Settings → Lines**, the **Line 1** page appears. Click on **'Message Center'** to expand the 'Message Center' section. Enter values as highlighted in the areas marked with red-boxes in the screenshot below and leave other areas at default. Click **Save**.



8. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Verify that the Duo has successfully registered with Session Manager.
2. Verify basic telephony features by establishing calls between a Duo and another phone.
3. Call a Duo that currently has no voice messages, and leave a message. Verify that the message waiting indicator (i.e., envelop icon) appears on the Duo LCD display. Call the voicemail system and retrieve voice messages. Verify that after hearing all messages, that the message waiting indicator is extinguished.

9. Conclusion

These Application Notes illustrate the procedures necessary for configuring the Polycom SoundStation IP to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All feature functionality test cases described in **Section 2.1** were passed.

10. Additional References

Product documentation for the Avaya products may be found at:
https://support.avaya.com
Product documentation for the Polycom Soundstation IP products may be found at:
http://www.polycom.com

[1] *Administering Avaya Aura® Communication Manager Server Options*, July 2012, Release 6.2, Issue 3.0, Document Number 03-603479.
[2] *Administering Avaya Aura® Session Manager*, July 2012, Release 6.2, Document Number 03-603324.
[3] Polycom SoundStation IP Series Documents:
Administrator's Guide for the Polycom® UC Software
http://support.polycom.com/PolycomService/support/us/support/voice/soundstation_ip_series

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.