Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Polycom VVX 500/600 running UC software release 5.0.0.7403 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager Release 6.3 - Issue 1.0

Abstract

These Application Notes describe a solution for supporting interoperability between the Polycom VVX 500/600 running UC software release 5.0.0.7403 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager release 6.3. Emphasis of the testing was to verify voice calls of VVX 500/600 as a SIP endpoint 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 VVX 500/600 (hereafter referred to as VVX 500/600) with a SIP infrastructure consisting of Avaya Aura® Session Manager (hereafter referred to as Session Manager) and Avaya Aura® Communication Manager (hereafter referred to as Communication Manager). During compliance testing, VVX 500/600 successfully registered with Session Manager, established calls with other Avaya telephones and all the applicable telephony features were executed on the VVX 500/600, where applicable, to ensure the interoperability with Communication Manager.

2. General Test Approach and Test Results

The general test approach was to have the VVX 500/600 register to Session Manager. Calls were then placed from Avaya telephone clients/users to and from the VVX 500/600. Other telephony features such as busy, hold, DTMF, transfer, conference, video (where applicable) 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 VVX 500/600 to Session Manager.
- Call establishment of VVX 500/600 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), and Do not Disturb (DND).
- Codec negotiation – G.711, G.729 and G.722.
- Incoming and Outgoing calls to VVX 500/600 from PSTN.
- Video call between two VVX 500/600 phones.

Note: Based on the micro-processor type, VVX 500 and VVX 600 belong to the same family and therefore the test results of VVX 500 also holds good for VVX 600. During compliance testing, only VVX 500 was tested.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. VVX 500/600 was registered to Session Manager successfully. Calls have been made between Communication Manager telephones and VVX 500/600 with clear voice path. All executed test cases passed with the following observations,

- On Communication Manager only the option of G.722 – 64 is available and since this option is not available from the VVX 500/600 codec list, this codec option could not be tested.
- Call Forward on Busy (CFB) has to be configured on Communication Manager at the set level and not through the Polycom Web Configuration Utility. However Call Forward Unconditional (CFU) and Call Forward No Answer (CFNA) can be configured using the Polycom Web Configuration Utility.

2.3. Support

Technical support for the Polycom VVX 500/600 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 Aura® Messaging providing voice mail service for the SIP endpoints.

The VVX 500/600 registers with Session Manager and is configured as an Off-PBX Station (OPS) on Communication Manager. Polycom Web Configuration Utility is used to manage the configuration of the VVX 500/600 phone.

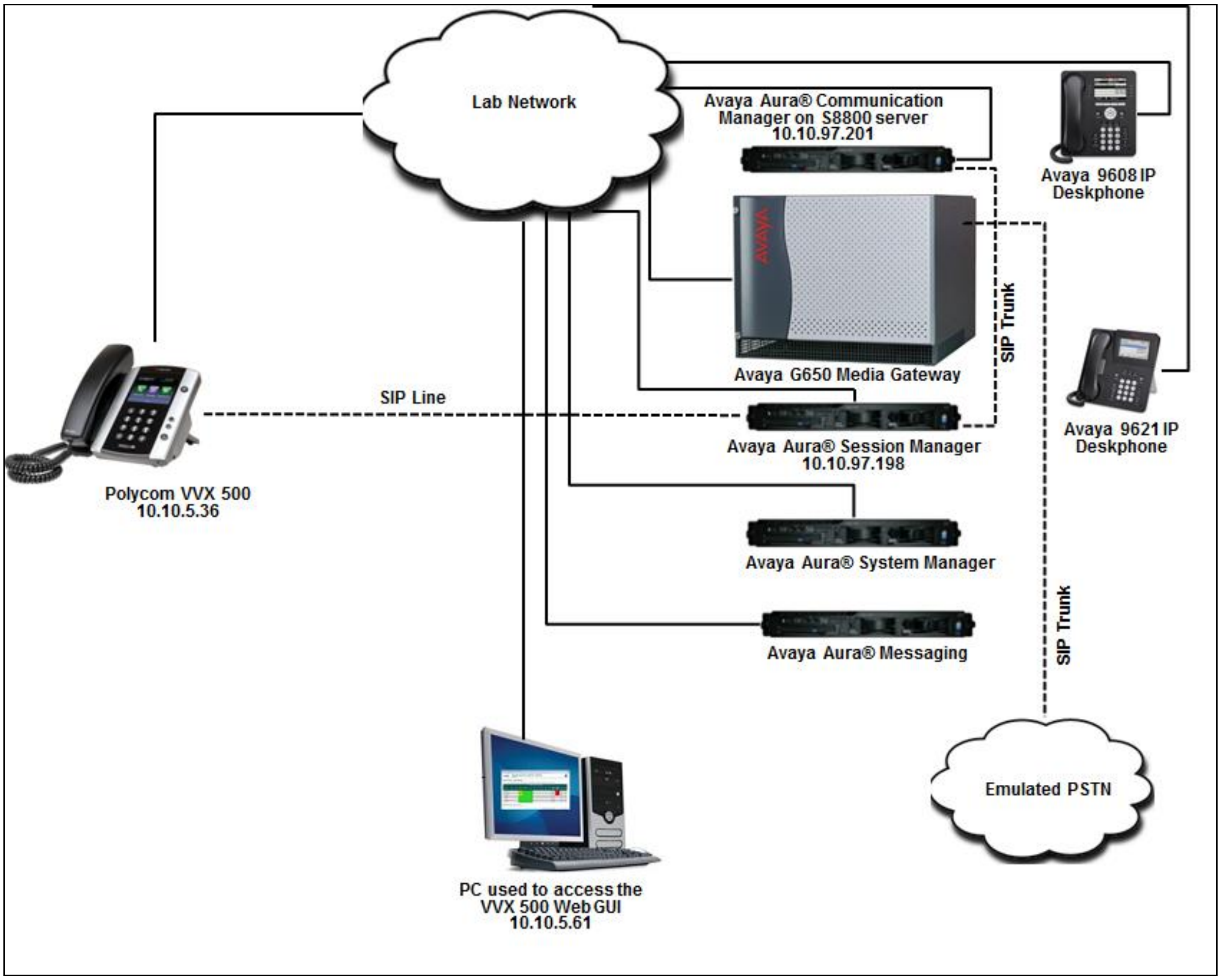


Figure 1: Network Configuration Diagram

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 S8800 Server and G650 Media Gateway	6.3-03.0.124.0
Avaya Aura® System Manager running on an Avaya S8800 Server	6.3.0-FP2
Avaya Aura® Session Manager running on S8800 Server.	6.3.2.0.632023
Avaya Aura® Messaging	6.1
Avaya 9620G IP (SIP) Telephone	6.2.0
Avaya 9608 IP (H.323) Telephone	6.0.2
Polycom UC Software for VVX 500	5.0.0.7403
Polycom Web Configuration Utility	Windows XP Professional OS

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring an Off-PBX Station (OPS) that can be used for VVX 500/600 and configuring a SIP trunk between Communication Manager and Session Manager. **Section 5.3** covers the station configuration that will be used by VVX 500/600. 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: 65000 213
                                     Maximum Stations: 41000 37
                             Maximum XMOBILE Stations: 41000 0
                   Maximum Off-PBX Telephones - EC500: 41000 4
                   Maximum Off-PBX Telephones -   OPS: 41000 24
                   Maximum Off-PBX Telephones - PBFMC: 41000 0
                   Maximum Off-PBX Telephones - PVFMC: 41000 0
                   Maximum Off-PBX Telephones - SCCAN: 0     0
                        Maximum Survivable Processors: 313   1




        (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: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 6
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 1
                       Maximum Administered SIP Trunks: 24000 130
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   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 S8800 Server processor, and Session Manager. The host names will be used throughout the other configuration screens of Communication Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
AES62               10.10.98.17
AVAYARDTT           10.10.98.68
CLAN1               10.10.97.217
CLAN2               10.10.97.238
DevCM3              10.10.4.9
GW                  10.10.97.193
InteropSM62         10.10.1.11
LSP-1               10.10.4.22
MedPro1             10.10.97.218
MedPro2             10.10.97.233
SM61                10.10.97.198
Server-1            10.10.97.19
default             0.0.0.0
procr               10.10.97.201
procr6              ::

( 15 of 15   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, **Intra-region IP-IP Direct Audio and Inter-region 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 G650 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 '1') is specified in the SIP signaling group.

```
display ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: bvwdev.com
    Name:                     Stub Network Region: n
MEDIA PARAMETERS              Intra-region IP-IP Direct Audio: yes
      Codec Set: 1            Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                    IP Audio Hairpinning? n
   UDP Port Max: 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 VVX 500/600. 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 VVX 500/600.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729             n         2        20
 3: G.722-64K                   2        20
 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 Communication Manager Processor Interface (procr) 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 TCP 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. Retain default values for all other fields.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SM61
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

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): 30
```

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 **Group Number** field, and specify the **Number of Members** supported by this SIP trunk group. Configure an appropriate **TAC** value. Retain default values for all other fields.

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

Group Number: 1                    Group Type: sip          CDR Reports: n
  Group Name: Private trunk               COR: 1        TN: 1        TAC: #001
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     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.

```
display trunk-group 1                                           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
```

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 '53' and whose calls are routed over any trunk group, including SIP trunk group "1", have the number sent to the far-end for display purposes.

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

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  5              1                           5     Total Administered: 1
       Maximum Entries: 540
```

5.3. Configure Stations

Use the **add station** command to add a station for each VVX 500/600 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**.

```
display station 53113                                             Page   1 of   6
                                     STATION

Extension: 53113                         Lock Messages? n               BCC: 0
     Type: 9620SIP                       Security Code:                  TN: 1
     Port: S00006                      Coverage Path 1:                 COR: 1
     Name: 53113, Moto                 Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 53113

        Display Language: english                  Button Modules: 0

          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 VVX 500/600.

```
display station 53113                                             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

                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 53113         Always Use? n IP Audio Hairpinning? n
```

Use the **change off-pbx-telephone station-mapping** command to map Communication Manager extensions (e.g., 53113) 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 53113                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 53113           OPS           -     53113            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 53113                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 53113         OPS      3        both         all         none
```

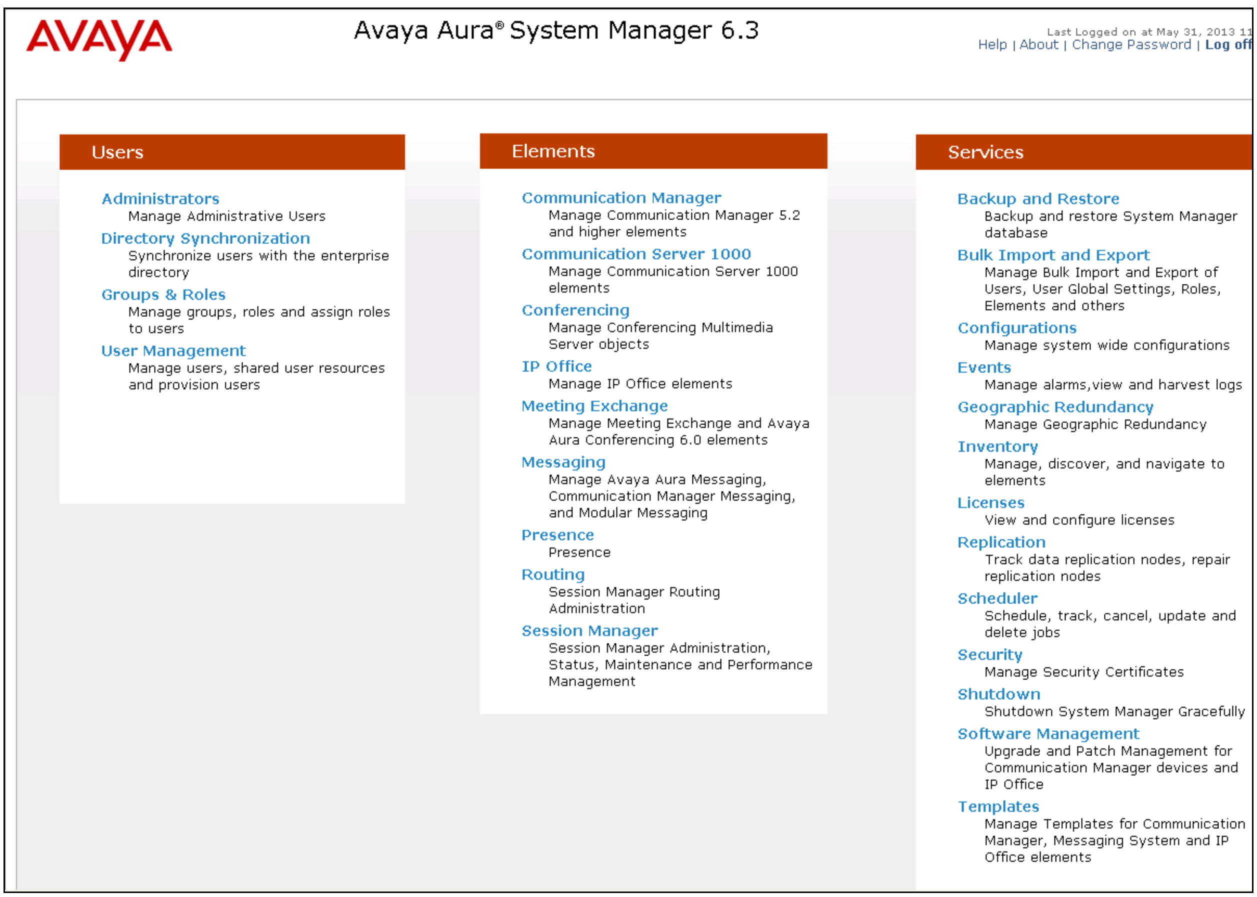
6. Configure Avaya Aura® Session Manager

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.

The main screen of System Manager is seen as shown below.



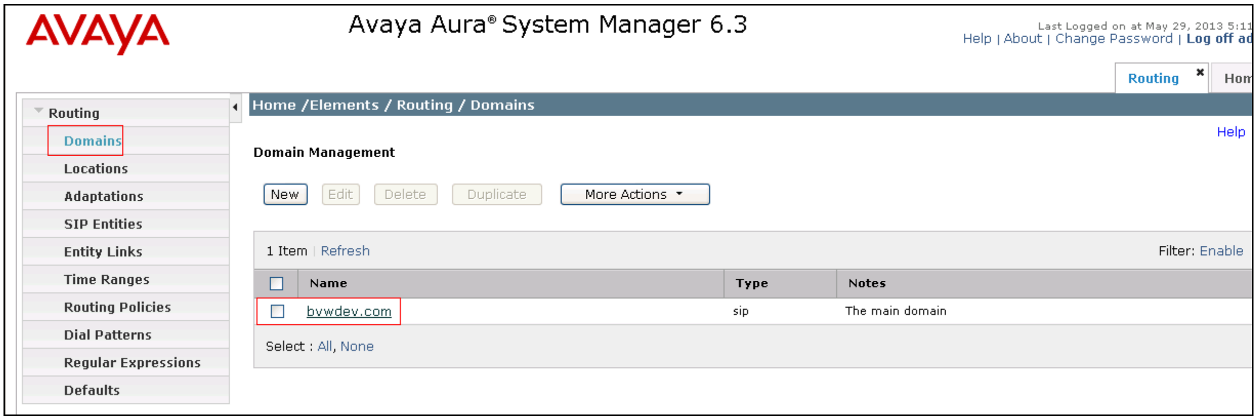
6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Navigate to **Routing → 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** (not shown).

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 (not shown). The following screen will then be shown. Fill in the following:

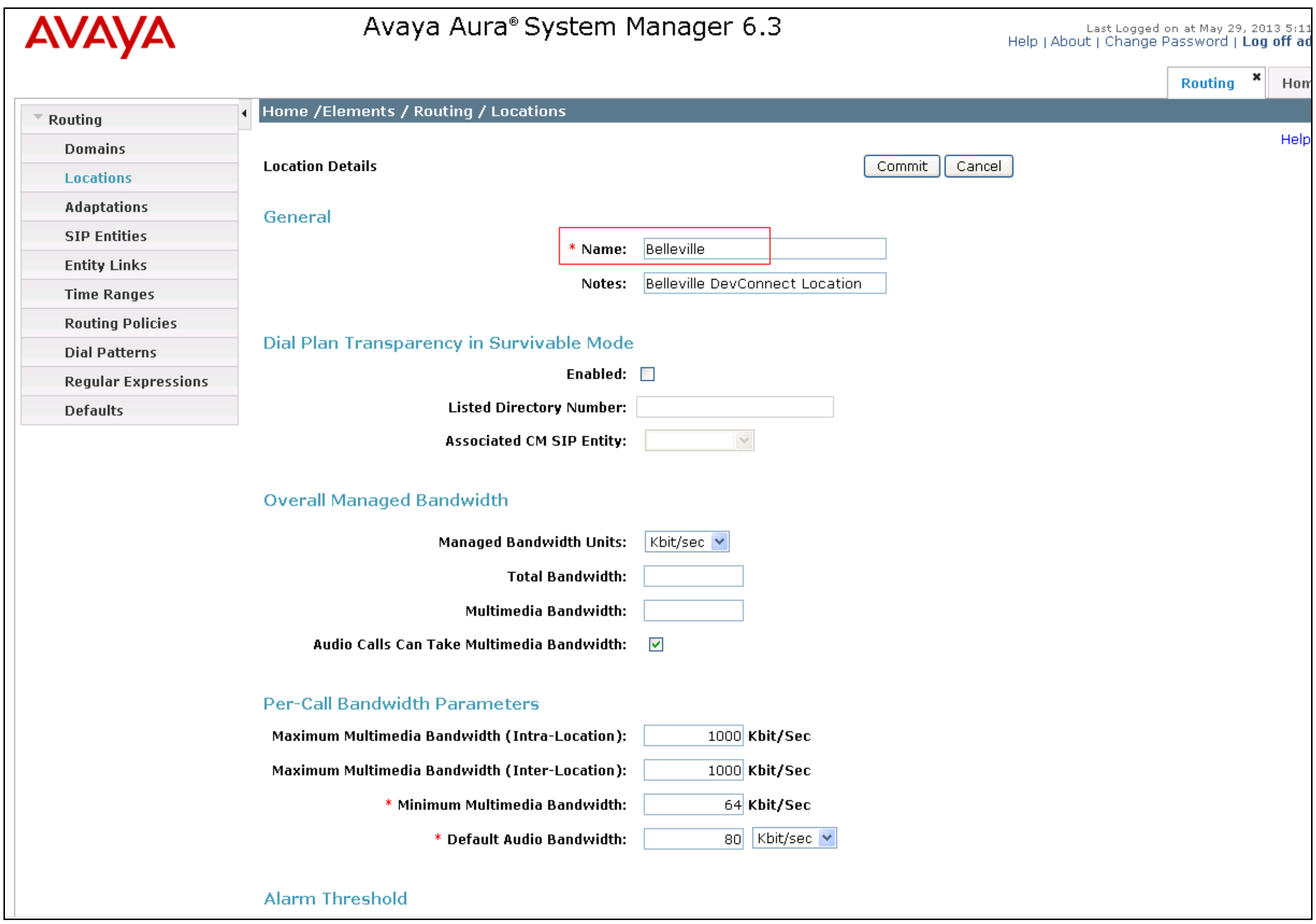
Under *General*:

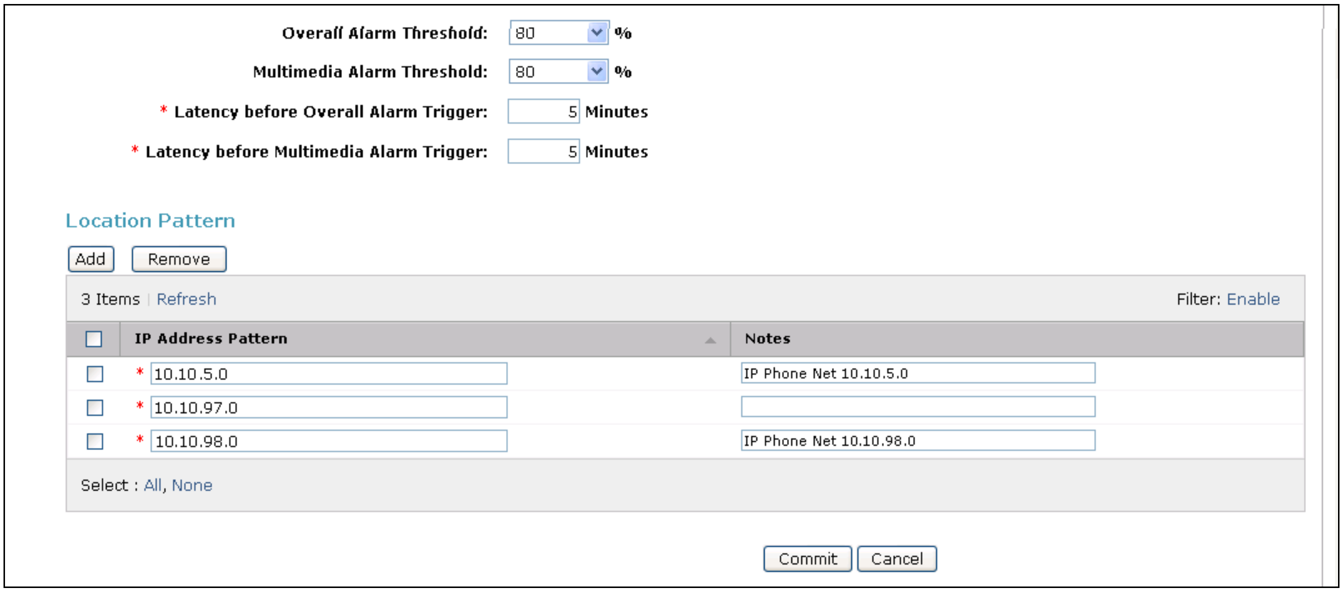
- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

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 *Belleville* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition. Retain default values for all other fields.




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
3 Items | Refresh
Filter: Enable
IP Address Pattern
Notes
* 10.10.5.0
IP Phone Net 10.10.5.0
* 10.10.97.0
* 10.10.98.0
IP Phone Net 10.10.98.0
Select : All, None
Commit
Cancel

6.3. Add SIP Entities

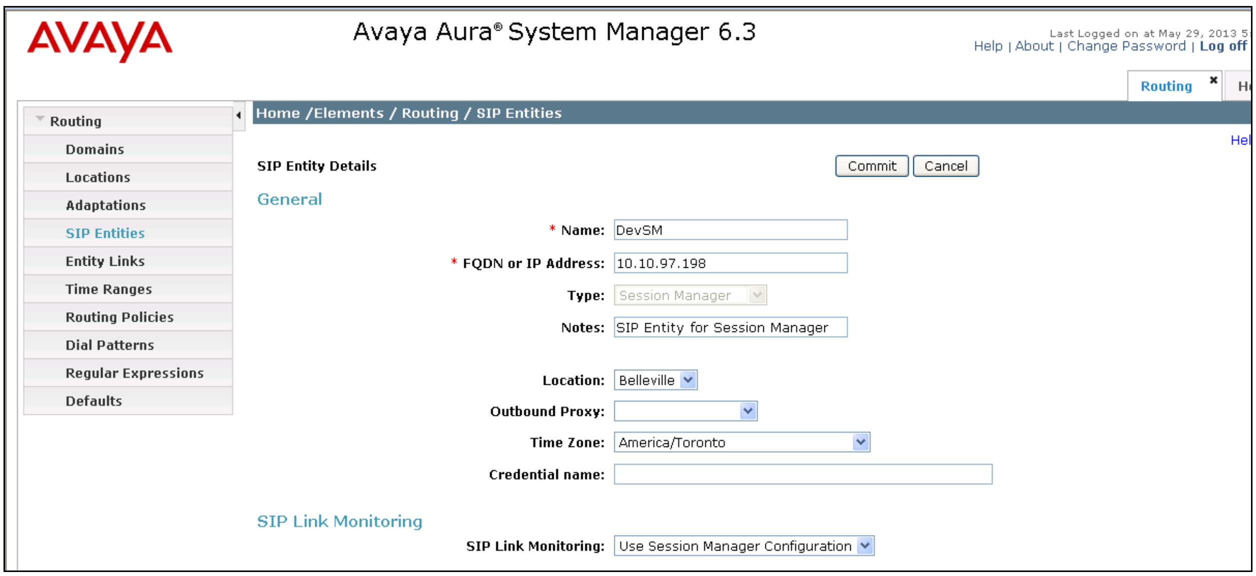
In the sample configuration, a SIP Entity is added for Session Manager and Communication Manager.

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 (not shown). The following screen is displayed. Fill in the following:

Under *General*:

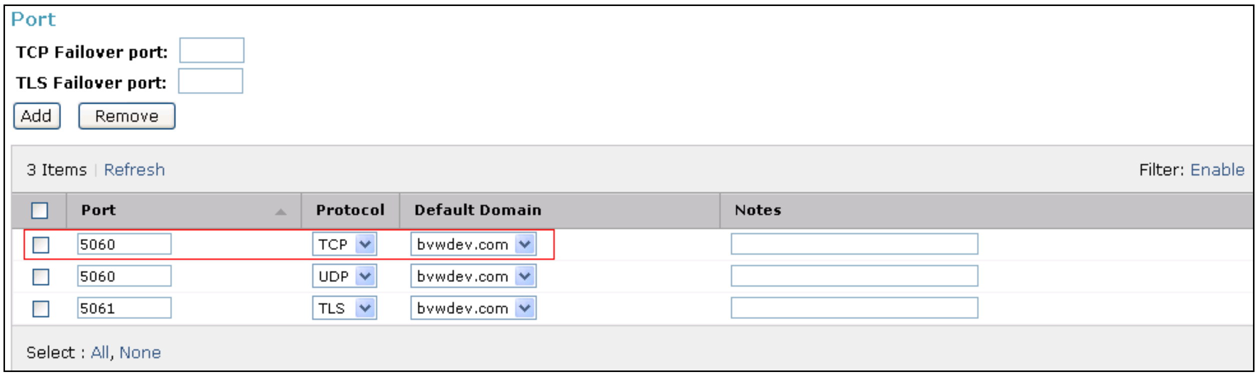
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Specify *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.



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. During compliance testing only TCP was used.
- **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.



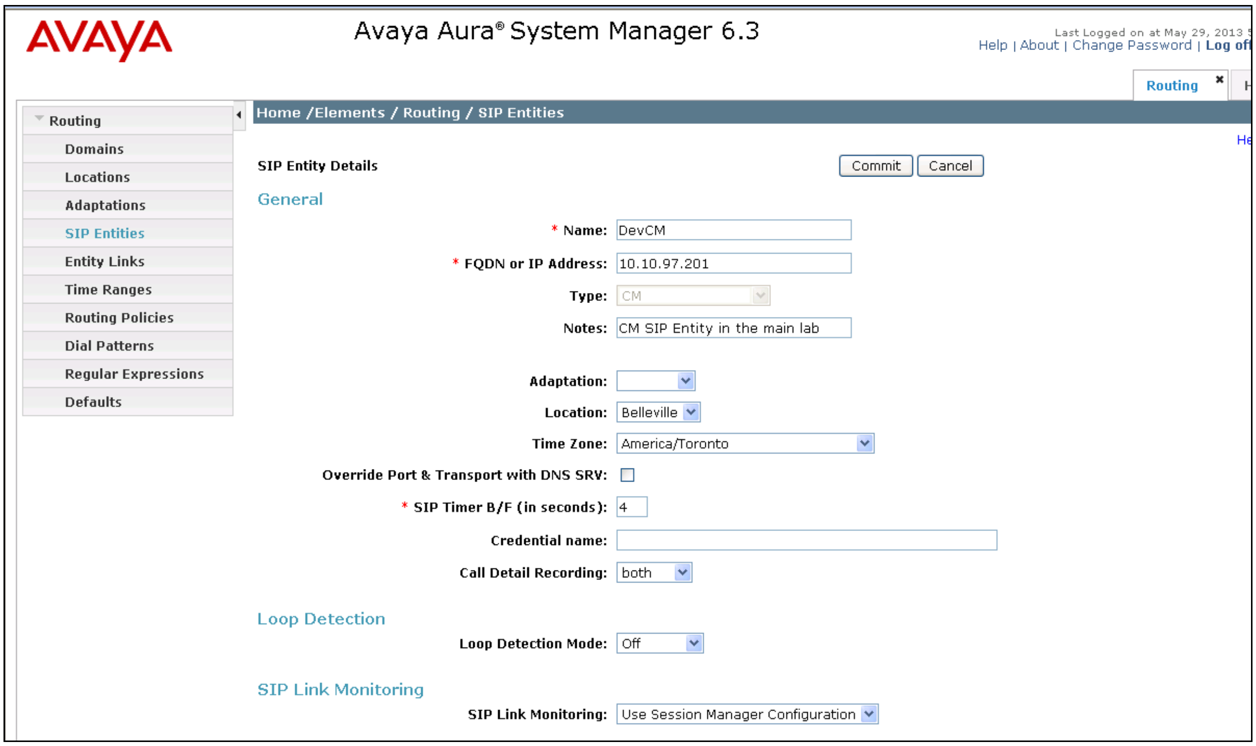
6.3.2. Avaya Aura® Communication Manager

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 (not shown). 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., procr interface) on the telephony system.
- **Type:** Specify *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Retain default values for all other fields. Click **Commit** to save each SIP Entity definition.

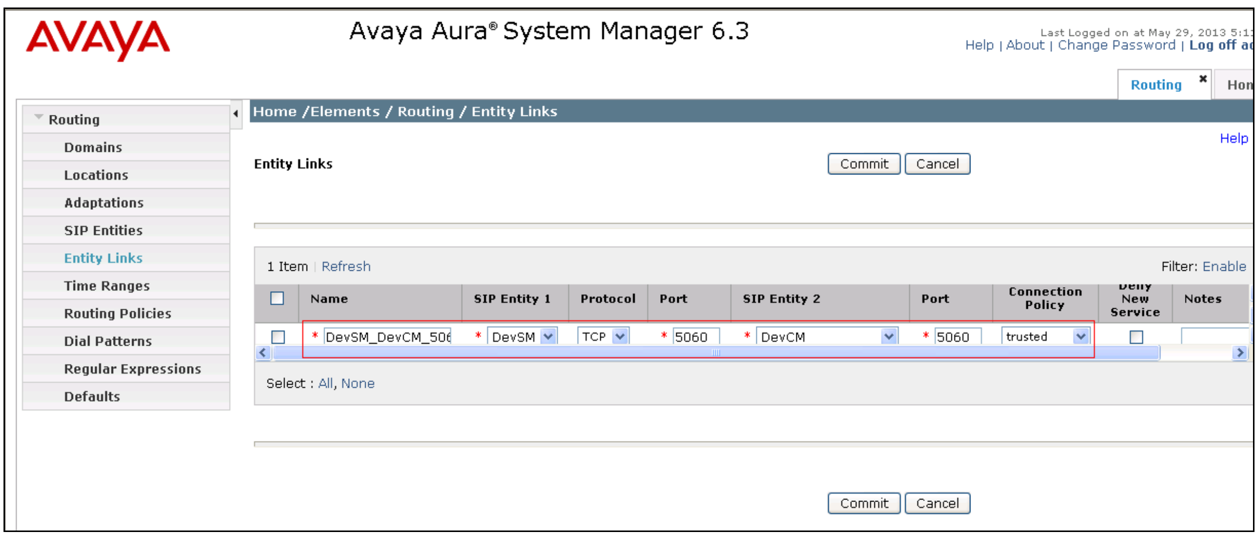


6.4. Add Entity Link

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 (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name
- **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.
- **Connection Policy** 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.

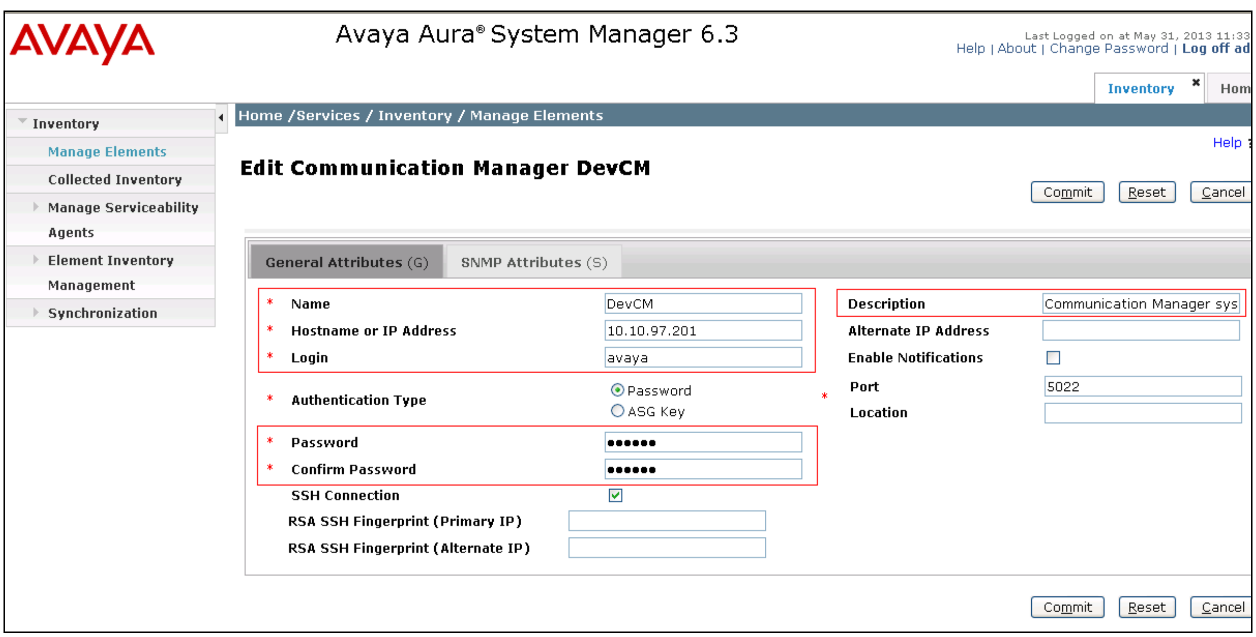
To define Communication Manager as a managed element, under **Services** (refer to screen shot in **Section 6.3**) navigate to **Inventory → Manage Elements** on the left and click on the **New** button on the right (not shown). In the **Application Type** field that is displayed (not shown), select *Communication Manager*.

Screen below shows an already added Communication Manager. Enter the values as follows and retain default values for all other fields.

Under *General Attributes (G)*:

- **Name:** Enter an identifier for Communication Manager.
- **Description:** Enter an appropriate description.
- **Hostname or IP Address:** Enter the IP address of the administration interface for Communication Manager.
- **Login:** A login name.
- **Password:** Enter password.
- **Confirm Password:** Confirm above entered password.

Click **Commit** to save the settings.

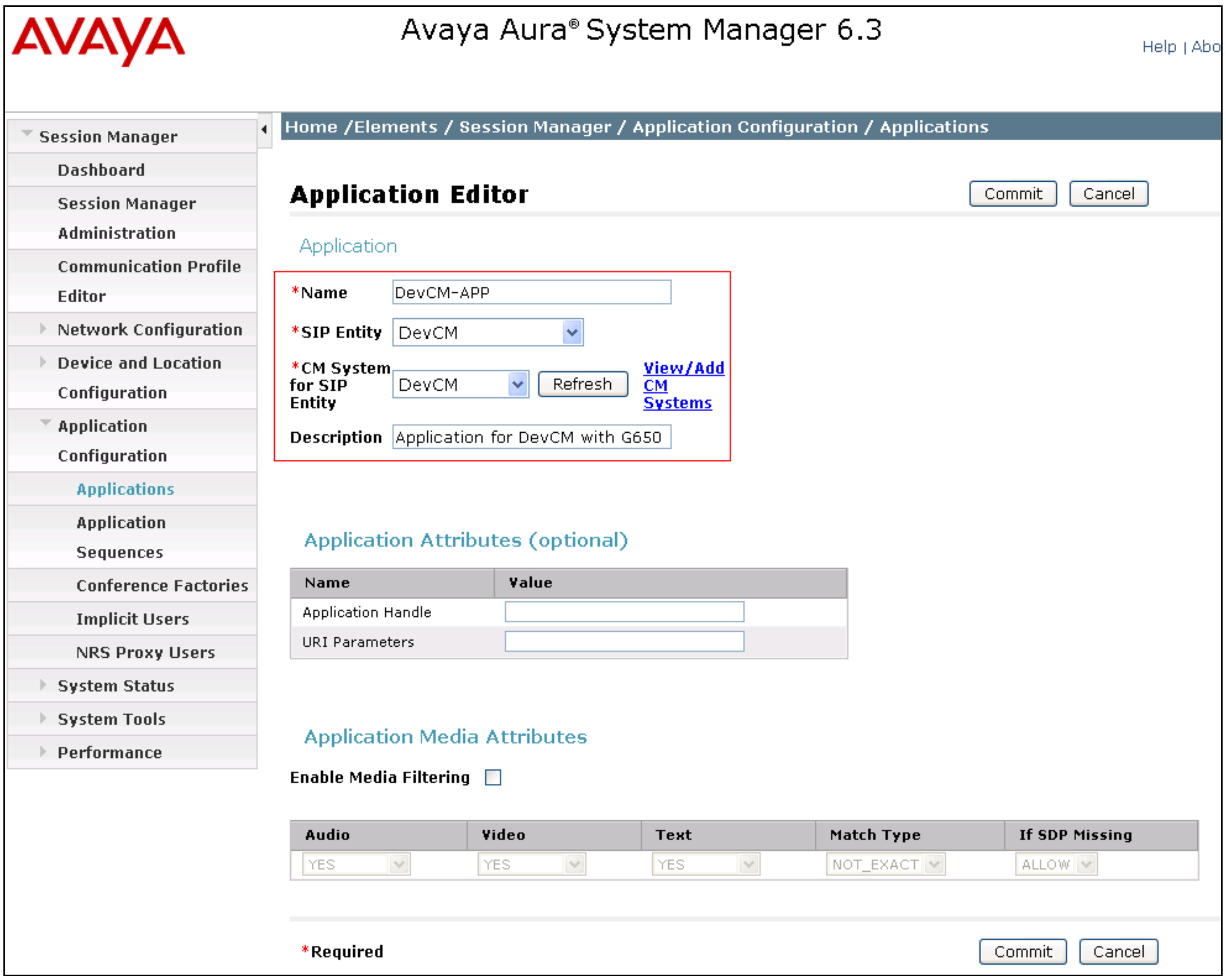


6.6. Add Application Sequence

Define an application for Communication Manager. Under **Elements** (refer to screen shot in **Section 6.3**) navigate to **Session Manager → Application Configuration →Applications** on the left and click on the **New** button on the right (not shown). Fill in the following fields:

- **Name:** An appropriate name.
- **SIP Entity:** Select the Communication Manager SIP entity.
- **CM System for SIP Entity:** Select the Communication Manager managed element.
- **Description:** An appropriate description.

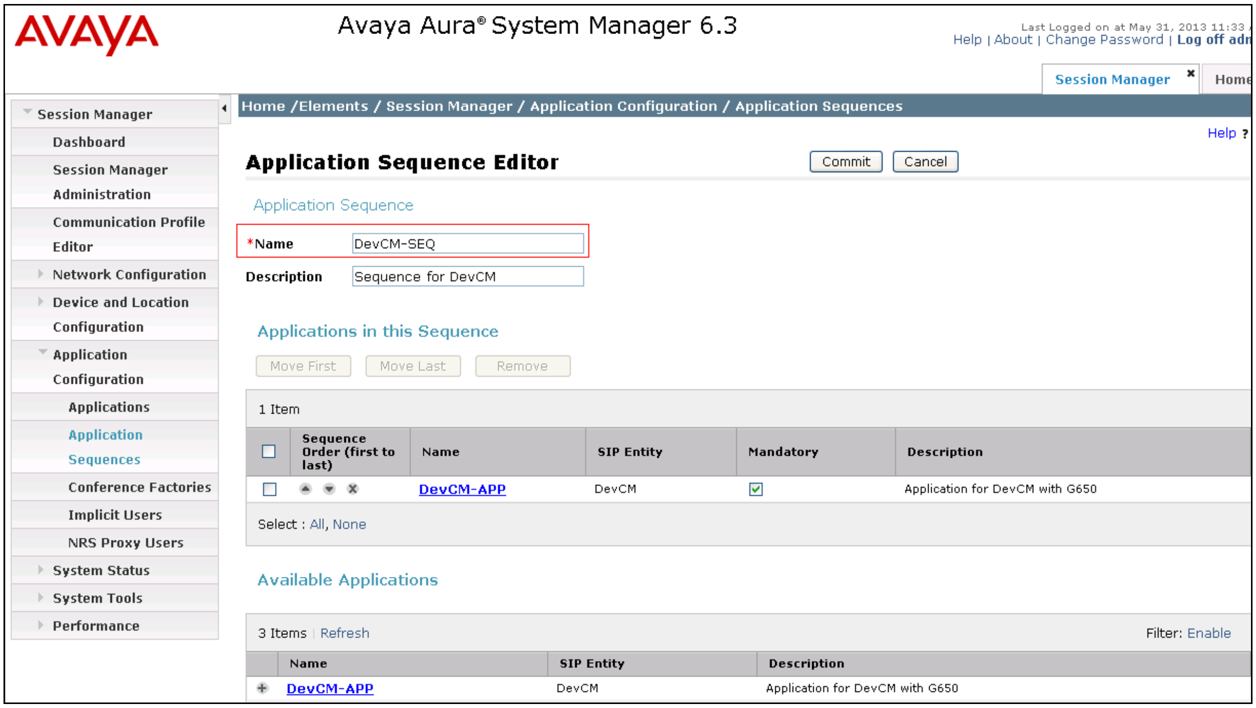
Click **Commit** to save the Application definition.



Next, define the **Application Sequences** for Communication Manager as shown below. Under **Elements** (refer to screen shot in **Section 6.3**) navigate to **Session Manager → Application Configuration →Application Sequences** on the left and click on the **New** button on the right (not shown). Fill in the following fields:

Enter a descriptive name in the **Name** field.

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



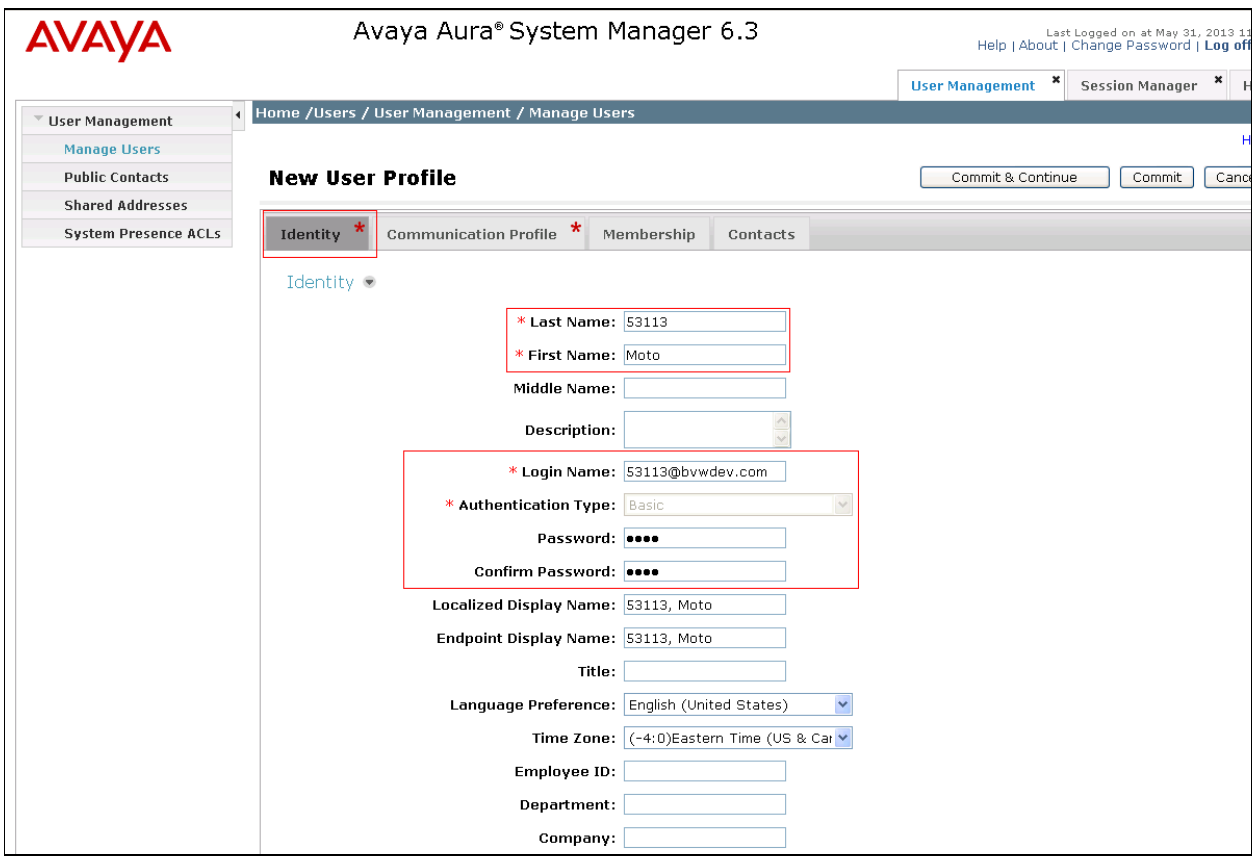
6.7. Add SIP Users

Add a SIP user corresponding to the VVX 500/600 as defined in **Section 5.3.** Alternatively, use the option to automatically generate the SIP stations on Communication Manager Feature/Evolution Server when adding a new SIP user. Under **Users** (refer to screen shot in **Section 6.3**) navigate to **User Management → Manage Users** (not shown) on the left and click on the **New** button on the right (not shown).

Under the **Identity** tab 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., 53113@bvwdev.com).
- **Authentication Type:** Select *Basic (by default)***.**
- **Password:** Password to be used by the SIP User.
- **Confirm Password:** Re-enter the password from above.

The screen below shows the information when adding a new SIP user during compliance testing.



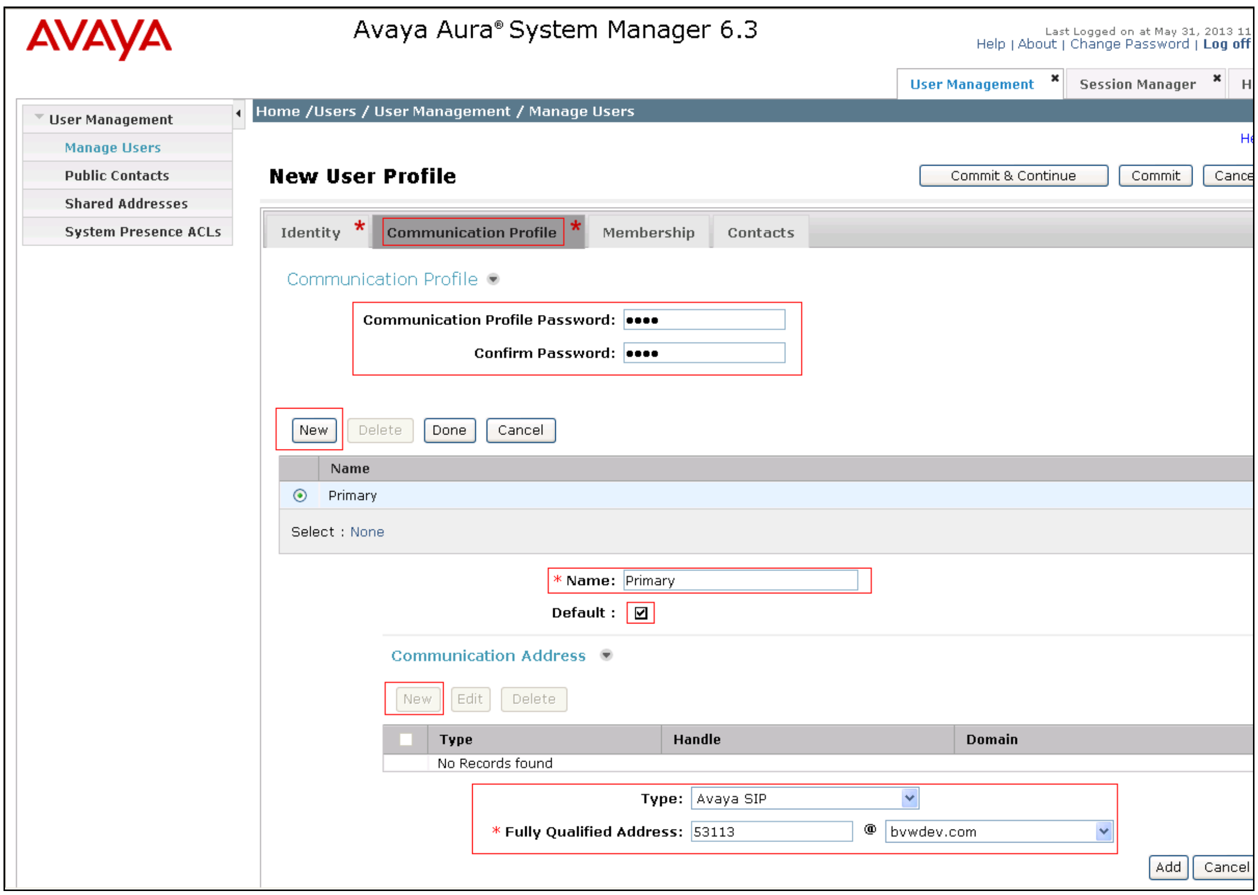
Click on the **Communication Profile** tab and select **New** to define a **Communication Profile** for the new SIP user. Enter a password in the **Communication Profile Password** and **Confirm Password** fields. Enter values for the following required fields:

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

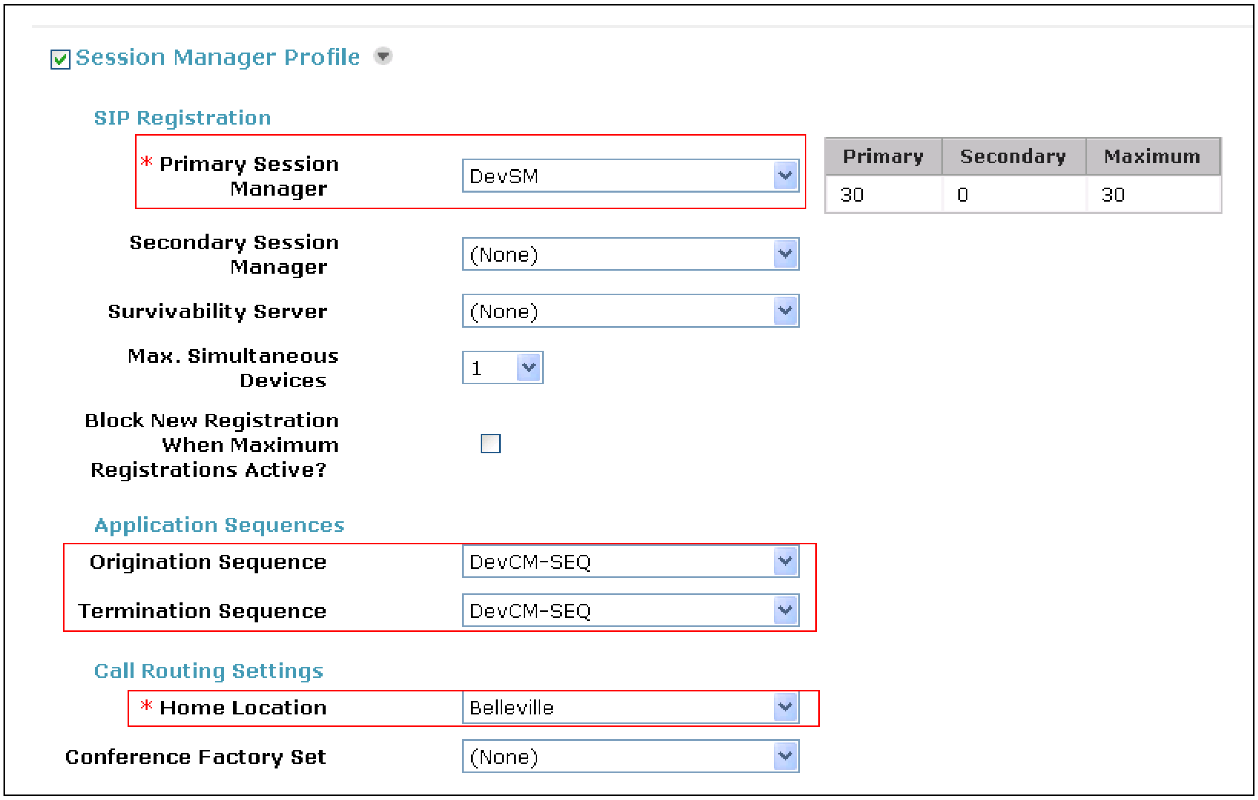
The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.



In the **Session Manager Profile** section, enter the following values,

- Under **SIP Registration**, select the Session Manager from the drop down list for the **Primary Session Manager** field.
- Assign the **Application Sequence** defined in **Section 6.6** 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.
- Select the required Home Location value from the drop down menu.

Retain default values for all other fields.

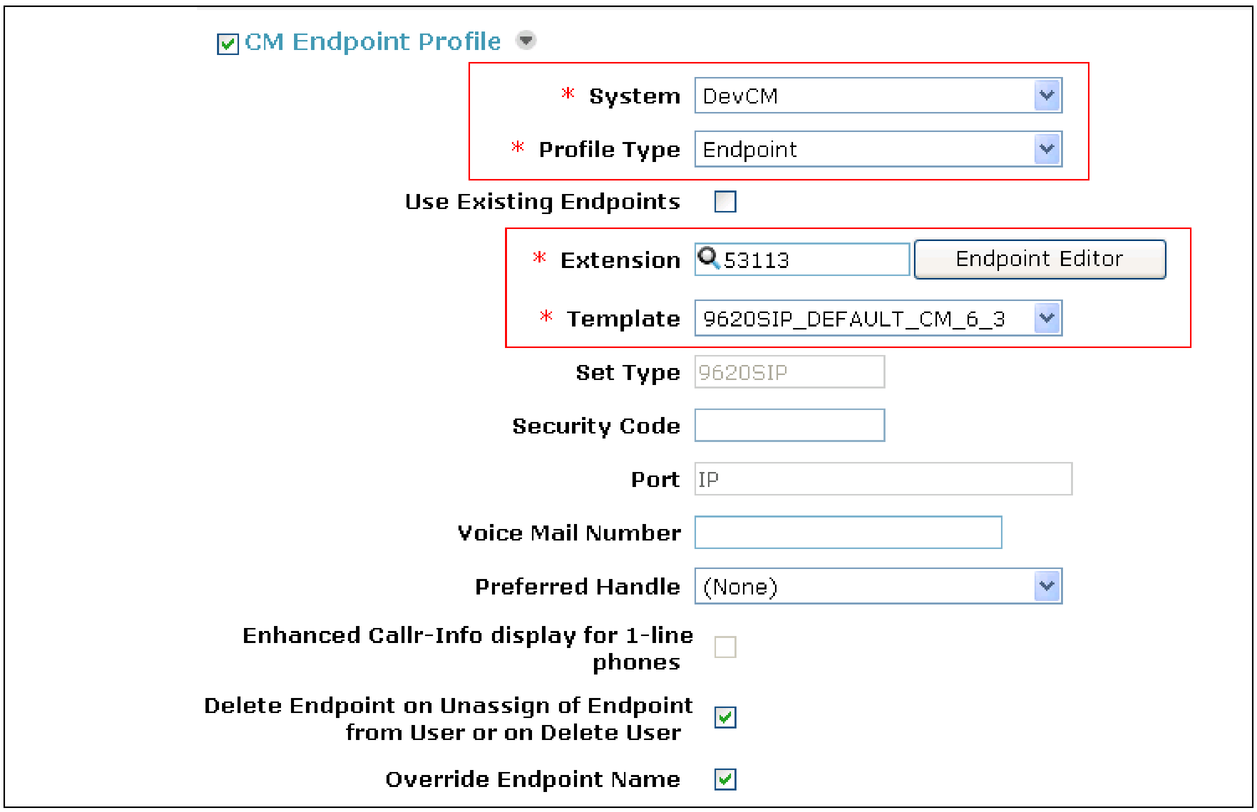


In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type:** Select Endpoint.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.

Retain default values for all other fields.

The screen below shows the configuration used for compliance testing.



Click **Commit** (not shown) to save the User Profile.

6.8. Add Session Manager

To complete the configuration, adding Session Manager will provide the linkage between System Manager and Session Manager. Under **Elements** (refer to screen shot in **Section 6.3**) navigate to **Session Manager → Session Manager Administration** (not shown). Then click **New** (not shown), and fill in the fields as described below and shown in the following screen:

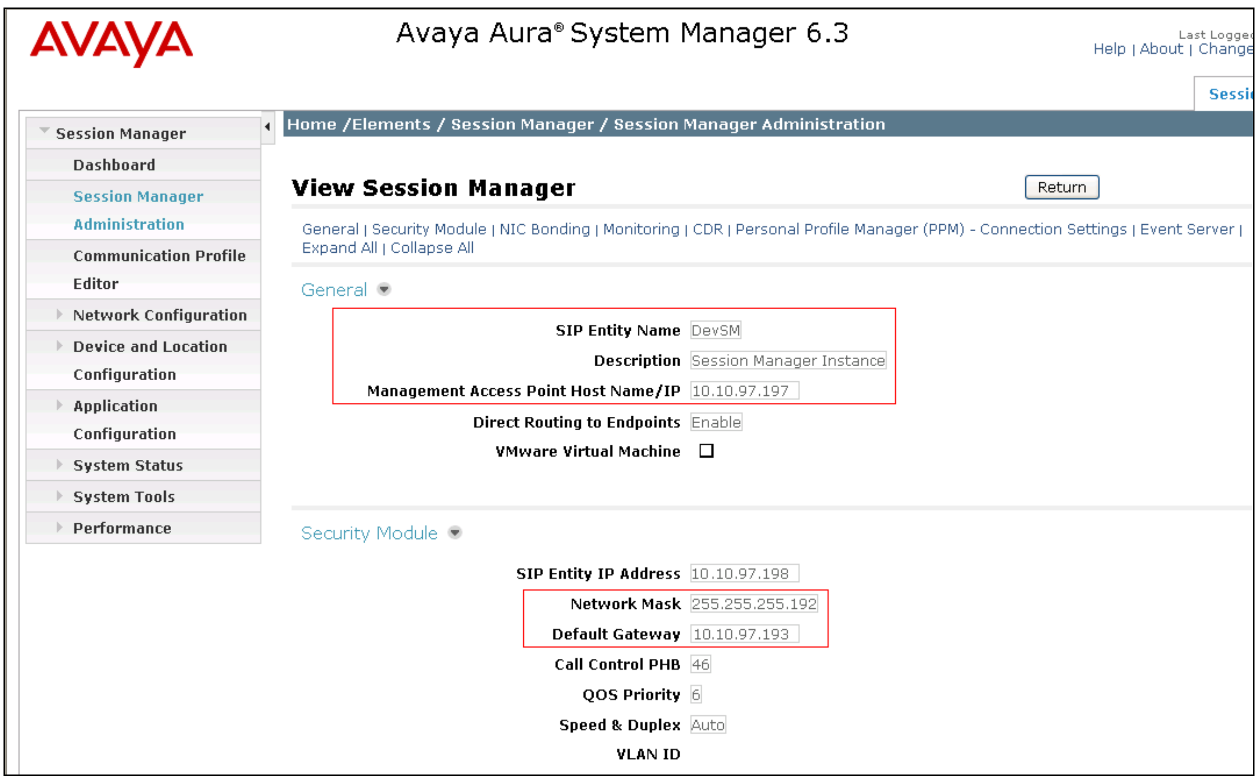
Under *General*:

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

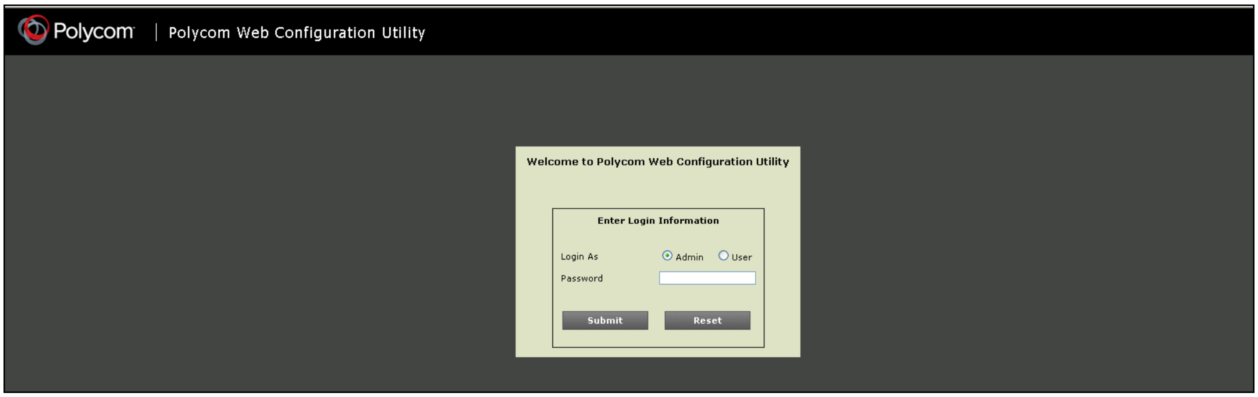
Retain default values for the remaining fields. Click **Save** to add this Session Manager (not shown).



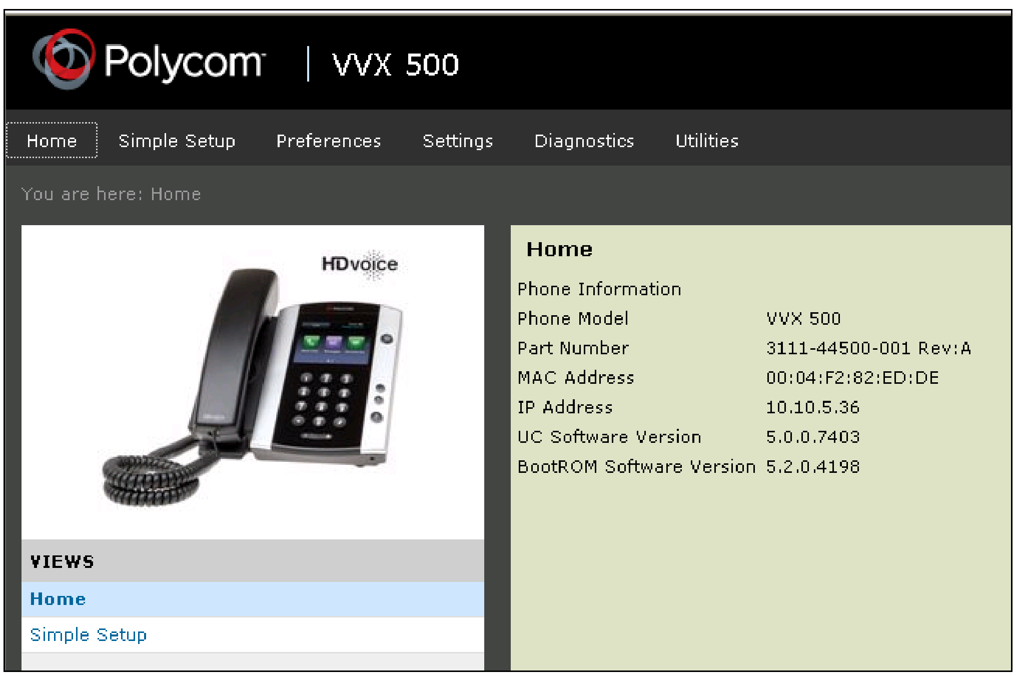
7. Polycom Web Configuration Utility

This section shows how to log in to the home page of Polycom Web Configuration Utility that is required to configure the VVX 500/600 phone.

Find the IP address assigned to the VVX 500/600 phone and type it into the URL address bar of a web browser. The web configuration utility login interface will be displayed as shown below. Select the **Admin** radio button and type in the default password of **456**.



Click **Submit**, the homepage of the Polycom VVX 500 is shown below.

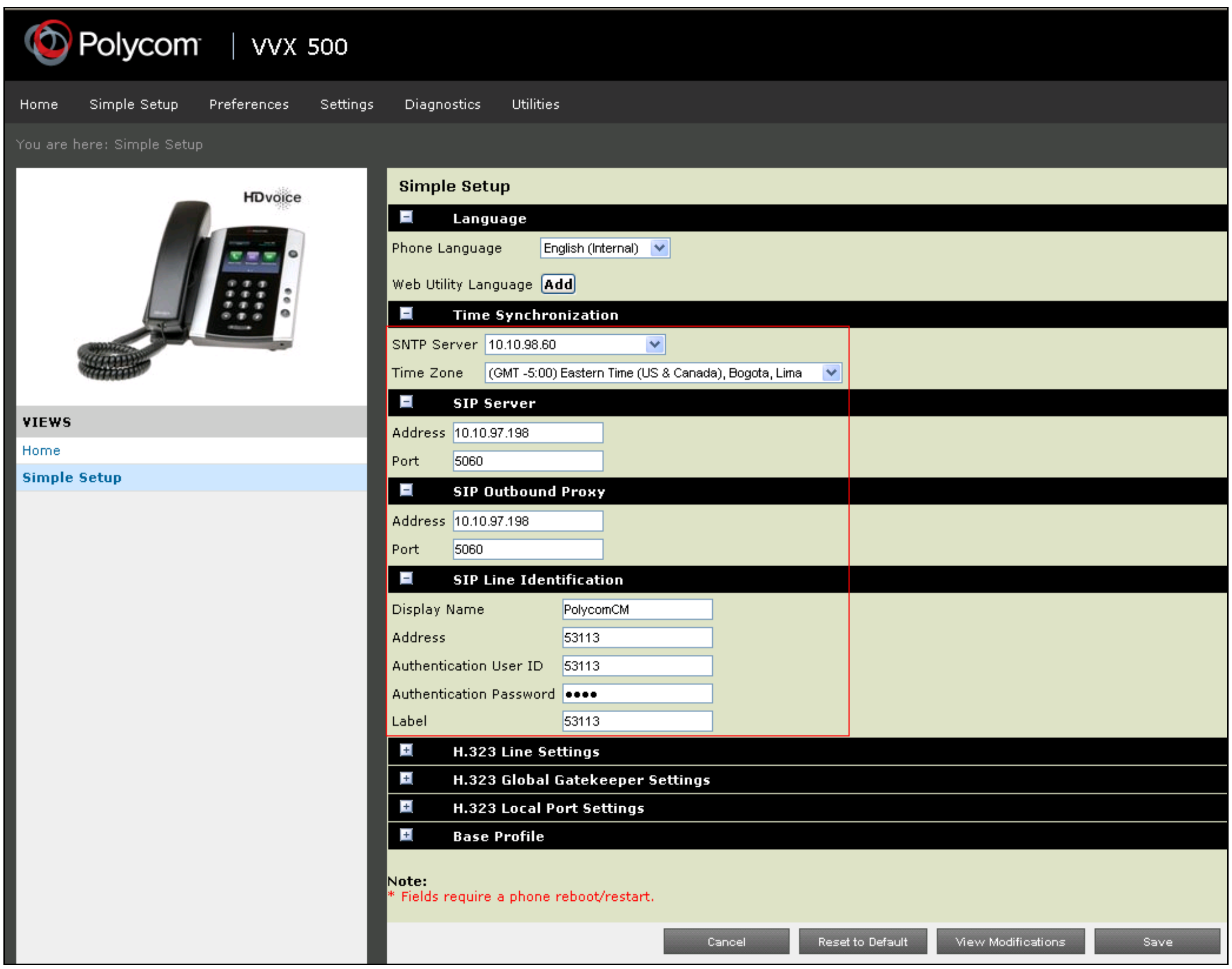


7.1. Configure the Lines for Polycom VVX 500/600

This section shows how to configure the VVX 500/600 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 the following values,

- **Phone Language**: English (internal)
- **Time Zone**: Select time zone for the region.
- Under **SIP Server** section, **Address**: 10.10.97.198 and **Port**: 5060; configured in **Section 6.3.1**.
- Under **SIP Outbound Proxy** section, **Address**: 10.10.97.198 and **Port**: 5060; configured in **Section 6.3.1**
- Under the **SIP Line Identification** section, **Display Name**: an appropriate name, **Address**: 53113, **Authentication User ID**: 53113 and **Authentication Password**: 1234; configured in **Section 6.7**

Click on **Save**.



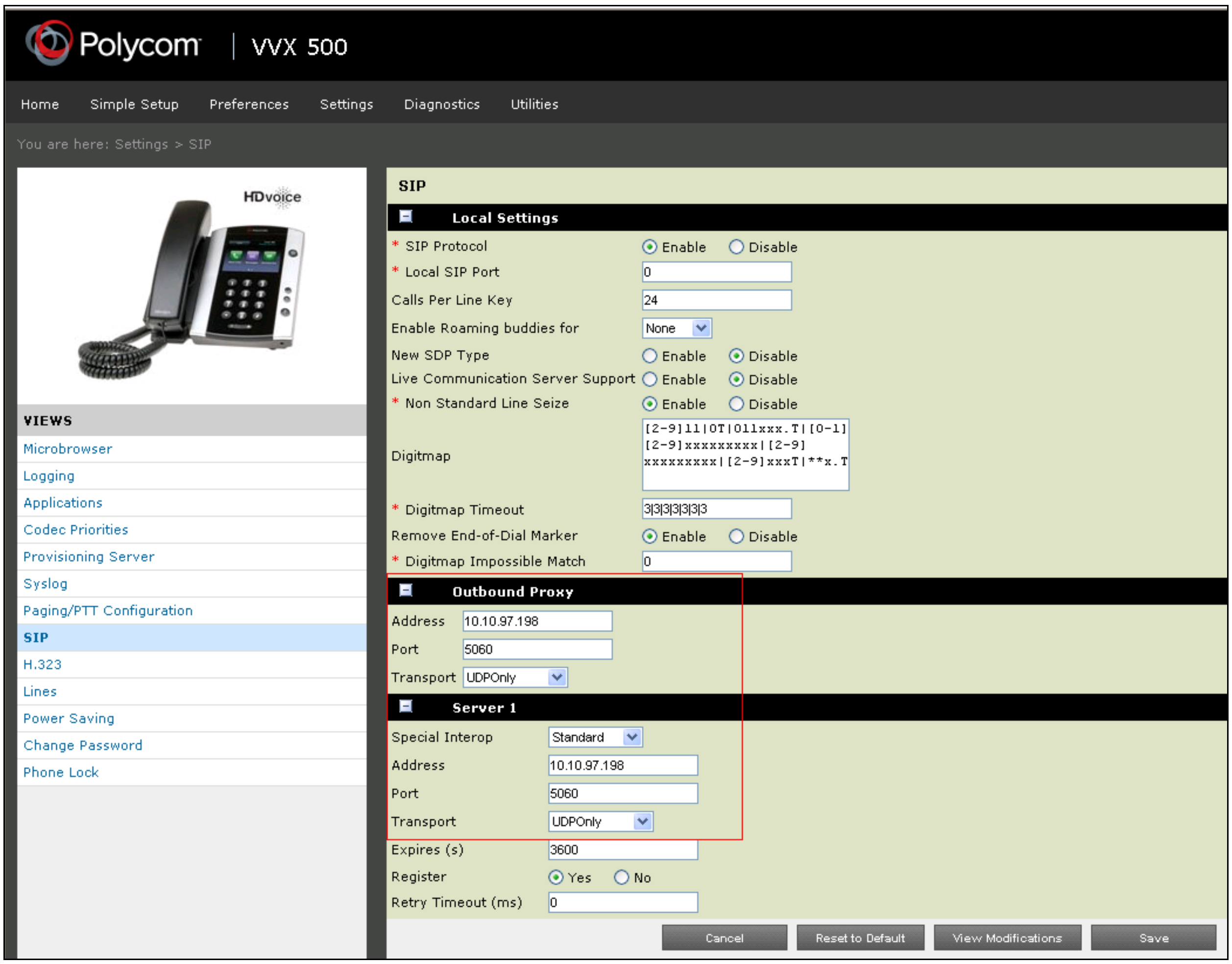
7.2. SIP Settings

This section shows how to set SIP parameters for VVX 500/600.

On the homepage of VVX 500/600, navigate to menu **Settings → SIP** (not shown), **SIP** screen is shown below. Enter the following values and retain rest at default.

- Under the **Outbound Proxy** section, **Address**: 10.10.97.198 and **Port**: 5060; configured in **Section 6.3.1 Transport**: UDPOnly.
- Under the **Server1** section, **Address**: 10.10.97.198 and **Port**: 5060; configured in **Section 6.3.1**.**Transport**: UDPOnly.

Click on **Save**.



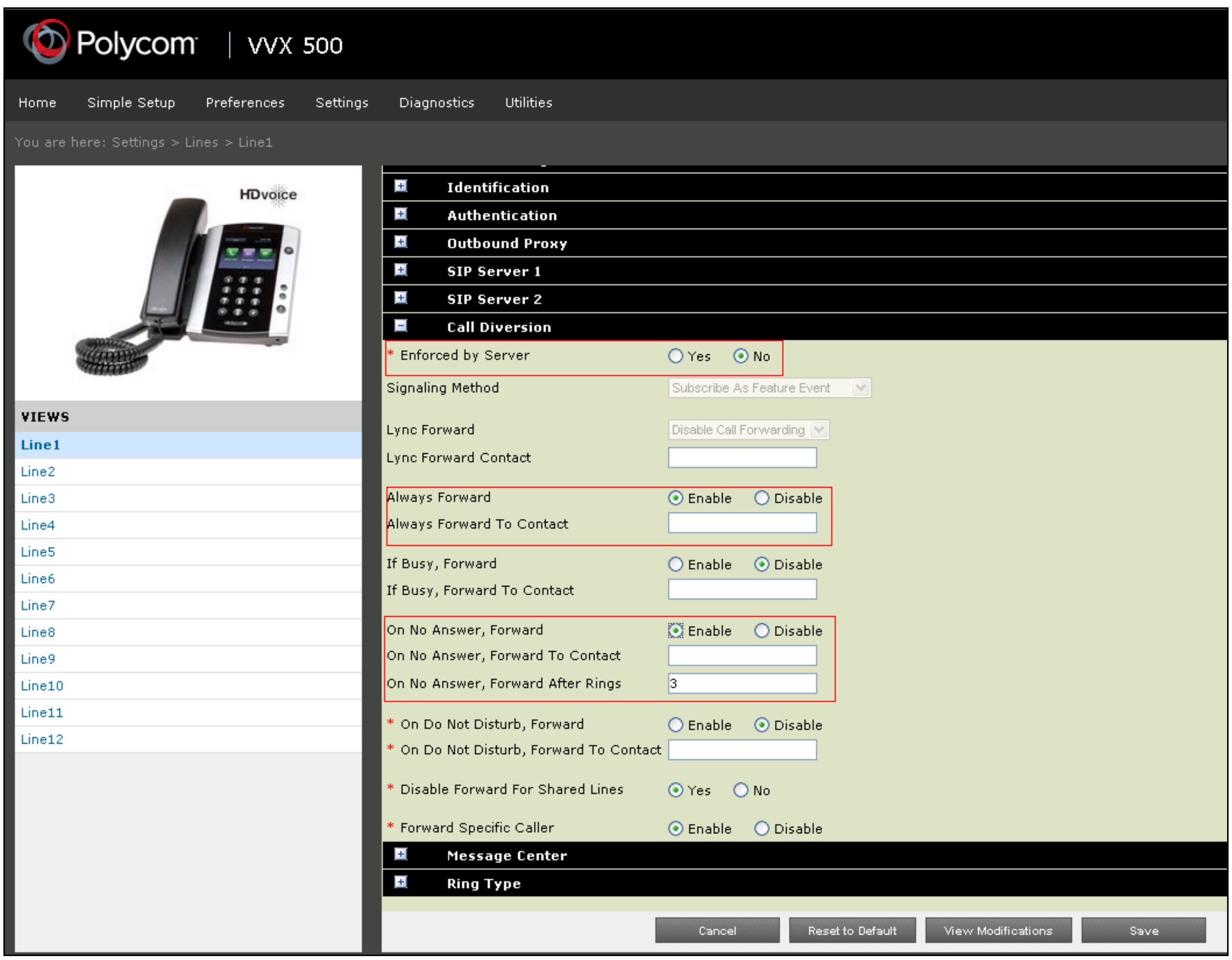
7.3. Local Call Forward Settings

This section shows how to set up call forward settings for Polycom VVX 500/600.

On the homepage of Polycom VVX 500/600, navigate to menu **Settings → Lines** (not shown). **Line1** screen is shown below. Enter the following values and retain rest at default.

- Under the **Call Diversion** section, ensure that the **Enforced by Server** radio button is No.
- **Always Forward**: Enable and configure an appropriate Directory Number (DN) for the **Always Forward To Contact** field.
- **On No Answer, Forward:** Enable and configure an appropriate Directory Number (DN) for the **On No Answer, Forward to Contact** field. Configure an appropriate value on the **On No Answer, forward After Rings** field.

Click on **Save**. As mentioned in **Section 2.2**, **If Busy, Forward** option does not function if configured here and has to be configured on the Communication Manager at the set level.



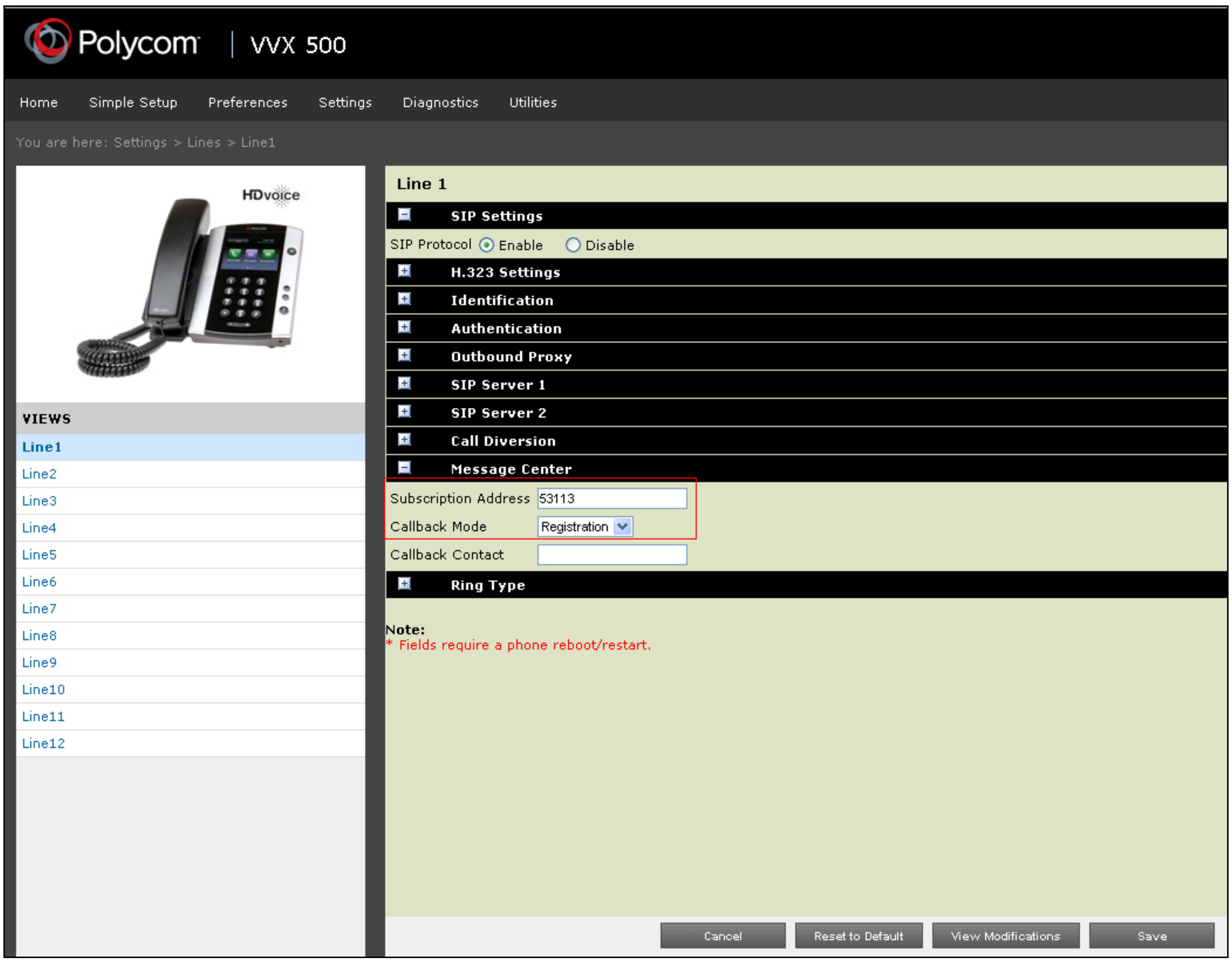
7.4. Configuring Message Center for Message Waiting Indicator

This section shows how to set up activation of MWI for Polycom VVX 500/600.

On the homepage of Polycom VVX 500/600, navigate to menu **Settings → Lines** (not shown). **Line1** screen is shown below. Enter the following values and retain rest at default.

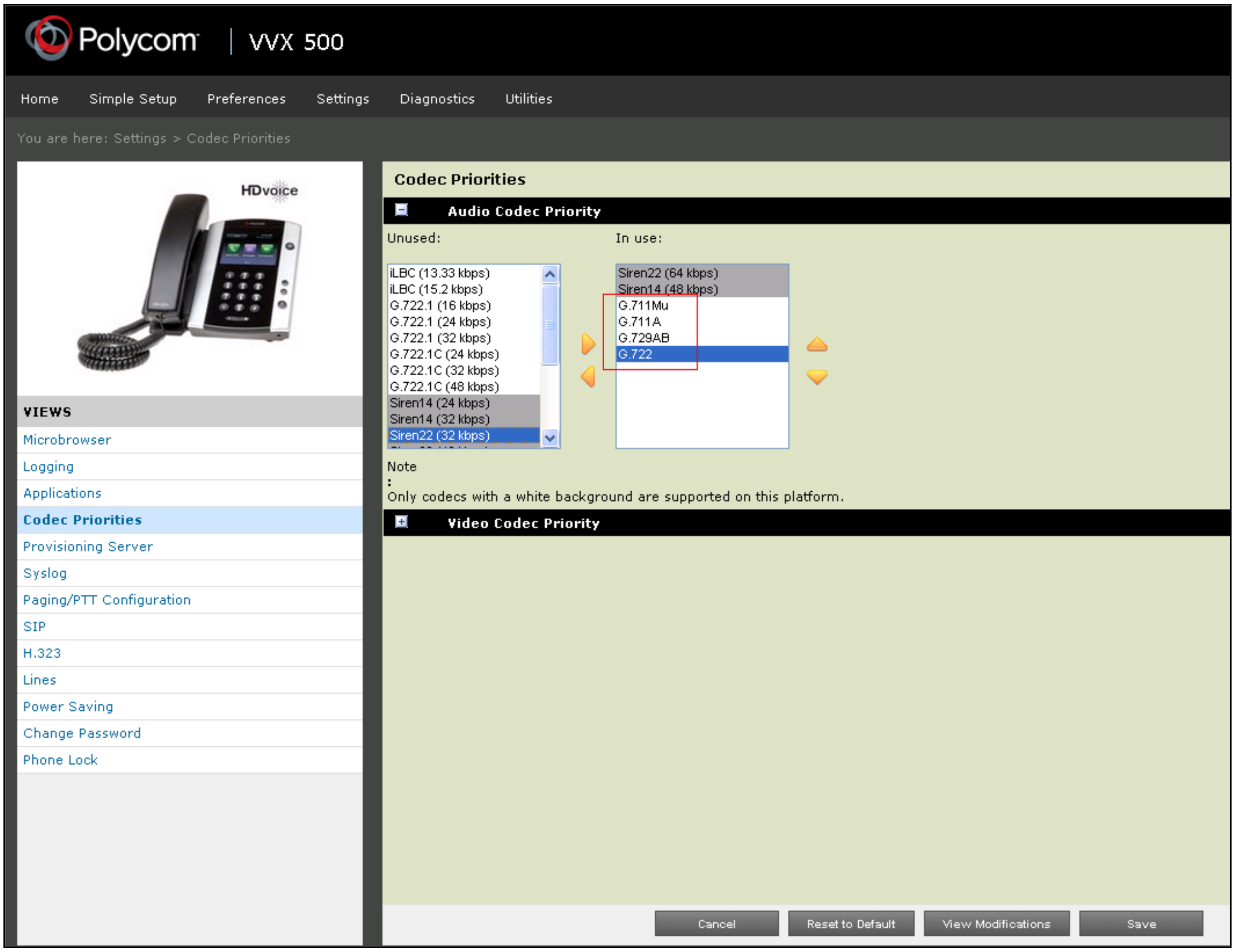
- Under the **Message Center** section, configure an appropriate Directory Number (DN) for the **Subscription Address** field. During compliance testing 53113 was the DN configured.
- Select **Registration** from the drop down menu for the **Callback Mode**

Click on **Save**.



7.5. Codec Settings

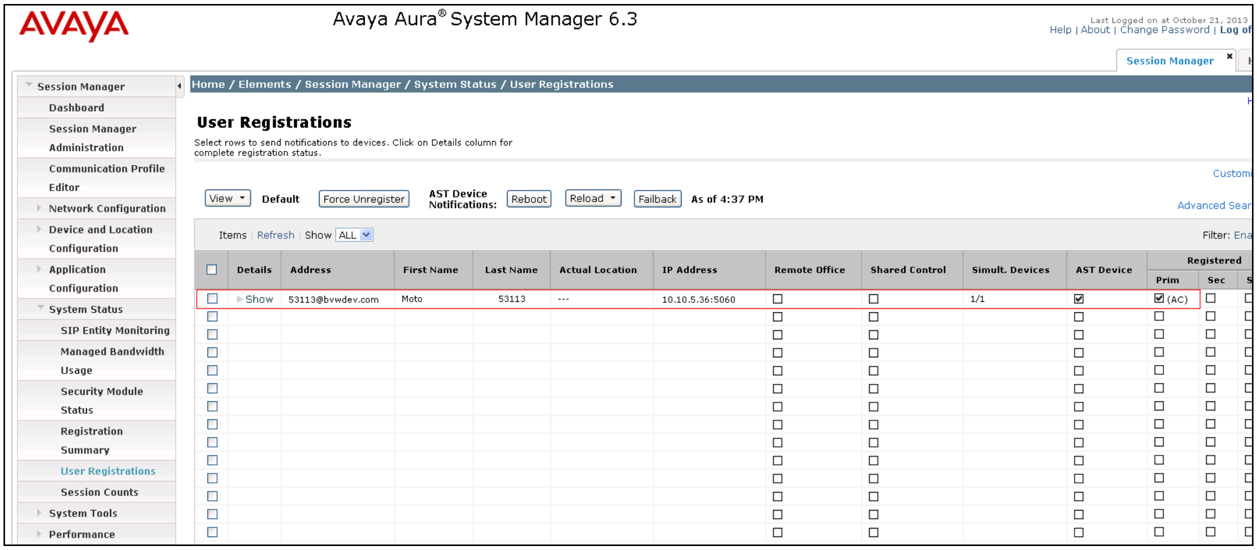
On the homepage of Polycom VVX 500/600, navigate to menu **Settings → Audio Codec Priority** (not shown). Select the codec list as shown below. Click **Save**.



8. Verification Steps

From the main screen of System Manager as shown in **Section 6.0**, select **Session Manager** (not shown).

From the Session Manager screen shown below, navigate to **System Status → User Registrations** to see a list of phones registered to the Session Manager.



From the physical phone display of VVX 500/600 navigate to **Menu → Settings → Status → Lines** (not shown). Verify that the Lines information shows the successful registration of the VVX 500/600 phone to the Session Manager.

Place a call from and to the VVX 500/600 and verify that the call is established with 2-way speech path. Verify basic telephony features by establishing calls between VVX 500/600 and phones on Communication Manager.

9. Conclusion

These Application Notes illustrate the procedures necessary for configuring the Polycom VVX 500/600 to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All feature functionality test cases described in **Section 2.1** were passed along with the observations noted in **Section 2.2**.

10. Additional References

Product documentation for the Avaya products may be found at:
https://support.avaya.com

Product documentation for the Polycom VVX family of phones may be found at:
http://support.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 VVX 500/600 Documents:
http://support.polycom.com/PolycomService/support/us/support/voice/index.html

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.