Avaya Solution & Interoperability Test Lab

Application Notes for Configuration of Avaya Communication Server 1000 Release 7.5 and Vocera Communication System Release 4.1SP7 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate Vocera Communication System release 4.1SP7 with Avaya Communication Server 1000 release 7.5 via SIP trunk configured on Avaya Aura® Session Manager release 6.1.

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 detailed configurations of Avaya Communication Server 1000 release 7.5 (hereafter referred to as Avaya CS1000) and Vocera Communication System release 4.1SP7 (hereafter referred to as Vocera Server). During the compliance testing, the Vocera Server was tested to make sure all supported telephony features properly functioned and interoperated with Avaya CS1000 via SIP trunk through the Avaya Aura® Session Manager Release 6.1.

2. General Test Approach and Test Results

The general test approach was to have different telephone types of Avaya CS1000 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 Trunk 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 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 observations:

- The Avaya CS1000 SIP phone could not perform transfer a call from one Vocera to another Vocera badge. Applying patch MPLR31794 to resolve this issue.
- Conference button on the Avaya CS1000 IP phone is not available if Ring Again No Answer feature is enabled on the phone which hosts conference. The scenario happens when Vocera is being invited to join the conference. The work around is not to provision the Ring Again No Answer on the IP phone. Work Item has been raised to track the issue to resolution.
- The Vocera Server can't parse the INVITE coming from Avaya CS1000 which has SDP encapsulated in the MIME part. Using Session Manager to create Adaptation module to remove the MIME part and provide the standard SDP to Vocera. See **Section 5.1.3**.

2.3 Support

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

- **Phone:** +1 408-882-5100
- **Email:** support@vocera.com
- **Web:** http://www.vocera.com/about/support.aspx

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya CS1000 SIP trunk network that includes the following Avaya products:

- Vocera Communication System connected to Avaya Aura® Session Manager via SIP trunk.
- Avaya Aura® Session Manager.
- Avaya Communication Server 1000 connected to the Avaya Aura® Session Manager via SIP trunk.
- Avaya SIP phone, IP soft and hard phones, and TDM phones
- Emulated PSTN over PRI trunk.

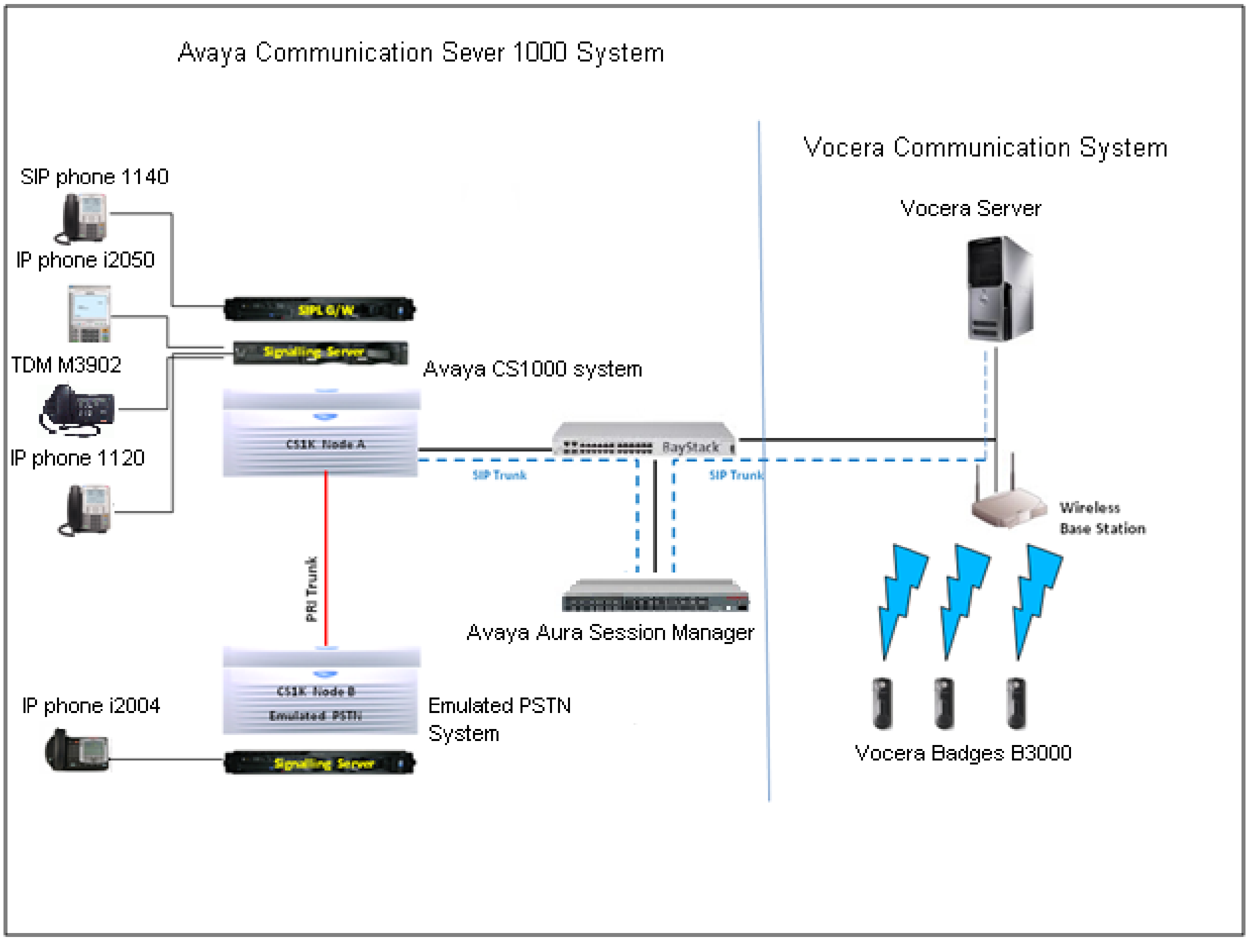

Figure 1: Avaya Communication Server 1000 Network with the Vocera Communication System connecting to Avaya Session Manager via SIP trunk.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Session Manager running on Avaya S8800 server	6.1 (6.1.5.0.615006)
Avaya Communication Server 1000E Call Server Signaling Server SIP Line Gateway	 7.50Q GA plus latest DEPLIST 7.50.17 GA plus latest Service Update 7.50.17 GA plus latest Service Update
Avaya CS1000 IP Phones	1120, i2050, i2004
Avaya CS1000 SIP Phones	1140
Avaya TDM phone	3902
Vocera Communication System Vocera Server Wireless badge Wireless base station	 4.1 SP7 B3000 N/A

5. Configure Avaya System

These Application Notes assume that Session Manager and Avaya CS1000 are installed, configured and operational. For detailed information on how to configure and administer the Avaya Systems, please refer to the **Section 9**.

The following section will describe how to configure the SIP trunk from Vocera Server, Session Manager and Avaya CS1000.

5.1 Configure the Avaya Aura® Session Manager

Log in to the **System Manager** with appropriate credential (not shown), the **System Manager** home page as show in **Figure 2** bellow:

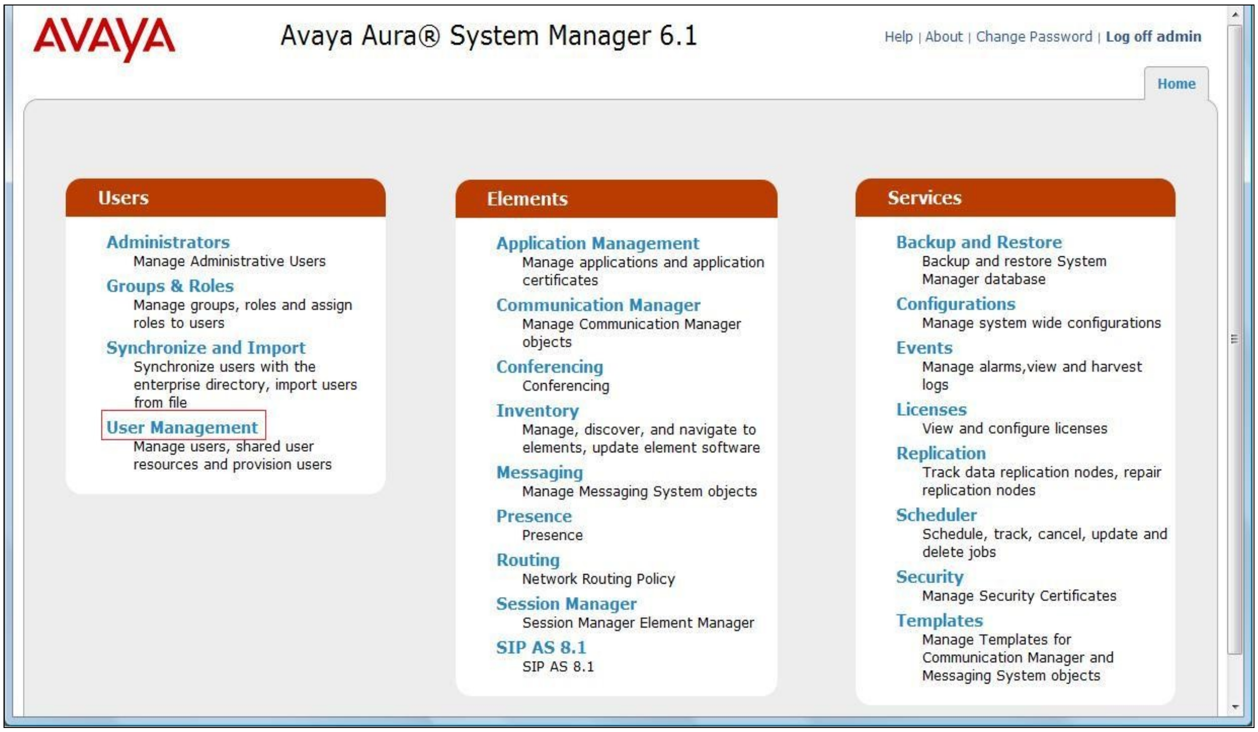

Figure 2: System Manager Home Page

Navigate to **Elements → Routing**, the **Introduction to Network Routing Policy** page will appear as shown in **Figure 3. Figure 3** shows 8 steps to configure a Network Routing Policy for SIP trunk connectivity between Vocera Server, Session Manager and Avaya CS1000.

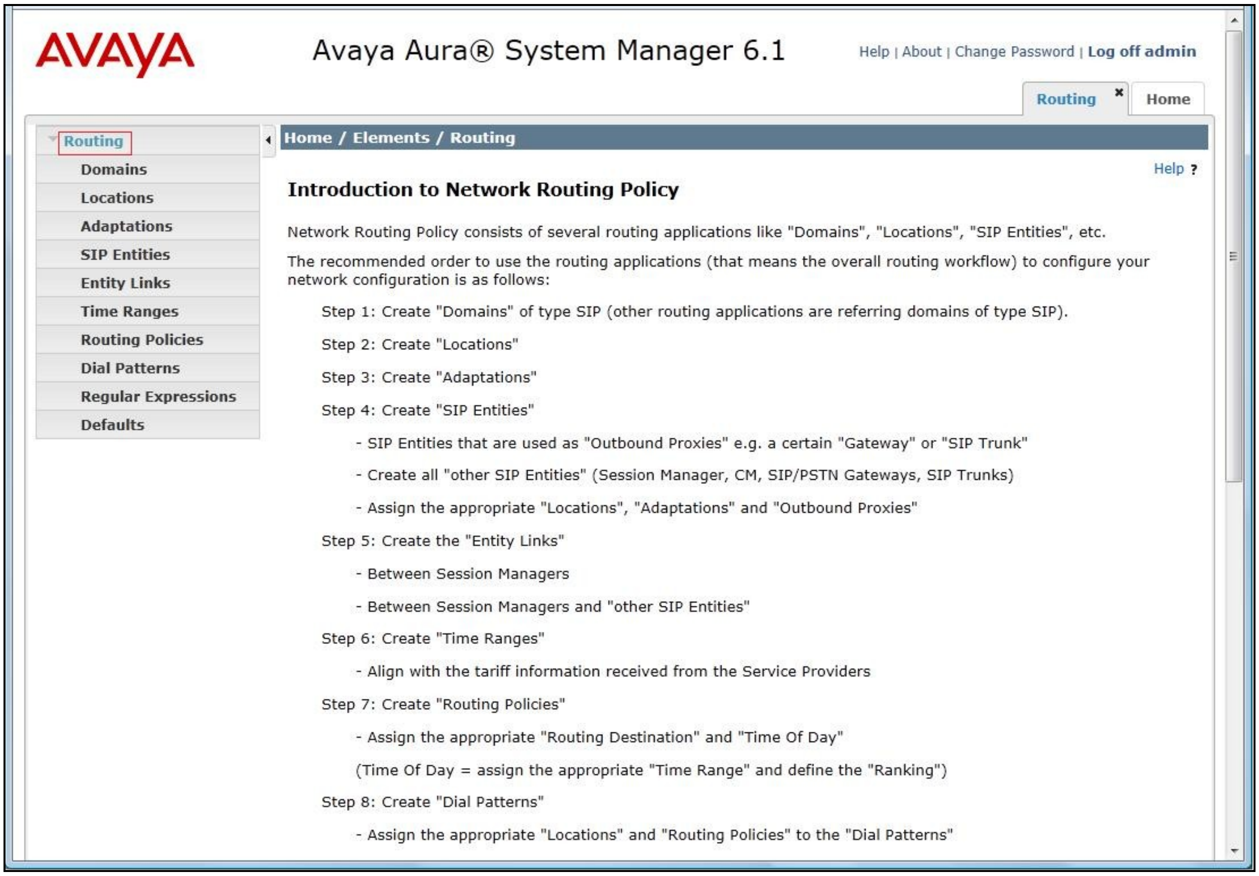

Figure 3: New User Profile

5.1.1 Create SIP Domains

From the left menu column, click on the **Domains → New, Domain Management** page will appear. Enter domain **Name, Type** and **Notes** as shown in **Figure 4**. Click **Commit**.

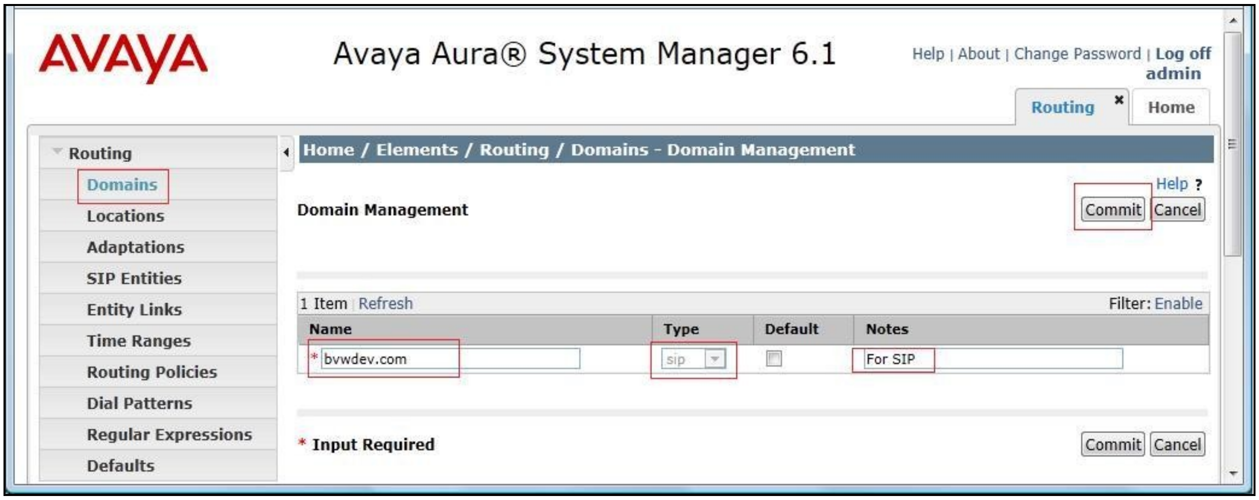

Figure 4: Domain Management

5.1.2 Create Locations

From the left menu column, click on the **Locations → New, Location Details** page will appear as shown in **Figure 5**. Enter location **Name**, **Managed Bandwidth Units**, **Total Bandwidth** and others are at default. Click **Commit**.

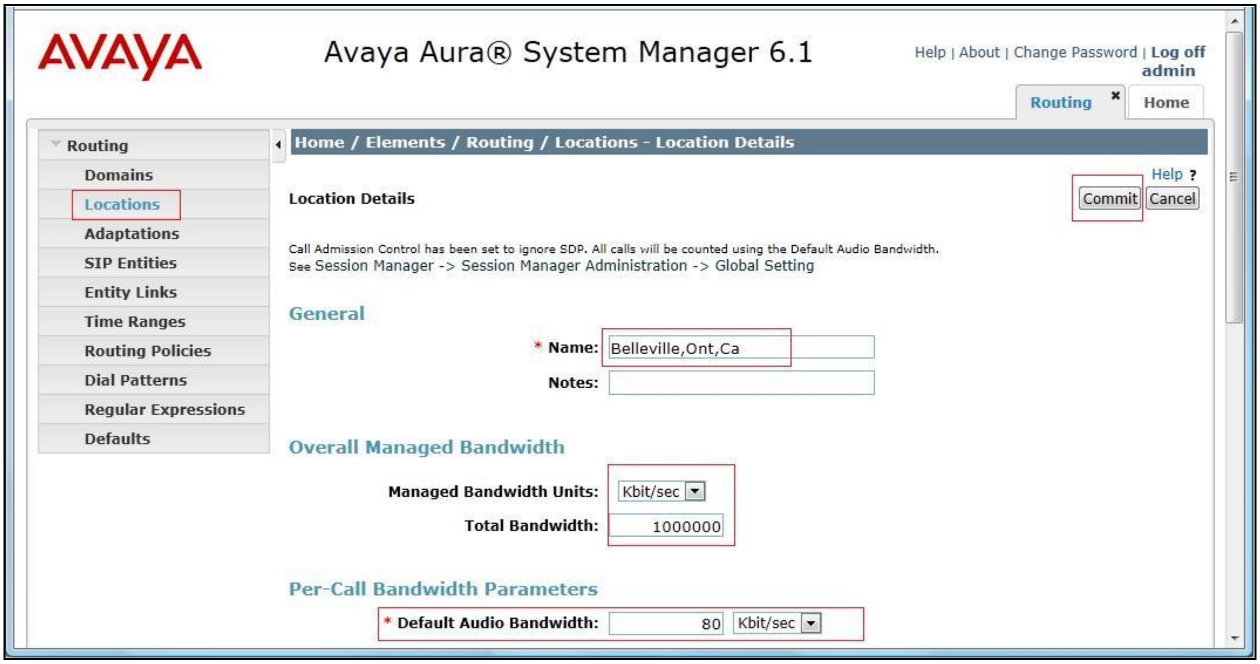

Figure 5: Location Details

5.1.3 Create Adaptations

From the left menu column, click on the **Adaptations → New, Adaptation Details** page will appear as shown in **Figure 6**. Enter **Adaptation name**, **Module name** and **Module parameter** as shown in red-box. Others are left at default. Click **Commit**.

Note: This adaptation module is required to remove the MIME encapsulated SDP of the INVITE message sending out from the Avaya CS1000 to Vocera Server.

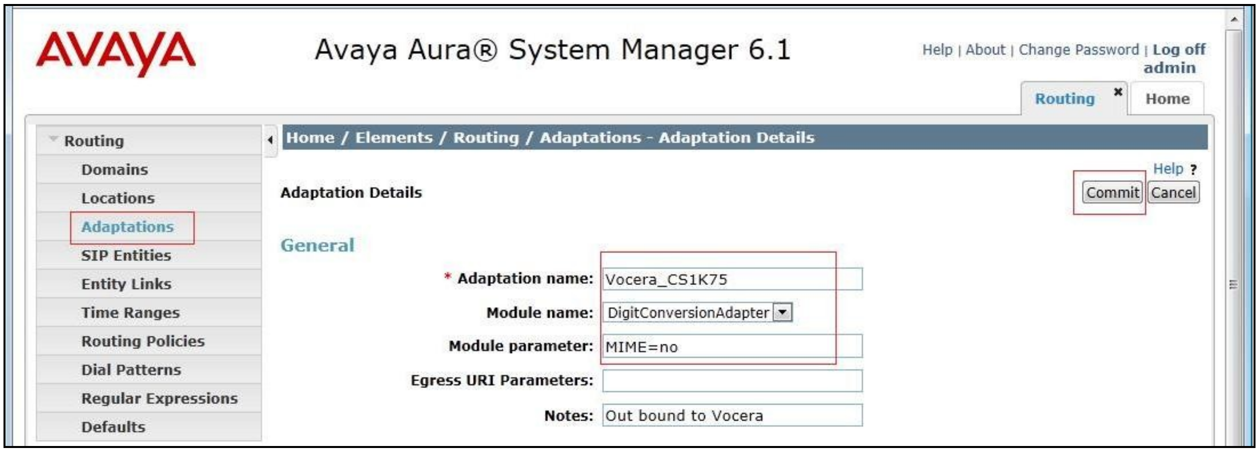

Figure 6: Adaptation Details

5.1.4 Create SIP Entities

This section describes how to create a SIP Entity ***DevASM*** for Session Manager. From the left menu column, click on the **SIP Entities → New,** the **SIP Entity Details** page will appear as shown in **Figure 7.** Enter **SIP Entity Name** ***DevASM***, **FQDN or IP Address** is ***Session Manager IP Address***, **Type** is ***Session Manager***, **Notes** (optional), **Location** and **Time Zone** are as shown in red-box. Others are left at defaults.
At the **Port** section, click on **Add** button to add 2 ports of ***5060, TCP*** and ***UDP*** protocol, and **Default Domain** is ***bvwdev.com*** as it was created in **Section 5.1.1**. Click **Commit**.

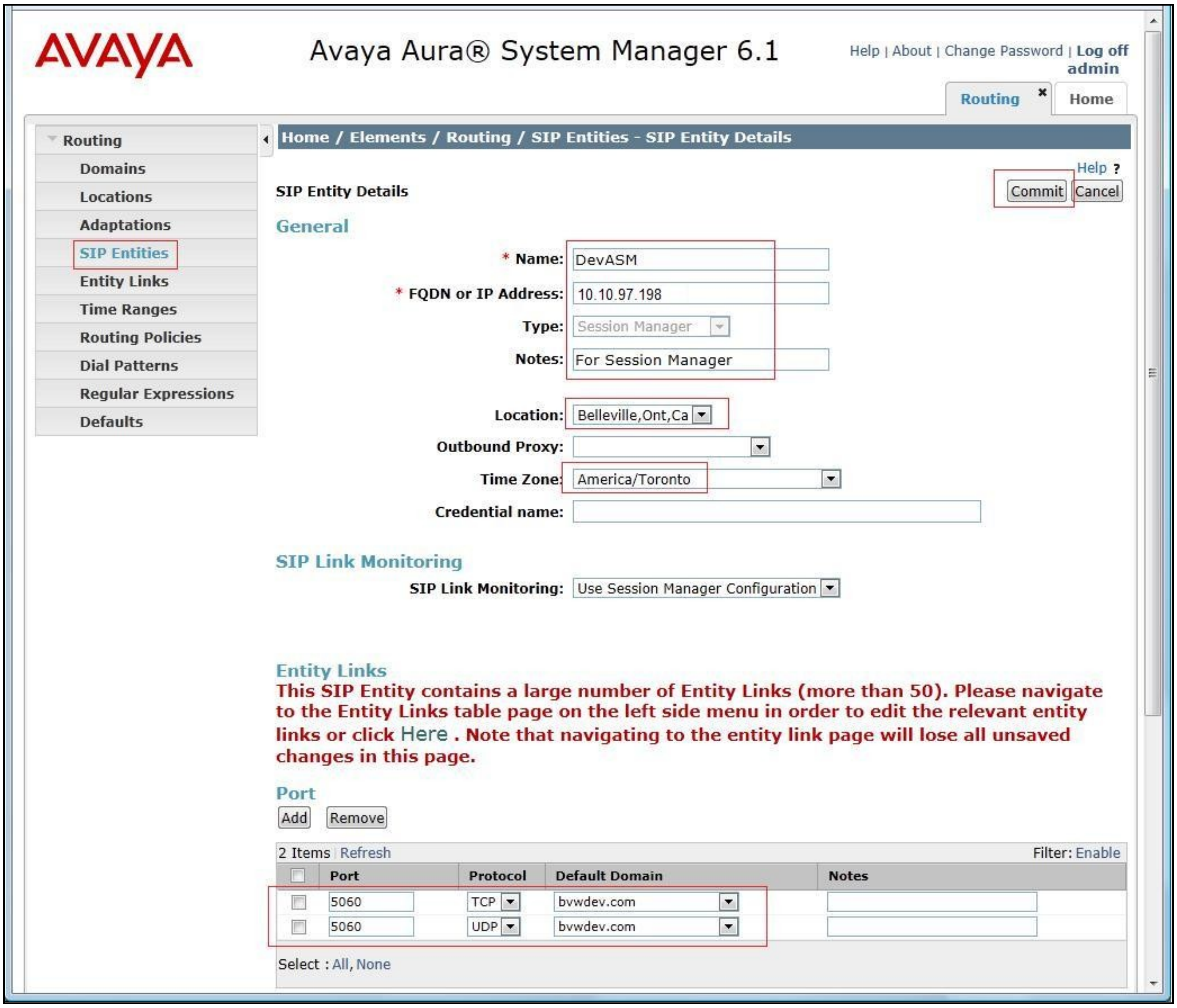

Figure 7: Avaya Session Manager SIP Entity Details

This section describes how to create a SIP Entity for Vocera Server. From the left menu column, click on the **SIP Entities → New,** the **SIP Entitiy Details** page will appear as shown in **Figure 8.** Enter **SIP Entity Name**, **FQDN or IP Address**, **Type**, **Notes**, **Adaptation**, **Location**, **Time Zone** and **SIP Timer B/F** as shown in red-box. Others are left at defaults. Click **Commit**.

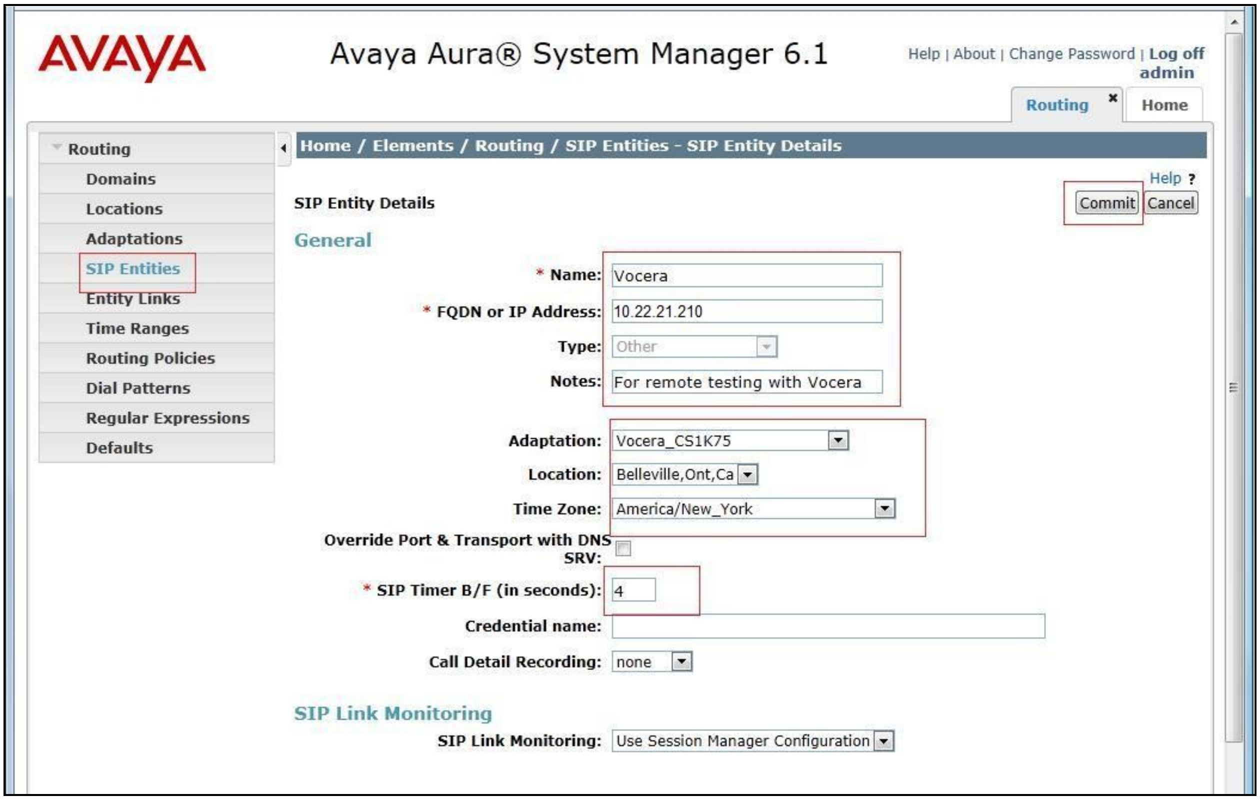

Figure 8: Vocera Server SIP Entity Details

This section describes how to create the SIP Entity for Avaya CS1000 .From the left menu column, click on the **SIP Entities → New, the SIP Entities Details** page will appear as shown in **Figure 9.** Enter **SIP Entity Name**, **FQDN or IP Address**, **Type**, **Notes**, **Location**, **Time Zone** and **SIP Timer B/F** as shown in red-box. Others are left at defaults. Click **Commit**.

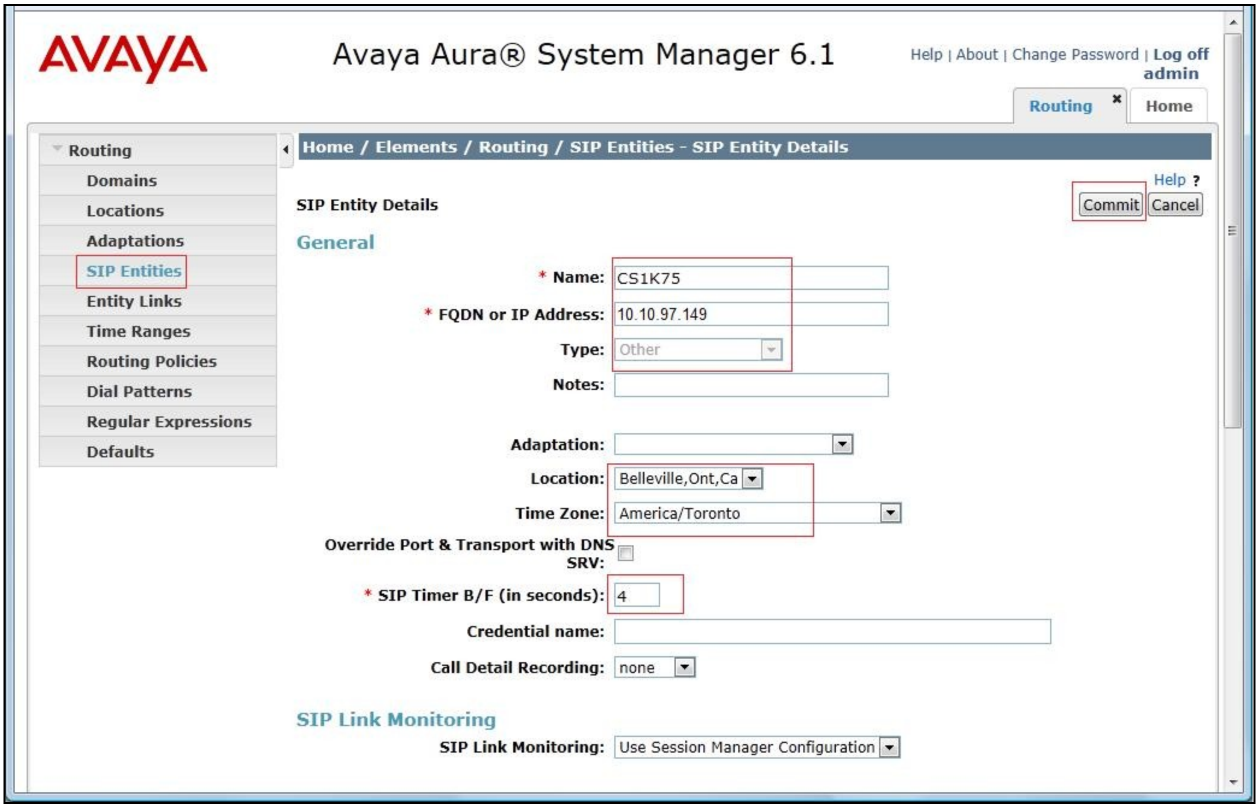

Figure 9: Avaya CS1000 SIP Entity Details

5.1.5 Create the Entity Links

A trusted entity link must be created between Session Manager and Vocera Server using TCP protocol. From the left menu column, click on the **Entity Links → New**, the **Entity Links** page will appear as shown in **Figure 10**. Enter entity link **Name**. Choose ***DevASM*** as **SIP Entity 1** from dropdown menu which was created in **Section 5.1.4**. Choose **Protocol *TCP*** and **Port *5060***. Choose ***Vocera*** as **SIP Entity 2** and **Port *5060***. Check box **Trusted**. Click **Commit**.

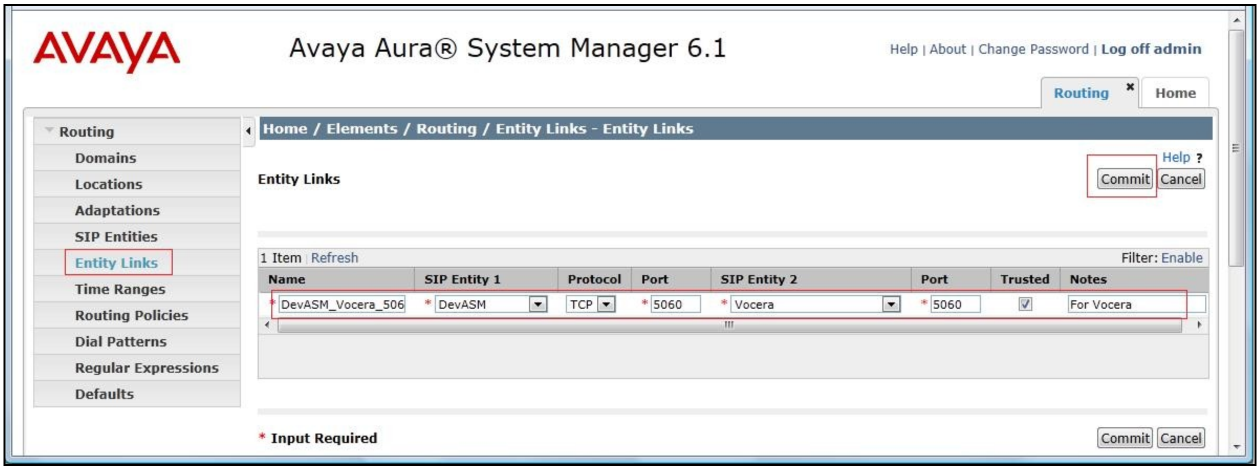

Figure 10: Vocera and Session Manager Entity Links (TCP)

A trusted entity link must be created between Session Manager and Vocera Server using UDP protocol. From the left menu column, click on the **Entity Links → New**, the **Entity Links** page will appear as shown in **Figure 11**. Enter entity link **Name**. Choose ***DevASM*** as **SIP Entity 1** from dropdown menu which was created in **Section 5.1.4**. Choose **Protocol *UDP*** and **Port *5060***. Choose ***Vocera*** as **SIP Entity 2** and **Port *5060***. Check box **Trusted**. Click **Commit**.

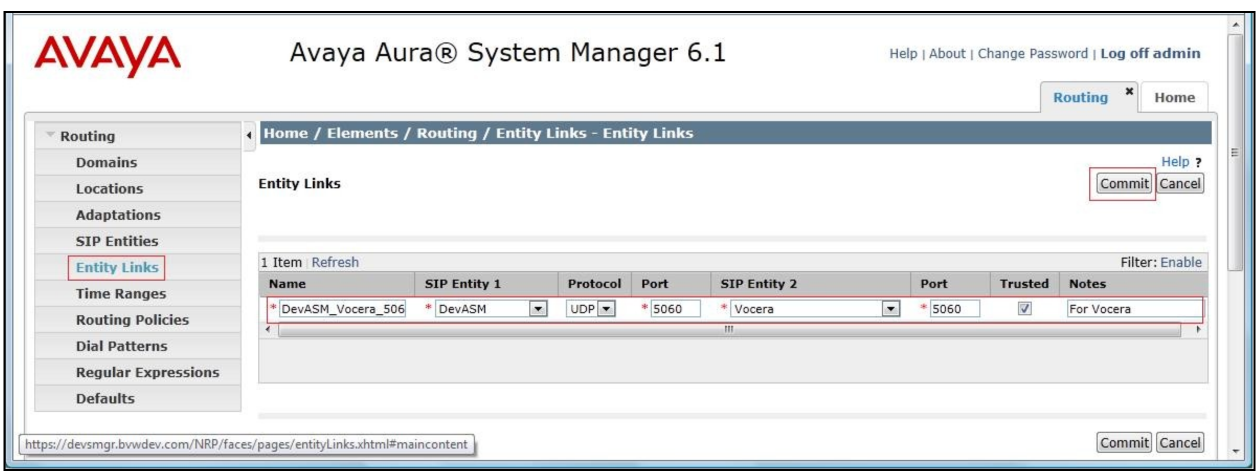

Figure 11: Vocera and Session Manager Entity Links (UDP)

A trusted entity link must be created between Session Manager and Avaya CS1000 using TCP protocol. From the left menu column, click on the **Entity Links → New**, the **Entity Links** page will appear as shown in **Figure 12**. Enter entity link **Name**. Choose ***DevASM*** as **SIP Entity 1** from dropdown menu which was created in **Section 5.1.4**. Choose **Protocol *TCP*** and **Port *5060***. Choose ***CS1K75*** as **SIP Entity 2** and **Port *5060***. Check box **Trusted**. Click **Commit**.

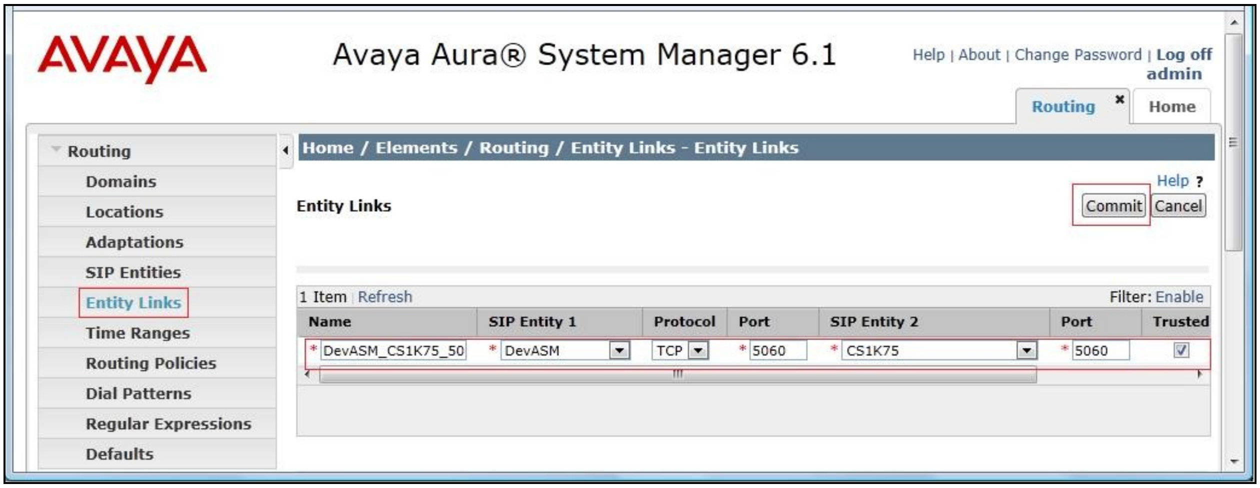

Figure 12: Avaya CS1000 and Session Manager Entity Link (TCP)

A trusted entity link must be created between Session Manager and Avaya CS1000 using UDP protocol. From the left menu column, click on the **Entity Links → New**, the **Entity Links** page will appear as shown in **Figure 13**. Enter entity link **Name**. Choose ***DevASM*** as **SIP Entity 1** from dropdown menu which was created in **Section 5.1.4**. Choose **Protocol *UDP*** and **Port *5060***. Choose ***CS1K75*** as **SIP Entity 2** and **Port *5060***. Check box **Trusted**. Click **Commit.**

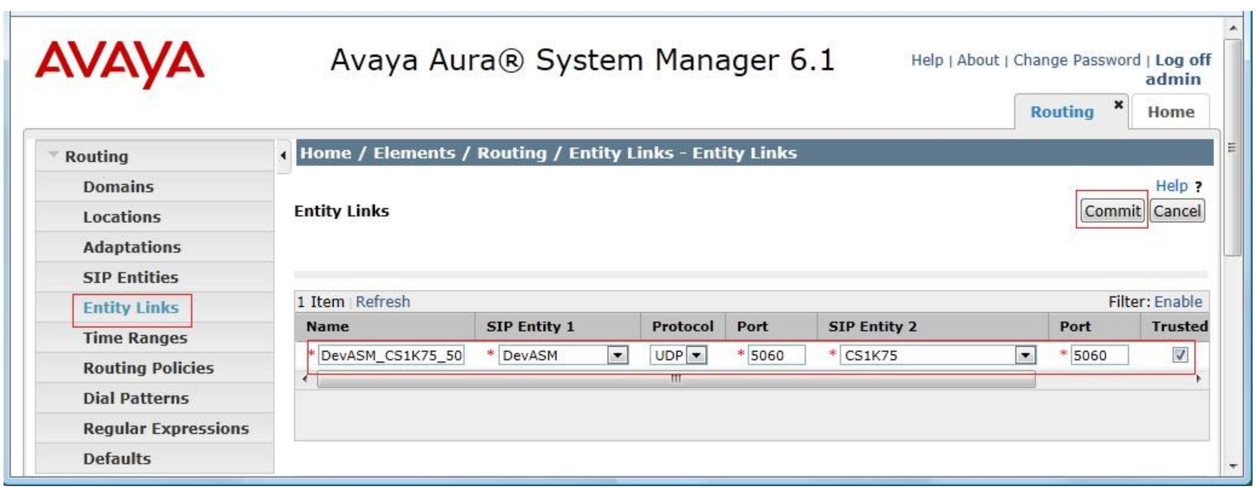

Figure 13: Avaya CS1000 and Session Manager Entity Link (UDP)

5.1.6 Create Time Ranges

From the left menu column, click on the **Time Ranges → New**, the **Time Ranges** page will appear as shown in **Figure 14**. Enter **Name** of the time range and others as shown in red-box.

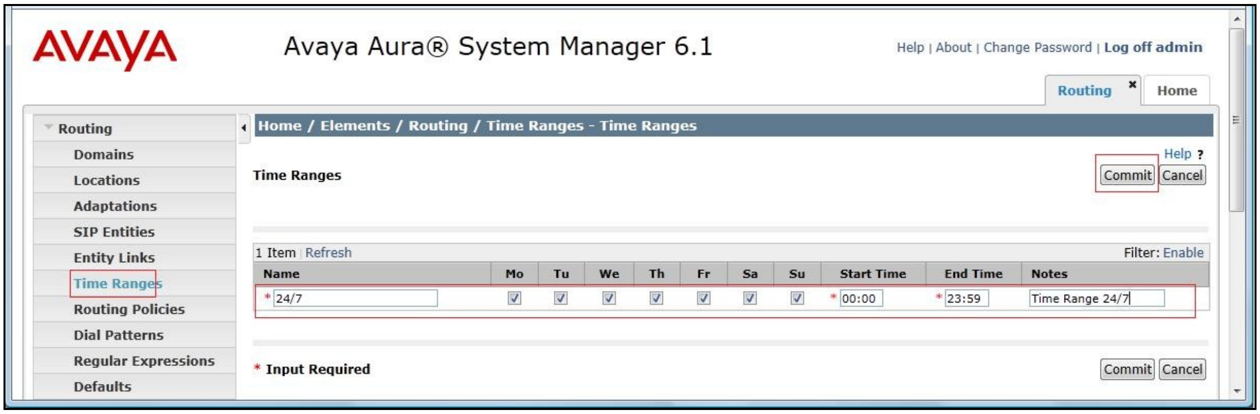

Figure 14: Time Ranges

5.1.7 Create Routing Policies

From the left menu column, click on **Routing Policies → New**, the **Routing Policy** page will appear as shown in **Figure 15**. Enter policy **Name** and **Notes** (optional).

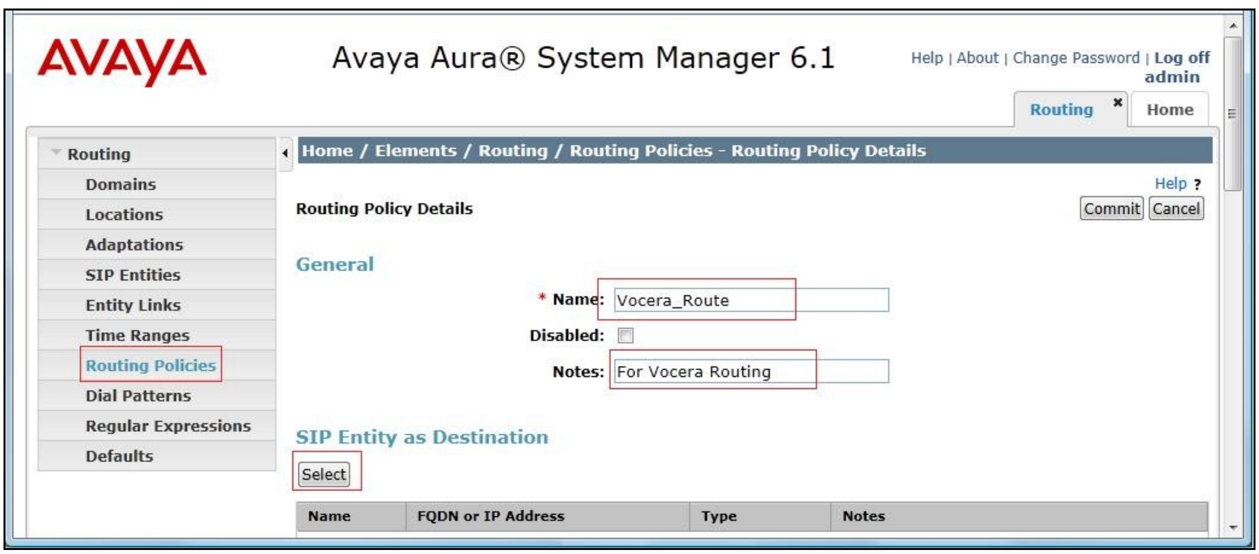

Figure 15: Routing Policy Details

From the **SIP Entity as Destination** as shown in **Figure 15**, click on **Select** button to select the SIP entity **Name *Vocera,*** which was created in **Section 5.1.4** as shown in **Figure 16**. Click **Commit**.

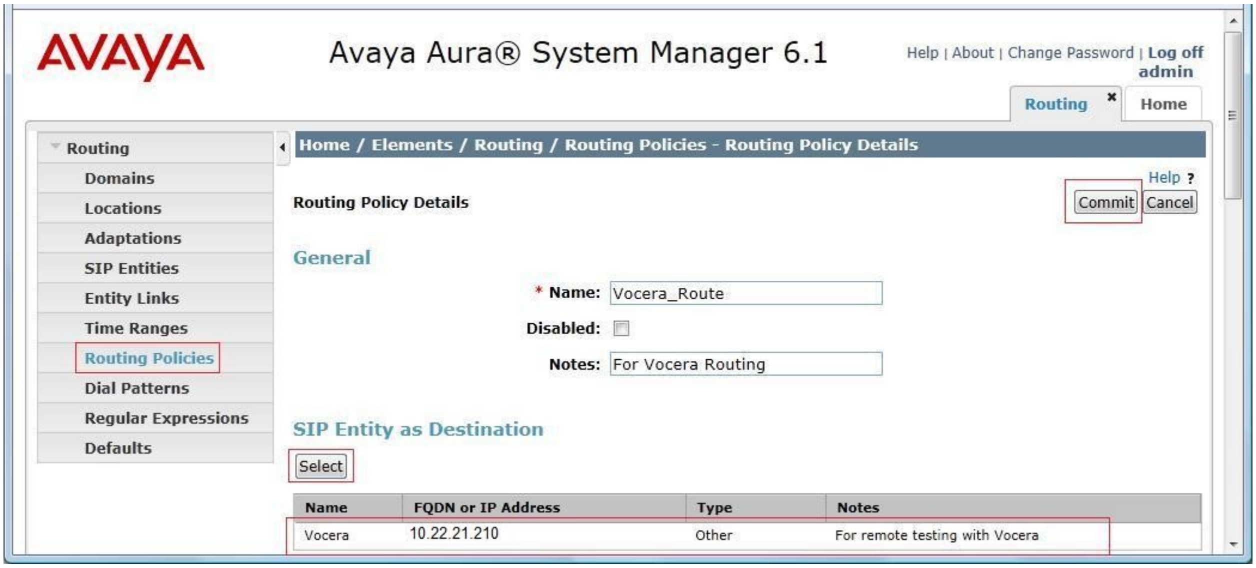

Figure 16: Select SIP Entity as Destination

5.1.8 Create Dial Patterns

From the left menu column, click on **Dial Patterns → New**, the **Dial Pattern Details** page will appear as shown in **Figure 17**. Enter dial **Pattern**, **Min** and **Max** values as shown in red-box. Choose **SIP domain** as it was created in **Section 5.1.1**. Click on **Add** button to add **Originating Locations and Routing Policies** for the newly created dial pattern.

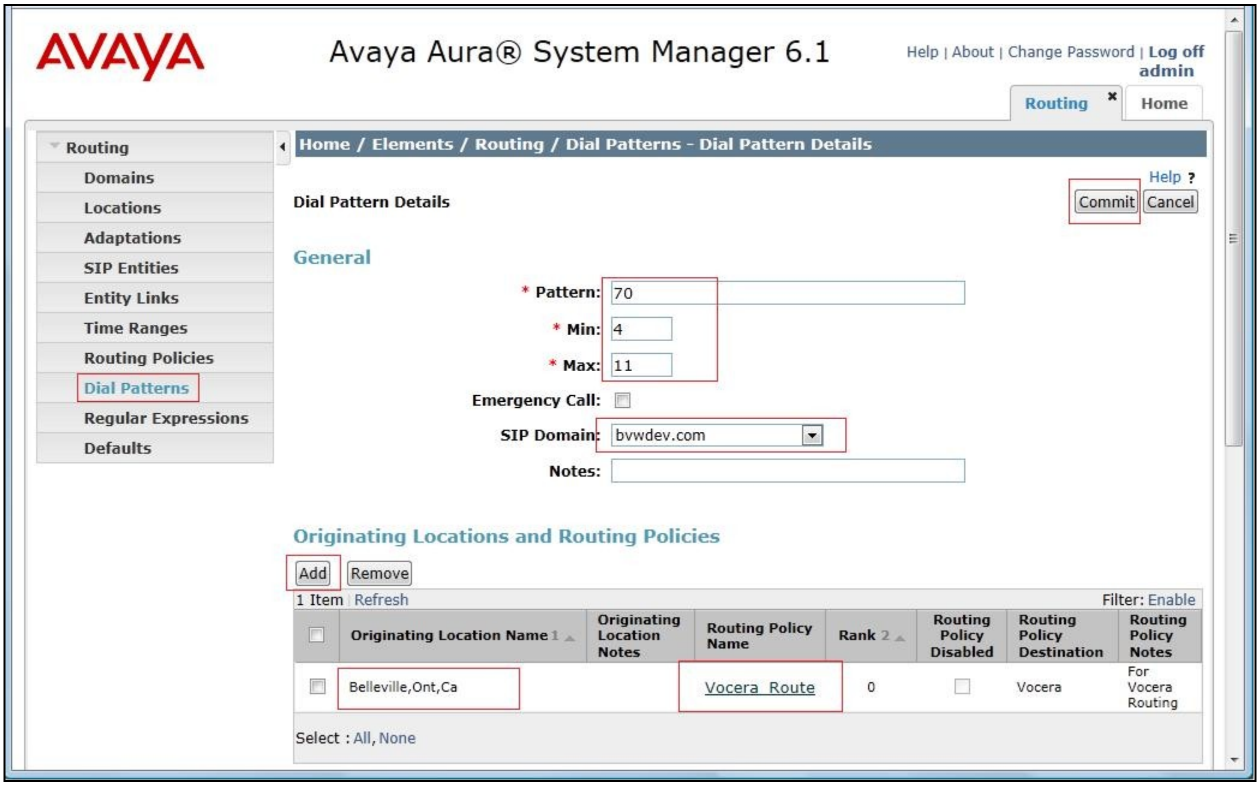

Figure 17: Dial Pattern Details

5.2 Configure Avaya CS1000

The assumption is that the route/trunk and dialing plan of the Avaya CS1000 have been configured. This section only describes the details on how to configure Avaya CS1000 to connect to the Session Manager via SIP Signaling Gateway using the Element Manager.

Prerequisites:

- An Avaya CS1000 server which has been:
  - Installed with Avaya CS1000 Release 7.5 Linux Base.
  - Joined Avaya CS1000 Release 7.5 Security Domain.
  - Deployed with SIP Trunk Application.
  - For more information on Avaya CS1000 installation, maintenance, and upgrades, see **Section 9**.

- The following software packages are enabled in the key-code.
- If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP	406	SIP Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

Log on the UCM Common Services of the Avaya CS1000, using the Microsoft Internet Explorer 6.0260 or later to access the UCM by addressing the IP address or FQDN (Full Qualified Domain Name) of the UCM and then input the username/password which was defined during the primary security server setup.

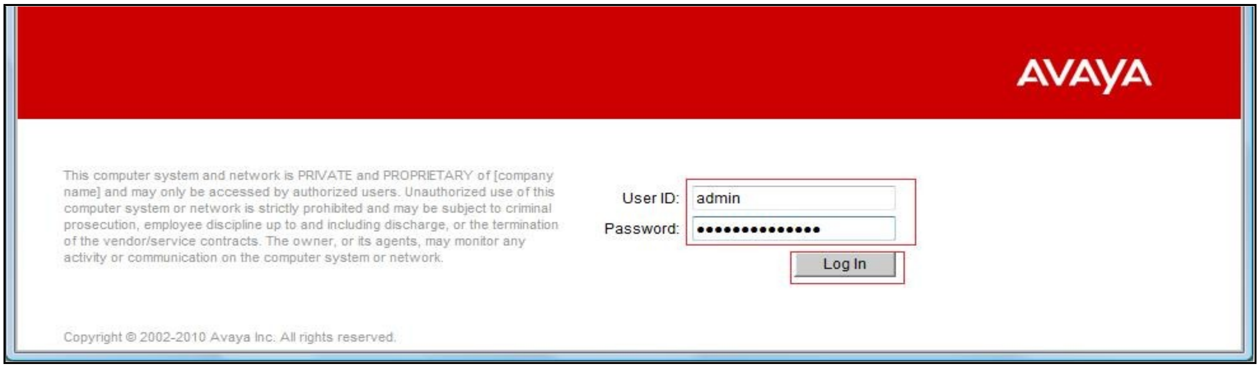

Figure 18: Log On Screen of UCM

After log on to the UCM, the **Avaya Unified Communications Management** is as shown in **Figure 19.**

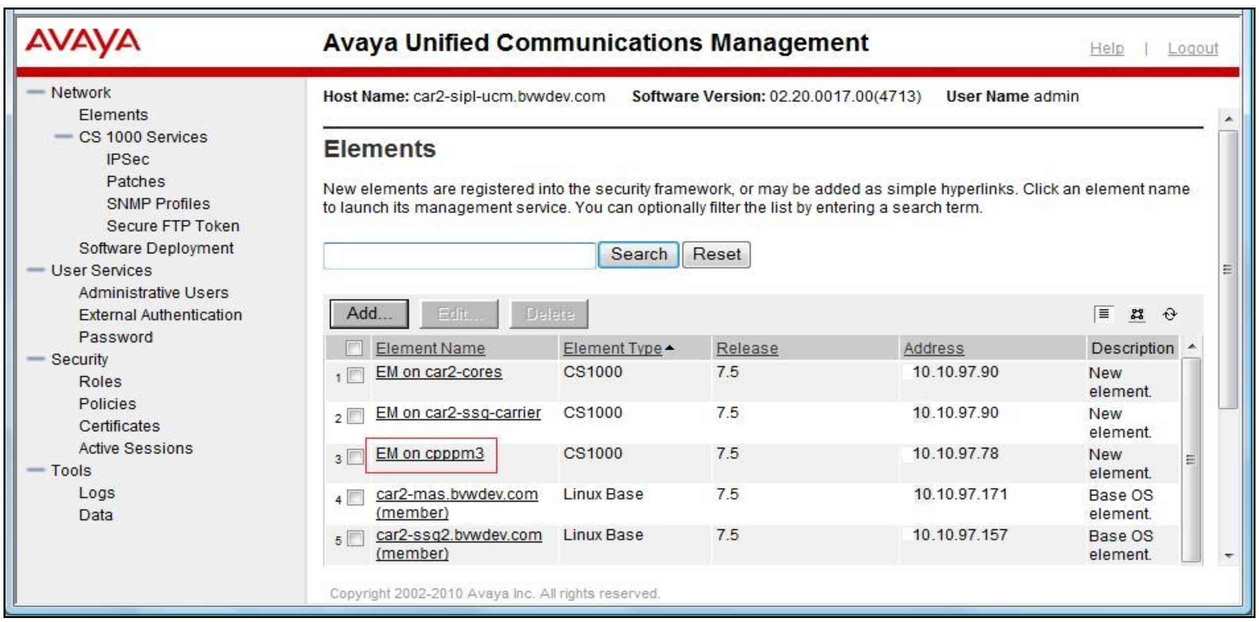

Figure 19: Avaya CS1000 Unified Communications Management

From **Figure 19**, click on the Avaya CS1000 CS element highlighted in red-box, the **System Overview** (EM) home page will appear as shown in **Figure 20**.

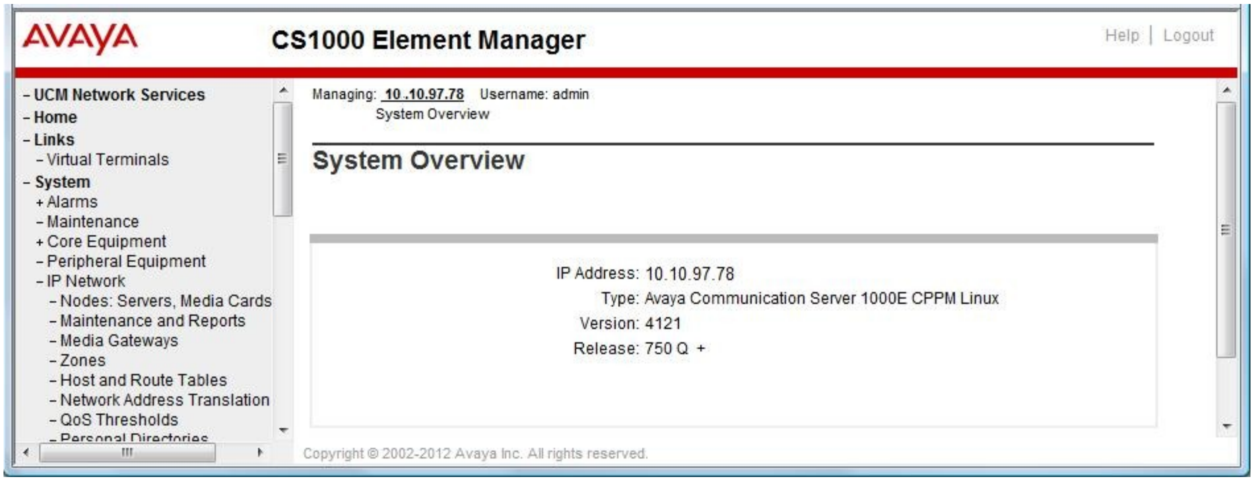

Figure 20: Element Manager Home Page

From the left menu column of the EM page, navigate to **System → IP Network → Nodes ID: Server Media Cards**. The **Node ID Telephone** page will appear (not shown).
Click on the **Node ID # 511,** which is **the *LTPS, Gateway (SIPGw)***. The **Node Details** page will appear as shown in **Figure 21.**

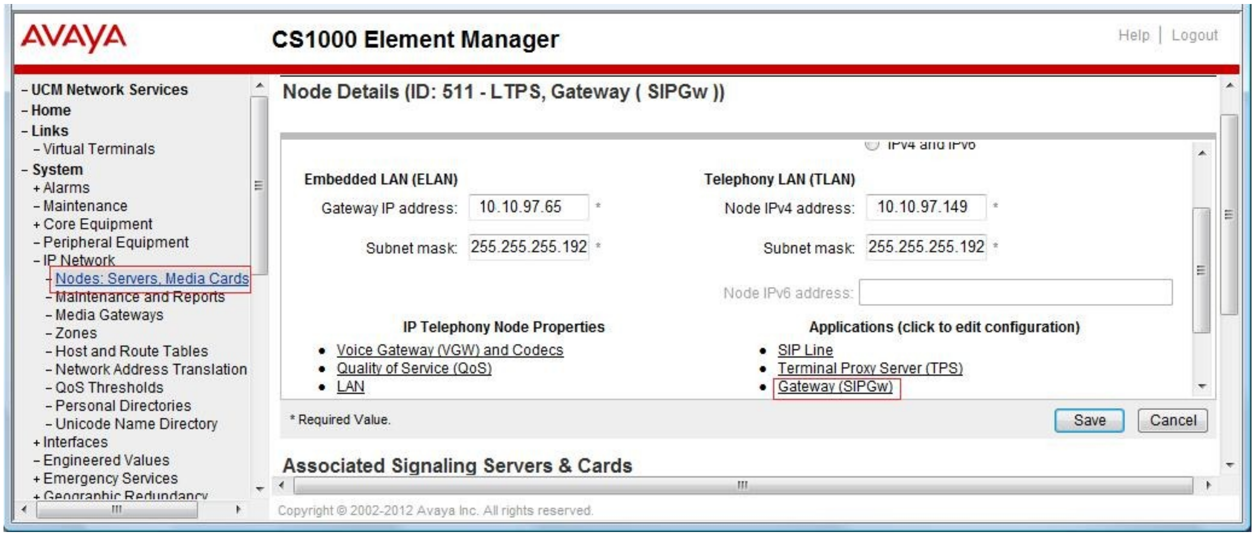

Figure 21: Node Details

Scroll down under the **Applications** column, click on the **Gateway (SIPGw)** link, the **Virtual Trunk Gateway Configuration Details** page will appear as shown in **Figure 22 and 23**. Enter the information highlighted in the red-box for the **General** and **SIP Gateway Settings**. Others are left at default. Click **Save**.
Note: **SIP domain name** should be matched with what was created in **Section 5.1.1**.

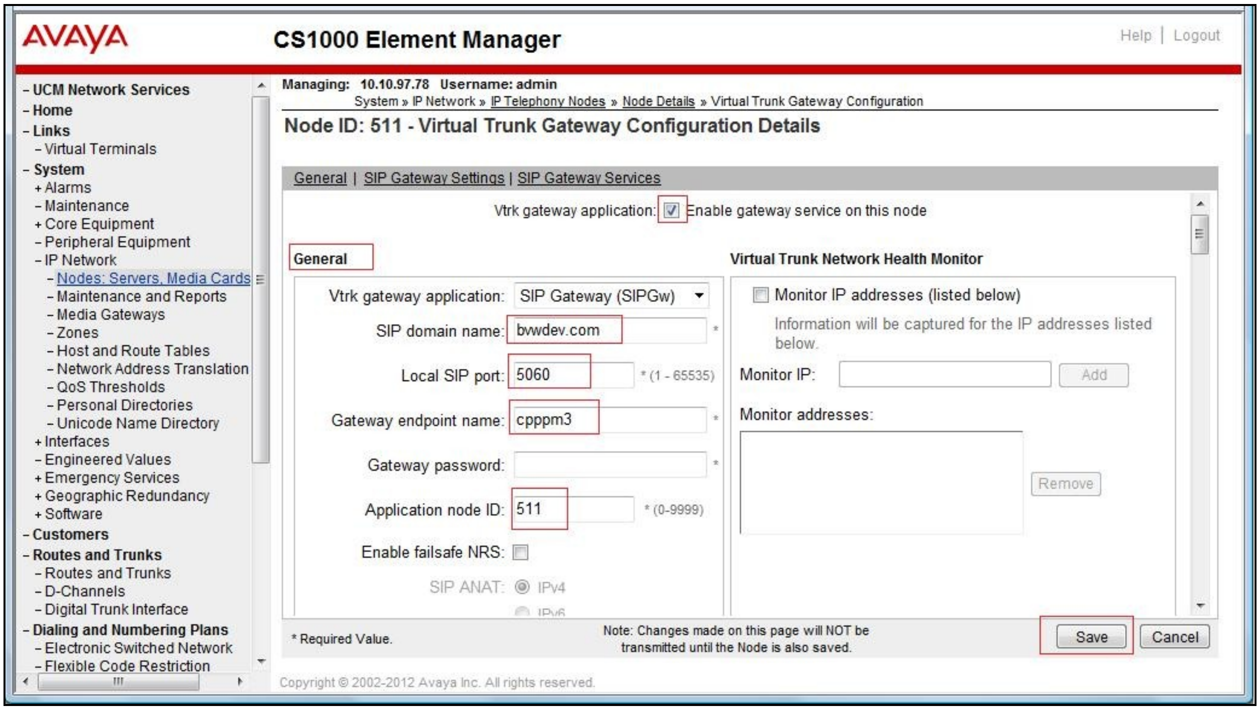

Figure 22: Virtual Trunk Gateway Configuration Details (General)

The **Primary TLAN IP** address is the IP address used in **Section 5.4**, the Session Manager IP address.

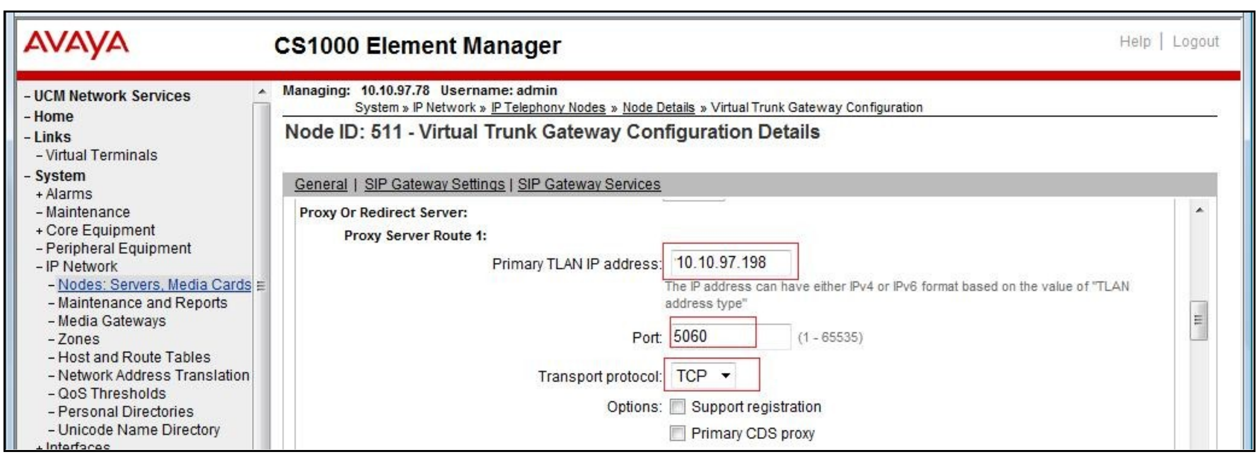

Figure 23: Virtual Trunk Gateway Configuration Details (SIP Gateway Settings)

On the same page, as shown in **Figure 22**, scroll-down the parameters box to the **SIP URI Map** section. Under the **Public E.164 Domain Names**

- **Special Number**: leave this SIP URI field as blank
- **Unknown**: leave this SIP URI field as blank
- **Vacant number**: leave this SIP URI field as blank
- **National**: leave this SIP URI field as blank

The remaining fields can be left at their default values as shown in **Figure 24**. Click **Save**. **Note**: This will remove the phone context information in the SIP invite URL.

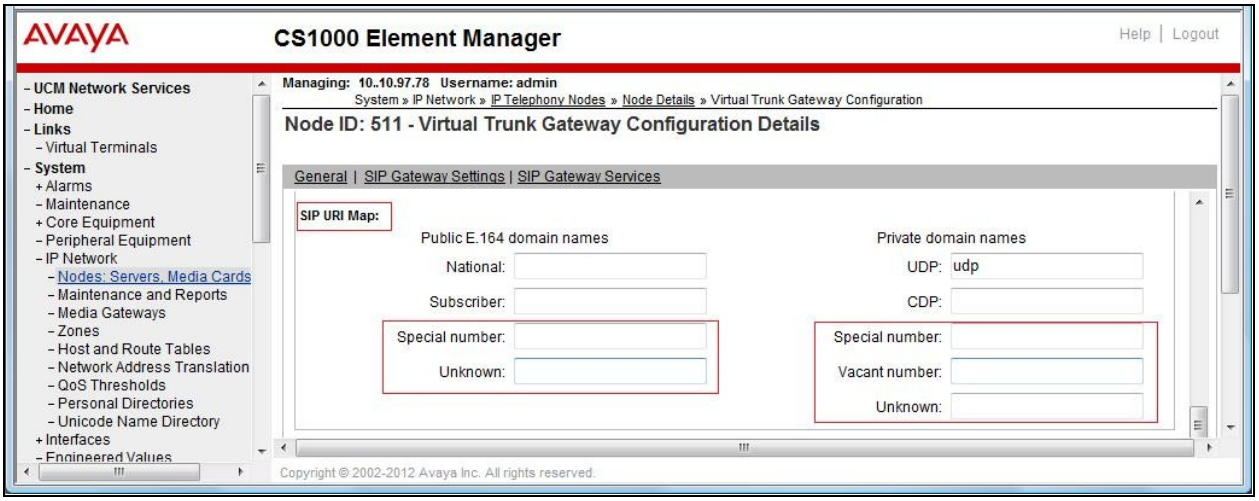

Figure 24: Virtual Trunk Gateway Configuration Details (SIP URI Map)

After click **Save** button, the system will bring back the **Node ID** page (not shown). Then click **Save** button on the **Node ID** page and that will take the user to the **Node Saved** page (not shown). Click on the **Transfer Now** button, when it finished, it will bring user to **Synchronize Configuration Files** page (not shown). Then click **Start Sync** button to complete the configuration saved process.

6. Configure Vocera Server

This section describes how to configure the Vocera Communication System to inter-work with Avaya CS1000.

6.1 Configuring Vocera SIP Connectivity to Avaya CS1000

Open the Vocera Communication Systems web page by entering 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 (not shown), then click on the **Vocera Administration Console** link to get to the console web page as shown in **Figure 25**.

Figure 25: Vocera Administration Console

Input the user name and password to log on the **Console page**, click on the **Log In** button to log in. The screen shown in **Figure 26** will appear with the **Status Monitor** menu page as default.

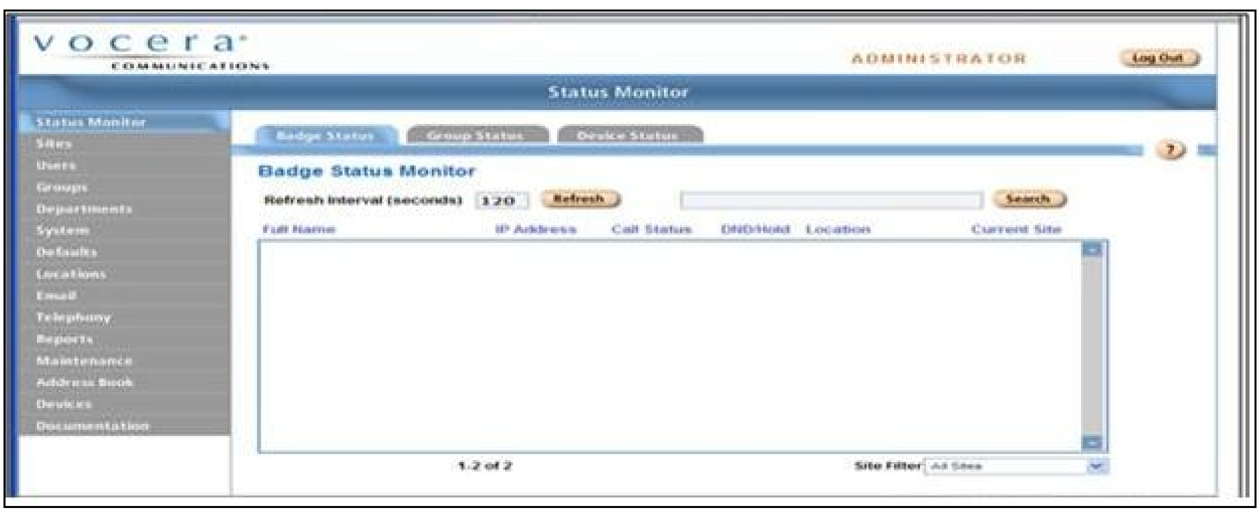

Figure 26: Administrator Console Web Page

For all the details on the configuration of Vocera Communication System, user can click on the **Documentation** option on the left column menu. 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 menu column. The **Telephony** page will appear with the **Basic Info** menu tab being selected as default, as shown in **Figure 26**. Fill in the details of the highlighted attributes in the red-boxes. Others fields are at default. Then Click **Save Changes** button.

Figure 26: Telephony Configuration

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 **Figure 27**. Then click **Save Changes** button.

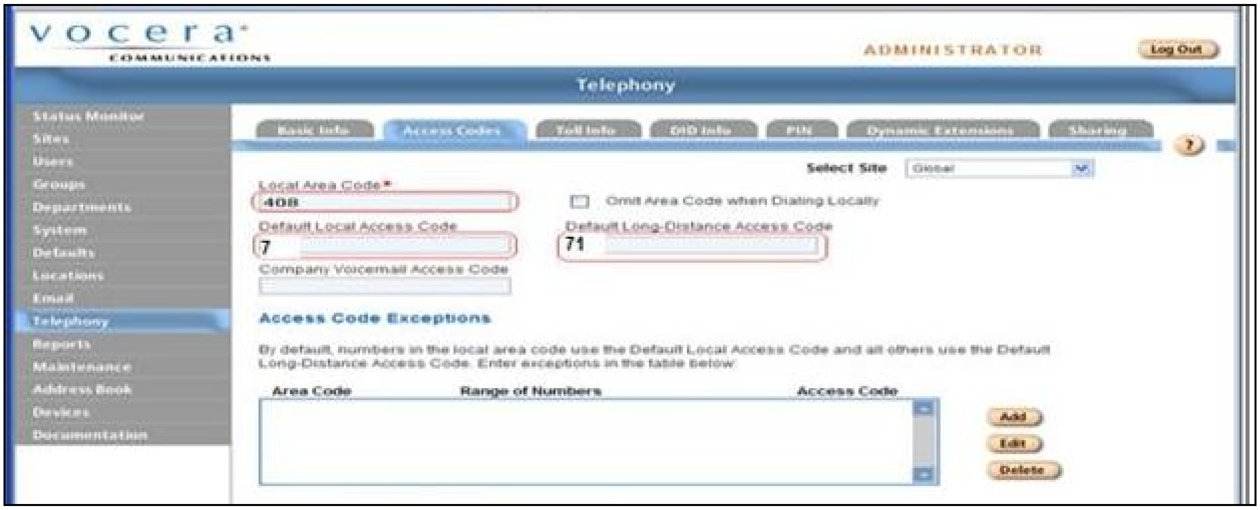

Figure 27: Access Codes Configuration

6.2 Configure Users on the Vocera Server

To configure the users on the Vocera Server to be able to send to and receive calls from the Avaya CS1000, as shown in **Figure 27** above, click on the **Users** menu option. The **Users** page will appear as shown in **Figure 28**.

Figure 28: User Page

To add a user, click **Add New User** button, the user **Info** detail configuration page will appear as shown in **Figure 29**. 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**.

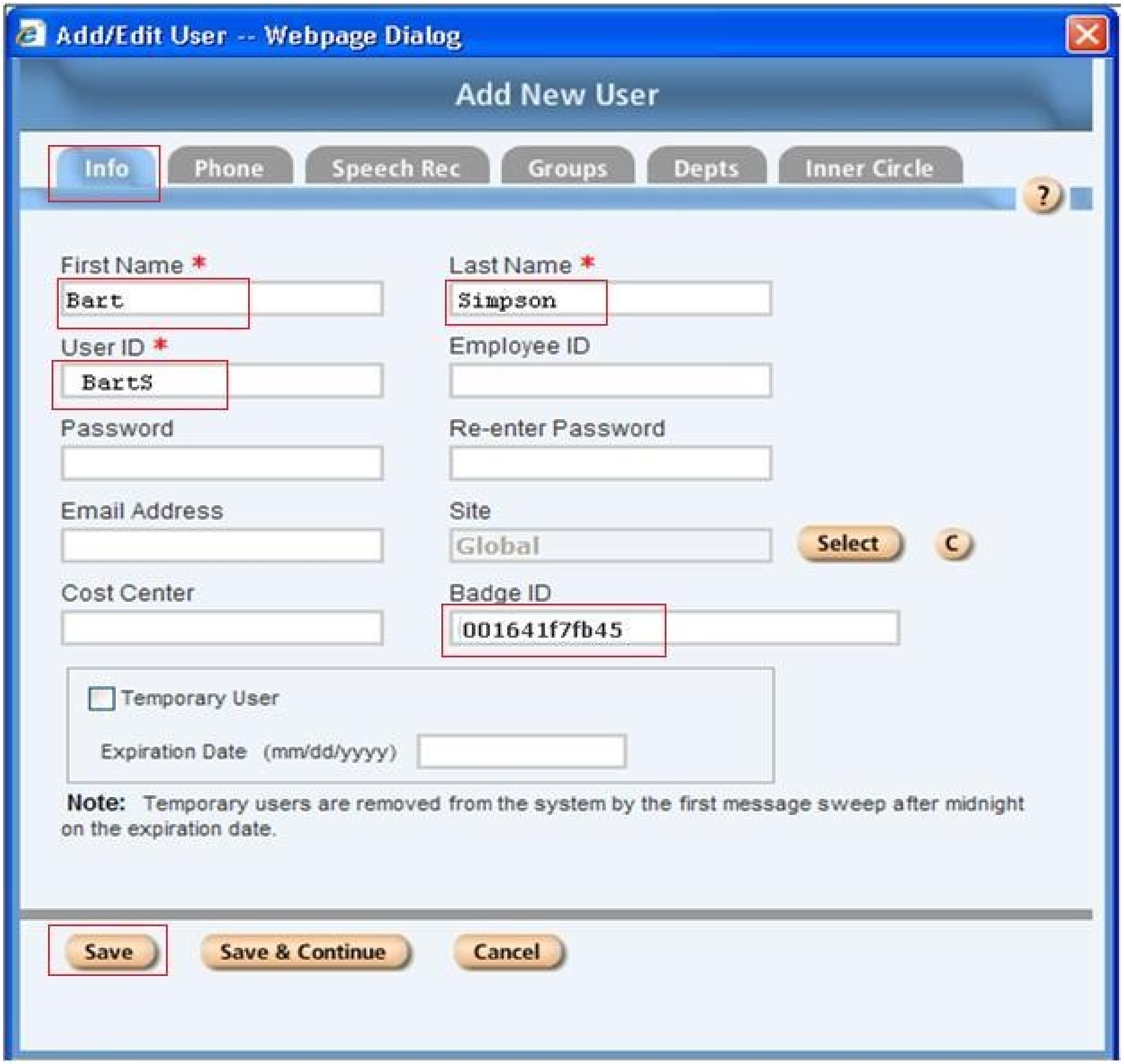

Figure 29: System VOIP Configuration

From the **Add New User** page, click on the **Phone** tab to configure user specific phone number information such as **Desk phone or Extension, Home phone,** as shown in **Figure 30**. Others fields are optional. Click **Save**.

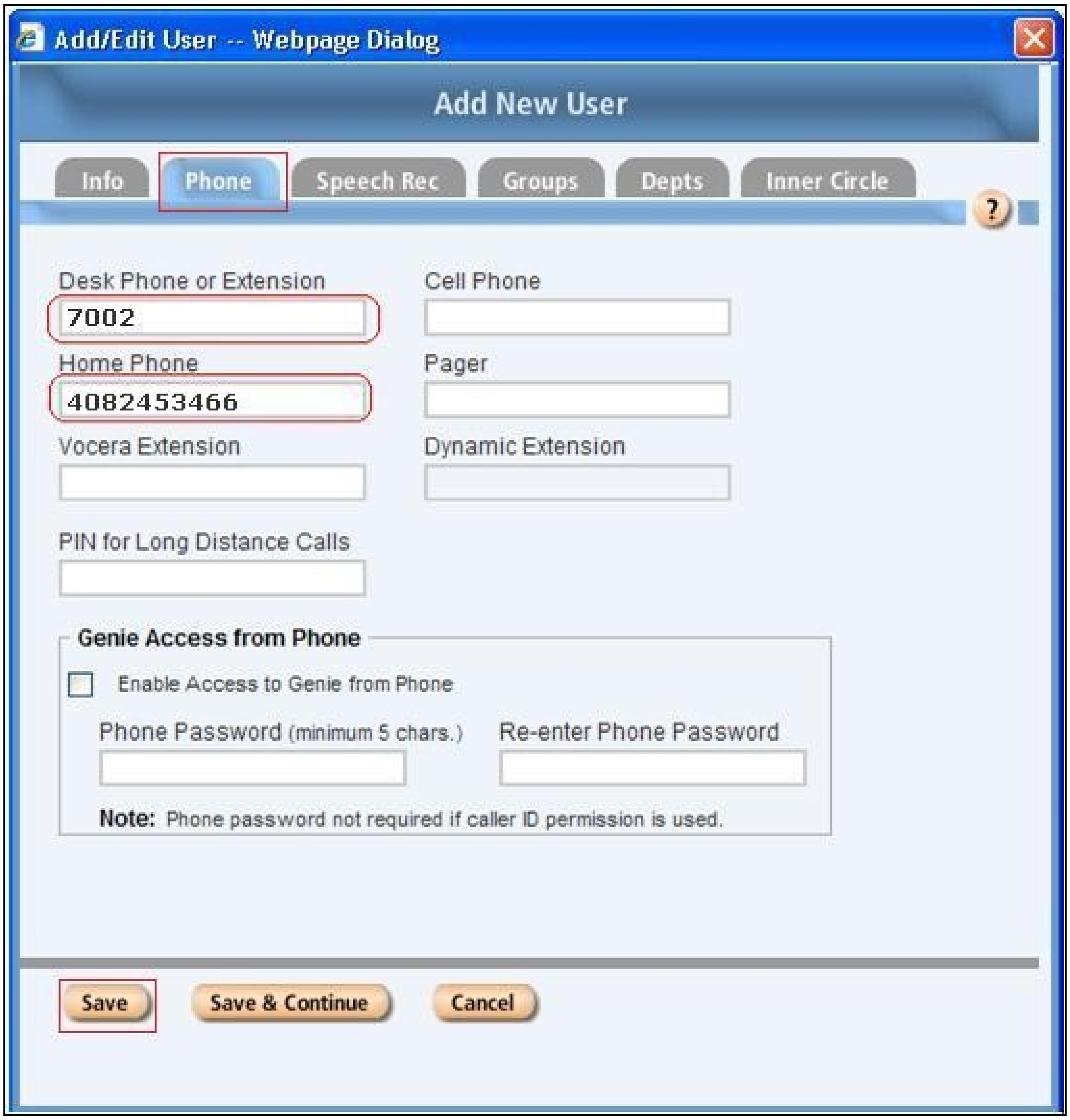

Figure 30: Phone Configuration

Click on the **Group** tab to assign a newly create 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 belonged to the **Site *Global*** (not shown).
For detail configuration on how these **Groups** and **Sites** are configured, please refer to the **Administration Guide** by clicking on the **Documentation** option menu on the left menu panel (not shown), under the **Administration and Configuration**.

7. Verification Steps

The following are typical steps to verify the interoperability between the Vocera Server and Avaya CS1000, please also refer to the **Figure 1** for more detail.

- Step 1: Place a call from an IP phone of Avaya CS1000 to the Vocera Server by entering the assigned DN number.
- Step 2: A voice greeting from the Vocera Server should be heard on the IP phone telling the caller to speak a full name or an extension of the callee.
- Step 3: When a spoken full name or an extension of the callee is received by the Vocera Server, it will redirect the call to the wireless badge associated with the assigned extension.
- Step 4: The user on the Vocera Server will hear a Attendant voice asking if the user would like to pick up the call. User presses the big circular button on the badge, to accept the call.
- Step 5: Verify that there are clear 2-way voice path between Avaya IP phone and the Vocera wireless badge.

8. Conclusion

These Application Notes have described the administration steps required to integrate the Vocera Communication System with the Avaya Communication Server 1000 via SIP trunk configured on the Avaya Aura® Session Manager. All test cases passed with observations noted in **Section 2.2**.

9. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.

[2] *Communication Server 1000 Installation and Commissioning*, April 2012, Release 7.5, Issue 05.08, Document Number NN43041-310.

[3] *Signaling Server IP Line Applications Fundamentals for Avaya Communication Server 1000 (Avaya CS 1000)*, April 2012, Release 7.5, Issue 03.11, Document Number NN43001-125.

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.