



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.

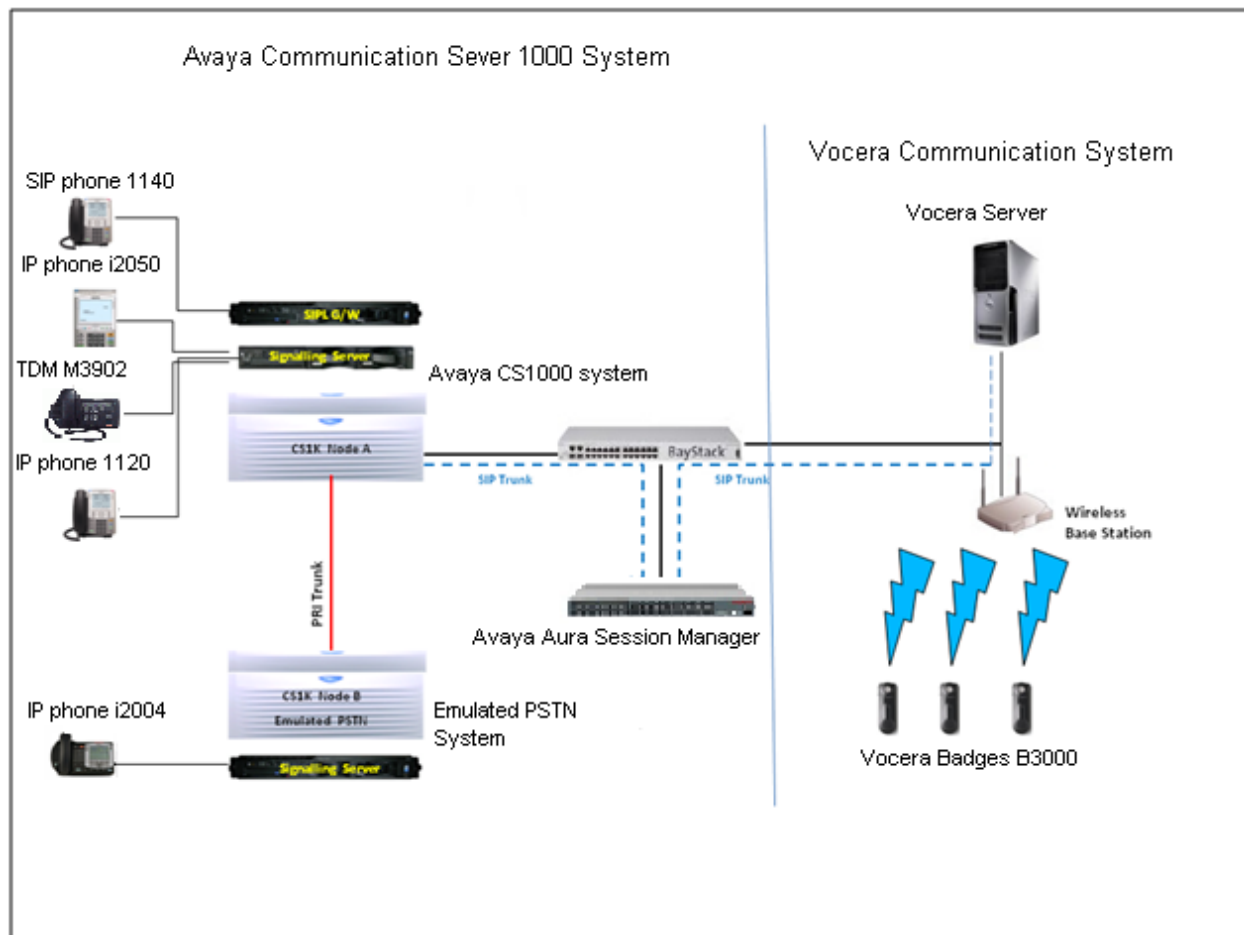


Figure 1: Avaya Communication Server 1000 Network with the Vocera Communication System connecting to Avaya Session Manager via SIP trunk.

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** below:

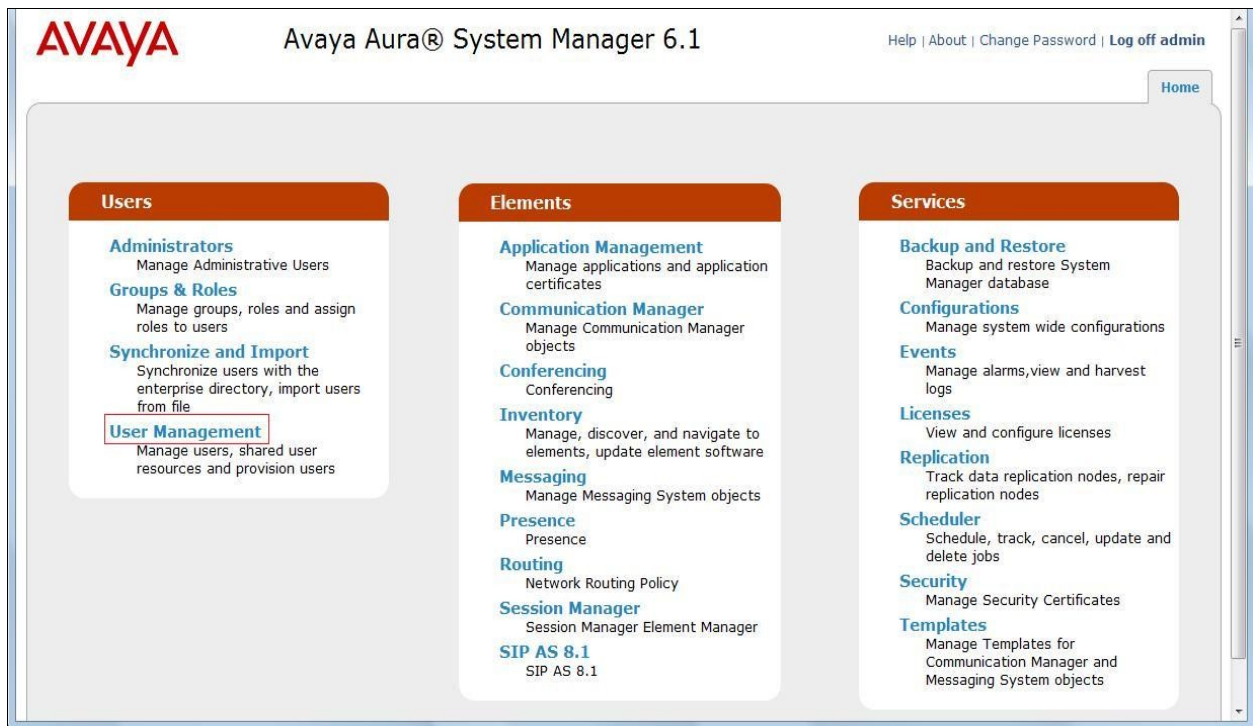


Figure 2: System Manager Home Page

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.

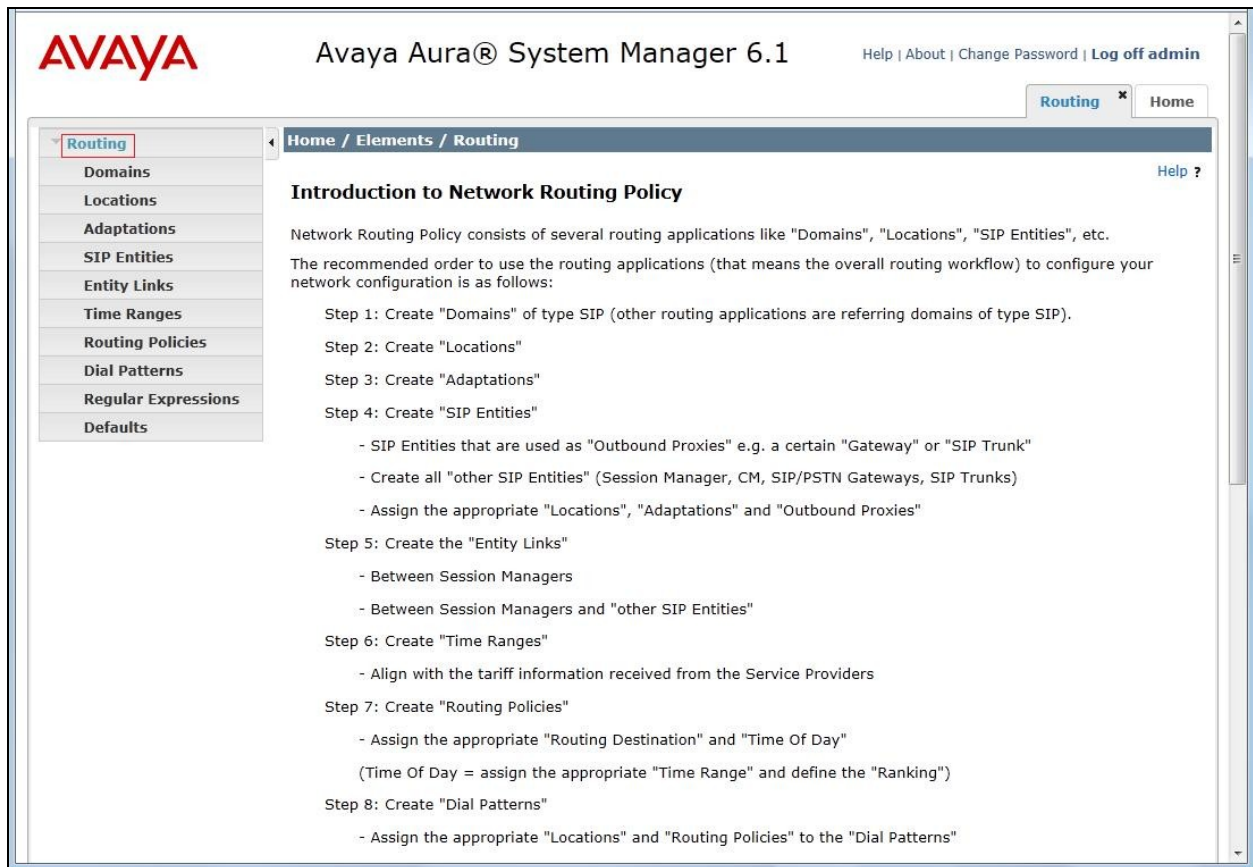


Figure 3: New User Profile

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**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with the following items: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The 'Domains' item is highlighted. The main content area is titled 'Domain Management' and shows a table with the following columns: Name, Type, Default, and Notes. The table contains one row with the following data: Name: bvwddev.com, Type: sip, Default: (checkbox), and Notes: For SIP. The 'Commit' button is highlighted in the top right corner of the main content area.

Name	Type	Default	Notes
* bvwddev.com	sip	<input type="checkbox"/>	For SIP

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**.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Locations - Location Details

Location Details

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. See Session Manager -> Session Manager Administration -> Global Setting

Help ?

Commit Cancel

General

* Name: Belleville, Ont, Ca

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth: 1000000

Per-Call Bandwidth Parameters

* Default Audio Bandwidth: 80 Kbit/sec

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.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with the following items: Routing, Domains, Locations, Adaptations (highlighted with a red box), SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and 'General'. It contains the following fields and controls:

- Adaptation name:** Vocera_CS1K75
- Module name:** DigitConversionAdapter (dropdown menu)
- Module parameter:** MIME=no
- Egress URI Parameters:** (empty field)
- Notes:** Out bound to Vocera

Buttons for 'Commit', 'Cancel', and 'Help ?' are located in the top right corner of the form area.

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 *bvwddev.com* as it was created in **Section 5.1.1**. Click **Commit**.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details Help ?

General

* Name: DevASM

* FQDN or IP Address: 10.10.97.198

Type: Session Manager

Notes: For Session Manager

Location: Belleville, Ont, Ca

Outbound Proxy:

Time Zone: America/Toronto

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Entity Links

This SIP Entity contains a large number of Entity Links (more than 50). Please navigate to the Entity Links table page on the left side menu in order to edit the relevant entity links or click [Here](#). Note that navigating to the entity link page will lose all unsaved changes in this page.

Port

Add Remove

2 Items Refresh Filter: Enable

	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	bvwddev.com	
<input type="checkbox"/>	5060	UDP	bvwddev.com	

Select : All, None

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 Entity 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**.

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The left-hand navigation menu is expanded, showing 'SIP Entities' as the selected option. The main content area is titled 'SIP Entity Details' and includes a breadcrumb trail: 'Home / Elements / Routing / SIP Entities - SIP Entity Details'. The 'General' tab is active, showing the following configuration fields:

- Name:** Vocera
- * FQDN or IP Address:** 10.22.21.210
- Type:** Other
- Notes:** For remote testing with Vocera
- Adaptation:** Vocera_CS1K75
- Location:** Belleville, Ont, Ca
- Time Zone:** America/New_York
- Override Port & Transport with DNS SRV:** ☐
- * SIP Timer B/F (in seconds):** 4
- Credential name:** (empty field)
- Call Detail Recording:** none
- SIP Link Monitoring:** Use Session Manager Configuration

The 'Commit' button is highlighted in the top right corner of the form area.

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**.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing × **Home**

Routing × **Home**

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

General

* **Name:** CS1K75

* **FQDN or IP Address:** 10.10.97.149

Type: Other

Notes:

Adaptation:

Location: Belleville, Ont, Ca

Time Zone: America/Toronto

Override Port & Transport with DNS SRV:

* **SIP Timer B/F (in seconds):** 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Commit **Cancel** **Help ?**

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**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with 'Entity Links' highlighted. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The row shows 'DevASM_Vocera_506' as the Name, 'DevASM' as SIP Entity 1, 'TCP' as Protocol, '5060' as Port, 'Vocera' as SIP Entity 2, '5060' as Port, and 'For Vocera' as Notes. The 'Trusted' checkbox is checked. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* DevASM_Vocera_506	* DevASM	TCP	* 5060	* Vocera	* 5060	<input checked="" type="checkbox"/>	For Vocera

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**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with 'Entity Links' highlighted. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The row shows 'DevASM_Vocera_506' as the Name, 'DevASM' as SIP Entity 1, 'UDP' as Protocol, '5060' as Port, 'Vocera' as SIP Entity 2, '5060' as Port, and 'For Vocera' as Notes. The 'Trusted' checkbox is checked. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* DevASM_Vocera_506	* DevASM	UDP	* 5060	* Vocera	* 5060	<input checked="" type="checkbox"/>	For Vocera

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**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with 'Entity Links' highlighted. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Trusted. The row shows 'DevASM_CS1K75_50' as the Name, 'DevASM' as SIP Entity 1, 'TCP' as Protocol, '5060' as Port, 'CS1K75' as SIP Entity 2, '5060' as Port, and the 'Trusted' checkbox is checked. There are 'Commit' and 'Cancel' buttons at the top right.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
* DevASM_CS1K75_50	* DevASM	TCP	* 5060	* CS1K75	* 5060	<input checked="" type="checkbox"/>

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**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with 'Entity Links' highlighted. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Trusted. The row shows 'DevASM_CS1K75_50' as the Name, 'DevASM' as SIP Entity 1, 'UDP' as Protocol, '5060' as Port, 'CS1K75' as SIP Entity 2, '5060' as Port, and the 'Trusted' checkbox is checked. There are 'Commit' and 'Cancel' buttons at the top right.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
* DevASM_CS1K75_50	* DevASM	UDP	* 5060	* CS1K75	* 5060	<input checked="" type="checkbox"/>

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.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Time Ranges - Time Ranges

Time Ranges

Commit Help ? Cancel

1 Item Refresh Filter: Enable

Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
* 24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	* 00:00	* 23:59	Time Range 24/7

* Input Required

Commit Cancel

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).

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

Commit Help ? Cancel

General

* Name: Vocera_Route

Disabled: ☐

Notes: For Vocera Routing

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
------	--------------------	------	-------

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**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with the following items: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (highlighted), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Routing Policy Details' and includes a breadcrumb trail: 'Home / Elements / Routing / Routing Policies - Routing Policy Details'. The 'General' section contains the following fields: 'Name' (Vocera_Route), 'Disabled' (checkbox), and 'Notes' (For Vocera Routing). The 'SIP Entity as Destination' section is highlighted and contains a 'Select' button. Below this section is a table with the following data:

Name	FQDN or IP Address	Type	Notes
Vocera	10.22.21.210	Other	For remote testing with Vocera

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.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

Commit Help ? Cancel

General

* Pattern: 70

* Min: 4

* Max: 11

Emergency Call: ☐

SIP Domain: bvwddev.com

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville, Ont, Ca		Vocera Route	0	<input type="checkbox"/>	Vocera	For Vocera Routing

Select : All, None

Figure 17: Dial Pattern Details

5.2 Configure Avaya CS1000

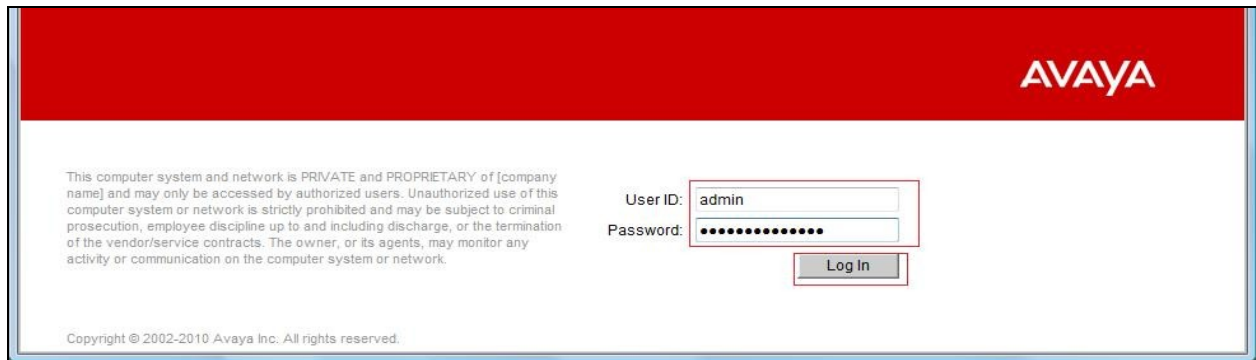
The assumption is that the route/trunk and dialing plan of the Avaya CS1000 have been configured. This section only describes the details on how to configure Avaya CS1000 to connect to the Session Manager via SIP Signaling Gateway using the Element Manager.

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	--

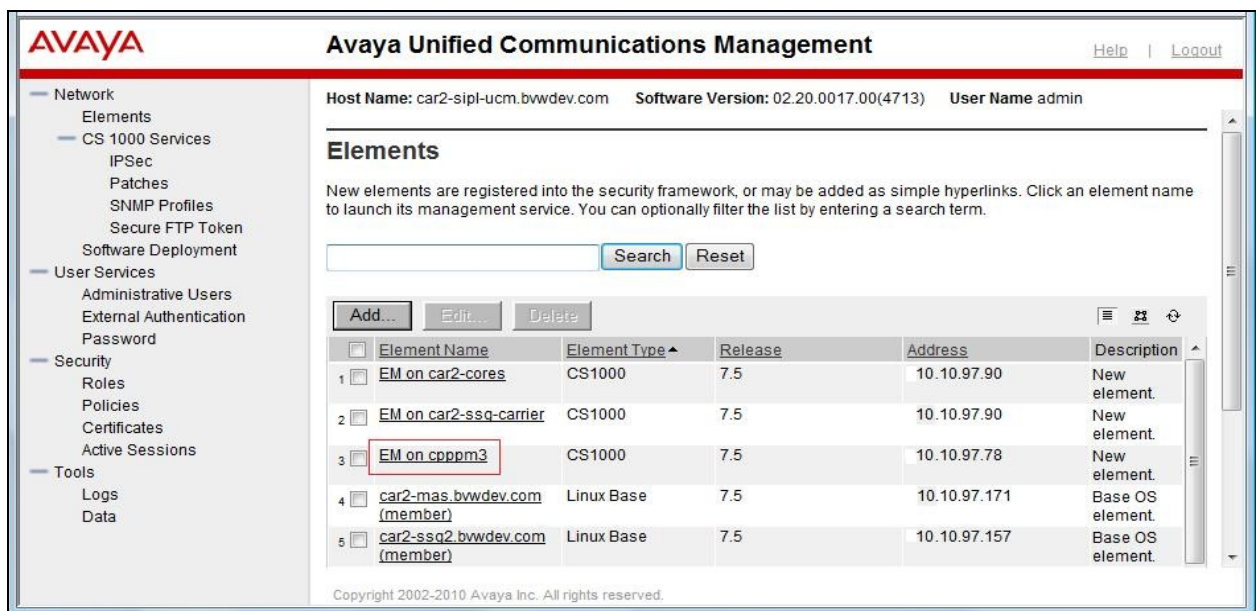
Log on the UCM Common Services of the Avaya CS1000, using the Microsoft Internet Explorer 6.0260 or later to access the UCM by addressing the IP address or FQDN (Full Qualified Domain Name) of the UCM and then input the username/password which was defined during the primary security server setup.



The screenshot shows the Avaya UCM login interface. At the top is a red header with the 'AVAYA' logo. Below the header, on the left, is a disclaimer: 'This computer system and network is PRIVATE and PROPRIETARY of [company name] and may only be accessed by authorized users. Unauthorized use of this computer system or network is strictly prohibited and may be subject to criminal prosecution, employee discipline up to and including discharge, or the termination of the vendor/service contracts. The owner, or its agents, may monitor any activity or communication on the computer system or network.' To the right of the disclaimer are input fields for 'User ID:' (containing 'admin') and 'Password:' (containing ten dots). Below these fields is a 'Log In' button. At the bottom left, it says 'Copyright © 2002-2010 Avaya Inc. All rights reserved.'

Figure 18: Log On Screen of UCM

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



The screenshot shows the Avaya Unified Communications Management dashboard. The top header has the 'AVAYA' logo, the title 'Avaya Unified Communications Management', and links for 'Help' and 'Logout'. Below the header, the dashboard displays 'Host Name: car2-sipl-ucm.bvwdev.com', 'Software Version: 02.20.0017.00(4713)', and 'User Name admin'. A left sidebar contains a navigation tree with categories: Network (Elements, CS 1000 Services, IPSec, Patches, SNMP Profiles, Secure FTP Token, Software Deployment), User Services (Administrative Users, External Authentication, Password), Security (Roles, Policies, Certificates, Active Sessions), and Tools (Logs, Data). The main content area is titled 'Elements' and includes a description: 'New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.' Below this is a search bar with 'Search' and 'Reset' buttons. A table lists elements with columns: Element Name, Element Type, Release, Address, and Description. The table contains five entries, with the third entry 'EM on cpppm3' highlighted by a red box. At the bottom, it says 'Copyright 2002-2010 Avaya Inc. All rights reserved.'

	Element Name	Element Type	Release	Address	Description
1	EM on car2-cores	CS1000	7.5	10.10.97.90	New element.
2	EM on car2-ssq-carrier	CS1000	7.5	10.10.97.90	New element.
3	EM on cpppm3	CS1000	7.5	10.10.97.78	New element.
4	car2-mas.bvwdev.com (member)	Linux Base	7.5	10.10.97.171	Base OS element.
5	car2-ssq2.bvwdev.com (member)	Linux Base	7.5	10.10.97.157	Base OS element.

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**.

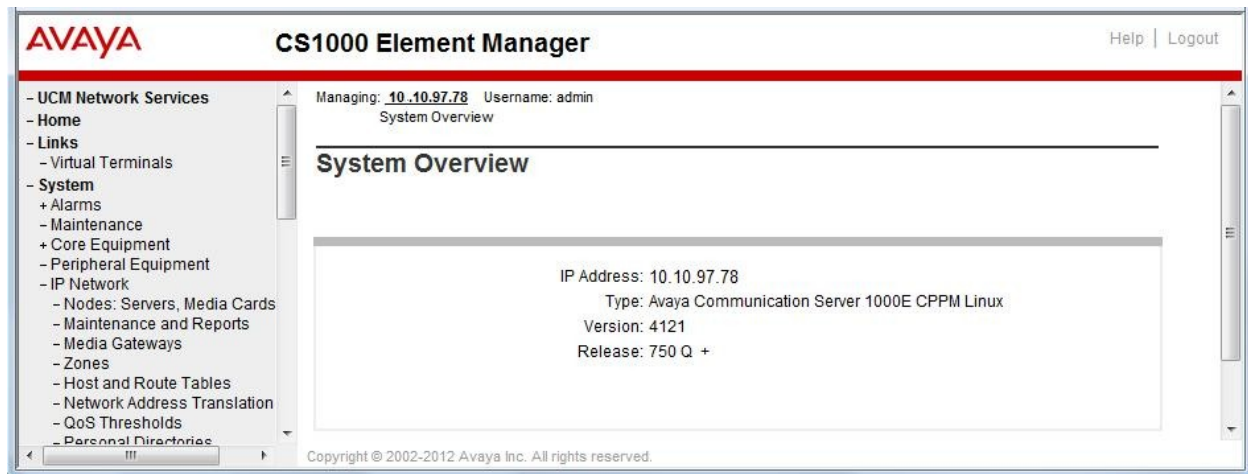


Figure 20: Element Manager Home Page

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**.

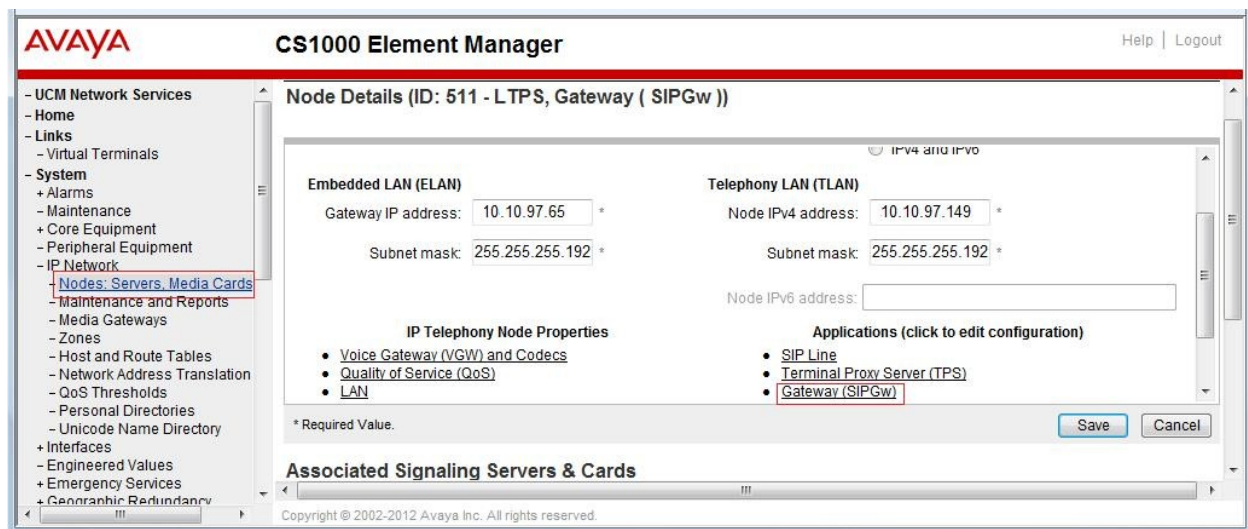


Figure 21: Node Details

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**.

AVAYA CS1000 Element Manager

Managing: 10.10.97.78 Username: admin

System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 511 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw)

SIP domain name: bwdev.com

Local SIP port: 5060 * (1 - 65535)

Gateway endpoint name: cpppm3

Gateway password: *

Application node ID: 511 * (0-9999)

Enable failsafe NRS: ☐

SIP ANAT: IPv4

Virtual Trunk Network Health Monitor

☒ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP: Add

Monitor addresses: Remove

* Required Value.

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

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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.

AVAYA CS1000 Element Manager

Managing: 10.10.97.78 Username: admin

System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 511 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.97.198

The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP

Options: ☒ Support registration ☐ Primary CDS proxy

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.

The screenshot shows the Vocera Communications Administrator interface. The left sidebar contains navigation links: Status Monitor, Sites, Users, Groups, Departments, System, Defaults, Locations, Email, Telephony (highlighted), Reports, Maintenance, Address Book, Devices, and Documentation. The main content area is titled 'Telephony' and has tabs for Basic Info, Access Codes, Toll Info, DID Info, PIN, Dynamic Extensions, and Sharing. The Basic Info tab is selected. The configuration fields are as follows:

- Enable Telephony Integration:** Checked (indicated by a red box).
- Vocera Hunt Group Numbers:**
 - Guest Access:** 7000 (highlighted with a red box).
 - Direct Access:** 7005 (highlighted with a red box).
- Number of Lines:** 6 (highlighted with a red box).
- Integration Type:** Digital (selected, highlighted with a red box). Note: Saving any changes to digital parameters will cause the telephony server to restart.
- IP Settings:**
 - Signaling Protocol:** SIP Version 2.0 (highlighted with a red box).
- SIP Settings:**
 - Call Signaling Address:** 10.10.97.198 (highlighted with a red box).
 - Calling Party Number:** 7000 (highlighted with a red box).

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**.

Type	Full Name	Site	User ID
	B. Nish	Global	nish
	C. mark	Global	mark
	D. adarsh	Global	adarsh
	g. mary	Global	mary
	S. bart	Global	bart
	S. Maggie	Global	Maggie
	S. Marge	Global	Marge
	Simpson, homer	Global	homer

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**.

The screenshot shows a web dialog titled "Add/Edit User -- Webpage Dialog" with a sub-header "Add New User". The "Info" tab is selected, and other tabs include "Phone", "Speech Rec", "Groups", "Depts", and "Inner Circle". The form contains the following fields and controls:

- First Name ***: Text box containing "Bart".
- Last Name ***: Text box containing "Simpson".
- User ID ***: Text box containing "BartS".
- Employee ID**: Empty text box.
- Password**: Empty text box.
- Re-enter Password**: Empty text box.
- Email Address**: Empty text box.
- Site**: Text box containing "Global", with a "Select" button and a "C" button next to it.
- Cost Center**: Empty text box.
- Badge ID**: Text box containing "001641f7fb45".
- ☐ **Temporary User**: A checkbox with the label "Temporary User".
- Expiration Date (mm/dd/yyyy)**: Empty text box.

A note at the bottom states: "Note: Temporary users are removed from the system by the first message sweep after midnight on the expiration date." At the bottom of the dialog are three buttons: "Save", "Save & Continue", and "Cancel".

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**.

The screenshot shows a web-based dialog box titled "Add/Edit User -- Webpage Dialog". Inside, there's a section titled "Add New User" with several tabs: "Info", "Phone", "Speech Rec", "Groups", "Depts", and "Inner Circle". The "Phone" tab is currently selected and highlighted with a red rectangular box. Below the tabs, there are several input fields for phone-related information. The "Desk Phone or Extension" field contains the number "7002", and the "Home Phone" field contains "4082453466". Other fields include "Cell Phone", "Pager", "Vocera Extension", "Dynamic Extension", and "PIN for Long Distance Calls". Below these fields is a section titled "Genie Access from Phone" which contains a checkbox labeled "Enable Access to Genie from Phone" (which is unchecked), two password fields labeled "Phone Password (minimum 5 chars.)" and "Re-enter Phone Password", and a note stating "Note: Phone password not required if caller ID permission is used." At the bottom of the dialog, there are three buttons: "Save", "Save & Continue", and "Cancel". The "Save" button is highlighted with a red rectangular box.

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>

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