Avaya Solution & Interoperability Test Lab

# Application Notes for configuring SIP TLS between Vocera Communication System Release 4.3 SP1 and Avaya Communication Server 1000E Release 7.5 and Avaya Aura® Session Manager 6.1 – Issue 1.0


## Abstract

These Application Notes describe the procedures for configuring SIP TLS between Avaya Communication Server 1000E and Avaya Aura® Session Manager and between Vocera Communication System and Avaya Aura® Session Manager, the solution uses Avaya Aura® Session Manager to route calls between Avaya Communication Server 1000E and Vocera Communication System. The overall objective of the interoperability compliance testing is to verify basic functions of Vocera system is able to work with Avaya Communication Server 1000E over SIP Trunk that is secured by TLS.

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 procedures to integrate Vocera Communication system with Avaya Communication Sever 1000E via SIP TLS that was configured on Avaya Aura® Session Manager. The Avaya Communication Server 1000E that was used for the testing is co-resident system which has Call Server, Signaling Server and Element Manager applications residing on the same CPPM card. The solution has Avaya Aura® Session Manager to provide SIP TLS and networking routing service to route calls between Avaya Communication Server 1000E and Vocera Communication system.

# 2. General Test Approach and Test Results

The general test approach was to have different telephone types of the Avaya Communication Server 1000E (hereafter referred as Avaya CS1000) to place a call to and from the Vocera Server and follow its voice instructions to verify other features of the Vocera Communication System such as: Basic call, transfer, conference and call forward.

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:

- SIP TLS is established successfully between the Vocera Server and Avaya CS1000 via the Session Manager
- Basic calls between the Vocera Server and different telephone types of Avaya CS1000 (SIP, non-SIP and emulated PSTN telephones).
- DTMF RFC2833 transmission.
- Conference and Transfer calls from different telephone types of Avaya CS1000 (SIP, non-SIP and emulated PSTN telephones) to the Vocera Server clients (wireless badge B3000) and vice versa.
- Call Forward (All Call, No Answer, and Busy) and Call Forward to voicemail with Message Waiting Indication (MWI) notification.
- Other telephony features: Busy, Hold and Retrieve calls.

## 2.2. Test Results

All test cases were passed with the following one observation.

- Conference button on the Avaya CS1000 IP Unistim phone is not available if Ring Again feature is enabled on the phone that hosts the conference. The scenario happens when the second Vocera badge user is being invited to join the conference hosted by Avaya CS1000 IP Unistim phone that already has the first call with the first Vocera badge. The work around is not to provision the Ring Again feature on the IP phone. Work Item has been raised to track the issue to resolution.

- SIP OPTIONS message to keep alive sent from Vocera server to Session Manager is not recognized by Session Manager. The recommendation is disable SIP OPTIONS message on Vocera server and let Session Manager send OPTIONS to keep the SIP Trunk up.

## 2.3. Support

For technical support on the Vocera product, contact Vocera Support via phone, email or website.

- **Phone:** +1 408-882-5700
- **Email:** support@vocera.com
- **Web:** http://www.vocera.com

## 3. Reference Configuration

**Figure 1** illustrates a sample configuration with an Avaya Communication Server 1000E SIP trunk network that includes the following Avaya products: Avaya Communication Server 1000E connected to the Avaya Aura® Session Manager via SIP TLS. Avaya Communication Server 1000E SIP Line server provides SIP registration for SIP Phone. Vocera Communication System connected to Avaya Aura® Session Manager via SIP TLS. Avaya 1140E SIP phone registers to Avaya Communication Server 1000E SIP Line server, IP Unistim 1110 and 1165E Unistim phones register to Avaya Communication Server 1000E TPS application that resides on the same server with SIP Virtual trunk application and Emulated PSTN over PRI trunk.

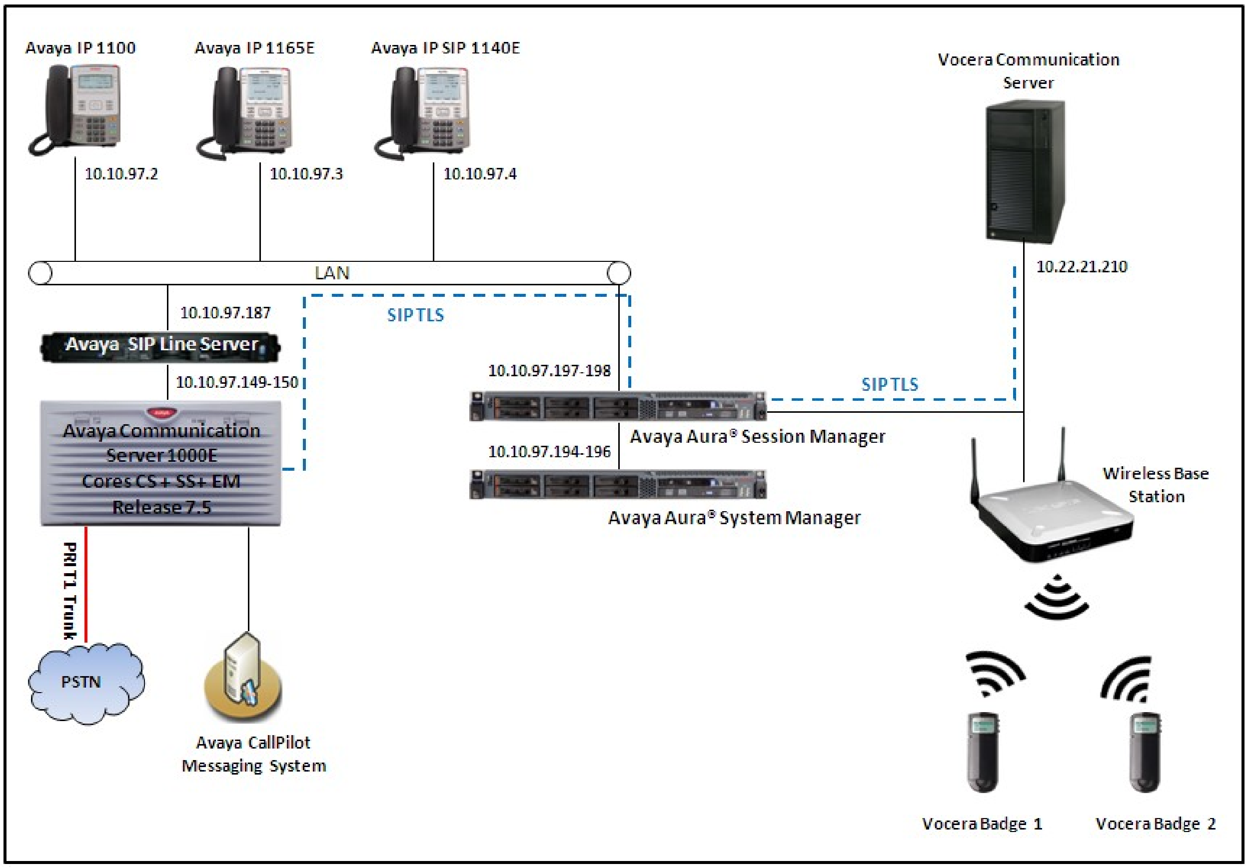



**Figure 1: Test Configuration Diagram**

# 4. Equipment and Software Validated

The following equipment and software were used for the compliance test:

| Equipment | Software |
|---|---|
| Avaya S8800 server running Avaya Aura® Session Manager Server | Avaya Aura® Session Manager 6.1 SP6 (Build No 6.1.6.0.616008) |
| Avaya S8800 server running Avaya Aura® System Manager Server | Avaya Aura® System Manager 6.1 SP6 (Build No: 6.1.0.0.7345-6.1.5.606 Software Update Revision No: 6.1.10.1.1774) |
| Avaya Communication Server 1000E/CPPM | Avaya Communication Server Release 7.5 Q+ Plus Deplist 1 (created: 2012-03-14) and Service Pack Linux (created 20120314) |
| Avaya CallPilot® 600i | Version 05.00.41.143 |
| Avaya IP Unistim Phone 1110 | Version 0623C8L |
| Avaya IP Unistim Phone 1165E | Version 0625C8L |
| Avaya IP SIP Phone 1140E | Version 4.03 |
| Vocera Server | Version 4.3 SP1 Build 2349 |

# 5. Configure Avaya Communication Server 1000E

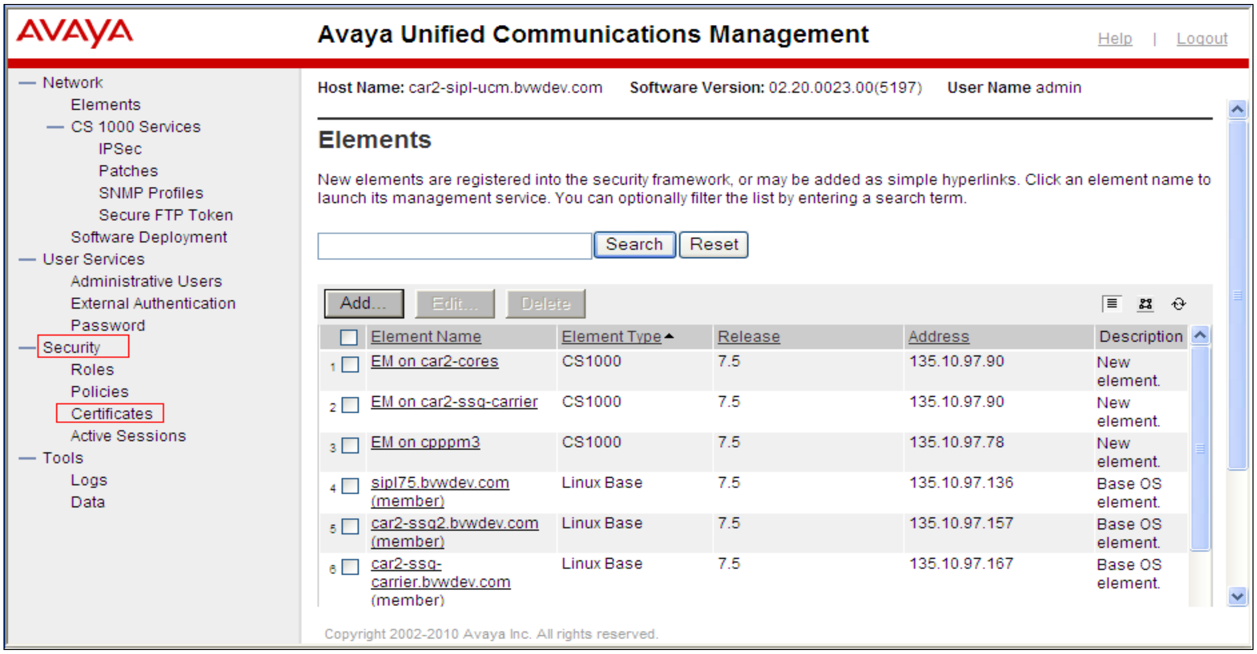
This document assumes that the Avaya Communication Server 1000E system that was used for the testing was already installed and configured for:

- Telephony Node ID ( Node 511)
- Customer ID ( customer 0)
- Zone
- D-channel for VoIP.
- SIP Route and Trunks
- Co-ordinate Dialing Plan (CDP) for local prefix of directory number in Avaya CS1000 starts from **54xxx** and prefix of CDP dialing plan for Vocera system is assigned to **732x** on the Avaya CS1000 system, this will be described in **Section 6.7**.

This section describes procedures of how to configure SIP Trunk secured by TLS between Avaya CS1000 SIP gateway and Session Manager.

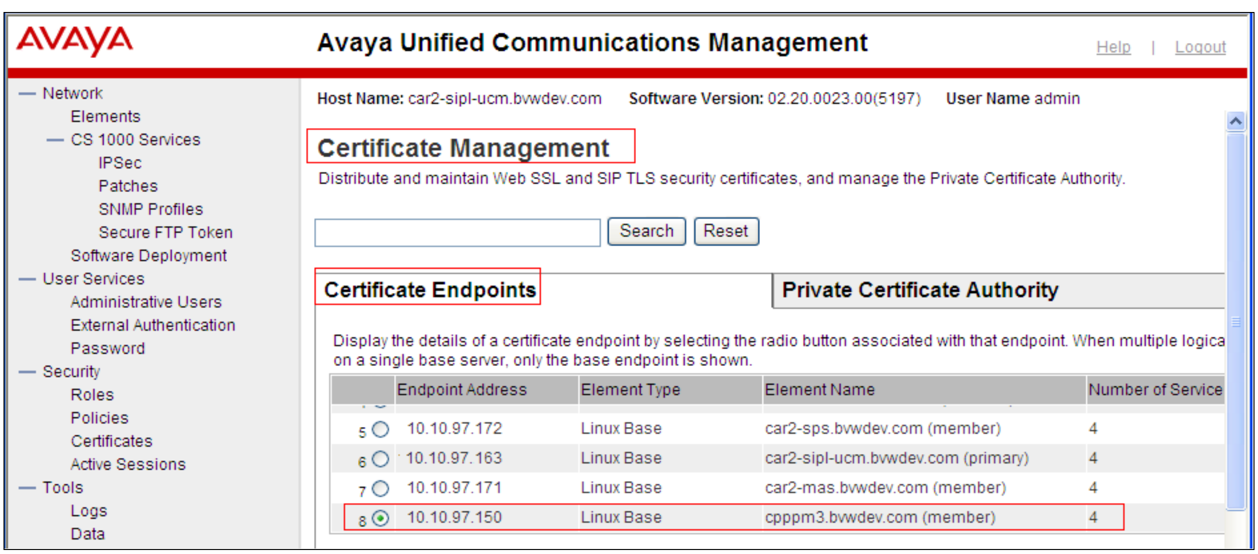
## 5.1. Create certificate For SIP TLS

Log in to Unified Communication Management (UCM) server that is managing the Avaya CS1000 system used to establish SIP TLS with Session Manager. The homepage of UCM is displayed as the screen below.

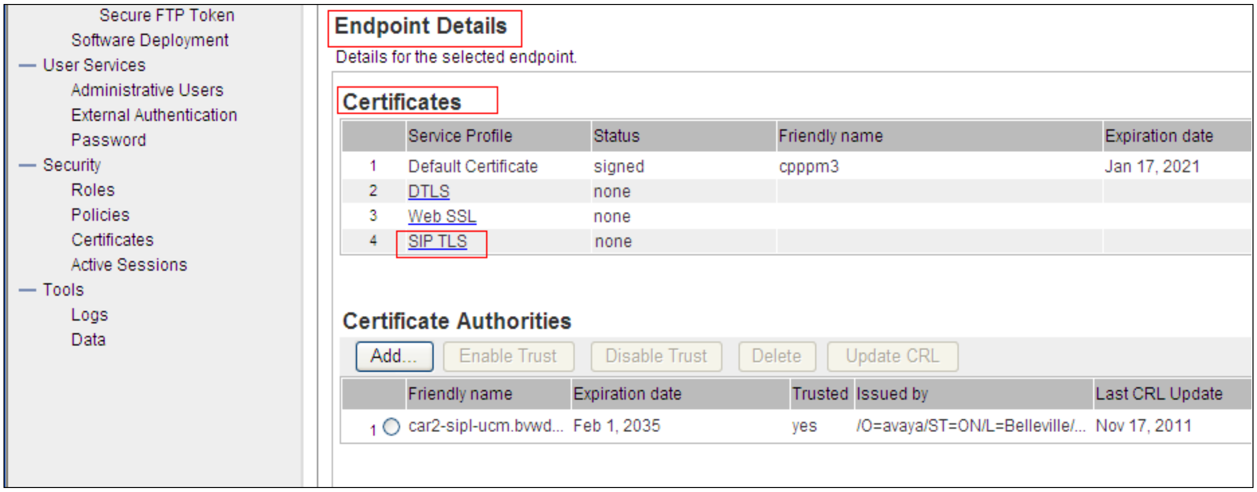



In the left navigation pane, select **Security → Certificates**, the **Certification Management** page is displayed in the right of UCM page.

Under **Certificate Endpoints** section, select an associated SIP gateway member that needs to be configured for the TLS certificate, in this document the SIP gateway member **10.10.97.150** is used to configure SIP TLS.

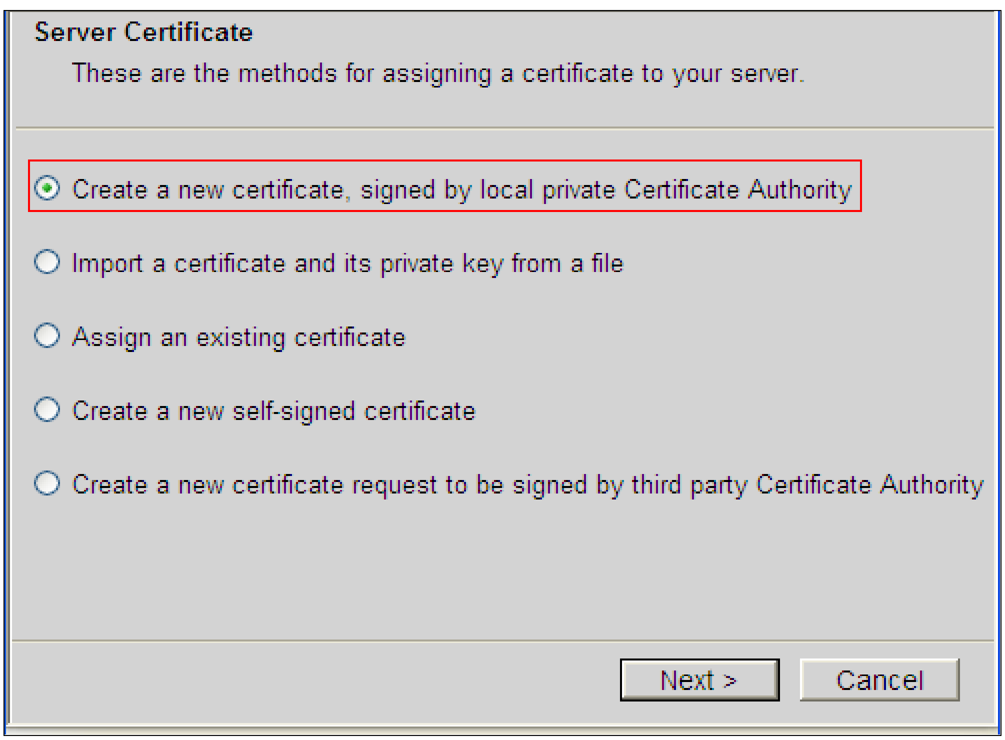


The **Endpoint Details** section of the SIP gateway **10.10.97.150** shows the current certificates installed in the **Certificates** window.

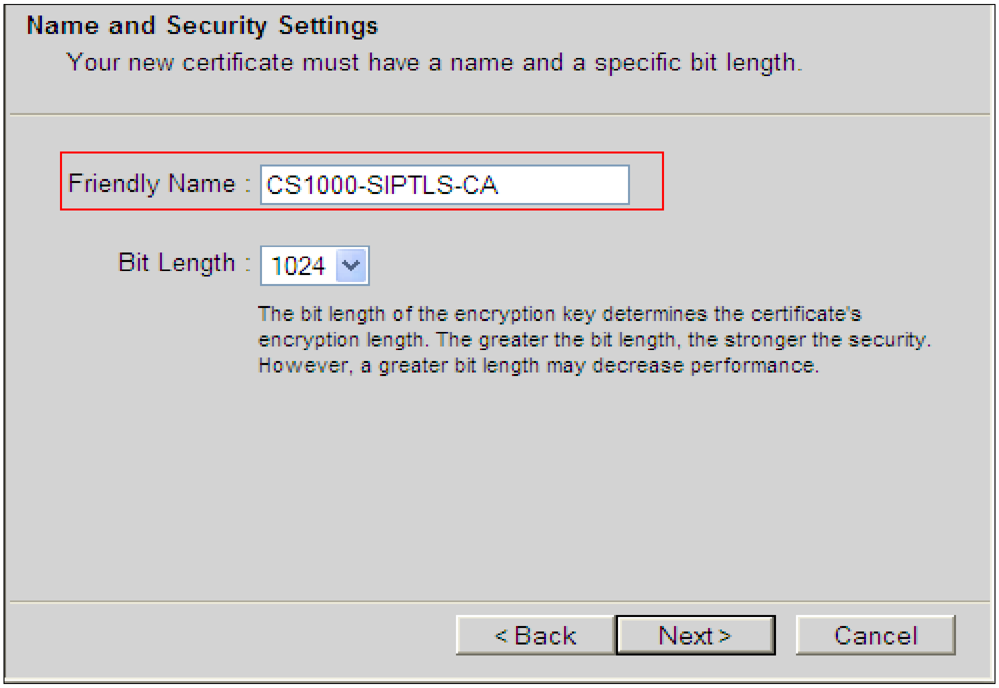



If in the **Status** column of **SIP TLS** service presents as **none**, this means the certificate for SIP TLS has not been configured for that SIP gateway. Click on the **SIP TLS** link to configure certificate for SIP TLS. Note: The certificate for SIP TLS will be created and signed by local UCM Certificate Authority (CA).

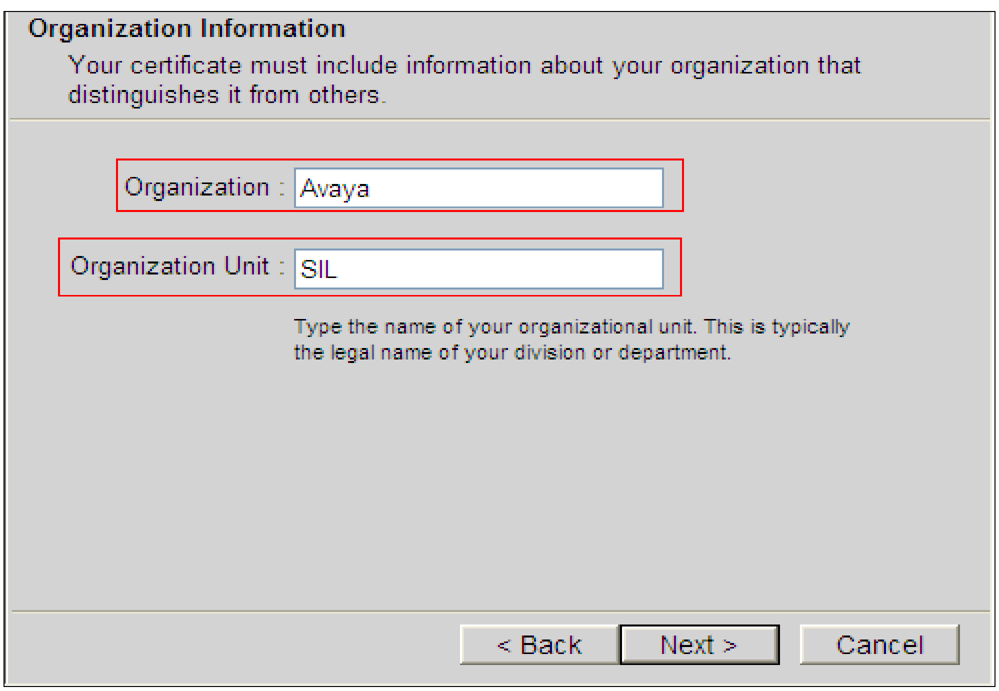
The **Server Certificate** window is displayed, select option "**Create a new certificate, signed by local private Certificate Authority**" and click on **Next** button to continue.



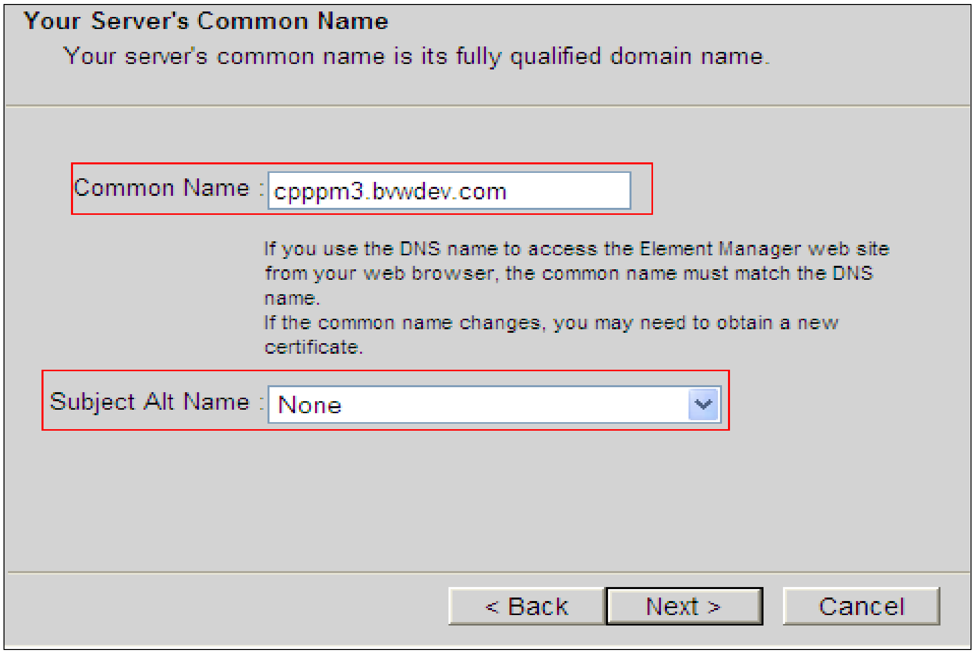
The **Name and Security Settings** page is displayed, enter a name in the **Friendly Name** field, e.g. "**CS1000-SIPTLS-CA**" and leave **Bit Length** field at default. Click on **Next** button to continue.



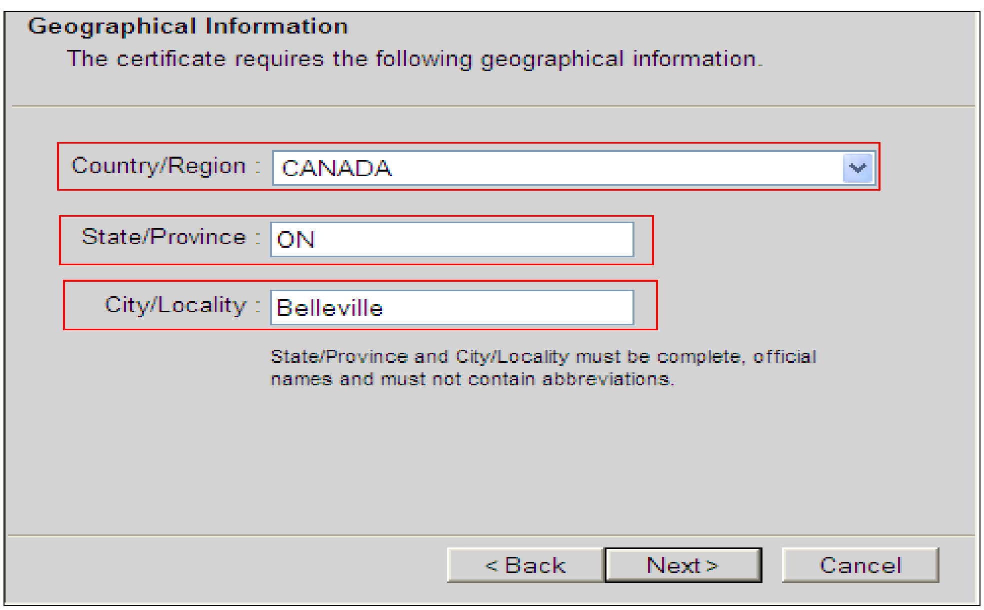

The **Organization Information** page is displayed, enter organization and organization unit names in the **Organization** and **Organization Unit** fields, e.g. "**Avaya**" and "**SIL**". Click **Next** button to continue.



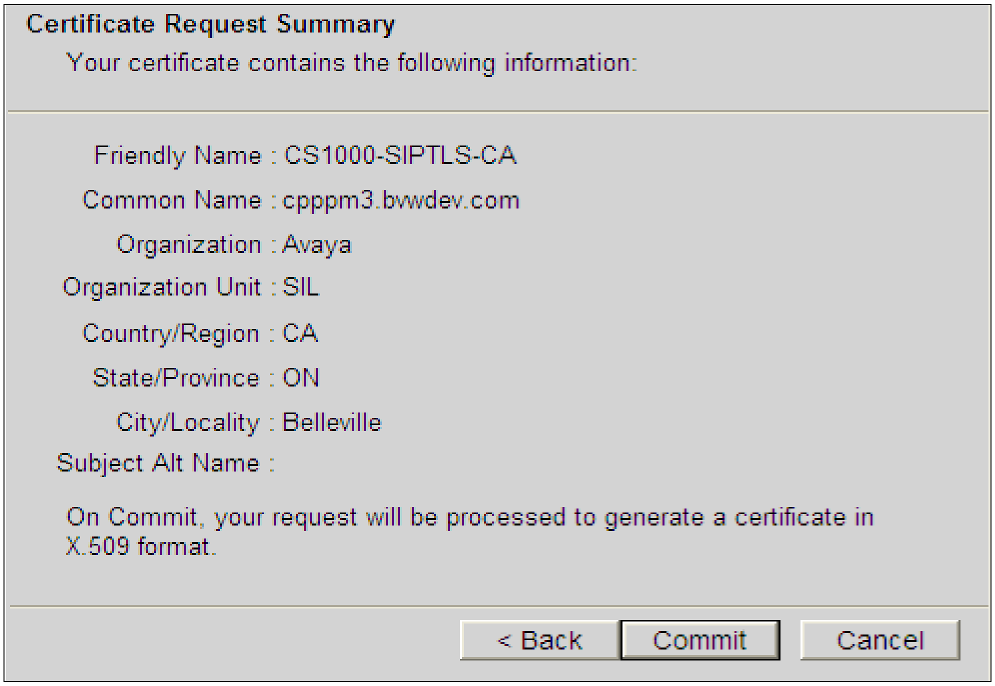
The **Your Server's Common Name** page is displayed, enter the FQDN of the SIP gateway in the **Common Name** field and leave the **Subject Alt Name** field at default. Note: The name in the **Common Name** field must be the FQDN or host name of the SIP gateway. Click **Next**.



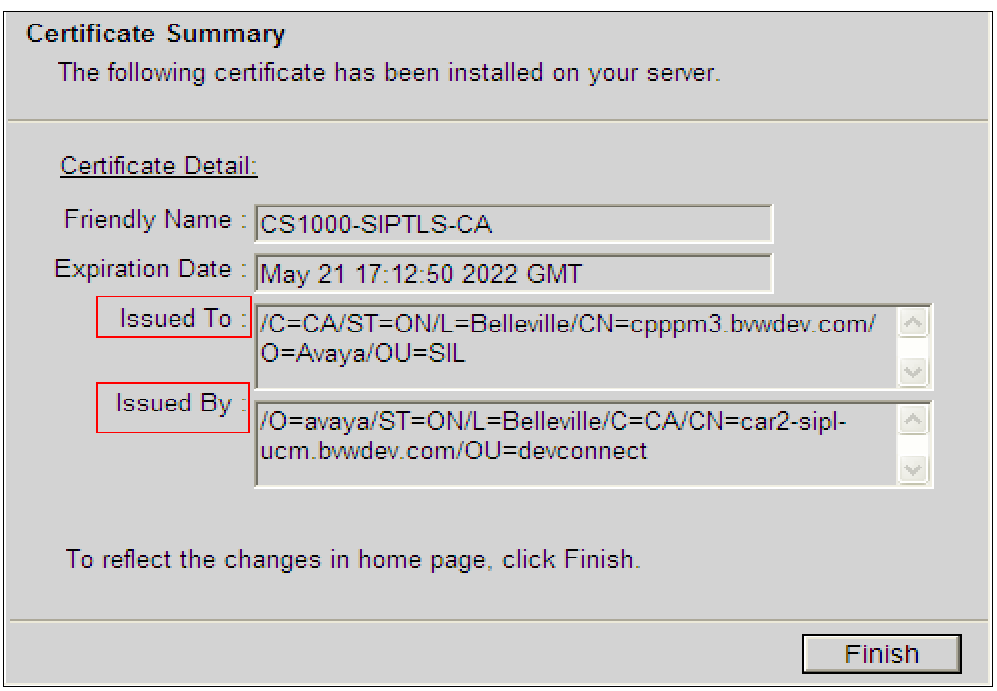

The **Geographical Information** page is displayed, enter country "**CANADA**", state "**ON**", and city "**Belleville**" where the server locates in the **Country/Regio**n, **State/Province**, and **City/locality** fields. Click **Next**.



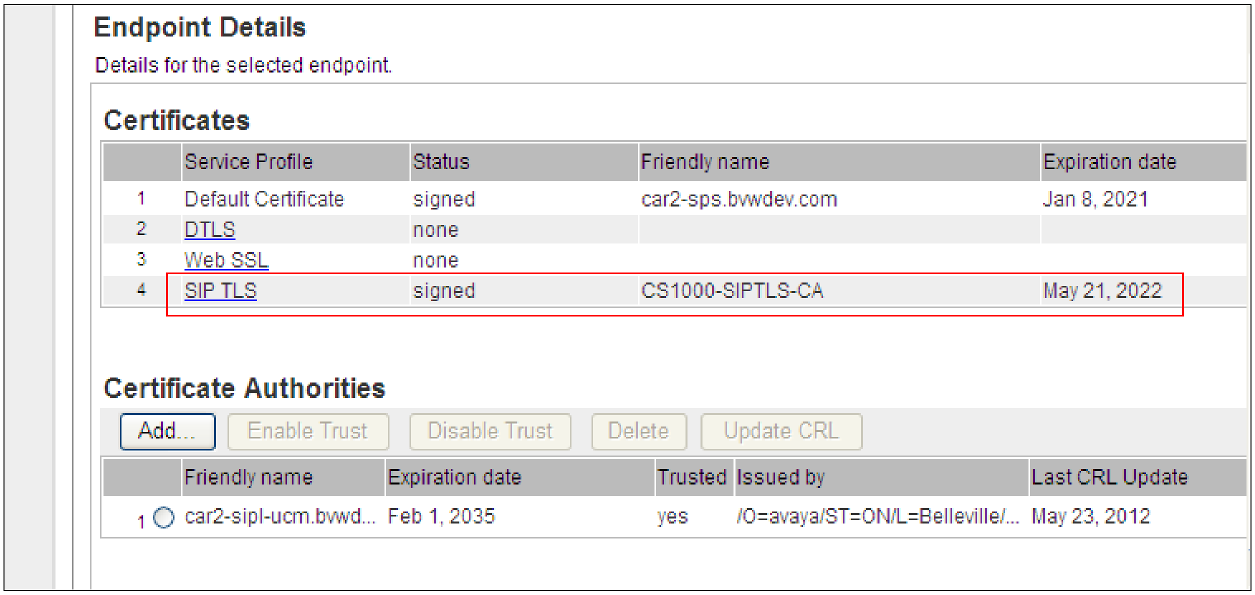
The **Certificate Request Summary** page is displayed, this display the summary of the certificate information that has been entered by the user in previous steps. Click **Commit** button to create the certificate.




The **Certificate Summary** page is displayed to indicate that the certificate is successfully created and signed by local private Certificate Authority, UCM. Click **Finish** button to complete and close the window.

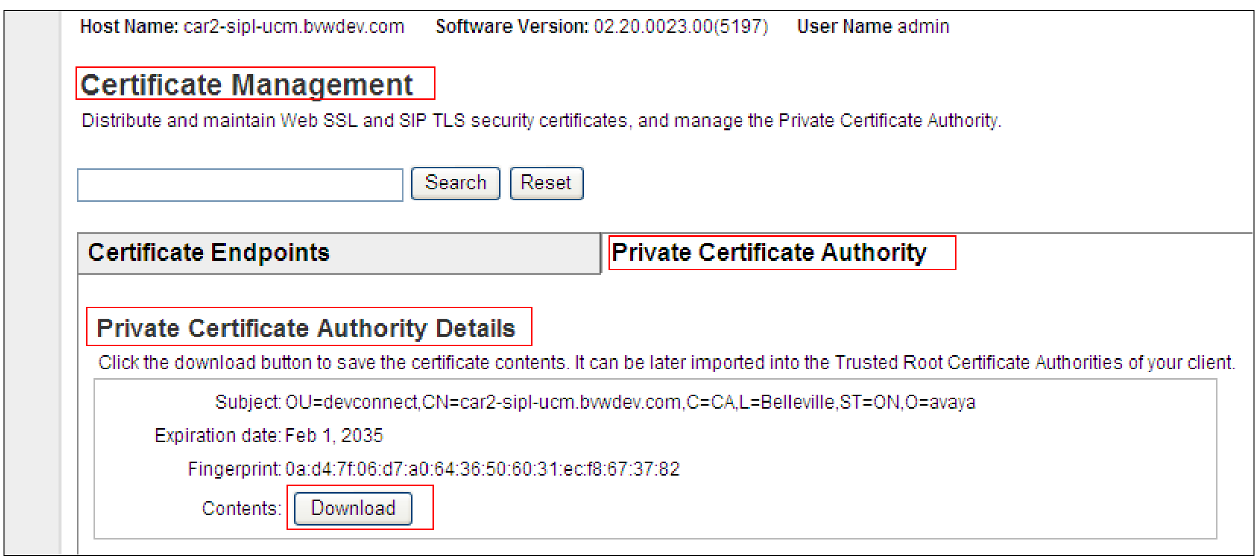


The screen below shows the new certificate for SIP TLS has been created and signed in the **Certificates** window of SIP gateway member **10.10.97.150**.

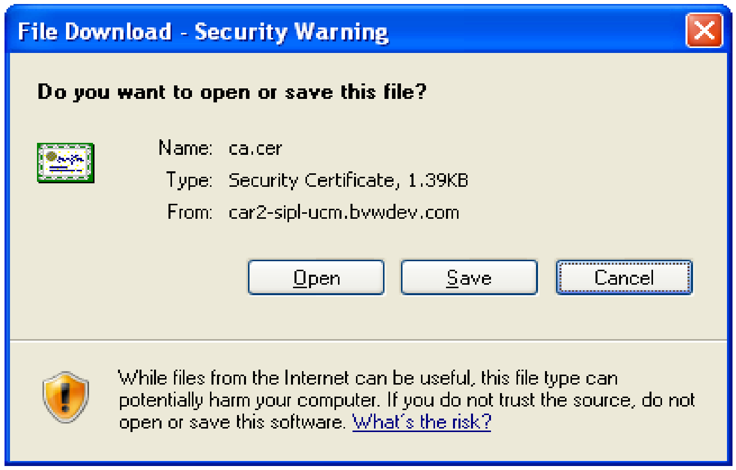



## 5.2. Download Local Private Certificate Authority of Avaya Communication Server 1000E

The certificate of local private Certificate Authority on UCM server which is used to sign for the SIP TLS certificate configured in **Section 5.**1 needs to be added to certificate store in Session Manager. This procedure shows how to obtain the CA. In the **Certificate Management** page, click on **Private Certificate Authority** tab and in the **Private Certificate Authority Details** window, click on **Download** button to download the CA certificate.



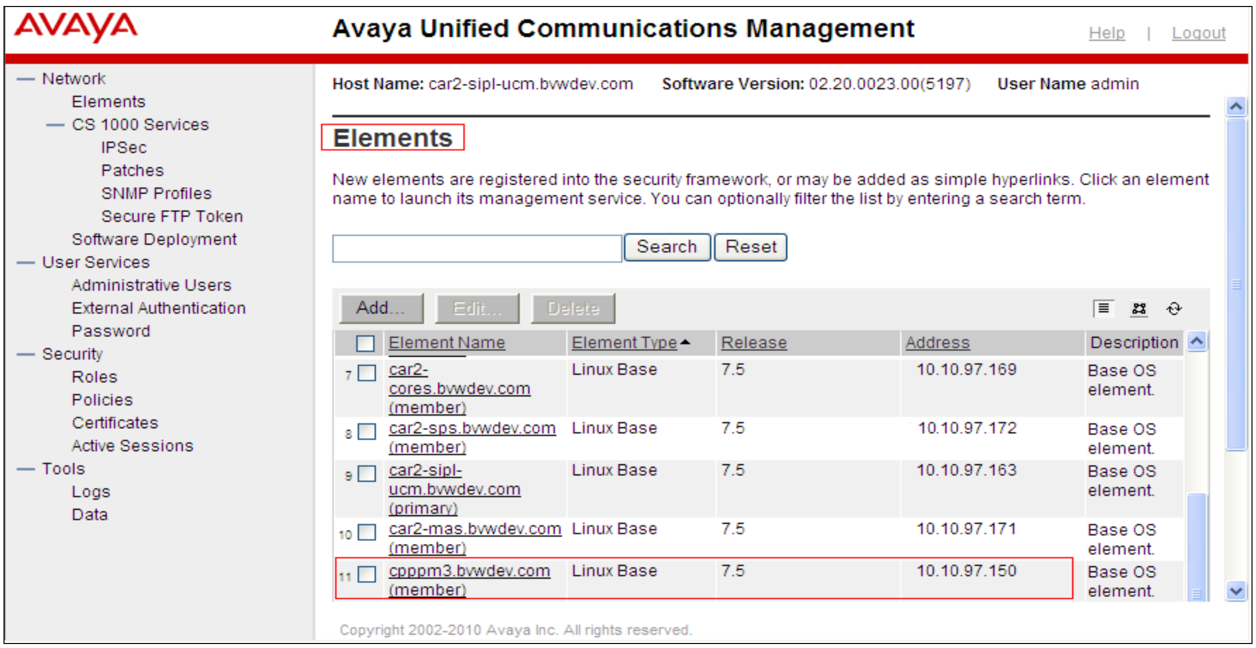
The popup window below is displayed; click **Save** button to save the CA certificate to local computer, this CA will be used to add to Session Manager in **Section 6.9**.



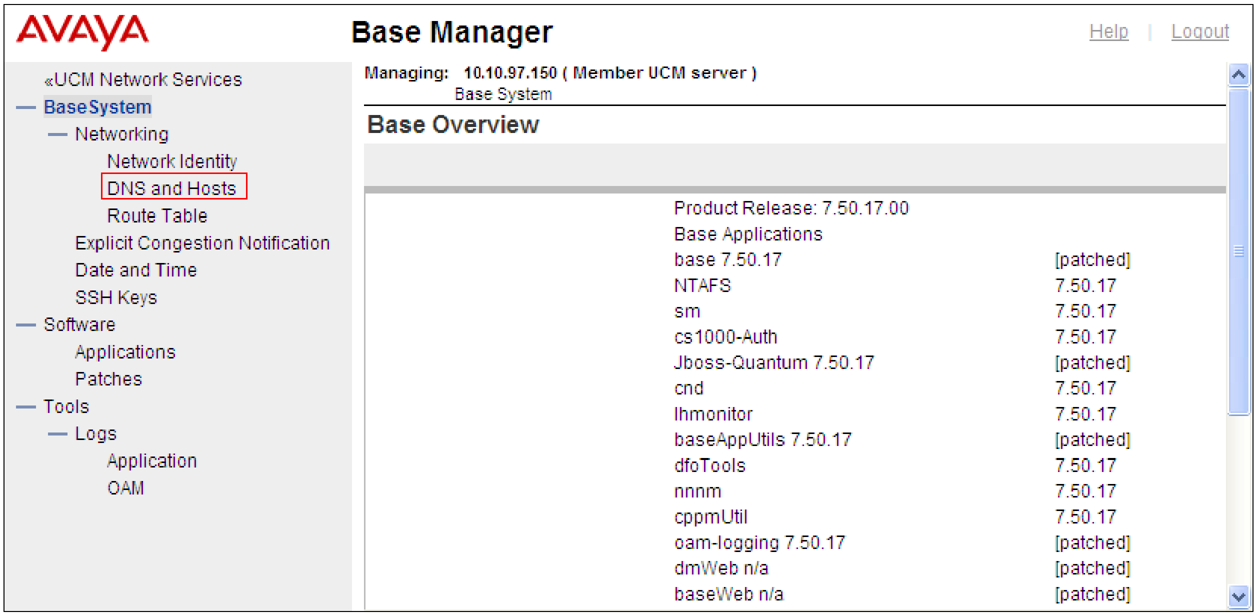
## 5.3. Add Common Name of Session Manager Certificate to Host File of SIP Gateway

When exchanging certificates between Avaya CS1000 SIP gateway and Session Manager to establish SIP TLS, Session Manager uses the default certificate that is shipped with the server and in the certificate of Session Manager, the **Common Name** is named as "**SM100**" and this name needs to be resolved to Session Manager signaling IP address in the host file of Avaya CS1000 SIP gateway.

In order to add an entry in the host file for SIP gateway server, log in to the UCM server, in the left navigation pane select **Elements**. The **Elements** page is displayed in the right, select Avaya CS1000 member server, in this case **cpppm3.bvwdev.com** with IP **10.10.97.150**.

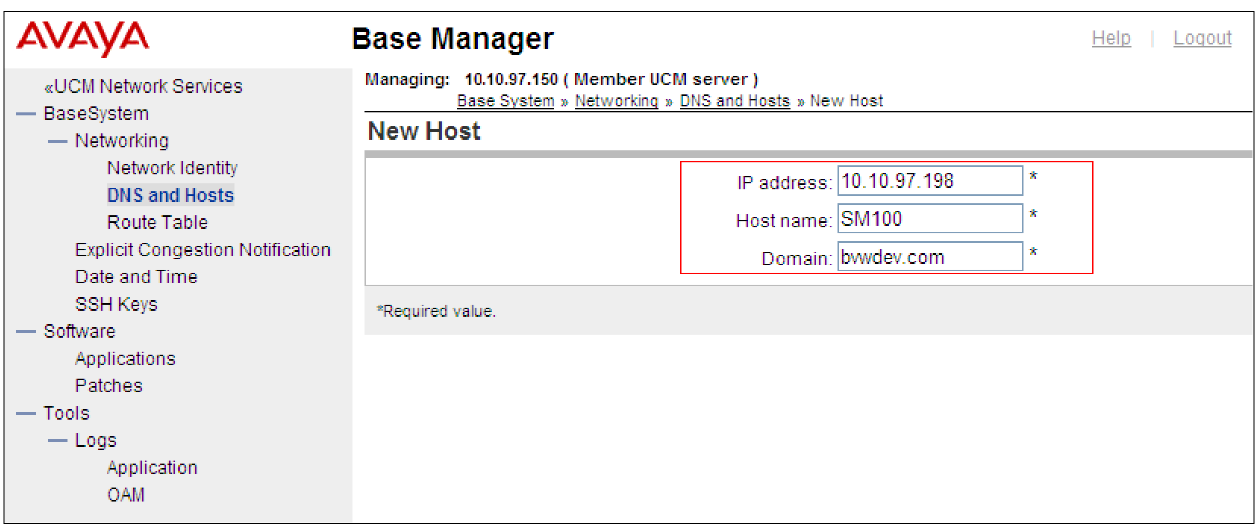


The **Base Manager** page of **cpppm.bvwdev.com** server is displayed.



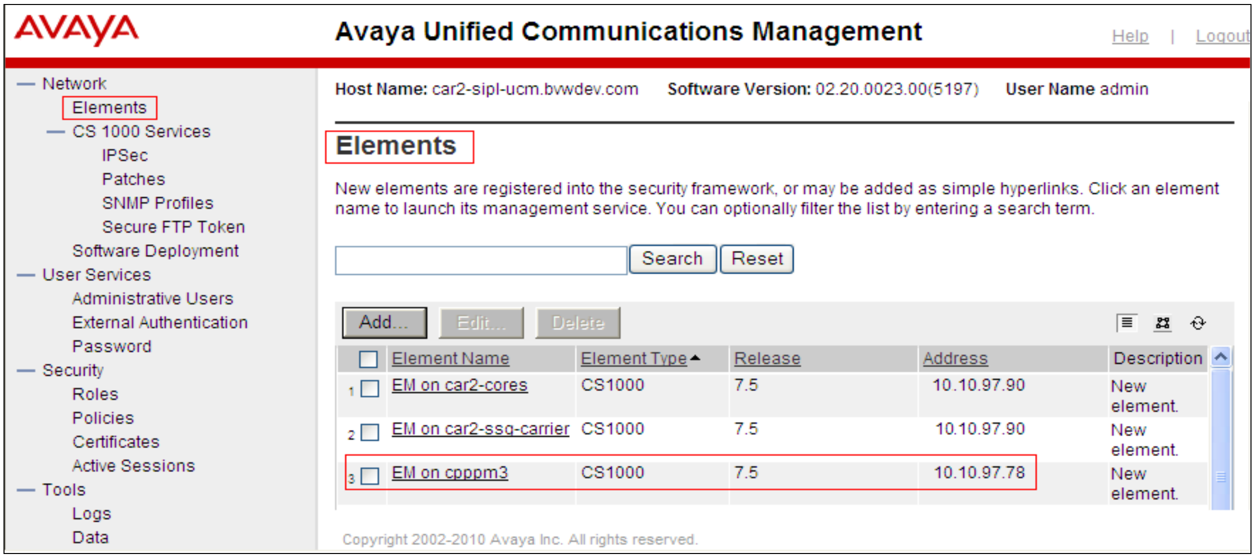

Click on **DNS and Hosts** tab in the left navigation pane in the screen above, **Domain Name Server (DNS)** page is displayed in the right (screen not shown), click on **Add** button under **Hosts** window in this page to add a new host (screen not shown).

The **New Host** page is displayed, enter Session Manager signaling IP **10.10.97.198** in the **IP Address** field, "**SM100**" in the **Host name** field and "**bvwdev.com**" in the **Domain** field. Click **Save** button to save changes.

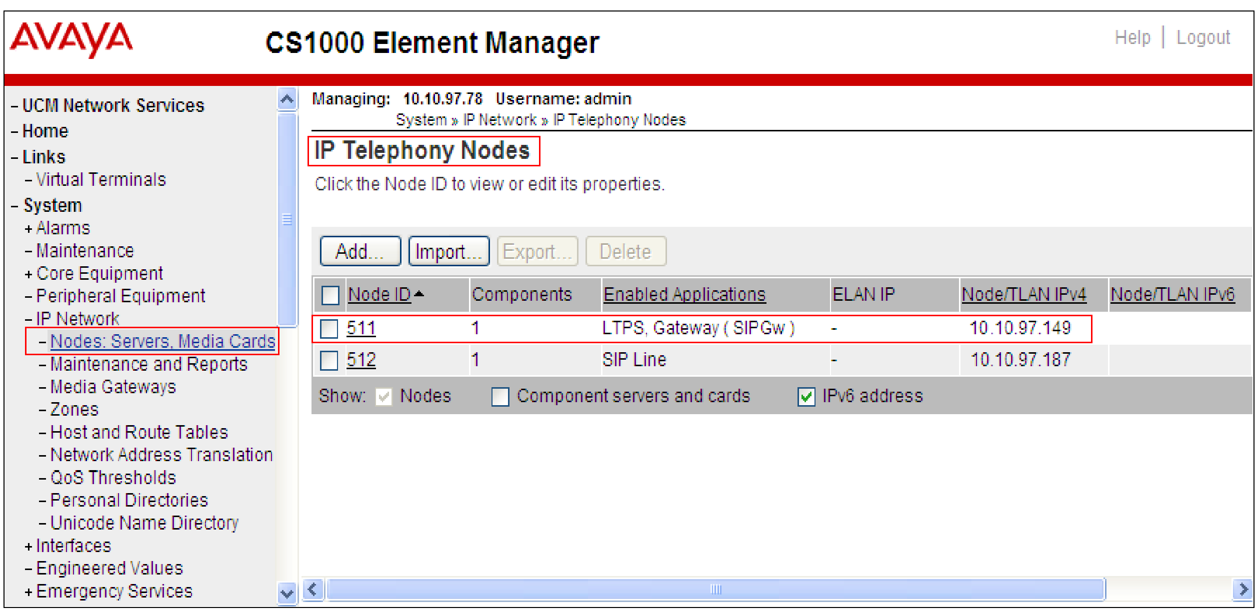


## 5.4. Configure Avaya Communication Server 1000E SIP Gateway using TLS

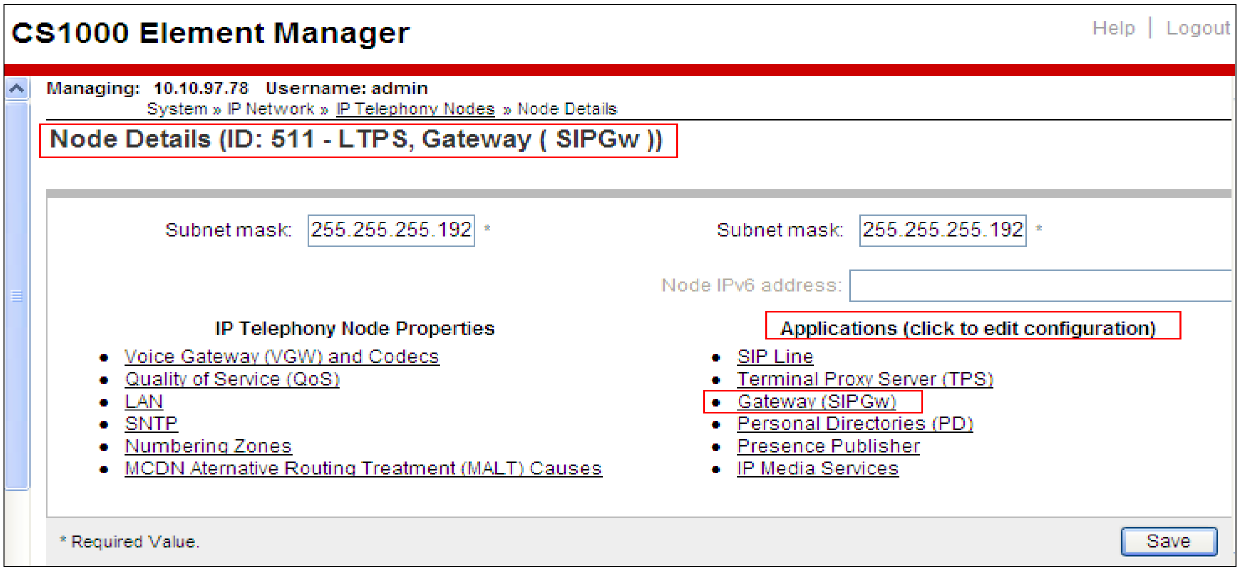
Access to Avaya CS1000 SIP Gateway Element Manager via UCM, from the homepage of UCM, navigate to **Network → Elements**, the Element page is displayed in the right, click **EM_on_cpppm3** which is the Element Manager of the Avaya CS1000 system that is to be administered.



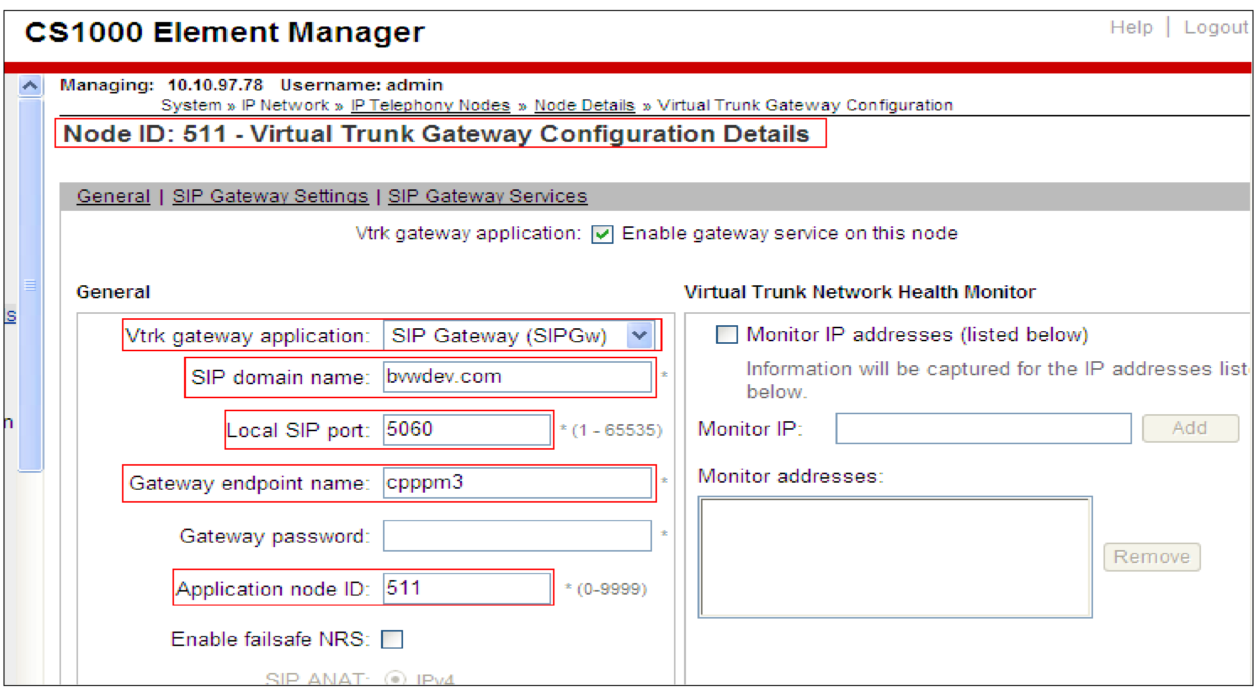

The CS1000 Element Manager page is displayed; in the left navigation pane select **System → IP Network → Nodes: Servers and Media Cards**, **IP Telephony Nodes** page is displayed in the right.



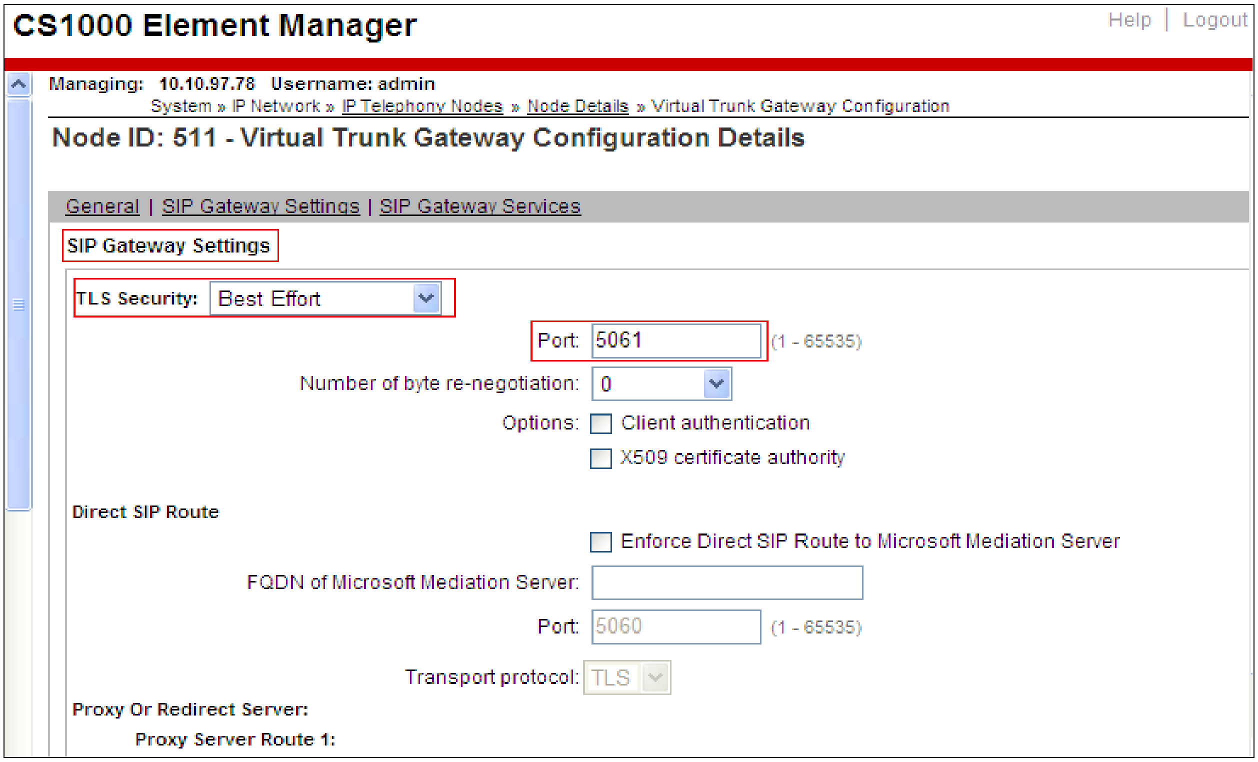
Select **Node 511** which has SIPGw and LTPS applications installed on this node in the **IP Telephony Nodes** page above. The **Node Details (ID: 511 - LTPS, Gateway ( SIPGw ))** section is displayed. Click on **Gateway (SIPGw)** application in the **Applications (click to edit configuration)** section to edit.



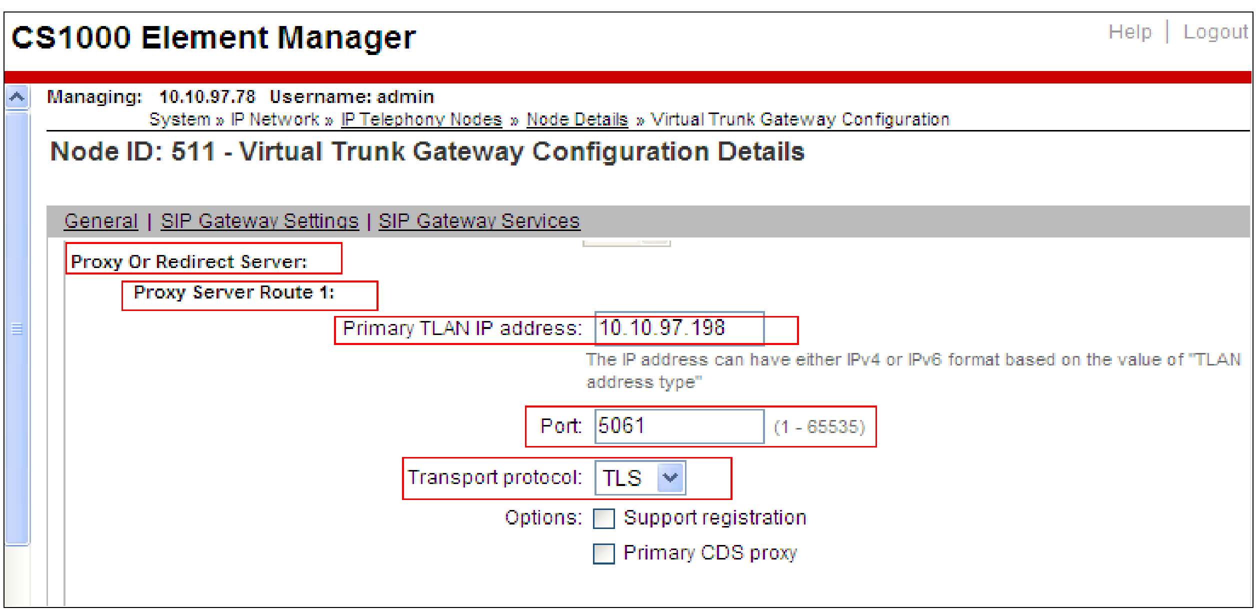

The **Node ID: 511 - Virtual Trunk Gateway Configuration Details** page is displayed, in the **General** section, select **SIP Gateway (SIPGw)** in **Vtrk gateway application** field, enter SIP domain "**bvwdev.com**" in **SIP domain name** field, this SIP domain will be defined in **Section 6.1**, **5060** in **Local SIP port** field, "**cpppm3**" in **Gateway endpoint name**, and **511** in **Application ID** field.



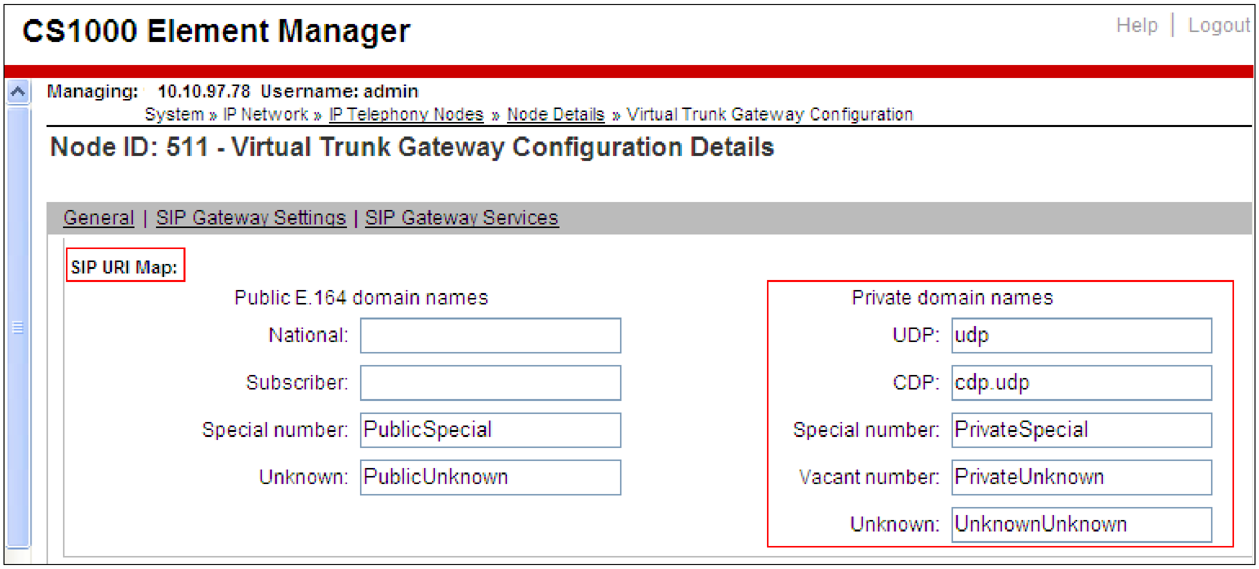
Continue to scroll down to **SIP Gateway Settings** section, select **Best Effort** in **TLS security** dropdown menu and enter port **5061** in **Port** field.




Scroll down to **Proxy Or Redirect Server** subsection of **SIP Gateway Settings** section, in the **Proxy Server Route 1**, enter signaling IP address of Session Manager **10.10.97.198** in **Primary TLAN IP address** field, **5061** in **Port** field, and select **TLS** from **Transport protocol** menu. Keep other values in the **Proxy Or Redirect Server** at default.

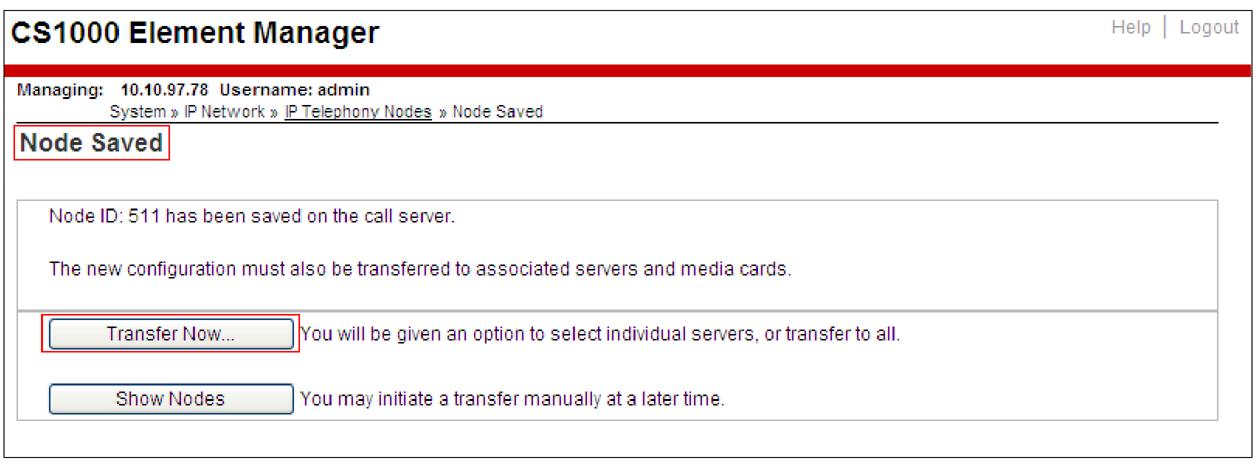


Scroll down to **SIP URI Map** subsection of the **SIP Gateway Settings** section, default values in **Private domain names** is displayed in the screen. If it is not same as displayed values, change it to default values.

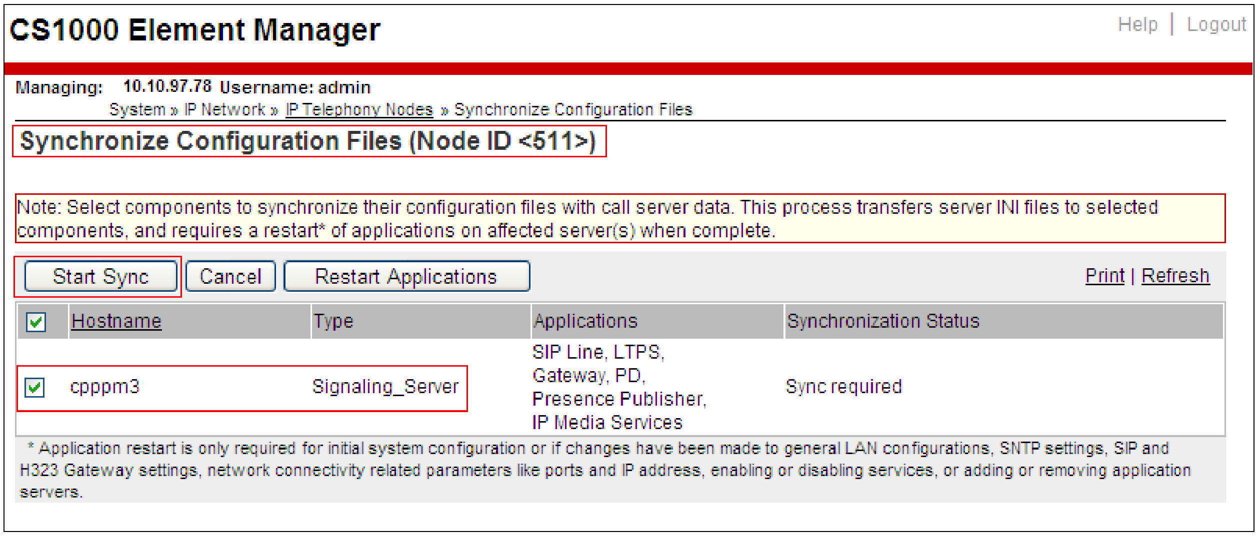



Scroll down to end of the **Node ID: 511 - Virtual Trunk Gateway Configuration Details** page, and select **Save** button to save changes in this page (screen not shown) and then click **Save** button in the **Node Details (ID: 511 - LTPS, Gateway ( SIPGw ))** page to save change in the **Node 511.**

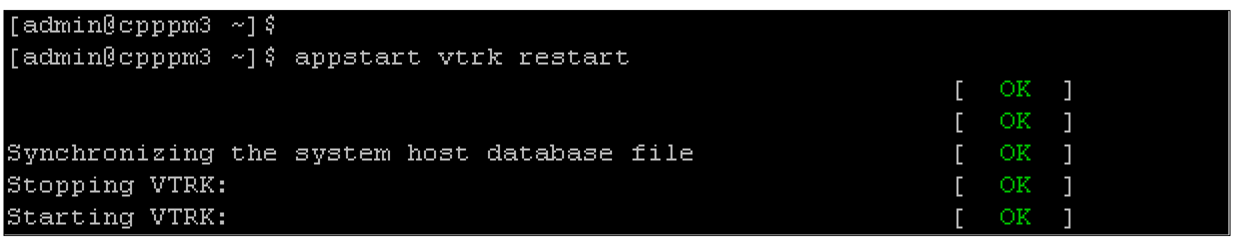
The **Node Saved** page is displayed, click on **Transfer Now** button to transfer changes to associated server, SIP Gateway **10.10.97.150**.



The changes also need to be synchronized with call server, **Synchronize Configuration Files (Node ID <511>)** page is displayed, select signaling **cpppm3** in **Hostname** column and then select **Start Sync** button to start synchronizing data between SIP Gateway and call server.




The SIP gateway needs to be restarted for configuration changes in the SIP gateway and for creating of new certificate of SIP TLS to take effect. In order to restart the SIP Gateway signaling, log in to Linux command line of your SIP gateway with administrator privileges and issue the command **appstart vtrk restart** as shown below:



# 6. Configure Avaya Aura® Session Manager

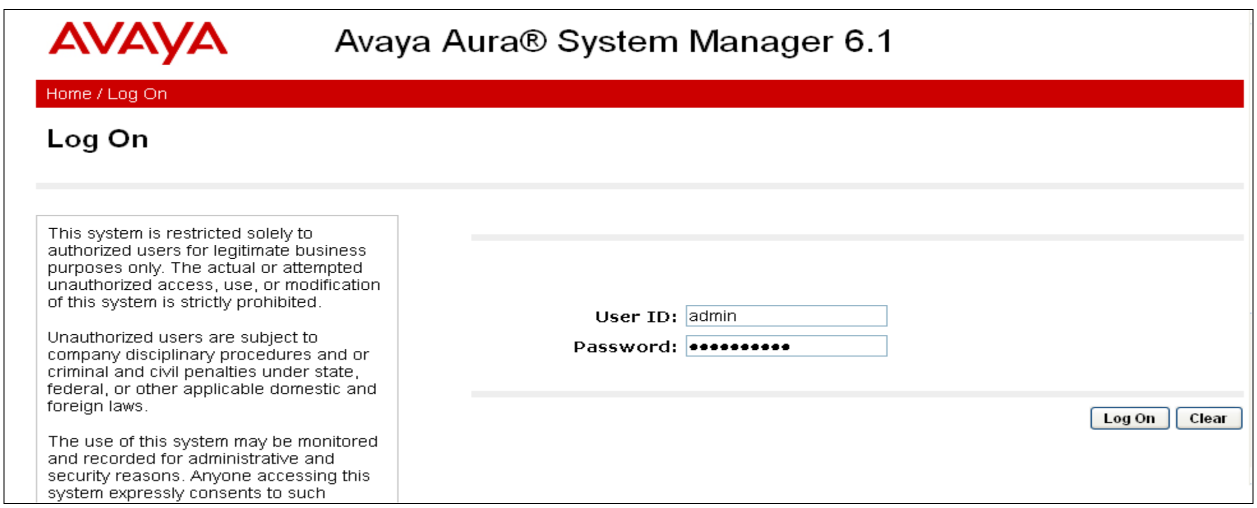
This section provides the procedures for configuring Session Manager. Session Manager is comprised of two functional components: The Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

This section assumes that Session Manager and System Manager have been installed, and network connectivity exists between the two platforms. The following steps describe the configuration needed for Session Manager.

- SIP Domains
- Locations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policy
- Dial Patterns
- Manage Element
- Configure SIP TLS certificate For CS1000 SIP Gateway and Vocera Server

## 6.1. Configure SIP Domain

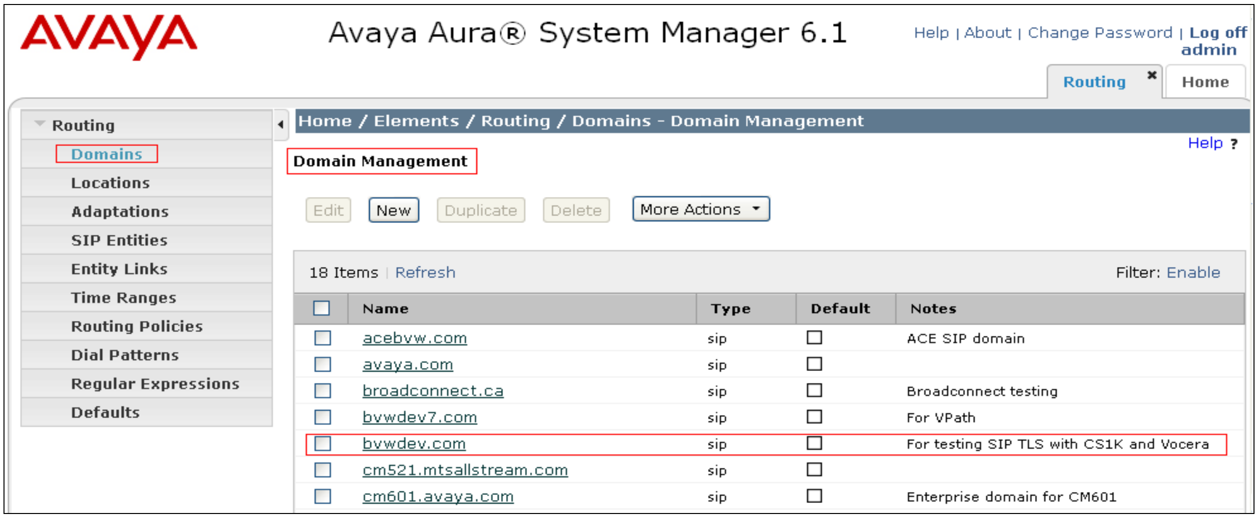
Launch a web browser, enter **https://<IP address of System Manager>** in the URL, and log in with the appropriate credentials.




Navigate to **Elements→Routing→Domains** and click on the **New** button to create a new SIP Domain (screen not shown). Enter the following values and use defaults for the remaining fields:

- **Name** –Enter the Authoritative Domain name specified in CS1000 SIP Gateway in **Section 5.4**, which is **bvwdev.com**.
- **Type** – Select **SIP**

Click **Commit** to save. The following screen shows the Domains page, listed is the newly created domain that was used during the compliance test.




## 6.2. Configure Locations

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

Navigate to **Routing→Locations**, and click on the **New** button to create a new SIP Entity location (screen not shown).

General section

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

- Enter a descriptive Location in the **Name** field (e.g. **Belleville**).
- Enter a description in the **Notes** field if desired.

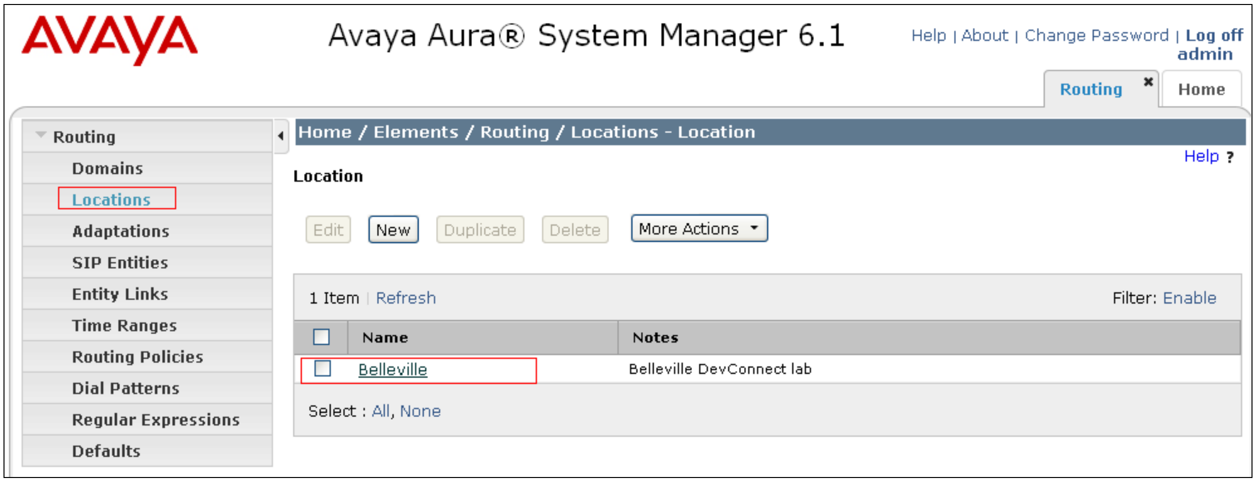
Location Pattern section

Click **Add** and enter the following values:

- The IP address information for the **IP address Pattern** (e.g. **10.10.97.***).
- A description in the **Notes** field if desired.

Repeat these steps in the Location Pattern section if the Location has multiple IP segments. Modify the remaining values on the form, if necessary; otherwise, use all the default values. Click on the **Commit** button.

Repeat all the steps for each new Location. The following screen shows the **Location** used during the compliance test.




## 6.3. Configure SIP Entities

A SIP Entity must be added for Session Manager and for each network component that has a SIP trunk. During the compliance test the following SIP Entities were configured:

- Session Manager
- Avaya CS1000 SIP Gateway
- Vocera Server

Navigate to **Routing →SIP Entities** and click on the **New** button to create a new SIP entity (screen not shown). Provide the following information:

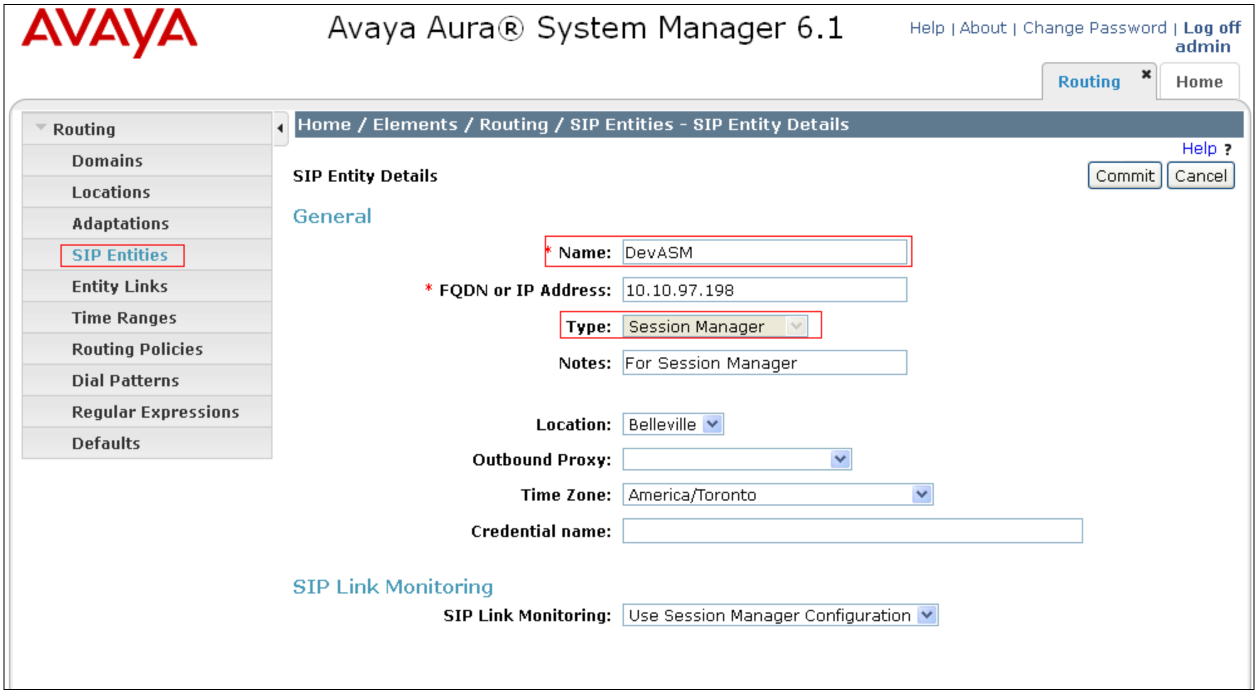
<u>General section</u>

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

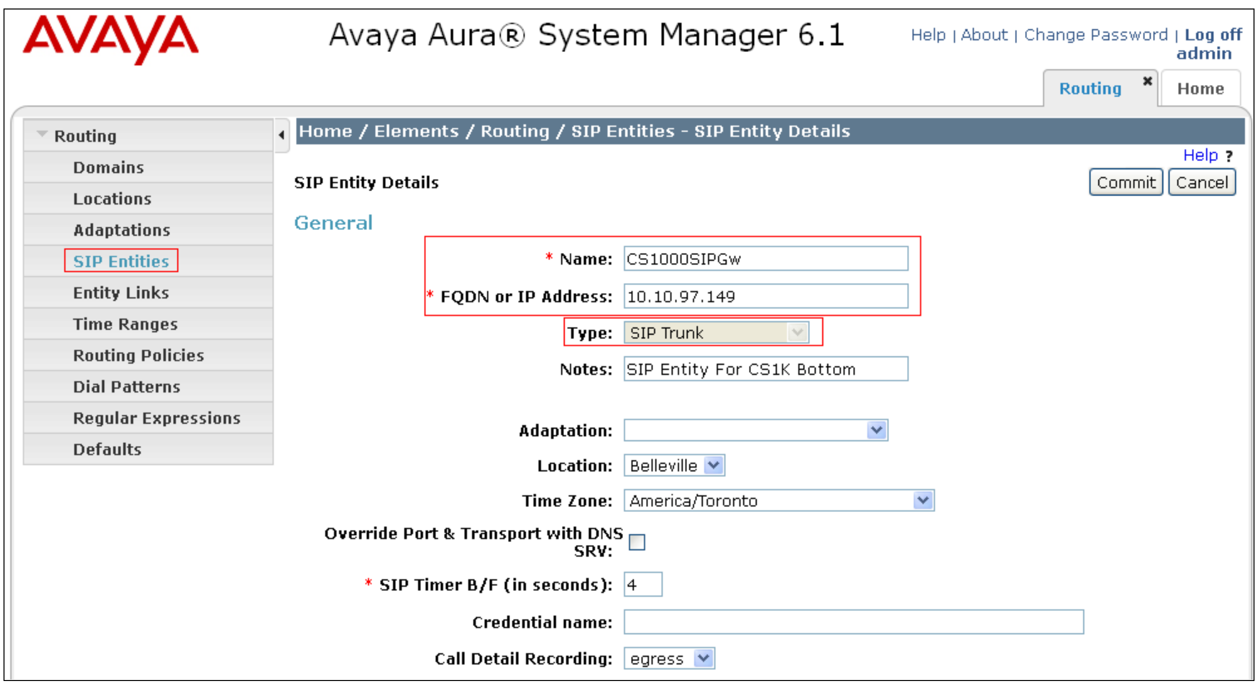
- **Name**: Enter a descriptive name.
- **FQDN or IP Address:** Enter the IP address of the signaling interface on each:
  - Avaya CS1000 SIP Gateway: 10.10.97.149
  - Signaling Session Manager: 10.10.97.198
  - Vocera server: 10.20.21.210
- From the **Type** drop down menu, select a type that best matches the SIP Entity:
  - For Avaya CS1000 SIP Gateway: select **SIP Trunk**
  - For Session Manager, select **Session Manager**
  - For Vocera Server, select **Other**
- Enter a description in the **Notes** field if desired.
- Select the appropriate time zone.
- Accept the other default values.

Click on the **Commit** button to save each SIP entity. Repeat all the steps for each new entity.

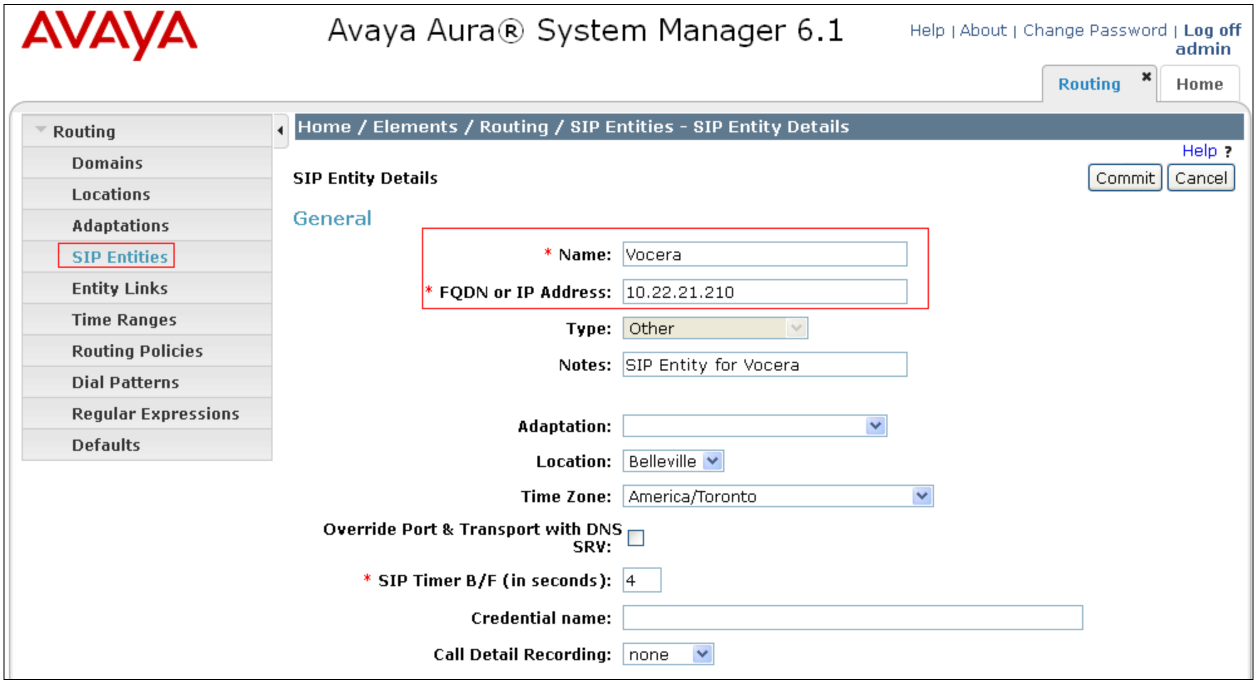
The screen below shows the detail of **Session Manger SIP Entity**.




The screen below shows the details of Avaya CS1000 SIP Entity.



The screen below shows the detail of **Vocera SIP Entity**.




## 6.4. Configure Entity Links

Entity Links define the connections between the SIP Entities (in this case, Avaya CS1000 SIP gateway and Vocera server) and Session Manager. In the compliance test, the following entity links are defined from Session Manager.

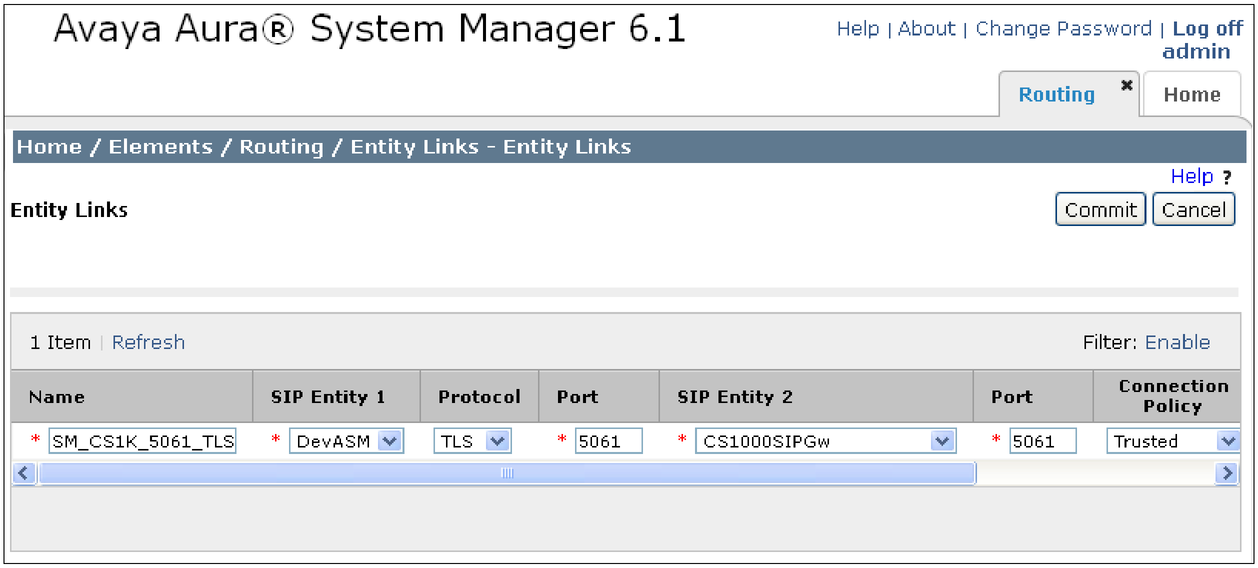
- Session Manager ⇔ Avaya CS1000 SIP Gateway
- Session Manager ⇔ Vocera Server

Navigate to **Routing →Entity Links** and click on the **New** button to create a new entity link (screen not shown). Provide the following information:

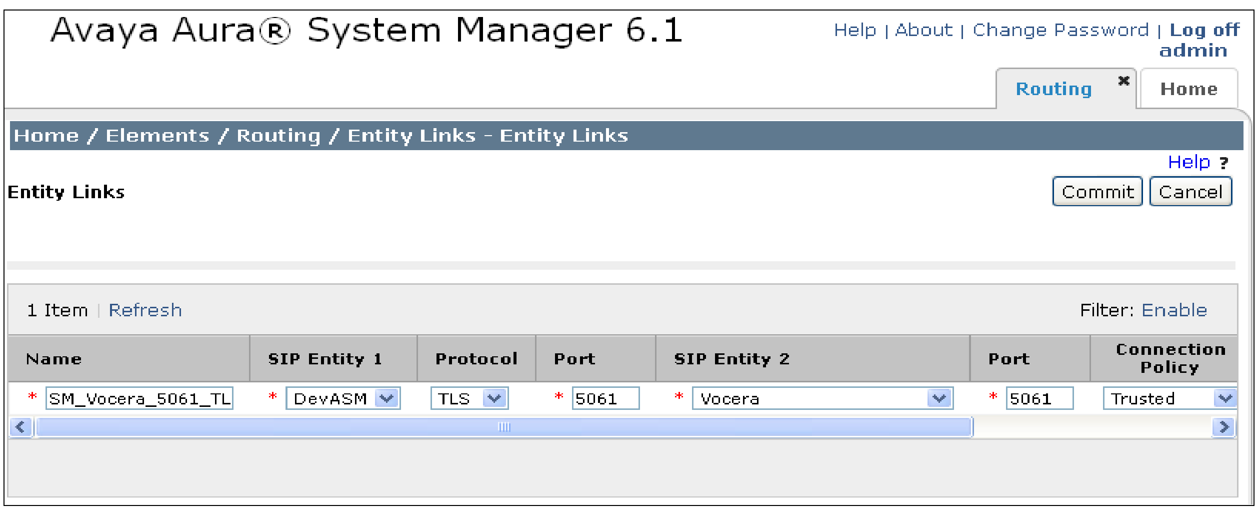
- **Name**: Enter a descriptive name.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity created in **Section** Error! Reference source not found. (e.g. **DevASM**).
- In the **Protocol** drop down menu, select the TLS protocol.
- In the **Port** field, enter the port to be used (e.g. **5061**).
- In the **SIP Entity 2** drop down menu, select **CS1000SIPGw** for the entity link between Session Manager and Avaya CS1000 SIP gateway and select **Vocera** for the Vocera entity.
- In the **Port** field, enter the port to be used (e.g. **5061**).
- Check the **Trusted** box.
- Enter a description in the **Notes** field if desired.

Click on the **Commit** button to save each Entity Link definition. Repeat all the steps for each new SIP Entity Link.

The newly created entity link between Session Manager and Avaya CS1000 SIP Gateway is shown below in the screen shot.




The newly created entity link between Session Manager and Vocera server is shown below in the screen shot.



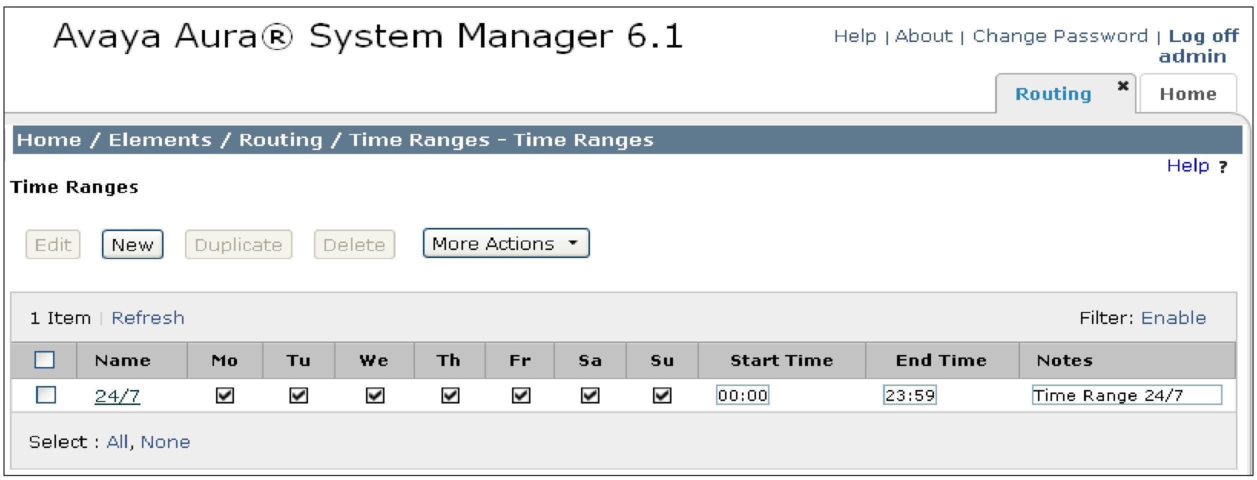
## 6.5. Time Ranges

Time Ranges define admission control criteria to be specified for Routing Policies (**Section** Error! Reference source not found.). In the reference configuration, no restrictions were used.

To add a Time Range, navigate to **Routing→Time Ranges**, and click on the **New** button (screen not shown). Provide the following information:

- Enter a descriptive name in the **Name** field (e.g. **24/7**).
- Check each day of the week.
- In the **Start Time** field, enter **00:00**.
- In the **End Time** field, enter **23:59**.
- Enter a description in the **Notes** field if desired.

Click the **Commit** button. The following screen shows the Time Range page used during the compliance test.




## 6.6. Configure Routing Policy

Routing Policies associate destination SIP Entities (**Section** Error! Reference source not found.) with Time of Day admission control parameters (**Section** Error! Reference source not found.) and Dial Patterns (**Section** Error! Reference source not found.). In the reference configuration, Routing Policies are defined for:

- Inbound calls to Avaya CS1000 SIP gateway.
- Inbound calls to Vocera server.

To add a Routing Policy, navigate to **Routing →Routing Policies** and click on the **New** button on the right (screen not shown). Provide the following information:

<u>General section</u>

- Enter a descriptive name in the **Name** field (e.g. **To_CS1K75_Bottom**, **To_Vocera**).
- Enter a description in the **Notes** field if desired.

SIP Entity as Destination section

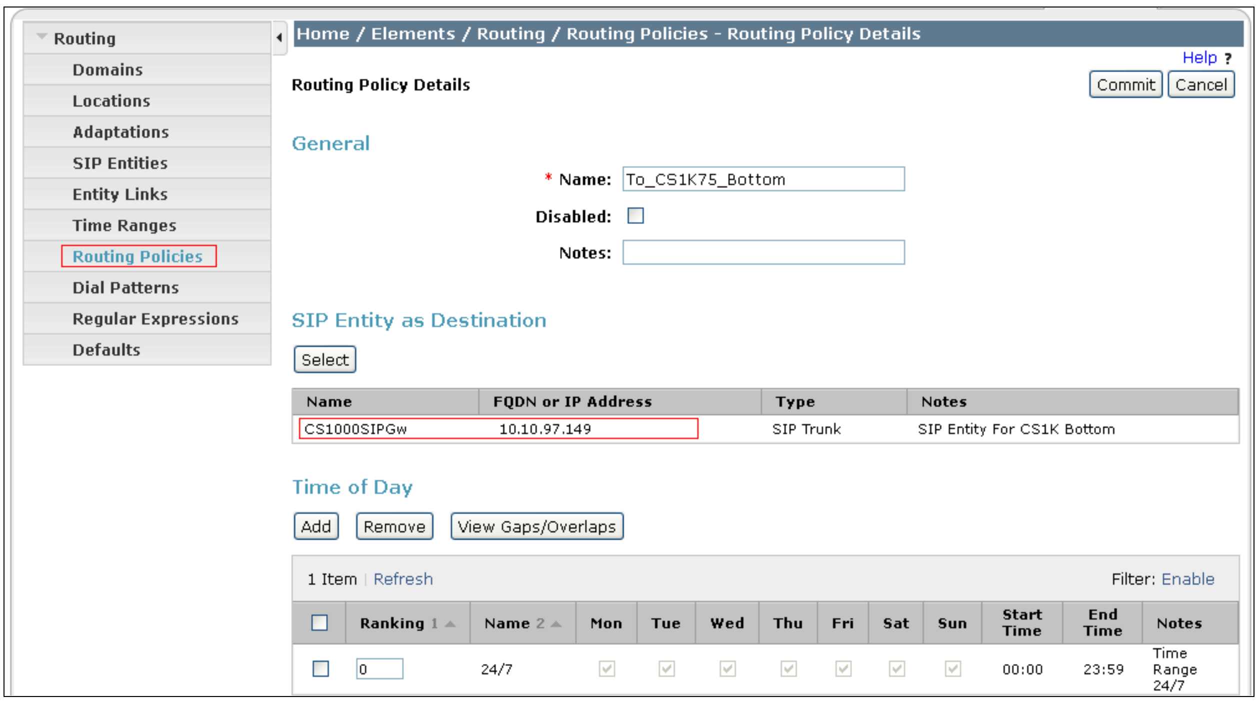
- Click the **Select** button.
- Select a SIP Entity that will be the destination for this call.
- Click the **Select** button and return to the Routing Policy Details form.

Time of Day section

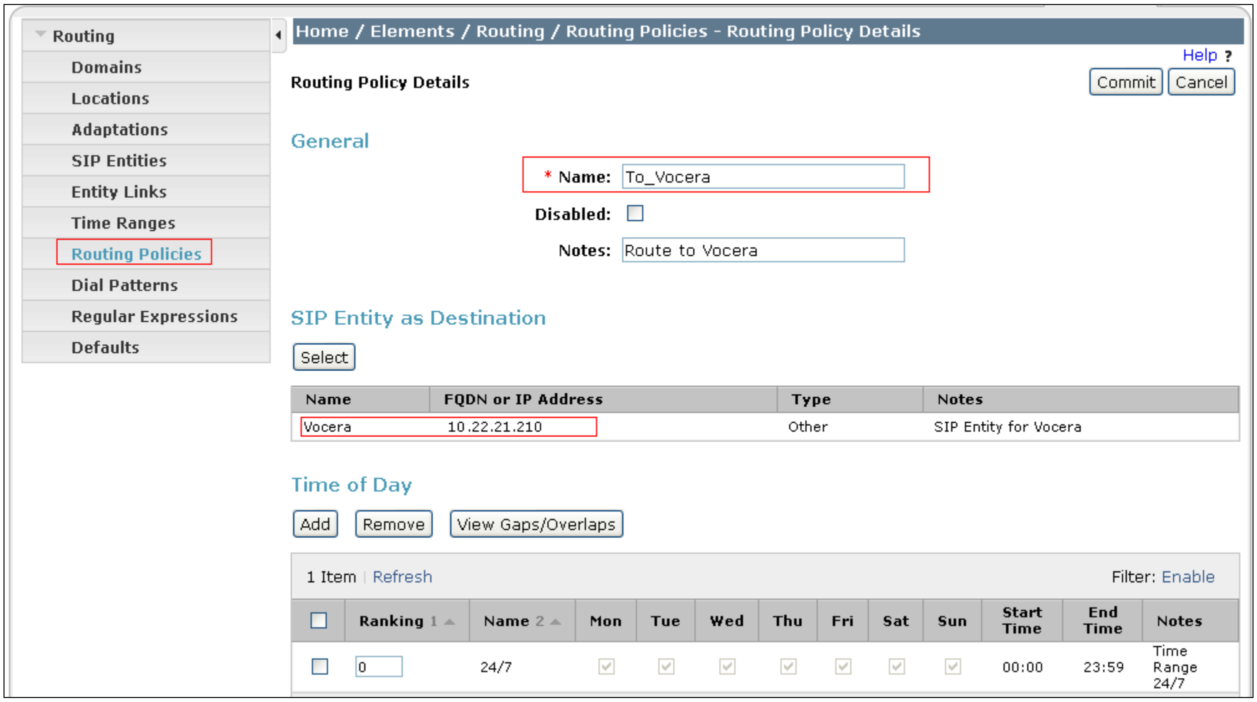
- Leave default values.

Click **Commit** to save Routing Policy definition. Repeat the steps for each new Routing Policy.

The following screen shows the Routing Policy used for Avaya CS1000 during the compliance test.



The following screen shows the Routing Policy used for Vocera during the compliance test




## 6.7. Configure Dial Patterns

Dial Patterns define digit strings to be matched for inbound and outbound calls. In the compliance test, the following dial patterns are defined from Session Manager.

- 54xxx – dial pattern used to route calls to Avaya CS1000.
- 732x – dial pattern used to route to Vocera.

To add a Dial Pattern, select **Routing →Dial Patterns** and click on the **New** button (screen not shown) on the right pane. Provide the following information:

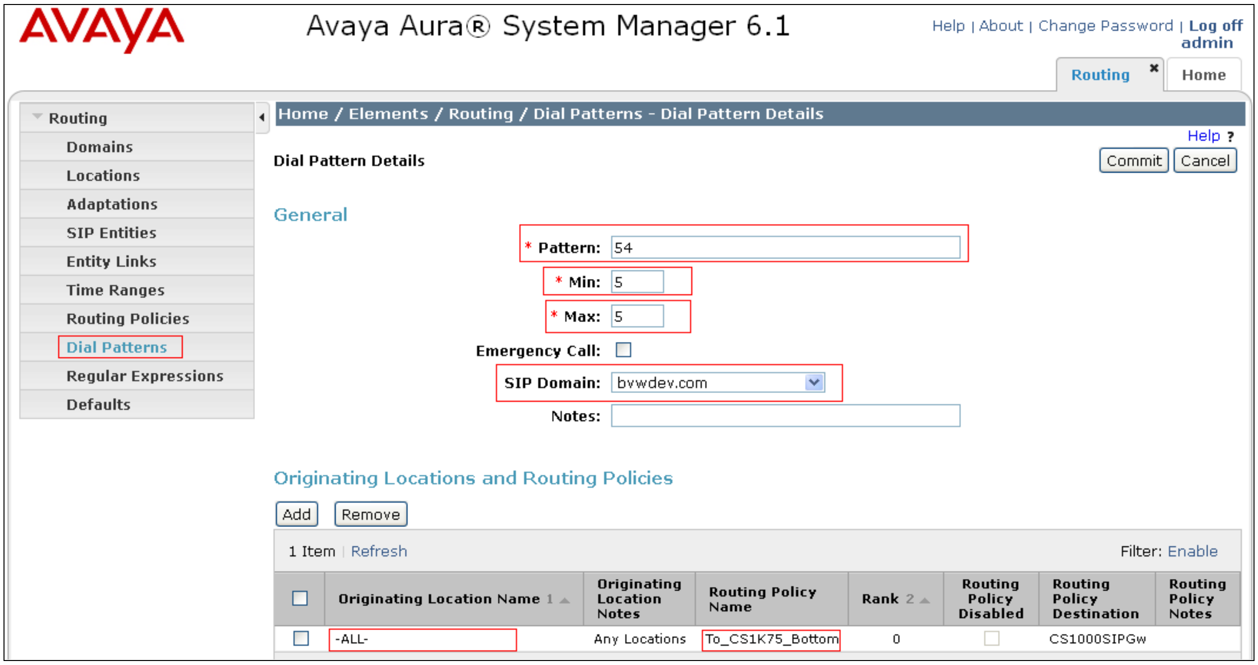
<u>General section</u>

- Enter a unique pattern in the **Pattern** field (e.g. **54**).
- In the **Min** field enter the minimum number of digits (e.g. **5**).
- In the **Max** field enter the maximum number of digits (e.g. **5**).
- In the **SIP Domain** drop down menu select the domain **bvwdev.com** defined in **Section 6.1**.

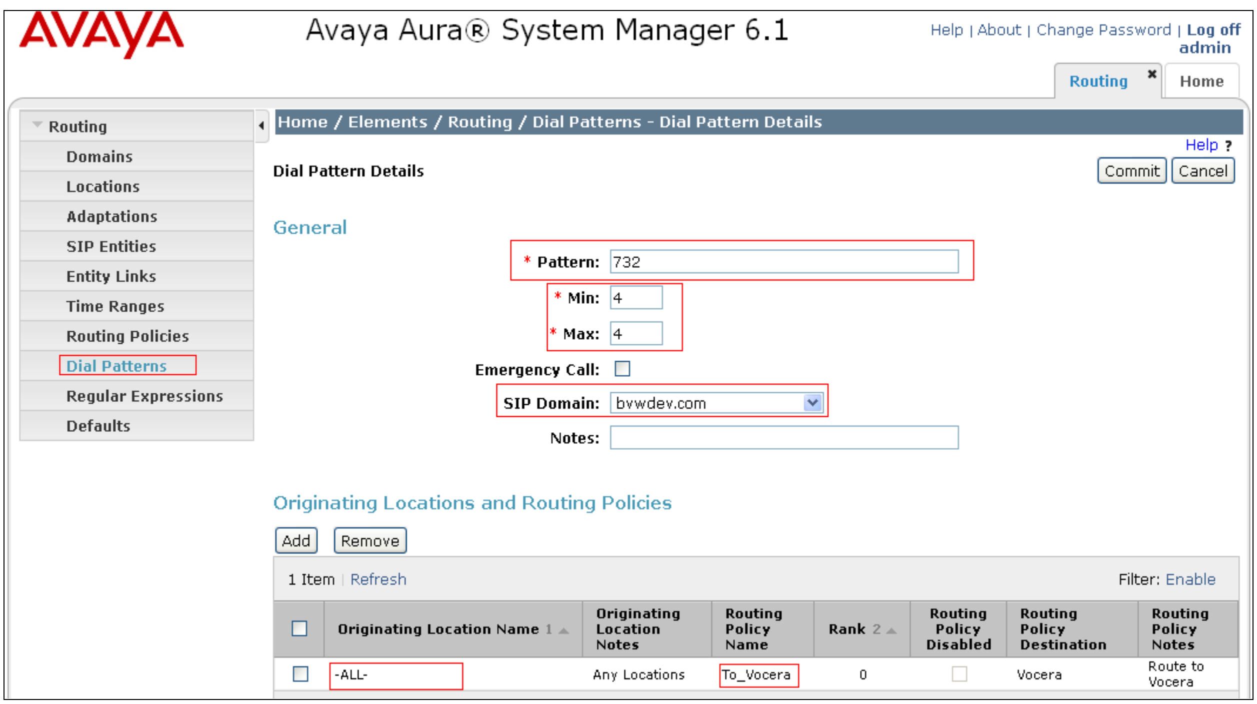
<u>Originating Locations and Routing Policies section</u>

- Click on the **Add** button and a window will open (screen not shown).
- Click on the box for the appropriate Originating Locations, and Routing Policies (see **Section 6.6**) that pertain to this Dial Pattern.
    - Select the Originating Location to apply the selected routing policies to **All**.
    - Select appropriate Routing Policies.
    - Click on the **Select** button and return to the **Dial Pattern** page.

Click the **Commit** button to save the new definition. Repeat steps for the remaining Dial Patterns. The following screen shows the dial pattern **54xxx** used to route calls to Avaya CS1000 system during the compliance test.




The following screen shows the dial pattern **732x** used to route calls to Vocera server during the compliance test.



## 6.8. Configure Manage Elements

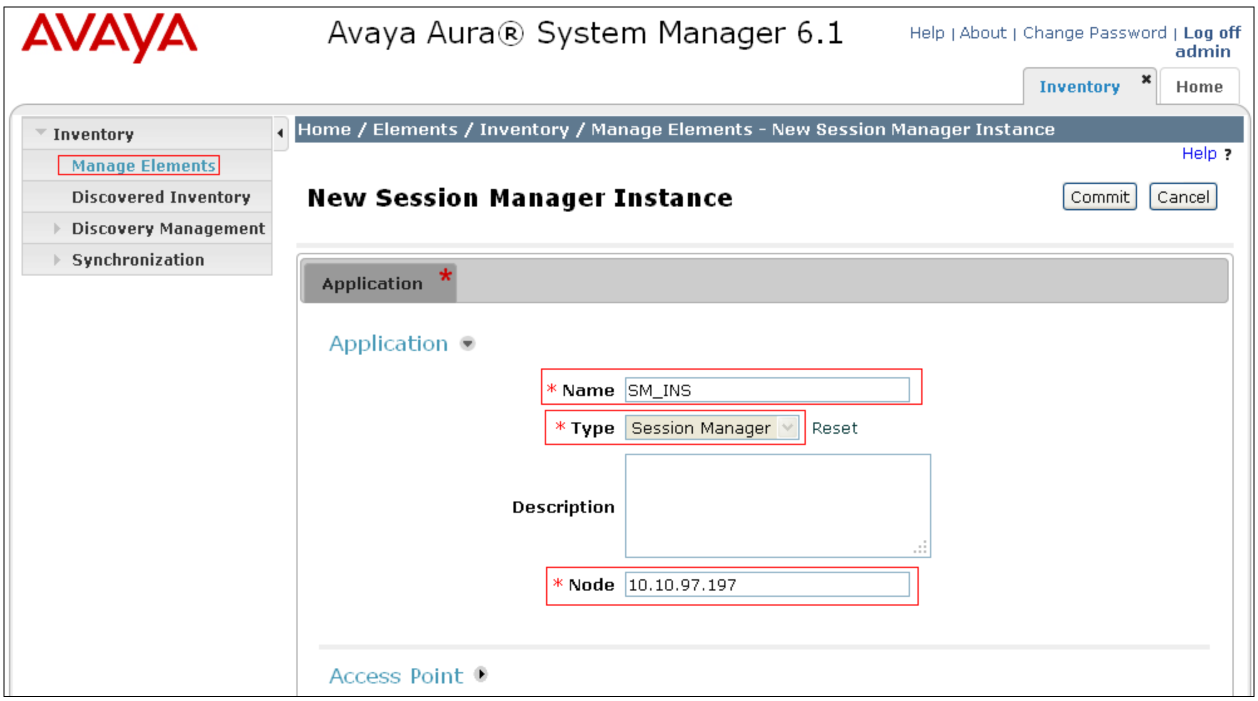
To define a new Manage Element, navigate to **Elements →Inventory→Manage Elements**. Click on the **New** button (screen not shown) to open the **New Entities Instance** page.

In the **New Entities Instance** Page

- In the **Type** field, select **Session Manager** using the drop-down menu and the **New Session Manager Instance** page opens (screen not shown).

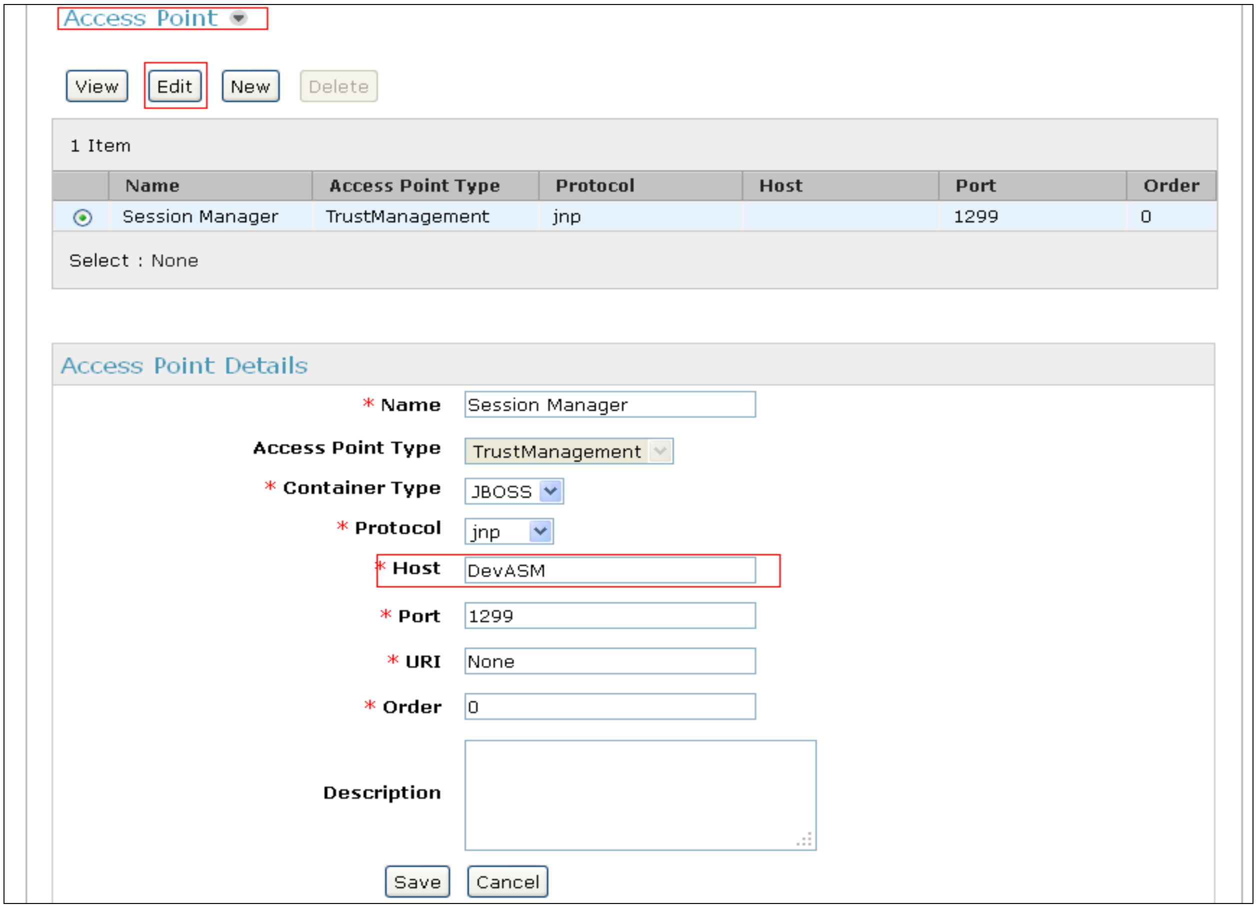
In the **New Session Manager Instance** page, provide the following information:

- Application section
  - **Name** – Enter name for Session Manager Instance, e.g. "**SM_INS**".
  - **Description -** Enter description if desired.
  - **Node** – Enter IP address of the administration interface, **10.10.97.197**.



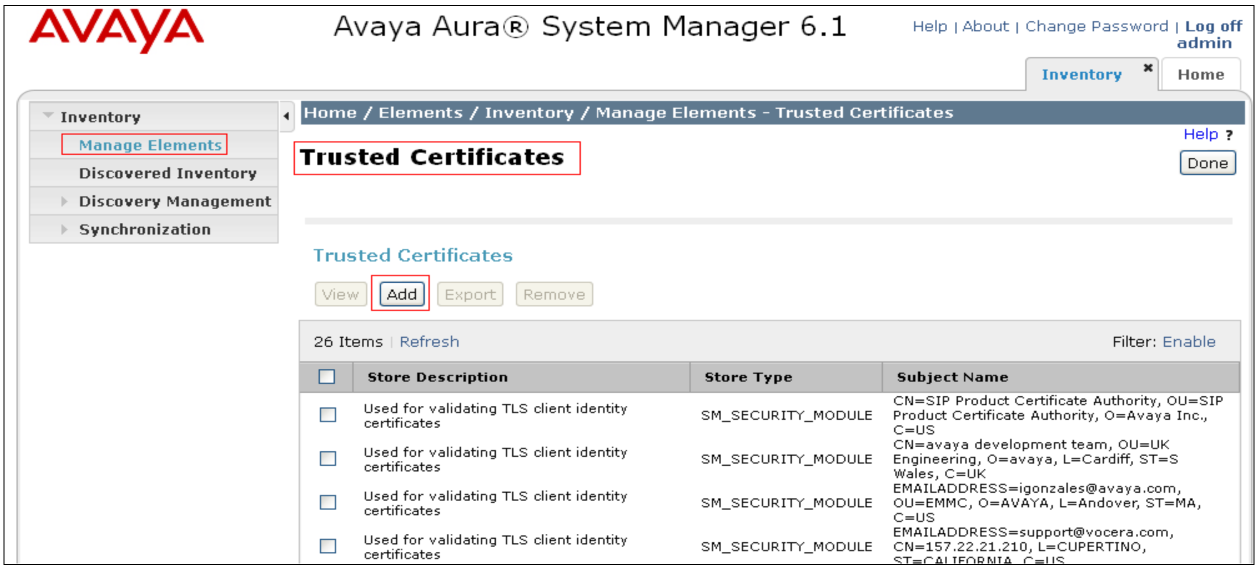

- Access Point section: Check on **Session Manager** radio button and then click **Edit** button to edit, in the **Access Point Details** section, enter host name of Session Manger server in the **Host** field, e.g. "**DevASM**" and keep other values at default. Click **Save** button to save changes.

Click **Commit** button in the **New Session Manager Instance** page to complete creation of new element of Session Manager.

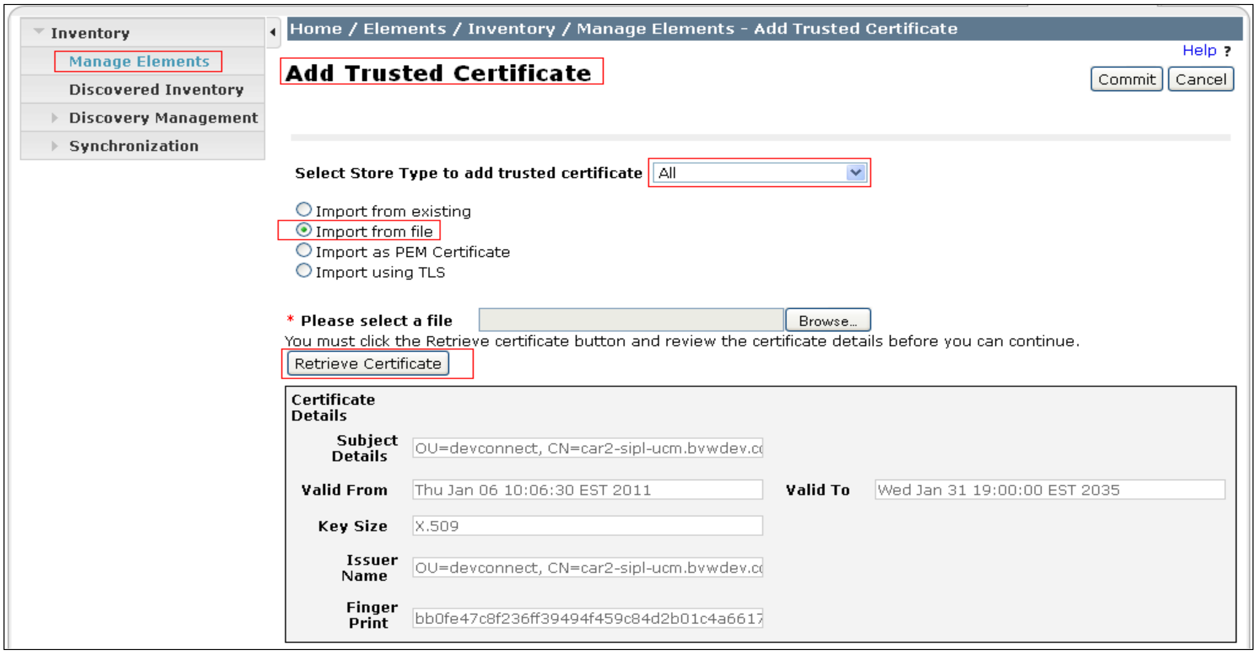

Access Point
View
Edit
New
Delete
1 Item
Name
Access Point Type
Protocol
Host
Port
Order
Session Manager
TrustManagement
jnp
1299
0
Select : None
Access Point Details
* Name
Session Manager
Access Point Type
TrustManagement
* Container Type
JBOSS
* Protocol
jnp
* Host
DevASM
* Port
1299
* URI
None
* Order
0
Description
Save
Cancel

## 6.9. Configure TLS for Avaya Communication Server 1000E SIP Gw and Vocera Server

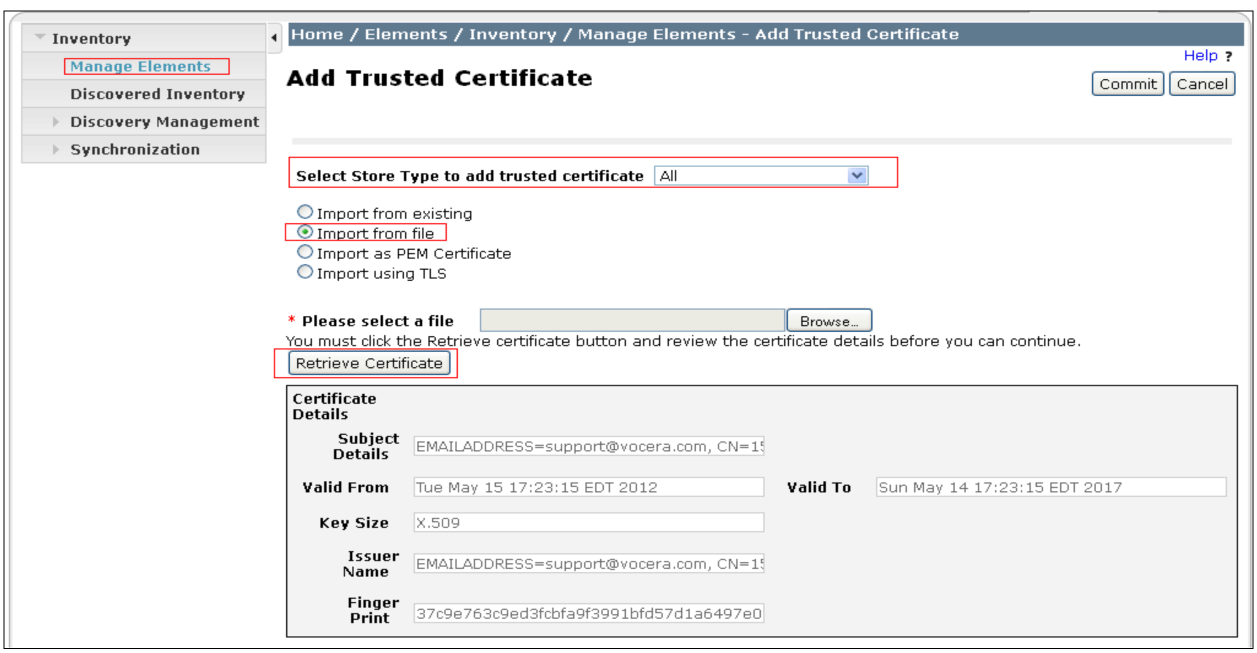
In System Manager, navigate to **Elements → Inventory→ Manage Elements**. Click the Session Manager **SM_INS** created in **Section 6.8** and select **More Actions → Configure Trusted Certificates** (not shown), the **Trusted Certificates** page is displayed as the screen below.



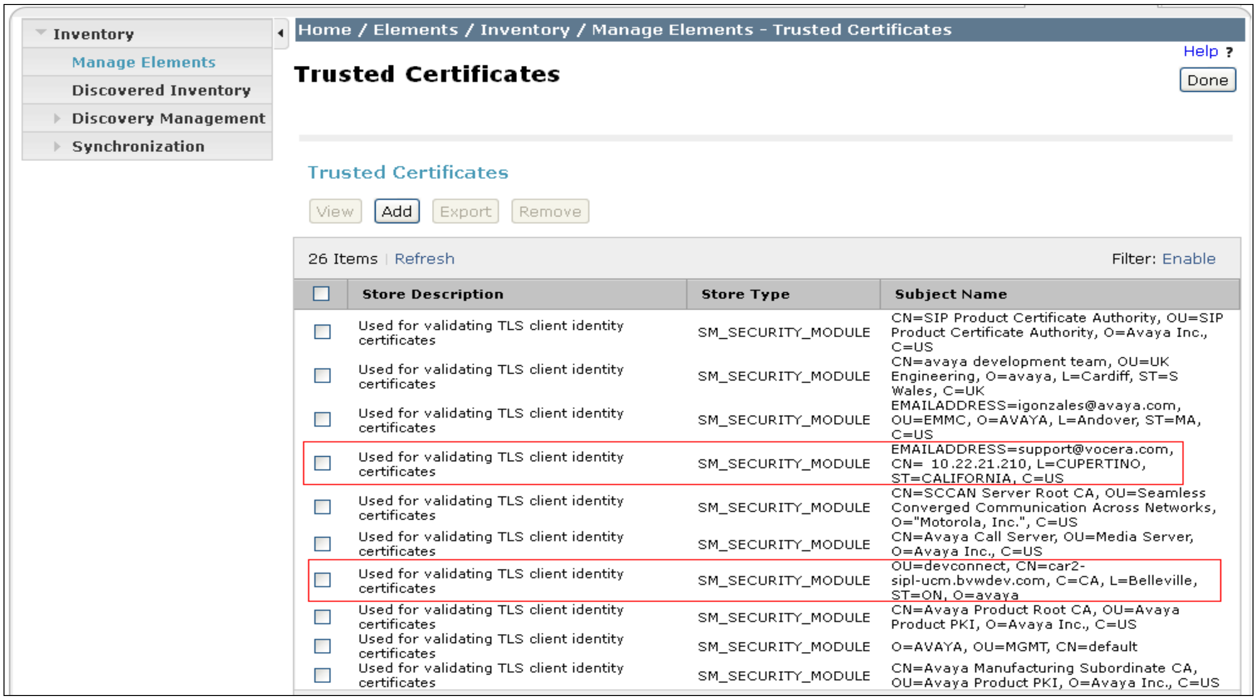
Click **Add** button as shown above to add the local CA certificate of Avaya CS1000 UCM that is already saved in **Section 5.2**. In the **Add Trusted Certificate** page, select "**All**" in the **Select Store Type to add trusted certificate** field, check on radio button to **Import from file**, the **Please Select a file** box is displayed, select **Browse** button to upload the local CA certificate of UCM and then click **Retrieve Certificate** button to retrieve. Click **Commit** button to save.



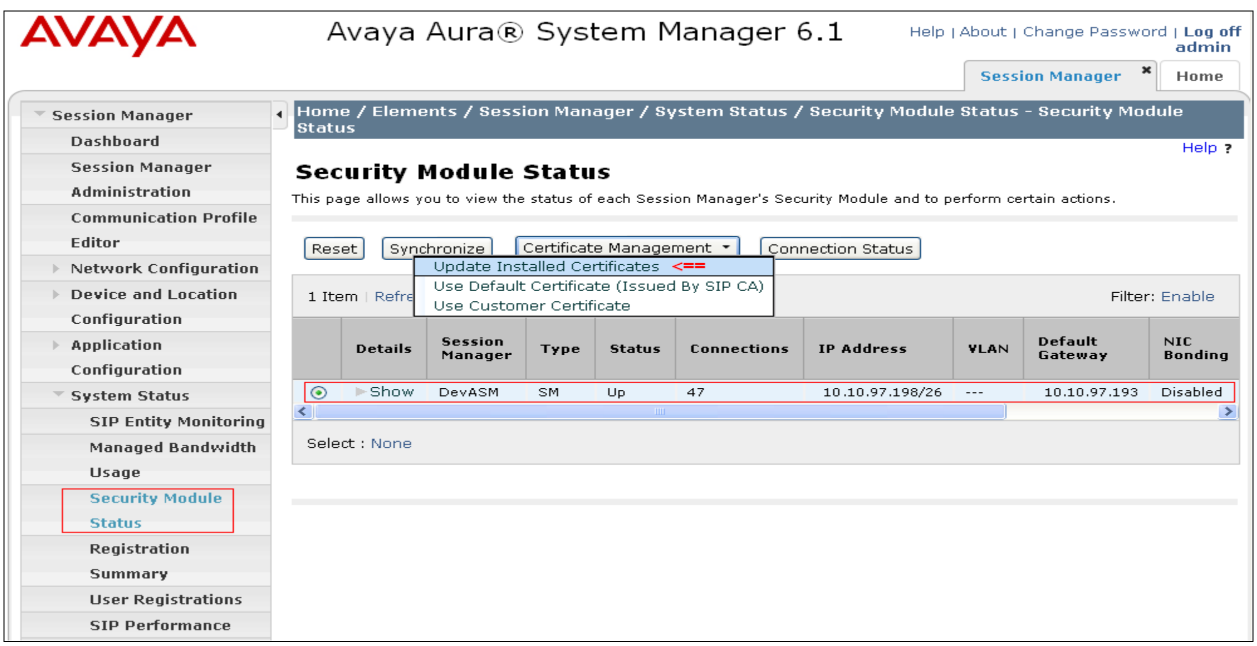

Repeat the same procedure above to add the self-signed certificate of Vocera server into the certificate store of **Session Manager**, the screen below shows the **Add Trusted Certificate** page while adding Vocera's certificate.



Return to **Manage Elements** page. Select **Session Manager** element and then select **Configure Trusted Certificates** from **More Actions** menu (not shown) to confirm the local Avaya CS1000 UCM and Vocera certificates were successfully added as shown below.




It is required to update the security certificates to the Session manager Security Module. Navigate to **Elements** → **Session Manager** → **System Status** → **Security Module Status**, select name of the Session Manager that needs to be updated, in this case **DevASM**, and select **Update Installed Certificates** from **Certificate Management** menu.



# 7. Configure Vocera System

This section assumes that Vocera system is already installed and configured by Vocera Engineer, the section describes procedure of how to configure the Vocera Communication System to inter-work with the Avaya CS1000 system and Session Manager.

## 7.1. Configure TLS Certificate

When installing Vocera server VSTG, it uses OpenSSL to generate a private key and a self-signed certificate in the **\vocera\telephony\vgw** folder. The certificate (**server.crt**) is set to expire 5 years after the date it was created and this certificate is used to upload to the store certificate in Session Manager as mentioned in **Section 6.9** for configuring TLS between Session Manager and Vocera server.

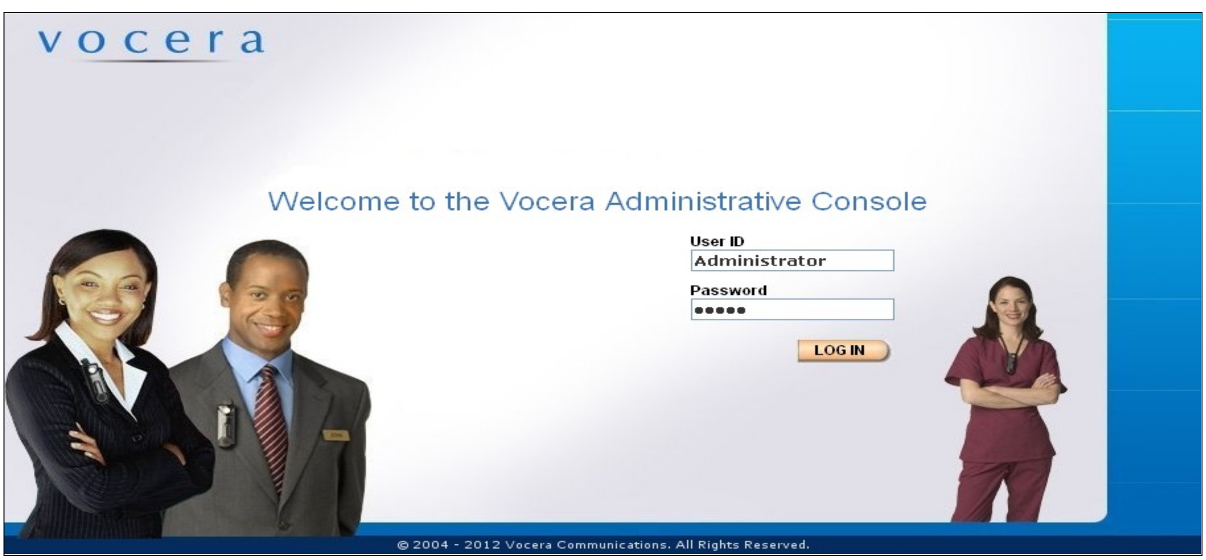
Note: If the 5-year VSTG certificate expires, a new certificate can be generated by running the **\vocera\telephony\certificate\cert.bat** batch file. However, if an upgrade VSTG is required at regular intervals, it should never need to generate a new certificate other than by running the VSTG installer.

To enable TLS transport between VSTG and Session Manager, complete the following tasks:

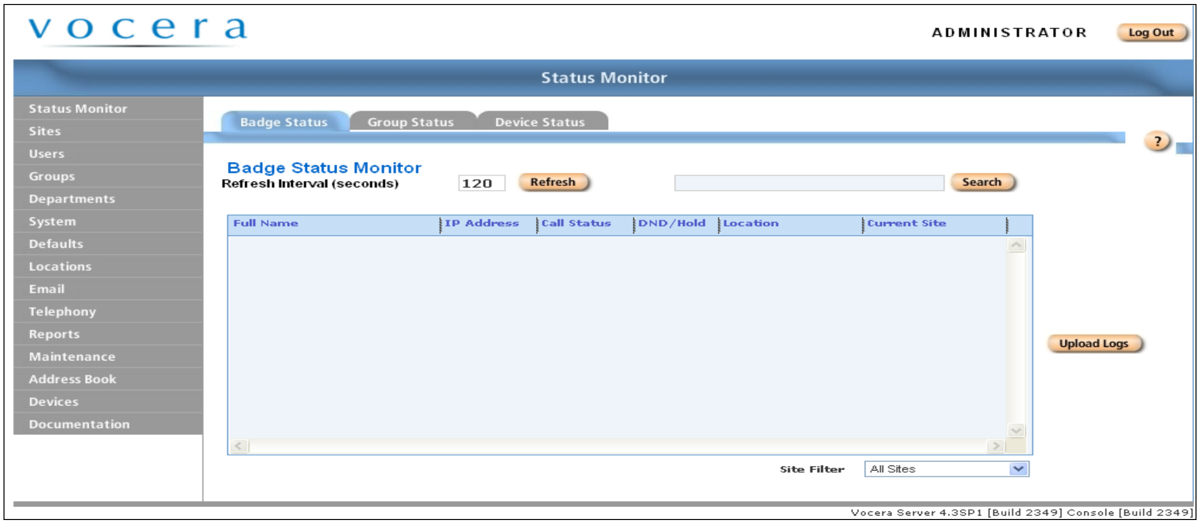
- Upload the TLS certificate from the following location on each Vocera SIP Telephony Gateway to Session Manger: **\vocera\telephony\vgw\server.crt**
- Set the following property in **c:\vocera\telephony\vgw\vgwproperties.txt** on each Vocera SIP Telephony Gateway: **VTGSIPTransport = tls**
- Restart each Vocera SIP Telephony Gateway.

## 7.2. Configure Vocera SIP Connectivity

Open the Vocera Communication Systems web page by addressing the IP address of the Vocera Server in the Microsoft Internet browser, http://10.22.21.210/. The **Welcome to the Vocera** page will appear (screen not shown), then click on the **Vocera Administration Console** link to get to the console web page as shown below.



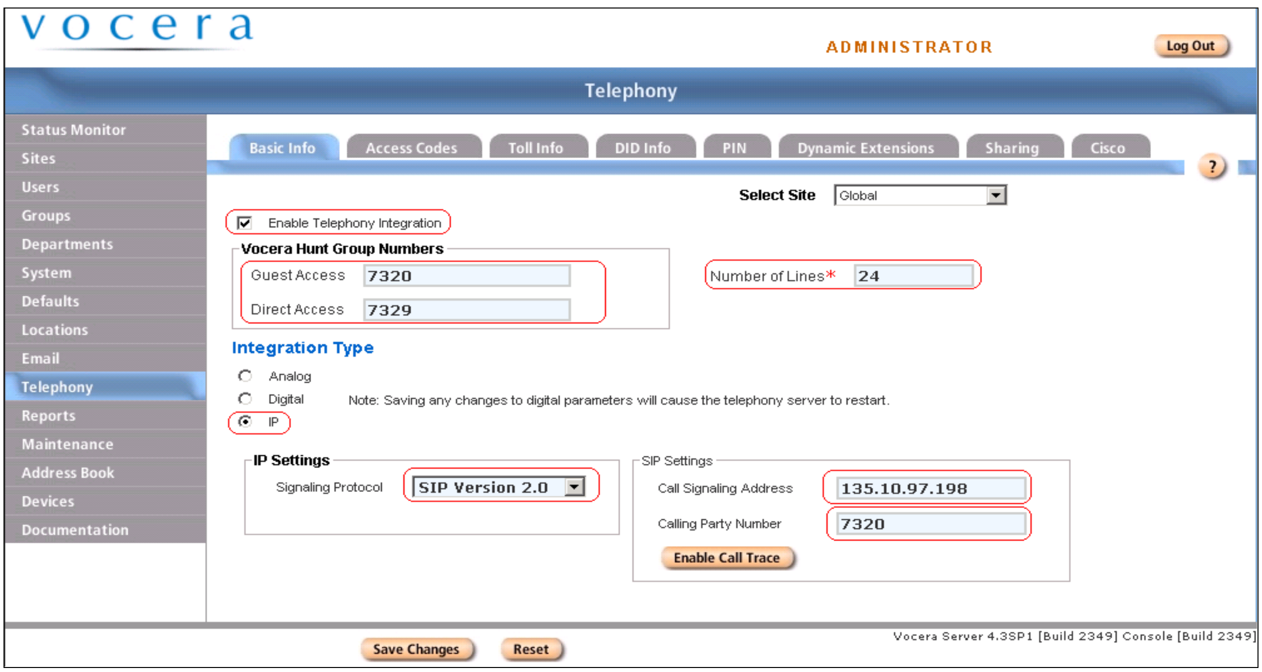
Input proper credentials to log on to the **Console page**, click on the **Log In** button to log in. The screen below shows the **Console** page with the **Status Monitor** menu page as default.



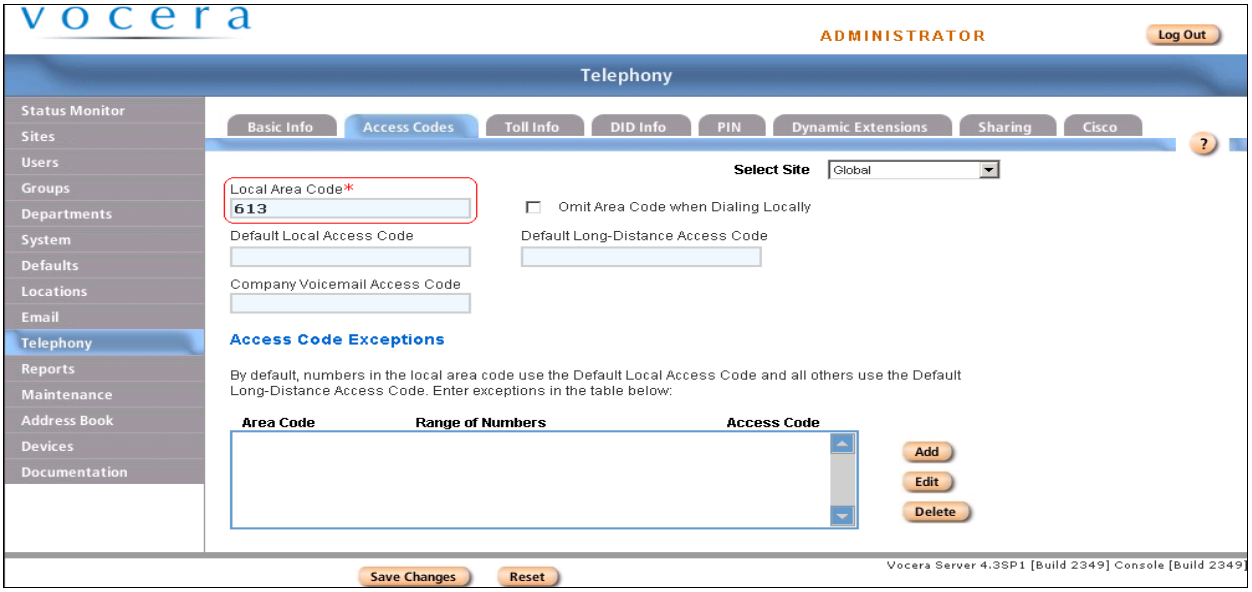

For all the details on the configuration of Vocera Communication System, user can click on the **Documentation** option on the left navigation pane. In the **Administration and Configuration** column, select on the **Telephony Configuration Guide** to view the details description of all the available attribute settings.

To configure the Vocera Server to work with the Avaya CS1000, click on the **Telephony** option on the left navigation pane. The **Telephony** page is displayed with the **Basic Info** menu tab being selected as default, as shown in the screen below. Fill in the details of the highlighted attributes in the red-boxes. Others fields leave at default. Then Click **Save Changes** button.

- **Enable Telephony Integration**: check the check box.
- **Vocera Hunt Group Numbers** section:
  - **Guess Access**: enter the DN "7320".
  - **Direct Access**: enter the DN "7329".
- **Number of Lines**: the default number is "24".
- **Integration Type**: select the "**IP**" radio button.
- **IP SIP Settings** section: select "SIP Version 2.0" in the drop down list.
- **SIP Settings section:**
  - **Call Signaling Address**: enter the signaling IP of Session Manager "10.10.97.198".
  - **Call Party Number**: enter the DN"7320". This DN will be displayed on the called party.

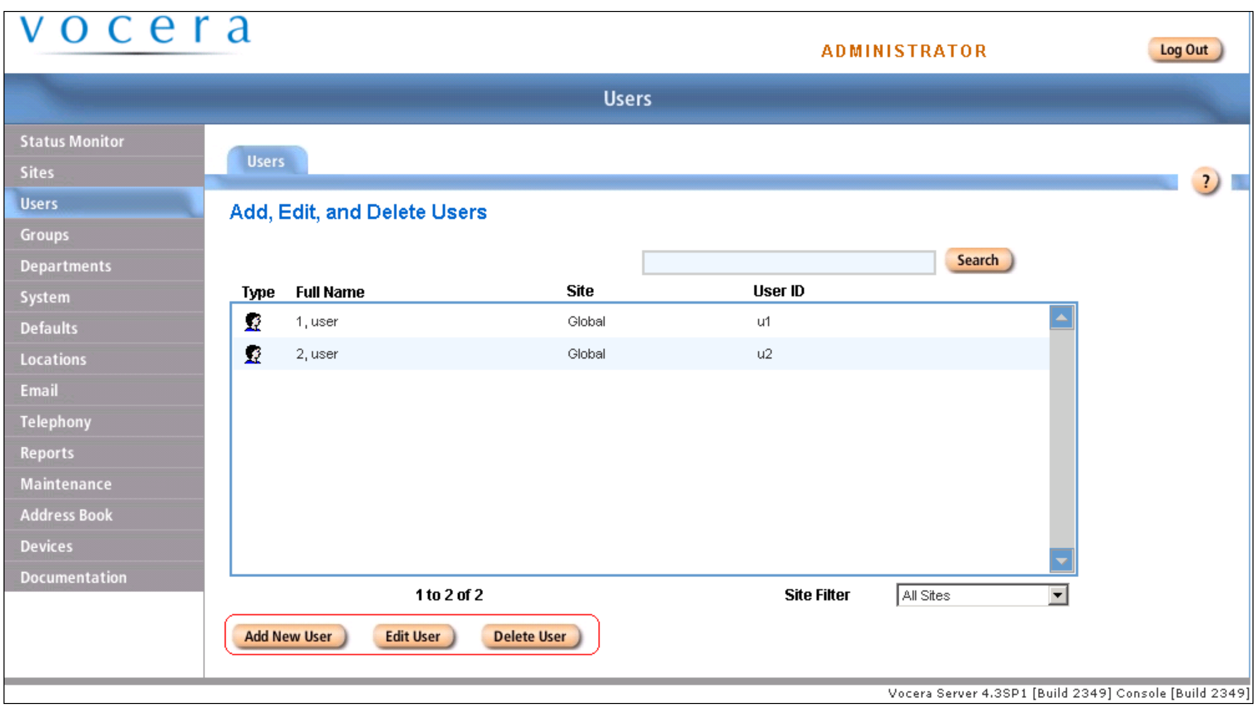


To configure the dialing rule on the Vocera Server, navigate to the **Access Codes** tab, fill in the red highlighted text box of the attributes as shown in the screen below. Then click **Save Changes** button.



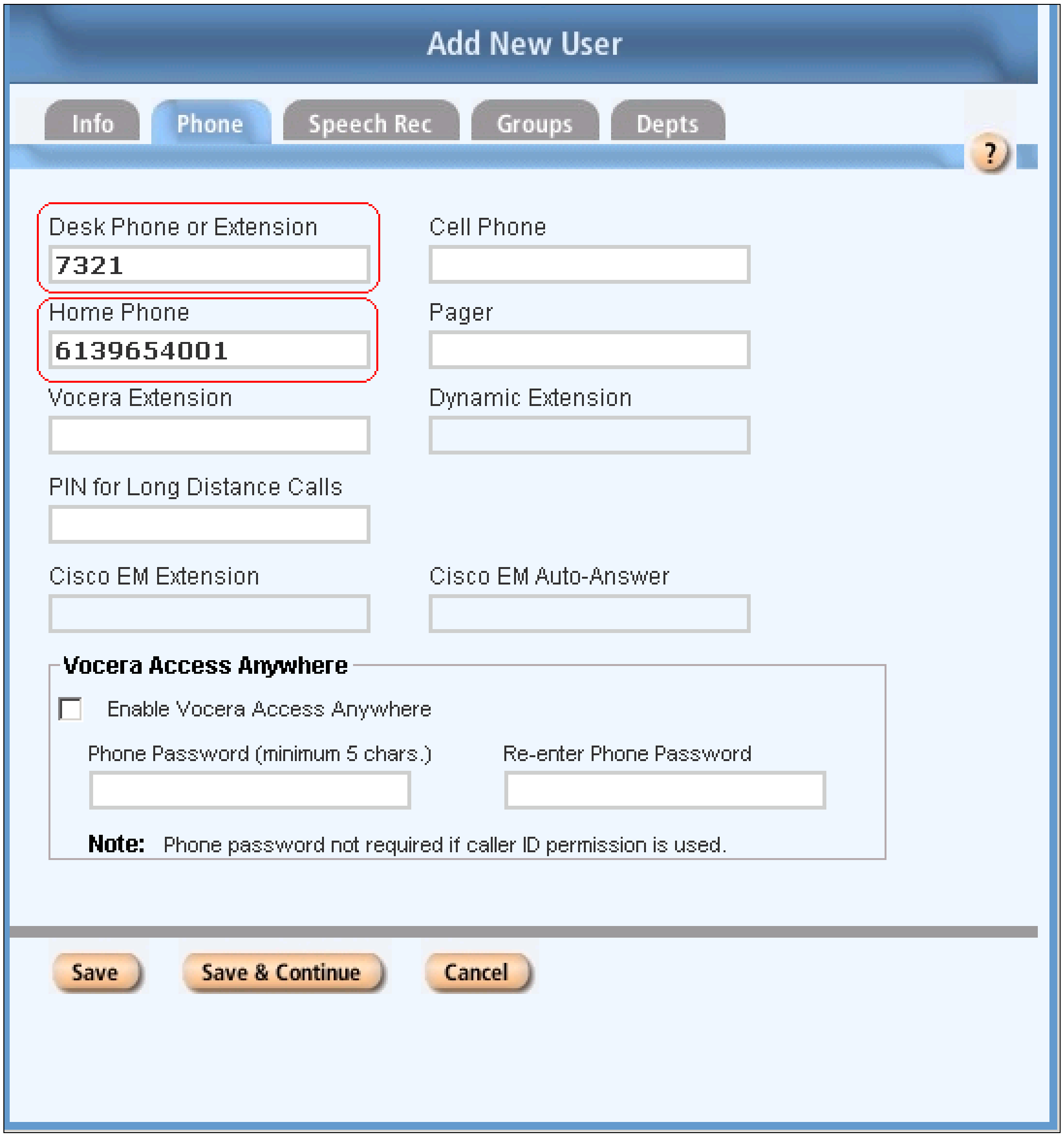

## 7.3. Configure Users

To configure the users on the Vocera Server to be able to send and receive calls from the Avaya CS1000, click on the **Users** menu option. The **Users** page is displayed as shown in the screen below.



To add a user, click **Add New User** button, the user **Info** detail configuration page is displayed. Fill in the required fields, which are indicated with the red stars.  The **Badge ID** field will be populated when the badge is registered to Vocera Server. Others are left at default. Click **Save**.

From the **Add New User** page, continue to click on the **Phone** tab to configure user specific phone number information such as **Desk phone or Extension, Home phone**. Others fields are optional. Click **Save**.



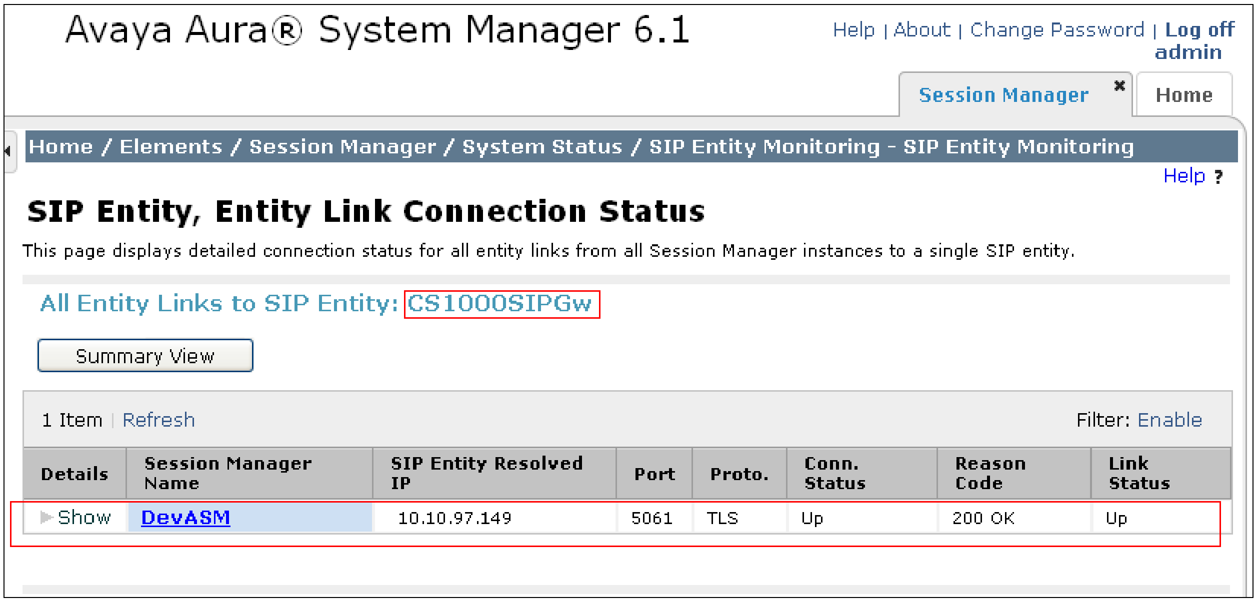

Click on the **Group** tab to assign the newly created user to a group with specific permission to use other call features on the Vocera Server. By default, in this example, every new user is assigned to the **Group Everyone** and belong to the **Site Global** (screen not shown).

For detailed configuration on how these **Groups** and **Sites** are configured, please refer to the **Administration Guide** by clicking on the **Documentation** option menu, under the **Administration and Configuration**.

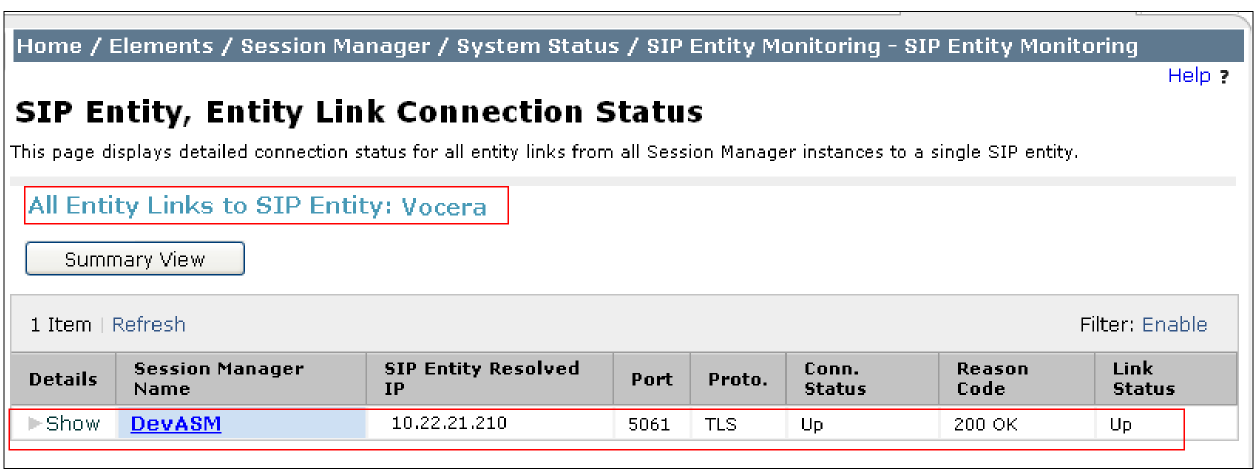
# 8. Verification Steps

The following typical steps are used to verify SIP TLS between Session Manager and Avaya CS1000 and between Session Manager and the Vocera server.

- Verify SIP TLS entity link status is up between Session Manager and Avaya CS1000 SIP gateway by navigating to **Elements →Session Manger → System Status → SIP Entity Monitoring** and select the Avaya CS1000 entity link.




- Repeat the same procedure to verify SIP TLS entity link between Session Manager and Vocera server.




- Place calls from Avaya CS1000 phone to Vocera wireless badge user and vice versa to make sure SIP TLS trunks between Avaya CS1000 SIP gateway and Session Manager and between Vocera server and Session Manger are established without any failures.

- Verify audio quality for calls established between Avaya CS1000 phones (Unistim and SIP phones) with Vocera wireless badge user.

# 9. Conclusion

These Application Notes have described the administration steps required to integrate the Vocera Communication System with the Avaya Communication Server 1000E via SIP TLS trunk configured on the Avaya Aura® Session Manager. All test cases passed with observations noted in **Section Error! Reference source not found.**.

# 10. Additional References

The following Avaya product documentation is available at http://support.avaya.com.


[1] *Administering Avaya Aura® Session Manager*, Release 6.1, November 2010, Issue 1.1, Document Number03-603324

[2] *Administering Avaya Aura® System Manager*, Release 6.1, November 2010

[3] *Avaya Communication Installation and Commissioning*, Doc# NN43041-310, Issue 05.04, Date May 2011.

[4] *Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals*, Doc # NN43001-116, Issue 05.11, Date June 2011.

[5] *Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals*, Doc # NN43001-509, Issue 03.02, Date June 2011.

[6] *Avaya Communication Server 1000 Element Manager System Reference - Administration*, Doc# NN43001-632, Issue 05.09, Date July 2011.


Product information for Vocera Communication System can be found at http://www.vocera.com/products/resources/documentation.aspx

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.