



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring CenturyLink SIP Trunking with the Avaya Communication Server 1000 Release 7.5 and Acme Packet Net-Net 3800 Session Border Controller Release 6.2 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of the Avaya Communication Server 1000 release 7.5 and the CenturyLink SIP Trunking. During the interoperability testing, Avaya Communication Server 1000 was able to interoperate with the CenturyLink Broadsoft via SIP trunk. This test was performed to verify SIP trunk features including basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls are placed in both directions with various set types.

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

This document provides a typical network configuration deployment of the Avaya Communication Server 1000 (hereafter referred to as CS1000) and the CenturyLink SIP Trunking (hereafter referred to as CenturyLink system). During the interoperability testing, all SIP trunk applicable feature test cases were executed to ensure the interoperability between the CenturyLink system and the Avaya CS1000.

2. General Test Approach and Test Results

The CS1000 system release 7.5 was connected to an Acme Packet 3800 Session Border Controller via the Network Routing Service/SIP Proxy Server (NRS/SPS). Then the Acme Packet 3800 was connected to the CenturyLink system via SIP trunk. Various call types were made from the CS1000 to the CenturyLink system and vice versa to ensure the interoperability between the CS1000 and the CenturyLink system.

2.1. Interoperability Compliance Testing

The focus of this testing is to verify that CS1000 release 7.5 can interoperate with the CenturyLink system. The following interoperability areas were covered.

- General call processing between CS1000 and CenturyLink systems including:
 - Codec (G.711 u-law)
 - Hold/Retrieve on both ends
 - CLID displayed
 - Ring-back tone
 - Speech path
 - Dialing plan support
 - Advanced features (Call on Mute, Call Park, Call Waiting)
 - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- RFC2833/DTMF on both direction
- SIP Transport UDP
- Thru dialing via PBX Call Pilot
- Voice Mail Server CallPilot (hosted on Avaya system)
- Early Media Transmission

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. All the applicable test cases were executed. However, the following observations were noted during the compliance testing:

01. Outbound call from CS1000 to PSTN, CLID Name was not delivered over CenturyLink SIP Trunk.

- Call scenario: Make an outbound SIP call from CS1000 phone to a PSTN phone.
- SIP observation: CS1000 sent "From" header with display name information.
- Expected result: Call is established with 2 way speech paths. CLID Number and Name are displayed correctly.
- Actual result: CLID Name was not delivered over CenturyLink SIP Trunk
- Recommendation: CenturyLink CLEC should populate CLID Name for the outgoing call to PSTN and CLID Name delivery needs to be supported by Local Exchange Carrier host PSTN test phones.

02. Inbound toll free call was not tested due to lacking of provisioning from CenturyLink.

- Call scenario: Make an inbound SIP call from PSTN phone to CS1000 toll free number.
- SIP observation: N/A.
- Expected result: Call is established with 2 way speech paths. CLID Number and Name are displayed correctly.
- Actual result: This case was not executed on CenturyLink production network.
- Recommendation: Need CenturyLink to provide a toll free number associate to SIP Trunk to CS1000.

03. Outbound call from CS1000 to PSTN with "Privacy: user", CLID Number was not delivered over CenturyLink SIP Trunk.

- Call scenario: Make an outbound SIP call from CS1000 phone to a PSTN phone with "Privacy: user".
- SIP observation: CS1000 sent "Privacy: user" header to CenturyLink.
- Expected result: Call is established with 2 way speech paths. PSTN does not display CLID Name of CS1000 phone, but it still displays CLID Number.
- Actual result: CLID Number was not delivered over CenturyLink SIP Trunk
- Recommendation: CenturyLink should support Calling Number Delivery with "Privacy: user".

04. Off-net call forward. CenturyLink should increase Trunk group timer on SIP interface and disable Symmetric-Latching on CenturyLink SBC

- Call scenario: Make a SIP call from PSTN_A to CS1000 phone, then CS1000 phone forward back to PSTN_B
- SIP observation: CenturyLink sent SIP/CANCEL to the 1st call leg.
- Expected result: Call is successfully forwarded. PSTN_1 talks to PSTN_2 with 2 way speech paths, CLID displayed correctly.
- Actual result: The call was dropped due to CenturyLink send SIP/CANCEL.
- Resolution: CenturyLink increased Trunk group timer on SIP interface and disabled Symmetric-Latching on CenturyLink SBC and the test case passed.

05. PSTN call to Voicemail hosted on CS1000 did not terminate correctly

- Call scenario: Make a SIP call from PSTN to Voicemail DN hosted on CS1000 phone, then PSTN inputs DTMF command to disconnect.
- SIP observation: CenturyLink sent SIP/500 Internal Server Error.
- Expected result: The call is successfully terminated, all trunks are released.
- Actual result: The call was terminated, all trunks were released. However, CenturyLink sent SIP/500 Internal Server Error.

- Resolution: No resolution is available at this time.

06. The call from PSTN to Voicemail hosted on CS1000 lost early media, duration was about 5 seconds.

- Call scenario: Make a SIP call from PSTN to Voicemail DN hosted on CS1000 phone.
- SIP observation: CS1000 sent 180/SDP to transmit the announcement.
- Expected result: The recorded announcement is fully (in length) transmitted to PSTN.
- Actual result: The early media was lost about 5 seconds.
- Resolution: Known issue against CenturyLink.

07. CenturyLink should support slow start.

- Call scenario: Make a SIP call from PSTN_1 to CS1000_A which answers and blind transfers to CS1000_B. CS1000_B then Call Forward No Answer to PSTN_2.
- SIP observation: CS1000 INVITE without SDP (slow start) to CenturyLink.
- Expected result: the call is established between PSTN_1 and PSTN_2 with 2-ways speech paths. The CLID is displayed correctly.
- Actual result: The call was failed. CenturyLink did not configure SIP Trunk to support slow start.
- Resolution: CenturyLink enabled slow start and the test case passed.

08. Off-net call transfer scenario, CenturyLink sends 500 Internal Server.

- Call scenario: Make a SIP call from PSTN_1 to CS1000 which will answer and blind transfer to PSTN_2.
- SIP observation: CenturyLink sends 500 Internal Server.
- Expected result: the call is established between PSTN_1 and PSTN_2 with 2-ways speech paths. The CLID is displayed correctly.
- Actual result: The call was failed due to CenturyLink sent 500 Internal Server Error.
- Resolution: This issue has been fixed by a patch p_30224.ntl, applied to CS1000 SIP Signaling Server. This patch was included in Service Update cs1000-vtrk-7.50.17.16-6.i386.000.ntl

09. CS1000 phone held/ retrieved an outgoing call causing the display to be changed.

- Call scenario: Make a SIP call from CS1000 phone to a PSTN, then during the call, CS1000 phone perform “hold/ retrieve”.
- SIP observation: N/A
- Expected result: Call is established with 2 way speech paths. Number and Name are displayed correctly with “hold/ retrieve” action
- Actual result: After retrieving the call, the telephone number previously displayed on CS1000 phone will be unavailable and replaced by Route ACOD – Trunk Channel ID.
- Resolution: This is a known issue of CS1000.

2.3. Support

For technical support on CenturyLink system, please contact CenturyLink technical support at:

- Toll Free: 1-877-290-5458
- <http://www.centurylink.com/Pages/Support/>

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing event between the CS1000 and CenturyLink System.

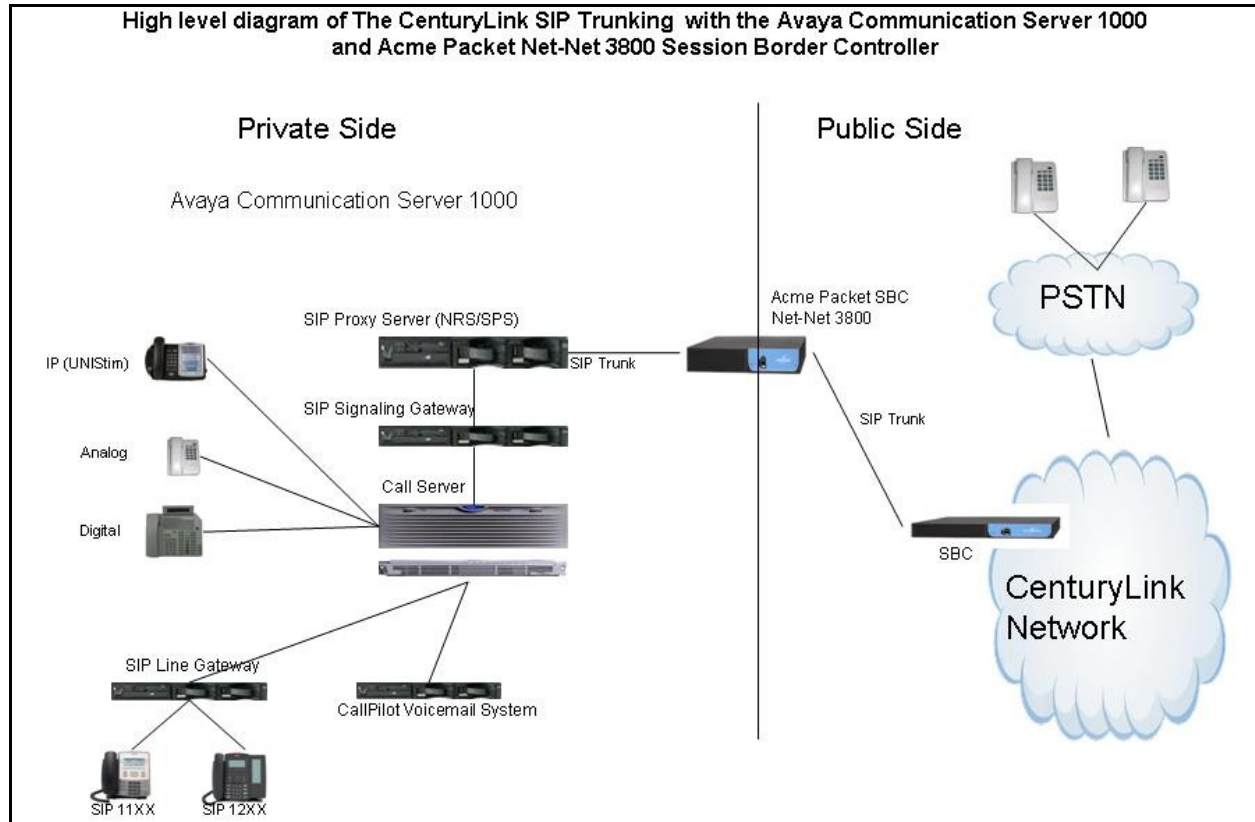


Figure 1- Network Diagram for Avaya CS1000 – CenturyLink System

The following assumptions were made for this lab test configuration.

1. CS1000 R7.5 software and implementation of latest patches
2. CenturyLink provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

During testing, the following activities were made to each test scenario:

1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved were checked for consistent and expected CLID, name and redirection information both prior to answer and after call establishment.

6. The speech path and messaging system were observed for timely and quality End to End tone audio path generation and application responses.
7. The call server maintenance terminal window were open during the test cases execution for the monitoring of BUG(s), ERR and AUD messages.
8. Speech path and display checked before and after calls were put on/off hold from each end.
9. Applicable files were screened on an hourly basis during the testing for message that may indicate technical issues. This refers to Avaya PBX files.
10. Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends.

4. Equipment and Software Validated

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

Avaya system:

System	Software/Loadware version
Avaya CS1000 7.5 (CPPM)	● Call Server: 7.50 Q GA ● SSG Server: 7.50.17 GA ● SLG Server: 7.50.17 GA ● NRS/SPS Server: 7.50.17 GA
Avaya phones	● 2002 p2: 0604DCJ (UNISTim) ● 2004 p2: 0604DCJ (UNISTim) ● 1140: 0625C6O (UNISTim) ● 1120: 0624C6O (UNISTim) ● 2007: 0621C6M (UNISTim) ● 1220: 062AC6O (UNISTim) ● SIP 1120, 1140: SIP12x0e04.00.04.00 ● SIP 1220,1240: SIP12x0e04.00.04.00
Acme Packet Net-Net 3800	● Firmware SCX6.2.0 MR-4 Patch 3 (Build 754)

CenturyLink system:

System	Software/Loadware version
Broadworks Broadsoft	● R16 sp2
Sonus GSX9000	● V07.02.05R000
Acme Packet Net-Net 4250	● R6.1

Additional software and patch lineup for the configuration and active patch list are listed as below.

Call Server: 7.50 Q GA plus latest DEPLIST – Issue: 01 Release: x2107.50, Created: 2011-03-15 10:26:33 (est)

SSG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50.17_20110118.ntl

SLG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50.17_20110118.ntl

NRS Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50.17_20110118.ntl

The output of “dstat” command on Call Server:

```
pdt> dstat
```

Call Server:

DepList name: core

Filename: /var/opt/nortel/cs/fs/u/patch/deplist/mcore_01.cpl

Issue : 01

Release : x2107.50

Created : 2011-05-10 09:43:40 (est)

Number of patches: 31

Patches Loaded: 31

Patches In-service: 31

pdt>

The output of “pstat” command on SSG Server:

```
[admin@car2-ssg-carrier ~]$ pstat
```

Product Release: 7.50.17.00

In system patches: 0

In System service updates: 8

PATCH#	IN_SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	07/01/11	NO	YES	cs1000-sps-7.50.17-01.i386.000
1	Yes	07/01/11	NO	YES	cs1000-baseWeb-7.50.17.01-1.i386.000
2	Yes	07/01/11	NO	YES	cs1000-shared-pbx-7.50.17-01.i386.000
4	Yes	07/01/11	NO	YES	cs1000-linuxbase-7.50.17.04-00.i386.000
5	Yes	07/01/11	NO	YES	cs1000-bcc-7.50.17.03-00.i386.000
6	Yes	07/01/11	NO	YES	cs1000-Jboss-Quantum-7.50.17.01-1.i386.000
7	Yes	07/01/11	NO	YES	cs1000-vtrk-7.50.17-11.i386.000
9	Yes	20/01/11	NO	YES	cs1000-dmWeb-7.50.17.04-00.i386.001

```
[admin@car2-ssg-carrier ~]$
```

The output of “pstat” command on NRS/SPS Server:

```
[admin2@car2-sps ~]$ pstat
```

Product Release: 7.50.17.00

In system patches: 0

In System service updates: 6

PATCH#	IN_SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	20/01/11	NO	YES	cs1000-sps-7.50.17-01.i386.000
1	Yes	07/01/11	NO	YES	cs1000-baseWeb-7.50.17.01-1.i386.000
2	Yes	20/01/11	NO	YES	cs1000-shared-pbx-7.50.17-01.i386.000
3	Yes	20/01/11	NO	YES	cs1000-dmWeb-7.50.17.04-00.i386.001
4	Yes	07/01/11	NO	YES	cs1000-linuxbase-7.50.17.04-00.i386.000
5	Yes	07/01/11	NO	YES	cs1000-Jboss-Quantum-7.50.17.01-1.i386.000

```
[admin2@car2-sps ~]$
```

5. Avaya Communication Server 1000 Configuration

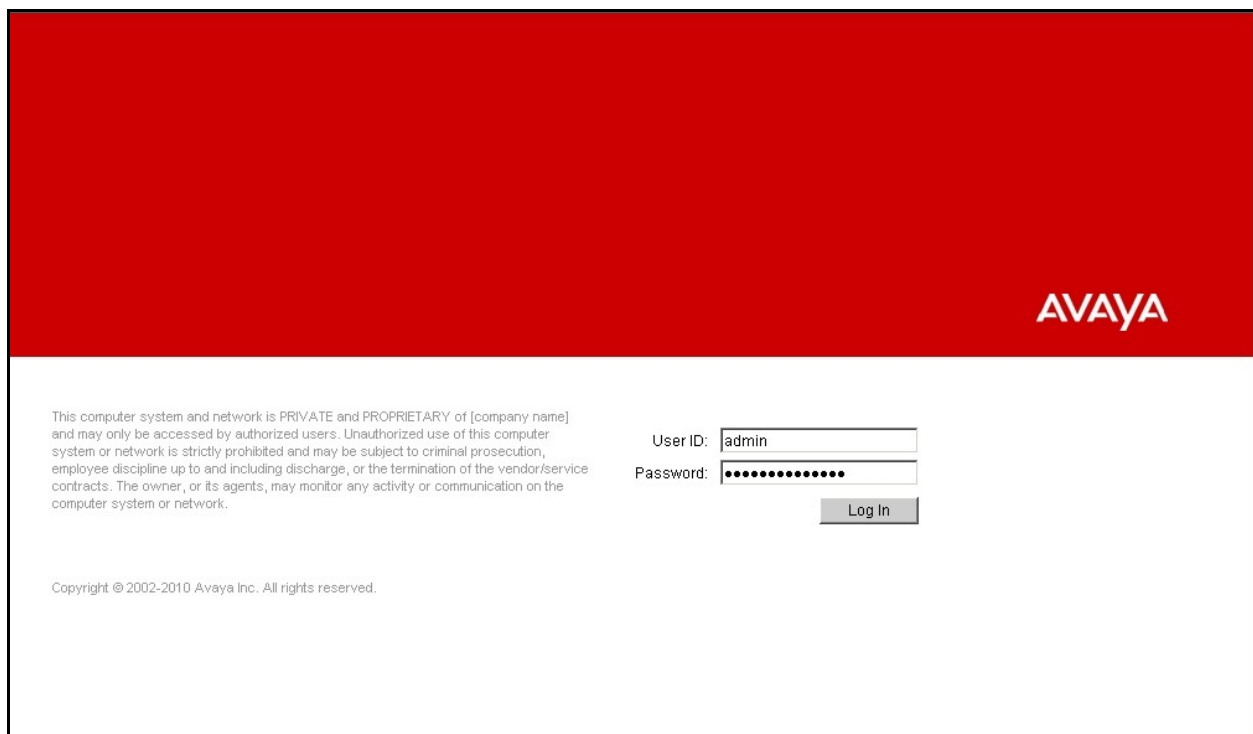
These Application Notes assume that the basic configuration has already been administered. For further information on Avaya Communications Server 1000, please consult references in **Section 9**.

The below procedures describe the configuration details of CS1000 with a SIP trunk to CenturyLink system.

5.1. Login to CS1000 System

5.1.1. Login Unified Communications Management (UCM) and Element Manager (EM)

a) Open an instance of a web browser and connect to the UCM GUI at the following address: `http://<UCM IP address>` as shown in **Figure 2**. Log in using an appropriate Username and Password.



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.

User ID:

Password:

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Figure 2 – Login Unified Communications Management

b) The **Unified Communications Management** screen is displayed. Click on the **Element Name** of the CS1000 Element as highlighted in red box as shown in **Figure 3**.

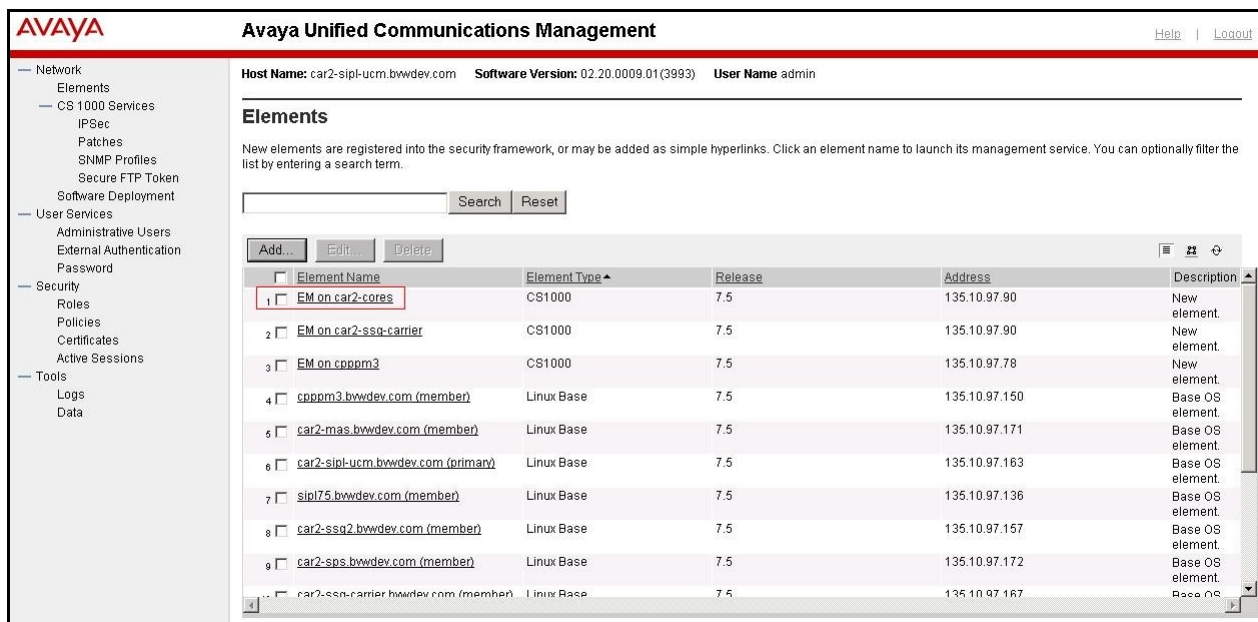


Figure 3 – Unified Communications Management

c) The CS1000 Element Manager System Overview page is displayed as shown in **Figure 4**.

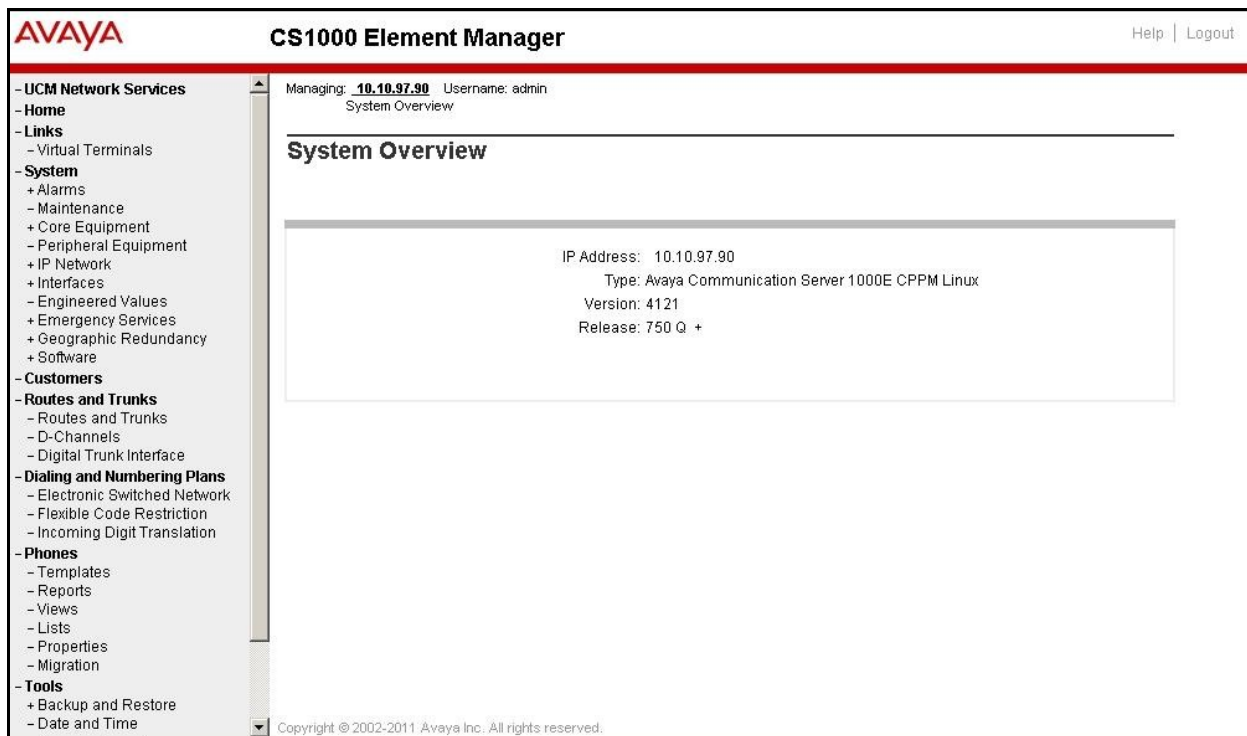


Figure 4 – Element Manager System Overview

5.1.2. Login to Call Server Command Line Interface (CLI)

- a) Using Putty, SSH to IP address of SSG Server with the admin account.
- b) Run the command “cslogin” and login with the appropriate admin account and password.
- c) Here are the logs.

```
login as: admin

Avaya Inc. Linux Base 7.50
The software and data stored on this system are the property of,
or licensed to, Avaya Inc. and are lawfully available only
to authorized users for approved purposes. Unauthorized access
to any software or data on this system is strictly prohibited and
punishable under appropriate laws. If you are not an authorized
user then do not try to login. This system may be monitored for
operational purposes at any time.

admin@10.10.97.168's password:
Last login: Thu Mar 10 17:38:16 2011 from 10.10.97.172

/usr/bin/xauth: creating new authority file /home/admin/.Xauthority
[admin@car2-ssg-carrier ~]$ cslogin

SEC054 A device has connected to, or disconnected from, a pseudo tty without authentica
ting
login
USERID? admin
PASS?
The software and data stored on this system are the property of,
or licensed to, Avaya Inc. and are lawfully available only to
authorized users for approved purposes. Unauthorized access to
any software or data on this system is strictly prohibited and
punishable under appropriate laws. If you are not an authorized
user then logout immediately. This system may be monitored for
operational purposes at any time.

.
TTY #09 LOGGED IN ADMIN 17:42 10/3/2011

>
```

5.2. Administer a Node IP Telephony

This section describes how to configure a Node IP Telephony on the CS1000.

5.2.1. Obtain Node IP address

These Application Notes assume that the basic configuration has already been administered and that a Node has already been created. This section describes the steps for configuring a Node (Node ID 2000) in CS1000 IP network to work with CenturyLink system. For further information on Avaya Communications Server 1000, please consult reference in **Section 9**.

a) Select **System** -> **IP Network** -> **Nodes: Servers, Media Cards**. **Figure 5** displays **IP Telephony Nodes** page. Then click on the Node ID of your CS1000 Element.

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
2000	1	LTPS, Gateway (SIPGw)	-	10.10.97.168		Synchronized
2001	1	LTPS, Gateway (SIPGw)	-	10.10.97.170		Synchronized
2002	1	SIP Line	-	10.10.97.164		Synchronized

Figure 5 – IP Telephony Nodes

b) The **Node Details** screen is displayed in **Figure 6** with the IP address of the CS1000 node. The **Node IP Address** is a virtual address which corresponds to the TLAN IP address of the Signaling Server, SIP Signaling Gateway. The SIP Signaling Gateway uses this **Node IP Address** to communicate with other components to process the SIP call.

Node Details (ID: 2000 - LTPS, Gateway (SIPGw))

Embedded LAN (ELAN)
 Gateway IP address: 10.10.97.65
 Subnet mask: 255.255.255.192

Telephony LAN (TLAN)
 Node IPv4 address: 135.10.97.168
 Subnet mask: 255.255.255.192
 Node IPv6 address:

IP Telephony Node Properties
 • Voice Gateway (Vgw) and Codecs
 • **Quality of Service (QoS)**
 • LAN
 • SNTP
 • Numbering Tables

Applications (click to edit configuration)
 • SIP Line
 • **Terminal Proxy Server (TPS)**
 • Gateway (SIPGw)
 • Personal Directories (PD)
 • Presence Publisher

Associated Signaling Servers & Cards

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
car2-ssg-carrier	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	135.10.97.89	10.10.97.167	Leader

Figure 6 –Node Details

5.2.2. Administer TPS

c) Continue from Section 5.2.1. On the **Node Details** page, select the **Terminal Proxy Server (TPS)** link as shown in Figure 6.

d) Check the **UNISlim Line Terminal Proxy Server** check box and then click **Save** as shown in Figure 7.

The screenshot displays the AVAYA CS1000 Element Manager interface. The top navigation bar shows the user is logged in as 'admin' with IP address '135.10.97.90'. The left sidebar contains a tree view of system components, including 'UCM Network Services', 'Home', 'Links', 'System', 'Alarms', 'Maintenance', 'Core Equipment', 'Peripheral Equipment', 'IP Network', 'Nodes: Servers, Media Cards', 'Maintenance and Reports', 'Media Gateways', 'Zones', 'Host and Route Tables', 'Network Address Translation (NAT)', 'QoS Thresholds', 'Personal Directories', 'Unicode Name Directory', 'Interfaces', 'Engineered Values', 'Emergency Services', 'Geographic Redundancy', 'Software', 'Customers', 'Routes and Trunks', 'Routes and Trunks', 'D-Channels', 'Digital Trunk Interface', 'Dialing and Numbering Plans', and 'Electronic Switched Network'. The main content area is titled 'Node ID: 2000 - UNISlim Line Terminal Proxy Server (LTPS) Configuration Details'. It features three tabs: 'Firmware', 'DTLS', and 'Network Connect Server'. The 'Firmware' tab is active, showing the 'UNISlim Line Terminal Proxy Server' section with a checked checkbox for 'Enable proxy service on this node'. Below this, the 'Firmware' section contains fields for 'IP address' (0.0.0.0), 'Full file path' (download/firmwa), 'Server Account/User ID', and 'Password'. The 'DTLS' section shows 'DTLS policy' set to 'Off' and two options: 'Client authentication' and 'Periodic re-keying'. The 'Network Connect Server' section shows the 'Primary network connect server (TL &N) IP address' set to '0.0.0.0'. At the bottom right, the 'Save' button is highlighted with a red box. A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.'

Figure 7 – TPS Configuration Details

5.2.3. Administer Quality of Service (QoS)

e) Continue from Section 5.2.1. On the **Node Details** page, select the **Quality of Service (QoS)** link as shown in Figure 6.

f) The default Diffserv values are as shown in Figure 8. Click the **Save** button.

AVAYA CS1000 Element Manager Help | Logout

Managing: 135.10.97.90 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Quality of Service (QoS)

Node ID: 2000 - Quality of Service (QoS)

Diffserv Codepoint (DSCP)

Enable Avaya automatic QoS: ☐

Control packets: (0-63)

Voice packets: (0-63)

VLAN tagging: ☐ 802.1Q support

802.1Q bits value (802.1P): (0-7)

* Required Value.

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

Save Cancel

Figure 8 – QoS Configuration Details

5.2.4. Synchronize the New Configuration

- g) Continue from Section 5.2.3, return to the **Node Details** page (Figure 6) and click on the **Save** button.
- h) The **Node Saved** screen is displayed. Click on the **Transfer Now** (not shown).
- i) The **Synchronize Configuration Files** screen is displayed. Check the Signaling Server check box and click on the **Start Sync** (not shown).
- j) When the synchronization completes, check the Signaling Server check box and click on the **Restart Applications** (not shown).

5.3. Administer Voice Codec

5.3.1. Enable Voice Codec, Node IP Telephony.

- a) Select **IP Network -> Nodes: Servers, Media Cards -> Configuration** from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of the CS1000 system. The **Node Details** screen is displayed. (See in Section 5.2.1 for more detail).
- b) On the **Node Details** page as shown in Figure 6, click on **Voice Gateway (VGW) and Codec**.
- c) The CenturyLink system only supports G711 with VAD disabled. The CenturyLink system does not support G729. Ensure that the Codec **G711** with unchecked **VAD**; uncheck Codec **G729A** checkboxes as shown in Figure 9. Then click on the **Save** button.

AVAYA CS1000 Element Manager Help | Logout

Managing: 135.10.97.90 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 2000 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Voice Codecs

Codec G711: ☒ Enabled (required)
 Voice payload size: 20 (milliseconds per frame)
 Voice playout (jitter buffer) delay: 40 80 (milliseconds)
 Nominal Maximum
 Maximum delay may be automatically adjusted based on nominal settings.
☒ Voice Activity Detection (VAD)

Codec G722: ☐ Enabled
 Voice payload size: 20 (milliseconds per frame)
 Voice playout (jitter buffer) delay: 40 80 (milliseconds)
 Nominal Maximum
 Maximum delay may be automatically adjusted based on nominal settings.

Codec G729: ☐ Enabled
 Voice payload size: 20 (milliseconds per frame)

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Figure 9 – Voice Gateway and Codec Configuration Details

d) Synchronize the new configuration (please refer to **Section 5.2.4** for more detail)

5.3.2. Enable Voice Codec on Media Gateways.

a) From the left menu of the Element Manager page in **Figure 9**, select **IP Network -> Media Gateways** menu item. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page.

b) In the following screen scroll down to the Codec **G711** and uncheck **VAD**; uncheck Codec **G729A** checkboxes as shown in **Figure 10**.

AVAYA CS1000 Element Manager Help | Logout

- UCM Network Services
 - Home
 - Links
 - Virtual Terminals
 - System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - IP Network
 - Nodes: Servers, Media Cards
 - Maintenance and Reports
 - **Media Gateways**
 - Zones
 - Host and Route Tables
 - Network Address Translation (NAT)
 - QoS Thresholds
 - Personal Directories
 - Unicode Name Directory
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
 - Customers
 - Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
 - Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
 - Phones
 - Templates
 - Reports
 - Views
 - Lists
 - Properties
 - Migration
 - Tools
 - + Backup and Restore
 - Date and Time
 - + Logs and reports
 - Security
 - + Passwords
 - + Policies
 - + Login Options

FAX configuration:

- FAX playout nominal delay: 100 (0 - 300 milliseconds)
- FAX no activity timeout: 20 (10 - 32000 milliseconds)
- FAX packet size: 30
- Codec: **G711** Select ☒
- Codec name: **G711**
- Voice payload size: 20 (ms/frame)
- Voice playout (jitter buffer) nominal delay: 40
- Modifications may cause changes to dependent settings
- Voice playout (jitter buffer) maximum delay: 80
- Modifications may cause changes to dependent settings
- VAD: ☐

Codec selection:

- + Codec: **G729A** Select ☐
- + Codec: **G723.1** Select ☐
- + Codec: **T38 FAX** Select ☒
- + QoS
- + Media Based CLID
- Call Server LAN

Embedded LAN (ELAN) configuration

- Geographic redundancy: ☐
- Primary call server IP address: 135.10.97.90
- Primary call server hostname: Primary_CS
- Signaling port: 15000
- Broadcast port: 15001 (1024 - 65535)

Telephony LAN (TLAN) configuration

- Signaling port: 5000
- Voice port: 5200 (1024 - 65535)

Routes

Add Remove

Click 'Add' to add routes to the IPMG

Save Cancel VGW Channels

** Mandatory fields of current configuration*

Figure 10 – Media Gateways Configuration Details

c) Then scroll down to the bottom of the page and click on the **Save** button.

5.4. Administer Zones and Bandwidth

This section describes the steps to create 2 zones: zone 10 for VGW and IP sets, and zone 255 for IP SIP Trunk.

5.4.1. Create a zone for IP phones (zone 10)

The following figures show how to configure a zone for IP sets and VGW for bandwidth management purposes. The bandwidth strategy can be adjusted to preference.

a) Select **IP Network** -> **Zones** configuration from the left pane, click on the **Bandwidth Zones** as shown in **Figure 11**.

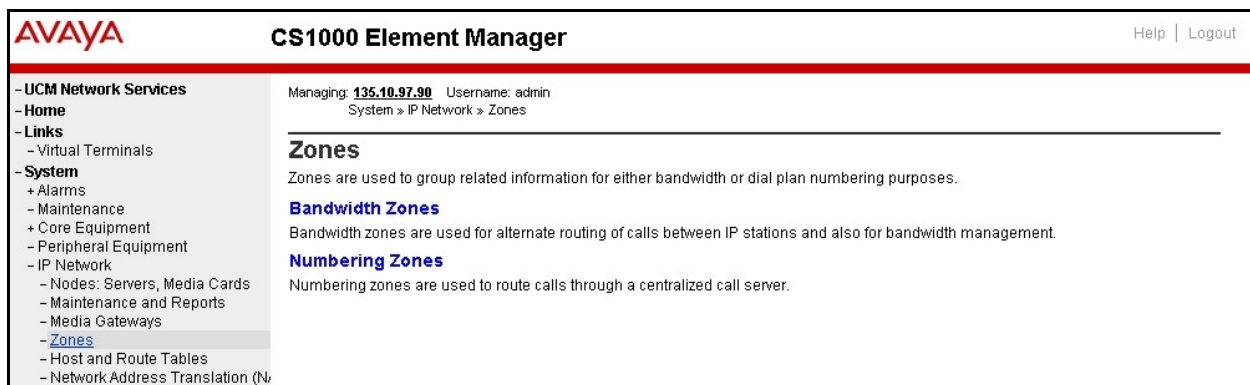


Figure 11 –Zones Page

b) The **Bandwidth Zones** screen is displayed as shown in **Figure 12**. Click **Add**.

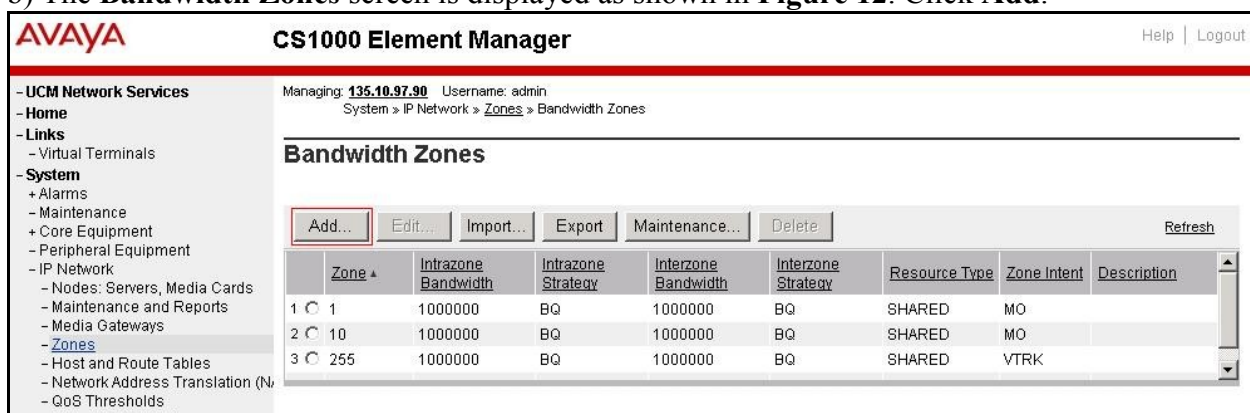


Figure 12 –Bandwidth Zones

c) Then in the **Add Bandwidth Zone** screen (not shown), click on **Zone Basic Property and Bandwidth Management**, select the values as shown (in red box) in **Figure 13** and click on the **Submit** button.

- **INTRA_STGY**: bandwidth configuration for local calls.
- **INTER_STGY**: bandwidth configuration for the calls over trunk.
- **BQ**: G711 is first choice and G729 is second choice.
- **BB**: G729 is first choice and G711 is second choice.
- **MO**: is used for IP phones, VGW
- **VTRK**: is used for virtual trunk.

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Managing: **135.10.97.90** Username: admin
System » IP Network » [Zones](#) » [Bandwidth Zones](#) » Zone Basic Property and Bandwidth Management

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	10 * (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	1000000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

* Required value.

Save

Cancel

Figure 13 –Bandwidth Management Configuration Details– IP phone

5.4.2. Create a zone for virtual SIP trunk (zone 255)

Follow Section 5.4.1 to create a zone for the virtual trunk. The difference is in **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk (Figure 14) and then click on the **Submit** button.

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Managing: **135.10.97.90** Username: admin
System » IP Network » [Zones](#) » [Bandwidth Zones](#) » Zone Basic Property and Bandwidth Management

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	255 * (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	1000000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

* Required value.

Save

Cancel

Figure 14 –Bandwidth Management Configuration Details–virtual trunk

5.5. Administer SIP Trunk Gateway

This section describes the steps for establishing a SIP IP connection between SIP Signalling Gateway (SSG) to NRS/SPS.

5.5.1. Integrated Services Digital Network (ISDN)

a) Select **Customers** in the left pane. The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. The **Customer 00 Edit** page will appear (not shown). Select the **Feature Packages** option from this page.

b) The screen is updated with a list of **Feature Packages** populated. Select **Integrated Services Digital Network** to edit its parameters. The screen is updated with parameters populated below **Integrated Services Digital Network**. Check the **Integrated Services Digital Network (ISDN)** checkbox, and retain the default values for all remaining fields (**Figure 15**). Scroll down to the bottom of the screen, and click on the **Save** button at the bottom of the page.

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 - + Passwords
 - Policies
 - + Login Options

Integrated Services Digital Network Package: 145

+ Dial Access Prefix on CLID table entry option

Integrated Services Digital Network: ☒

- Virtual private network identifier: 1 (1 - 16383)

- Private network identifier: 1 (1 - 16383)

- Node DN:

Multi-location business group: 0 (0 - 65535)

Business sub group consult-only: 65535 (0 - 65535)

Prefix 1:

Prefix 2:

Home number plan area code: (200 - 999)

Prefix for central office: (100 - 9999)

Home location code: (100 - 99999999)

Local steering code:

Calling number type: CLID feature displays the set's Prime DN

Redirection count for ISDN calls: 5

CLID information for incoming/outgoing calls: No manipulation is done

Public service telephone networks: ☐

Feature Packages

+ Flexible Services	Package: 152
+ Network Attendant Service	Package: 159
+ Flexible Numbering Plan	Package: 160
+ Trunk Failure Monitor	Package: 182
+ Radio Paging	Package: 187
+ DPNSS Network Services	Package: 231
+ M911 Enhancement Display	Package: 249
+ Called Party Control on Internal Calls	Package: 310
+ DPNSSI Message Waiting Indication	Package: 325
+ M3900 Product Enhancement	Package: 386
+ IP Media Services	Package: 422

Save Cancel

Figure 15 –Customer – ISDN Configurations

5.5.2. Administer SIP Trunk Gateway to NRS/SPS

a) Select **IP Network -> Nodes: Servers, Media Cards** configuration from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of this CS1000 system. The **Node Details** screen is displayed as shown in **Figure 6, Section 5.2.1**.

b) On the **Node Details** screen, select **Gateway (SIPGw)** (not shown).

c) Under **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following testing values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 16**. The parameters (highlighted in red boxes) are filled in, which were obtained when user creates a SIP profile on the NRS/SPS (these are shown in **Section 5.8.2**).

- Vtrk gateway application: SIP Gateway (SIPGw)
- SIP domain name: bvwddev.com
- Local SIP port: 5060
- Gateway endpoint name: car2-ssg-carrier
- Application node ID: 2000

AVAYA CS1000 Element Manager

Managing: 135.10.97.90 Username: admin

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

Node ID: 2000 - 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: bvwddev.com *

Local SIP port: 5060 * (1 - 65535)

Gateway endpoint name: car2-ssg-carrier *

Gateway password: *

Application node ID: 2000 * (0-9999)

Enable failsafe NRS: ☐

SIP ANAT: ☒ IPv4 ☐ IPv6

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

Figure 16 – Virtual Trunk Gateway Configuration Details Page 1

d) Click on the **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, enter the following values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 17**.

AVAYA CS1000 Element Manager Help | Logout

Managing: 135.10.97.90 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 2000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Proxy Or Redirect Server:

Proxy Server Route 1:

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

Port: (1 - 65535)

Transport protocol:

Options: ☒ Support registration
☐ Primary CDS proxy

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

Port: (1 - 65535)

Transport protocol:

Options: ☐ Support registration

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

Figure 17 – Virtual Trunk Gateway Configuration Details Page 2

e) On the same page as shown in **Figure 17**, scroll down to the **SIP URI Map** section (**Figure 18**).

Under the **Public E.164 Domain Names**, for:

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

Under the **Public E.164 Domain Names**, for:

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

Then click on the **Save** button.

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Dialing and Numbering Plans

Electronic Switched Network

Managing: 135.10.97.90 Username: admin

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

Node ID: 2000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

SIP URI Map:

Public E.164 domain names		Private domain names	
National:		UDP:	
Subscriber:		CDP:	
Special number:		Special number:	
Unknown:		Vacant number:	
		Unknown:	

SIP Gateway Services

SIP Converged Desktop:

Enable CD service

Service DN:

Used for making VTRK call from agent.

Converged telephone call forward DN:

RAN route for announce:

(route number 0 - 511)

Wait time before RAN queue:

(-1 - 32767 msec)

* Required Value.

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

Save Cancel

Figure 18 – Virtual Trunk Gateway Configuration Details Page 3

- f) **Synchronize** the new configuration (please refer to **Section 5.2.4**).
- g) After the configuration is completed on the NRS/SPS, the Gateway endpoint entry for the SSG (in this case is car2-ssg-carrier) shows the IP address of the SSG which is successfully registered to the NRS/SPS) (**Figure 19**). Please refer to **Section 5.8** for more details on the NRS/SPS configuration.

AVAYA Network Routing Service Manager Help | Logout

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 - Backup
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 - GK/NRS Data upgrade

Managing: ☒ Active database 135.10.97.92
☐ Standby database [Numbering Plans » Endpoints](#)

Search for Endpoints Hide

Enter an endpoint ID (use * for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID:

Limit results to Domain: All service domains / All L1 domains / All LD domains

Results per page: 50

Gateway Endpoints (2)

User Endpoints (0)

SIP phone context... Refresh

☐	ID ^	Supported Protocols	SIP mode:	Call Signaling IP	Description	# of Routing Entries	Context
☐	acme	Static SIP endpoint	Proxy Mode	10.10.97.184		15	bwwdev.com / udp / cdp
☐	car2-ssg-carrier	Dynamic SIP endpoint	Proxy Mode	10.10.97.168		1	bwwdev.com / udp / cdp

1 - 2 of 2 Gateway Endpoint(s) Page 1 of 1 First | Previous | Next | Last

Figure 19 – SSG is registered successfully to the NRS/SPS

5.5.3. Administer Virtual D-Channel

a) Select **Routes and Trunks -> D-Channels** from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list as shown in **Figure 20**. Click on **to Add** button.

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Managing: 135.10.97.90 Username: admin
Routes and Trunks » D-Channels

D-Channels

Maintenance

[D-Channel Diagnostics \(LD 96\)](#)
[Network and Peripheral Equipment \(LD 32, Virtual D-Channels\)](#)
[MSDL Diagnostics \(LD 96\)](#)
[TMDI Diagnostics \(LD 96\)](#)
[D-Channel Expansion Diagnostics \(LD 48\)](#)

Configuration

Choose a D-Channel Number: 0 and type: DCH to Add

- Channel: 100	Type: DCH	Card Type: DCIP	Description: carrier	Edit
- Channel: 101	Type: DCH	Card Type: DCIP	Description: enterprise	Edit
- Channel: 102	Type: DCH	Card Type: DCIP	Description: sipi	Edit

Figure 20 – D-Channels

b) The D-Channels 100 Property Configuration screen is displayed next as shown in **Figure 21**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **D channel Card Type (CTYP): D-Channel is over IP (DCIP)**
- **Designator (DES): A descriptive name**
- **Interface type for D-channel (IFC): Meridian Meridian1 (SL1)**
- **Meridian 1 node type: Slave to the controller (USR)**
- **Release ID of the switch at the far end (RLS): 25**
- **Advanced options (ADVOPT): check on Network Attendant Service Allowed**

Figure 21 – D-Channels Configuration Details

c) Click on the **Basic Options** and click on the **Edit** button at the **Remote Capabilities (RCAP)** attribute. The **Remote Capabilities Configuration** page will appear. Then check on the **ND2** and the **MWI** checkboxes as shown in **Figure 22** and **Figure 23**.

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D channel Card Type: DCIP

Designator: carrier

Recovery to Primary: ☐

PRI loop number for Backup D-channel:

User: Integrated Services Signaling Link Dedicated (ISLD)

Interface type for D-channel: Meridian Meridian1 (SL1)

Country: ETS 300 =102 basic protocol (ETSI)

D-Channel PRI loop number:

Primary Rate Interface: more PRI

Secondary PRI2 loops:

Meridian 1 node type: Slave to the controller (USR)

Release ID of the switch at the far end: 25

Central Office switch type: 100% compatible with Bellcore standard (STD)

Integrated Services Signaling Link Maximum: 4000 Range: 1 - 4000

Signalling server resource capacity: 3700 Range: 0 - 3700

Basic options (BSCOPT)

Primary D-channel for a backup DCH: Range: 0 - 254

- PINX customer number:

- Progress signal:

- Calling Line Identification:

- Output request Buffers: 32

- D-channel transmission Rate: 56 kb/s when LCMT is AMI (56K)

- Channel Negotiation option: No alternative acceptable, exclusive. (1)

- Remote Capabilities: Edit

+ - Change protocol timer value (TIMR)

- B channel Service messaging: ☐

+ Advanced options (ADVOPT)

+ Feature Packages

Submit Refresh Delete Cancel

Figure 22 – D-Channels Configuration Details

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Call transfer integer (CTI) ☐

Call transfer object (CTO) ☐

Diversion info. is sent using integer value (DV1I) ☐

Diversion info. is sent using object identifier (DV1O) ☐

Rerouting requests processed using integer value (DV2I) ☐

Rerouting requests processed using object identifier (DV2O) ☐

Diversion info. sent. rerouting requests processed (DV3I) ☐

EuroISDN - div. info sent. rerouting req. processed (DV3O) ☐

Call transfer notification and invocation to EuroISDN (ECTO) ☐

Malicious call identification (MCID) ☐

MCDN QSIG conversion (MQC) ☐

Remote D-channel is on a MSDL card (MSL) ☐

Message waiting interworking with DMS-100 (MWM) ☒

Network access data (NAC) ☐

Network call trace supported (NCT) ☐

Network name display method 1 (ND1) ☐

Network name display method 2 (ND2) ☒

Network name display method 3 (ND3) ☐

Name display - integer ID coding (NDI) ☐

Name display - object ID coding (NDO) ☐

Path replacement uses integer values (PRI) ☐

Path replacement uses object identifier (PRO) ☐

Release Link Trunks over IP (RLTI) ☐

Remote virtual queuing (RVQ) ☐

Trunk anti-tromboning operation (TAT) ☐

User to user service 1 (UUS1) ☐

NI-2 name display option. (NDS) ☐

Message waiting indication using integer values (QMWI) ☐

Message waiting indication using object identifier (QMW0) ☐

User to user signalling (UUI) ☐

Return - Remote Capabilities

Cancel

Figure 23 – Remote Capabilities Configuration Details

- d) Click on the **Return – Remote Capabilities** button.
- e) Click on the **Submit** button (not shown).

5.5.4. Administer Virtual Super-Loop

Select **System** -> **Core Equipments** -> **Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click “**Add**” button to create a new one as shown in **Figure 24**. In this example, Superloop 100 is being added and used.

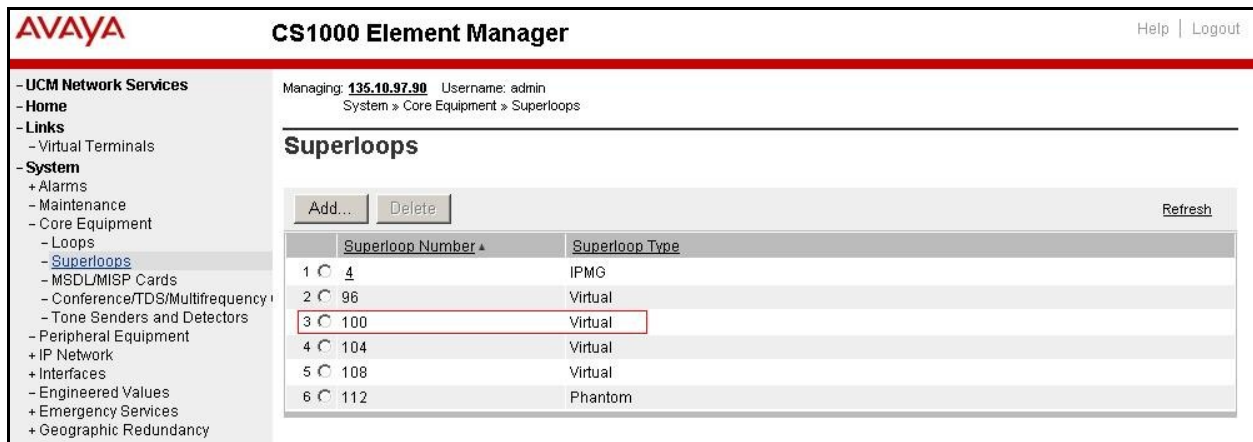


Figure 24 – Administer Virtual Super-Loop

5.5.5. Administer Virtual SIP Routes

a) Select **Routes and Trunks** -> **Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. In this example, **Customer 0** is being used. Click on the **Add route** button as shown in **Figure 25**.

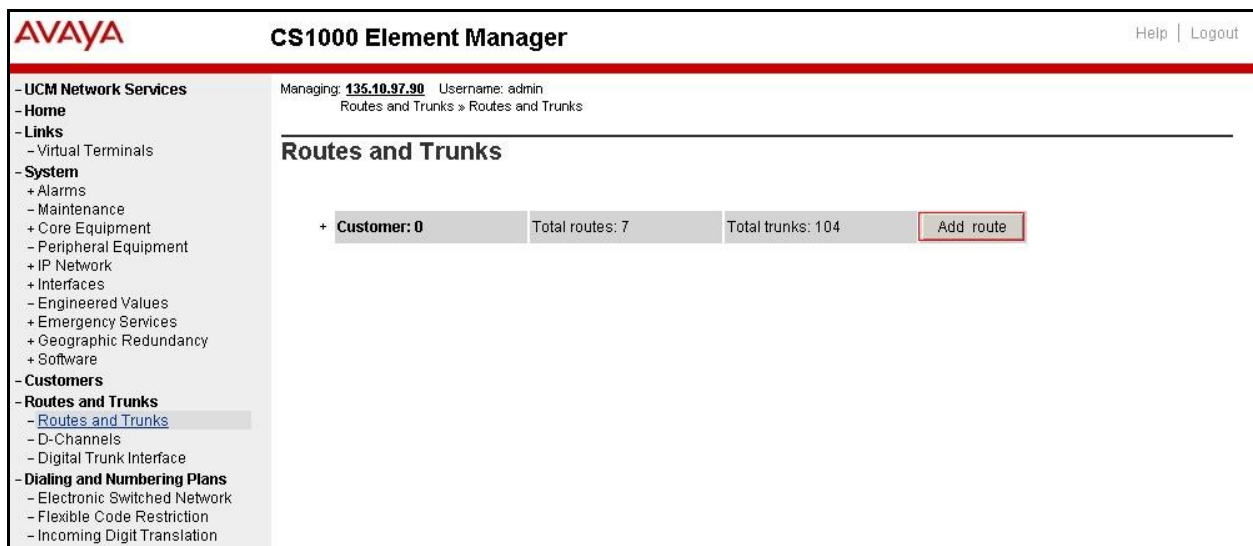


Figure 25 – Add route

b) The **Customer 0**, **New Route Configuration** screen is displayed next. Scroll down until the **Basic Configuration** Section is displayed and enter the following values for the specified fields, and retain the default values for the remaining fields as shown in **Figure 26**.

- **Route Number (ROUT):** Select an available route number.
- **Designator field for trunk (DES):** A descriptive text.
- **Trunk Type (TKTP):** TIE trunk data block (TIE)
- **Incoming and Outgoing trunk (ICOG):** Incoming and Outgoing (IAO)
- **Access Code for the trunk route (ACOD):** An available access code.

- Check the field **The route is for a virtual trunk route (VTRK)**, to enable four additional fields to appear.
- For the **Zone for codec selection and bandwidth management (ZONE)** field, enter 255 (created in Section 5.4.2).
- For the **Node ID of signaling server of this route (NODE)** field, enter the node number 2000 (created in Section 5.2.1).
- Select **SIP (SIP)** from the drop-down list for the **Protocol ID for the route (PCID)** field.
- Check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen.
 - o **Mode of operation (MODE)**: Route uses **ISDN Signalling Link (ISLD)**
 - o **D channel number (DCH)**: D-Channel number 100 (created in Section 5.5.3)
 - o **Network calling name allowed (NCNA)**: Check the field.
 - o **Network call redirection (NCRD)**: Check the field.
 - o **Insert ESN access code (INAC)**: Check the field.

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- Basic Configuration

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 00

Route number (ROUT): 100

Designator field for trunk (DES): CARRIER

Trunk type (TKTP): TIE

Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO)

Access code for the trunk route (ACOD): 8000

Trunk type M911P (M911P): ☐

The route is for a virtual trunk route (VTRK): ☒

- Zone for codec selection and bandwidth management (ZONE): 00255 (0 - 8000)

- Node ID of signaling server of this route (NODE): 2000 (0 - 9999)

- Protocol ID for the route (PCID): SIP (SIP)

- Print correlation ID in CDR for the route (CRID): ☐

Integrated services digital network option (ISDN): ☒

- Mode of operation (MODE): Route uses ISDN Signalling Link (ISLD)

- D channel number (DCH): 100 (0 - 254)

- Interface type for route (IFC): Meridian M1 (SL1)

- Private network identifier (PNI): 00001 (0 - 32700)

- Network calling name allowed (NCNA): ☒

- Network call redirection (NCRD): ☒

- Trunk route optimization (TRO): ☐

- Recognition of DTI2 ABCD FALT signal for ISL (FALT): ☐

- Channel type (CHTY): B-channel (BCH)

- Call type for outgoing direct dialed TIE route (CTYP): Unknown Call type (UKWN)

- Insert ESN access code (INAC): ☒

- Integrated service access route (ISAR): ☐

Figure 26 – Route Configuration Details Pages 1

- Click on **Basic Route Options**, check the **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)**, input **DCNO 1** for both **Day IDC Tree Number** and **Night IDC Tree Number** as shown in **Figure 27**.

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- Basic Route Options

Attendant announcement (ATAN) : No Attendant Announcement. (NO)

Billing number required (BILN) : ☐

Call detail recording (CDR) : ☒

- CDR records generated on incoming calls (INC) : ☒

- CDR record printing content option for redirected calls (LAST) : ☒

- Time to answer output in CDR (TTA) : ☐

- CDR ACD Q initial connection records to be generated (QREC) : ☒

- CDR on outgoing calls (OAL) : ☒

- CDR on outgoing toll calls (OTL) : ☐

- Answered call identification allowed (AIA) : ☒

- CDR timing starts on answer supervision of outgoing calls (OAN) : ☒

- outpulsed digits in CDR (OPD) : ☒

- Number of digits printed (NDP) : EXC 0 SCH2017

North American toll scheme (NATL) : ☒

Controls or timers (CNTL) : ☐

Conventional (Tie trunk only) (CNVT) : ☐

Incoming DID digit conversion on this route (IDC) : ☒

- Day IDC tree number (DCNO) : 1 (0 - 254)

- Night IDC tree number (NDNO) : 1 (0 - 254)

- Display external dialed digits (DEXT) : ☐

Multifrequency compelled or MFC signaling (MFC) : No MFC (NO)

Process notification networked calls (PNNC) : ☐

+ Network Options

+ General Options

+ Advanced Configurations

Submit Refresh Delete Cancel

Figure 27 – Route Configuration Details Pages 2

c) Click on the **Submit** button.

5.5.6. Administer Virtual Trunks

a) Continue **Section 5.5.5**, after click **Submit**, the **Routes and Trunks** screen is displayed and updated with the newly added route. In the example, Route 100 was being added. Click on the **Add trunk** button next to the newly added route 100 as shown in **Figure 28**.

AVAYA
CS1000 Element Manager
[Help](#) | [Logout](#)

- UCM Network Services
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- Links
- Virtual Terminals
- System
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- Maintenance
+ Core Equipment
- Peripheral Equipment
+ IP Network
+ Interfaces
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
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- Phones
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Managing: **135.10.97.90** Username: admin
Routes and Trunks » Routes and Trunks

Routes and Trunks

- Customer: 0	Total routes: 7	Total trunks: 104	Add route	
+ Route: 50	Type: MUS	Description: MUS	Edit	Add trunk
+ Route: 51	Type: MUS	Description: MUS51	Edit	Add trunk
+ Route: 52	Type: MUS	Description: MUS	Edit	Add trunk
+ Route: 53	Type: MUS	Description: MUS	Edit	Add trunk
+ Route: 100	Type: TIE	Description: CARRIER	Edit	Add trunk
+ Route: 101	Type: TIE	Description: ENTERPRISE	Edit	Add trunk
+ Route: 102	Type: TIE	Description: SIPL	Edit	Add trunk

Figure 28 – Route and Trunks

b) The **Customer 0, Route 100, Trunk 1 Property Configuration** screen is displayed in **Figure 29**. Enter the following values for the specified fields and retain the default values for the remaining fields. The Media Security (sRTP) has to be disabled at the trunk level by editing the **Class of Service (CLS)** at the bottom basic trunk configuration page. Click on the **Edit** button as shown in **Figure 29**.

- The **Multiple trunk input number (MTINPUT)** field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, 32 trunks were created.
- **Trunk data block (TYPE): IP Trunk (IPTI)**
- **Terminal Number (TN):** Available terminal number (created in Section 5.5.4)
- **Designator field for trunk (DES):** A descriptive text
- **Extended Trunk (XTRK): Virtual trunk (VTRK)**
- **Member number (RTMB):** Current route number and starting member
- **Start arrangement Incoming (STRI): Immediate (IMM)**
- **Start arrangement Outgoing (STRO): Immediate (IMM)**
- **Trunk Group Access Restriction (TGAR):** Desired trunk group access restriction level
- **Channel ID for this trunk (CHID):** An available starting channel ID

AVAYA **CS1000 Element Manager** Help | Logout

Managing: **135.10.97.90** Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 100

Customer 0, Route 100, Trunk type TIE trunk data block

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 - Peripheral Equipment
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 - Routes and Trunks
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 - Properties
 - Migration
- Tools
 - + Backup and Restore
 - Date and Time
 - + Logs and reports
- Security

- Basic Configuration

Multiple trunk input number: Range: 2 - 3700

Auto increment member number: ☒

Trunk data block:

Terminal number: *

Designator field for trunk:

Extended trunk:

Member number: *

Level 3 Signaling:

Card density:

Start arrangement Incoming:

Start arrangement Outgoing:

Trunk group access restriction:

Channel ID for this trunk:

Class of Service:

+ Advanced Trunk Configurations

* Required value.

Figure 29 – New Trunk Configuration Details

c) For **Media Security**, select **Media Security Best Try (MSBT)**. Enter the remaining values for the specified fields as shown in **Figure 30**. Scroll down to the bottom of the screen and click **Return Class of Service** and then click on the **Save** button (not shown).

AVAYA **CS1000 Element Manager** Help | Logout

Managing: **135.10.97.90** Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 100

Customer 0, Route 100, Trunk type TIE trunk data block

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 - Routes and Trunks
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- Security

- Echo Canceling:

- Hong Kong DTI:

- Loop Break Supervised COT:

- Make-break ratio for dial pulse:

- Manual Incoming:

- Media Security:

- Network Hook Flash Over M911P:

- Polarity:

- Priority:

- Restriction level:

- Reversed Ear Piece:

- Short or long line:

- Transmission Class of Service:

- Warning Tone:

- Reversed Ear Piece:

- ARF Supervised COT:

Figure 30 – Class of Service Configuration Details Page

5.5.7. Administer Calling Line Identification Entries

a) Select **Customers > 00 > ISDN and ESN Networking**. Click on **Calling Line Identification Entries** as shown in Figure 31.

AVAYA CS1000 Element Manager Help | Logout

Managing: **135.10.97.90** Username: admin
Customers » Customer 00 » Customer Details » ISDN and ESN Networking

ISDN and ESN Networking

General Properties

Flexible trunk to trunk connection option: **Connections restricted**

Flexible orbiting prevention timer: **8**

Country code: **1** (0 - 9999)
Code for processing the called number

National access code: **001**

International access code: **011**

Options: ☒ Transfer on ringing of supervised external trunks
☒ Connection of supervised external trunks

Network option: ☒ Coordinated dialing plan routing

Integrated services digital network: ☒

Microsoft converged office dialing plan: **Private dialing plan**

Private dialing plan for non-DID users: ☐ Coordinated dialing plan
☐ Uniform dialing plan

Calling Line Identification

Information for incoming/outgoing calls: **No manipulation is done**

Size: **256** (0 - 4000)

Country code: **1** (0 - 9999)
Code displayed as part of calling number

Calling Line Identification Entries

Save **Cancel**

Figure 31 – ISDN and ESN Networking

b) Click on **Add** as shown in Figure 32.

The screenshot shows the AVAYA CS1000 Element Manager interface. The top header includes the AVAYA logo, the title "CS1000 Element Manager", and links for "Help" and "Logout". The sidebar on the left contains a tree view with categories like "UCM Network Services", "Home", "Links", "System", "Customers", "Routes and Trunks", and "Digital Trunk Interface". The main content area is titled "Calling Line Identification Entries" and includes a "Search for CLID" section with input fields for "Start range" and "End range", a "Search" button, and a message: "'End range' should not exceed the CLID size specified". Below this is a section for "Calling Line Identification Entries" with "Add..." and "Delete" buttons, and a "Refresh" link.

Figure 32 – Calling Line Identification Page

c) Add entry **0** as shown in **Figure 33**

- **National Code**: leave as blank
- **Local Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits – 913324. This **Local Code** will be used for call display purpose of outbound international call configuration in **Section 5.6.6** in where the **Special Number 0** is associated with Call Type = Unknown.
- **Home Location Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 913324. This **Home Location Code** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 913324. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Calling Party Name Display**: Uncheck for **Roman characters**.

AVAYA

CS1000 Element Manager

[Help](#) | [Logout](#)

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 - + Software
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 - + Logs and reports
- Security
 - + Passwords
 - + Policies
 - + Login Options

Managing: **135.10.97.90** Username: admin
[Customers](#) » [Customer 00](#) » [Customer Details](#) » [ISDN and ESN Networking](#) » [Calling Line Identification Entries](#) » [Edit Calling Line Identification 0](#)

Edit Calling Line Identification 0

General Properties

National Code: (0 - 999999)
Code for national home number

Local Code: (1-12 digits)
Code for home local number or listed DN

Home Location Code: (1-7 digits)

Local Steering Code: (1-7 digits)

Use DN as DID: YES

Emergency Services Access

Emergency Local Code: (1-12 digits)
Code for home local number during Emergency calls

Emergency Options:
☐ Home national number for emergency services access calls
☒ Append the originating directory number for emergency services access calls

Calling Party Name Display

Roman characters: ☐

CPND Name:
first name, last name

Expected Length:

Display Format: First name, Last name

Save

Cancel

Figure 33 – Edit Calling Line Identification 0

d) Click on **Save**.

5.5.8. Enable External Trunk to Trunk Transferring

This section shows how to enable External Trunk to Trunk Transferring feature which is a mandatory configuration to make call transfer and conference work properly over SIP trunk.

- a) Login Call Server CLI (please refer to **Section 5.1.2** for more detail)
- b) Allow External Trunk To Trunk Transferring for **Customer Data Block** by using LD 15

```
>ld 15
CDB000
MEM AVAIL: (U/P): 35600176   USED U P: 8325631 954062   TOT: 44879869
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
...
TRNX YES
EXTT YES
...
```

5.6. Administer Dialing Plans

5.6.1. Define ESN Access Codes and Parameters (ESN)

- a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **ESN Access Code and Parameters (ESN)** as shown in **Figure 34**.

The screenshot shows the AVAYA CS1000 Element Manager interface. The top header includes the AVAYA logo, the title 'CS1000 Element Manager', and links for 'Help' and 'Logout'. Below the header, the main content area is titled 'Electronic Switched Network (ESN)'. On the left, a navigation pane lists various categories: UCM Network Services, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, and Tools. The 'Dialing and Numbering Plans' category is expanded, showing 'Electronic Switched Network' as the selected option. The main content area displays the configuration for 'Customer 00'. Under 'Network Control & Services', the 'ESN Access Codes and Parameters (ESN)' option is highlighted. Other options include 'Digit Manipulation Block (DGT)', 'Home Area Code (HNPA)', 'Flexible CLID Manipulation Block (CMDDB)', 'Free Calling Area Screening (FCAS)', 'Free Special Number Screening (FSNS)', 'Route List Block (RLB)', 'Incoming Trunk Group Exclusion (ITGE)', and 'Network Attendant Services (NAS)'. Under 'Coordinated Dialing Plan (CDP)', options include 'Local Steering Code (LSC)', 'Distant Steering Code (DSC)', and 'Trunk Steering Code (TSC)'. Under 'Numbering Plan (NET)', there are two sections: 'Access Code 1' and 'Access Code 2'. Each section lists options: 'Home Location Code (HLOC)', 'Location Code (LOC)', 'Numbering Plan Area Code (NPA)', 'Exchange (Central Office) Code (NXX)', 'Special Number (SPN)', and 'Network Speed Call Access Code (NSCL)'. The 'ESN Access Codes and Parameters (ESN)' option is highlighted in the 'Access Code 1' section.

Figure 34 – Electronic Switch Network (ESN)

b) In the **ESN Access Codes and Basic Parameters** page, define **NARS/ BARS Access Code 1** as shown in **Figure 35**.

Figure 35 – ESN Access Codes and Basic Parameters

c) Click **Submit** (not shown).

5.6.2. Associate NPA and SPN call to ESN Access Code 1

a) Login Call Server CLI (please refer to **Section 5.1.2** for more detail)

b) In LD 15, change Customer Net_Data block by disable NPA and SPN to be associated to Access Code 2. It means Access Code 1 will be used for NPA and SPN calls.

```
>ld 15
CDB000
MEM AVAIL: (U/P): 35600086   USED U P: 8325631 954152   TOT: 44879869
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
AC2 xNPA xSPN
FNP
CLID
...
```

c) Verify Customer Net_Data block by using LD 21

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTA
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
...
```

5.6.3. Digit Manipulation Block (DMI)

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Digit Manipulation Block (DGT)** as shown in **Figure 34**.

b) In the Choose a DMI Number field, select an available DMI from the drop-down list and click to **Add** as shown in **Figure 36**.

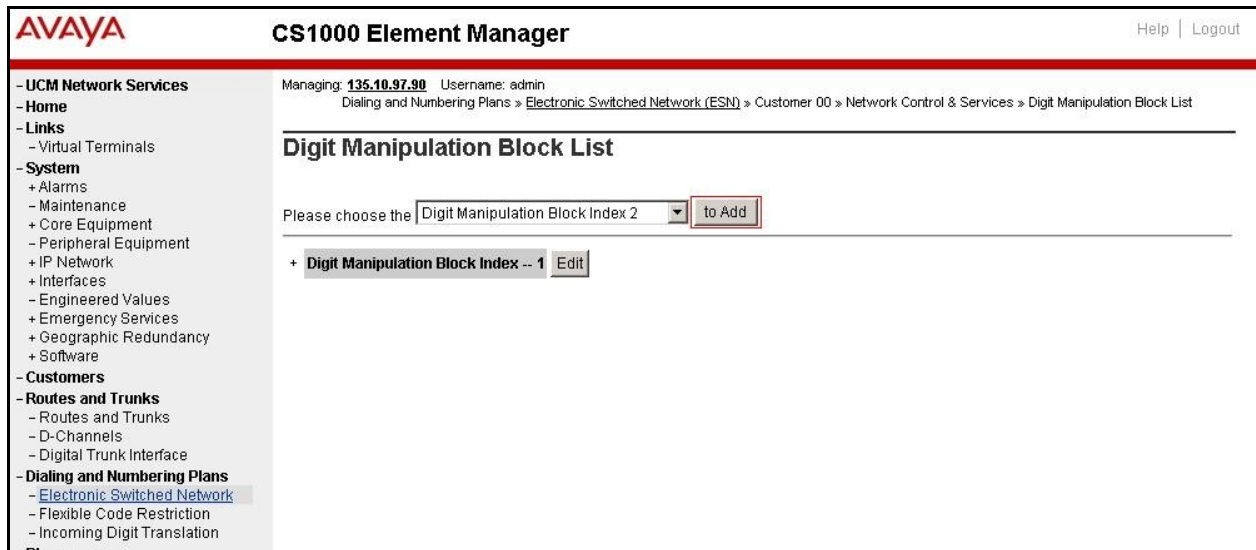


Figure 36 – Digit Manipulation Block List

c) Enter **0** for the **Number of leading digits to be Deleted (Del)** field and select **NPA (NPA)** for the **Call Type to be used by the manipulated digits (CTYP)** and then click **Submit** as shown in **Figure 37**.

Figure 37 – Digit Manipulation Block

5.6.4. Route List Block (RLB) (RLB 100)

This section shows how to add a RLB associated with the DMI created in Section 5.6.3.

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Route List Block (RLB)** as shown in Figure 34.

b) Select an available value in the textbox for the **route list index** and click on the “to Add” button (in this case is 100) (Figure 38).

Figure 38 – Route List Blocks

c) Enter the following values for the specified fields, and retain the default values for the remaining fields (Figure 39). Scroll down to the bottom of the screen, and click on the **Submit** button.

- **Route number (ROUT):** 100 (created in Section 5.5.5)

- **Digit Manipulation Index (DMI): 1** (created in **Section 5.6.3**)
- **Incoming CLID Table: 0** (created in **Section 5.5.7**)

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 - Security
 - + Passwords
 - + Policies
 - + Login Options

Indexes

Time of Day Schedule: 0

Facility Restriction Level: 0 (0 - 7)

Digit Manipulation Index: 1

ISL D-Channel Down Digit Manipulation Index: 0 (0 - 1999)

Free Calling Area Screening Index: 0

Free Special Number Screening Index: 0

Business Network Extension Route: ☐

Incoming CLID Table: 0 (0 - 255)

Options

Local Termination entry: ☐

Route Number: 100

Skip Conventional Signaling: ☐

Display Originator's Information: ☐

Use Tone Detector: ☐

Conversion to LDN: ☐

Expensive Route: ☐

Strategy on Congestion: No Reroute (NRR)

QSIG Alternate Routing Causes: QSIG Alternate Routing Cause 1

Preferred Routing: Preferred Route 1

ISDN Drop Back Busy: Drop Back Disabled (DBD)

ISDN Off-Hook Queuing Option: ☐

Off-Hook Queuing Allowed: ☐

Call Back Queuing Allowed: ☐

VNS Options

Entry is a VNS Route: ☐

Submit Cancel

Figure 39 – Route List Blocks Configuration Details

5.6.5. Inbound Call Digit Translation

This section describes the steps for receiving the calls from PSTN via the CenturyLink system.

a) Select **Dialing and Numbering Plans** -> **Incoming Digit Translation** from the left pane to display the **Incoming Digit Translation** screen. Click on the **Edit IDC** button (**Figure 40**).

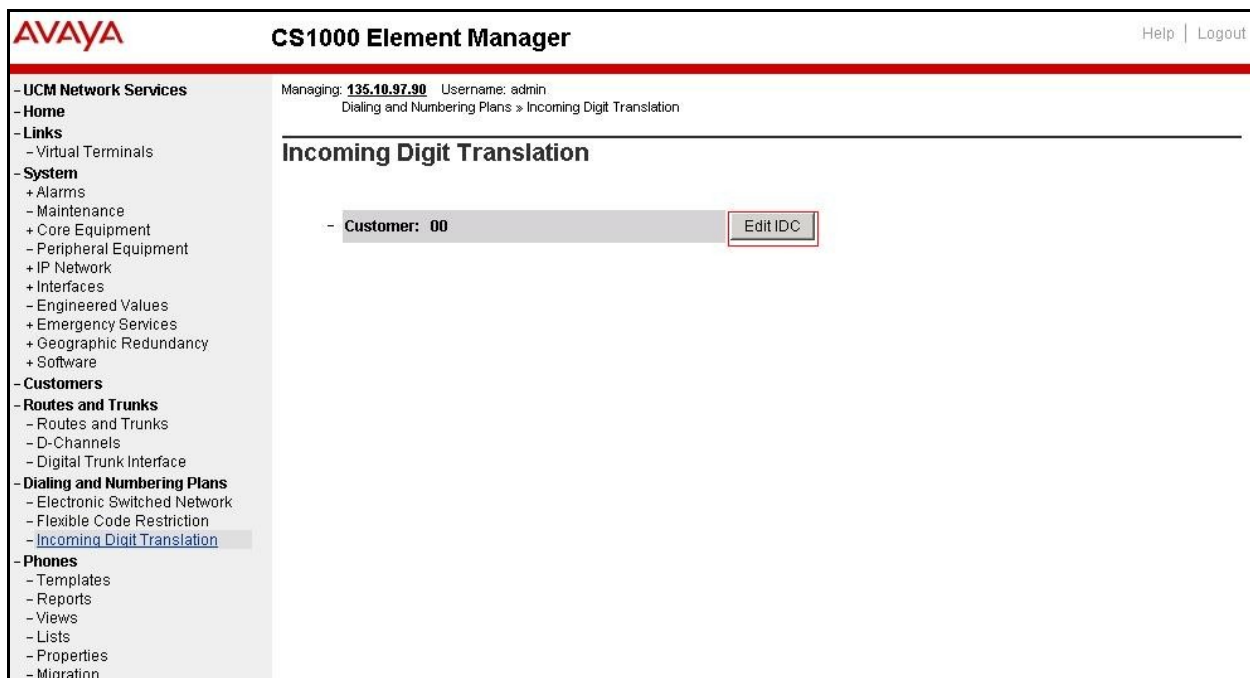


Figure 40 – Incoming Digit Translation

b) Click on the **New DCNO** to create the digit translation mechanism. In this example, Digit Conversion Tree Number (DCNO) 1 has been created as shown in **Figure 41**.

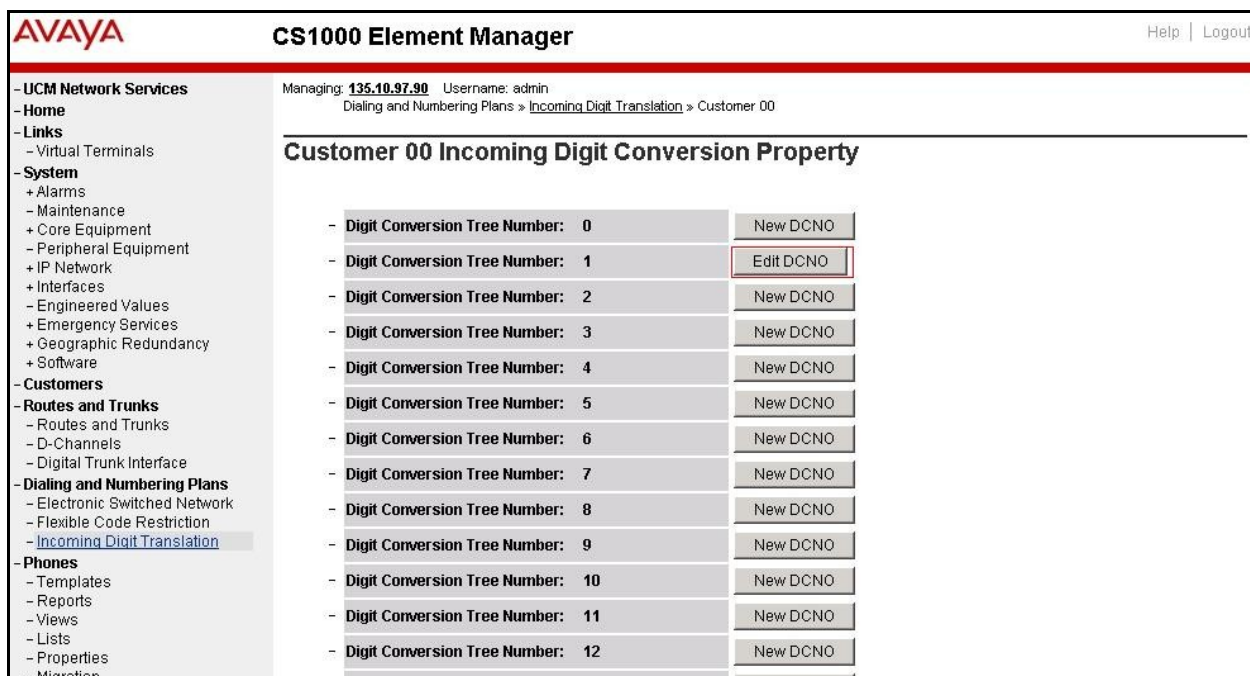


Figure 41 – Incoming Digit Conversion Property

c) Detail configuration of the **DCNO** is shown in **Figure 42**. The **Incoming Digits** can be added to map to the **Converted Digits** which would be the CS1000 system phones DN. This **DCNO** has been assigned to route 100 as shown in **Figure 27**.

In the following configuration, the incoming call from PSTN with the prefix 913324598x will be translated to CS1K DN 598x. The DID 9133245990 is translated to 3111 for Voicemail accessing purpose.

AVAYA CS1000 Element Manager

Managing: 135.10.97.90 Username: admin
Dialing and Numbering Plans > Incoming Digit Translation > Customer 00 > Digit Conversion Tree 1 Configuration

Digit Conversion Tree 1 Configuration

Regular IDC tree
Send calling party DID disabled

Buttons: Add... Delete IDC Delete IDC tree Refresh

	Incoming Digits	Converted Digits	CPND Name	CPND language
1	913324598	598		
2	9133245990	3111		

Figure 42 – Digit Conversion Tree Configuration

5.6.6. Outbound Call - Special Number Configuration.

There are special numbers which have been configured to be used for this testing such as; **0** to reach Service Provider operator, **0+10** digits to reach Service Provider operator assistant, **011** prefix for international call, **1** for national long distance call, **411**, **911** and so on.

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Special Number (SPN)** as shown in **Figure 34**.

b) Enter SPN and then click on the “to Add” button. **Figure 43** shows all the special numbers were used for this testing.

Special Number: 0

- **Flexible length:** 0 (flexible, unlimited and accept the character # to ending dial number)
- **CallType:** NONE
- **Route list index:** 100, created in **Section 5.6.4**

Special Number: 1

- **Flexible length:** 0 (flexible, unlimited and accept the character # to ending dial number)
- **CallType:** NATL
- **Route list index:** 100, created in **Section 5.6.4**

Special Number: 411

- **Flexible length:** 3
- **CallType:** NATL
- **Route list index:** 100, created in **Section 5.6.4**

Special Number: 911

- **Flexible length:** 3
- **CallType:** NATL
- **Route list index:** 100, created in **Section 5.6.4**

The screenshot displays the AVAYA CS1000 Element Manager web interface. The top header shows the AVAYA logo, the title "CS1000 Element Manager", and links for "Help" and "Logout". Below the header, a navigation menu on the left lists various system components like "UCM Network Services", "Home", "Links", "System", "Customers", "Routes and Trunks", "Dialing and Numbering Plans", "Phones", "Tools", and "Security". The main content area is titled "Special Number List" and shows a list of configured special numbers. At the top of this section, there is a form to "Please enter a Special Number" with an "Add" button. The list contains four entries, each with an "Edit" button:

- Special Number -- 0**: Flexible length: 0, International dialing plan: NO, Type of call that is defined by the special number: NONE, Route list index: 100.
- Special Number -- 1**: Flexible length: 0, Type of call that is defined by the special number: NATL, Route list index: 100.
- Special Number -- 411**: Flexible length: 3, Inhibit time-out handler: NO, Type of call that is defined by the special number: NATL, Route list index: 100.
- Special Number -- 911**: Flexible length: 3, Inhibit time-out handler: NO, Type of call that is defined by the special number: NATL, Route list index: 100.

Figure 43 – Special Number List

5.6.7. Outbound Call - Numbering Plan Area (NPA)

This section describes the creation of NPA numbers used in this testing configuration.

- Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Numbering Plan Area Code (NPA)** as shown in **Figure 34**.
- Enter area code desired in the textbox and click on the “to Add” button. **Figure 44** shows NPA numbers **613, 913, 318, 608** configured for this testing. These codes are associated to SIP route.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The top header shows the AVAYA logo, the title 'CS1000 Element Manager', and links for 'Help' and 'Logout'. Below the header, a navigation pane on the left lists various system services, with 'Dialing and Numbering Plans' and 'Electronic Switched Network' highlighted. The main content area shows the 'Numbering Plan Area Code List' page. At the top of this page, it indicates the user is managing IP address 135.10.97.90 as 'admin'. The breadcrumb trail shows the path: 'Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Numbering Plan (NET) > Access Code 1 > Numbering Plan Area Code List'. The page title is 'Numbering Plan Area Code List'. Below the title, there is a form to 'Please enter an area code' with a text input field and a 'to Add' button. A table below lists four configured NPAs: 318, 608, 613, and 913. Each entry includes an 'Edit' button, the 'Route List Index' (all set to 100), and the 'Incoming Trunk group Exclusion Index' (all set to NONE).

Numbering Plan Area Code	Route List Index	Incoming Trunk group Exclusion Index
318	100	NONE
608	100	NONE
613	100	NONE
913	100	NONE

Figure 44 – Numbering Plan Area Code List

5.7. Administer Phone

This section describes the creation of CS1000 clients used in this testing configuration.

5.7.1. Phone creation

- Refer to **Section 5.5.4** to create a virtual super-loop - **108** used for IP phone.
- Refer to **Section 5.4.1** to create a bandwidth zone - **10** for IP phone.
- Login Call Server CLI (please refer to **Section 5.1.2** for more detail).
- Create an IP phone by using LD 11.

REQ: prt
 TYPE: **2001p2**
 TN **108 0 0 1**
 DATE
 PAGE
 DES
 MODEL_NAME
 EMULATED

 DES **2001P2**
 TN **108 0 00 01** VIRTUAL
 TYPE 2001P2
 CDEN 8D
 CTYP XDLC
 CUST 0
 NUID
 NHTN
 CFG_ZONE **00010**
 CUR_ZONE 00010
 MRT
 ERL 12345
 ECL 0
 FDN
 TGAR 1
 LDN NO
 NCOS 7
 SGRP 0
 RNPG 0
 SCI 0
 SSU
 LNRS 16
 XLST
 SCPW
 SFLT NO
 CAC_MFC 0
 CLS_UNR FBD WTA LPR MTD FND HTD TDD CRPD
 MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
 POD SLKD CCSD SWD LNA CNDA
 CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
 ICDD CDMD LLCN MCTD CLBD AUTU
 GPUD DPUD DNDD CFXD ARHD CLTD ASCD
 CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
 UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
 DRDD EXR0
 USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
 FDSF NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
 MSNV FRA PKCH MWTD DVLD CROD ELCD
 CPND_LANG ENG
 HUNT
 PLEV 02
 PUID
 UPWD
 (con't)

DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 5989 0 MARP
CPND
CPND_LANG ROMAN
NAME Car2 5989
XPLN 13
DISPLAY_FMT FIRST, LAST
01
02
03
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 11 FEB 2011

NACT

5.7.2. Enable Privacy for Phone

In this section, it shows how to enable Privacy for a phone by changing its class of service (CLS). By modifying the configuration of the phone created in **Section 5.7.1**, the display of the outbound call will be changed appropriately.

a) To hide display name, set CLS to **namd**. CS1000 will include “Privacy:user” in SIP message header before sending to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2001p2
TN 108 0 0 0
SCH3928
TN 108 0 0 1
ECHG yes
ITEM cls namd
...
```

b) To hide display number, set CLS to **ddgd**. CS1000 will include “Privacy:id” in SIP message header before sending to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2001p2
TN 108 0 0 0
SCH3928
TN 108 0 0 1
ECHG yes
ITEM cls ddgd
...
```

c) To hide display name and number, set CLS to **namd, ddgd**. CS1000 will include “Privacy:id, user” in SIP message header before sending to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2001p2
TN 108 0 0 0
SCH3928
TN 108 0 0 1
ECHG yes
ITEM cls namd ddgd
...
```

d) To allow display name and number, set CLS to **nama, ddga**. CS1000 will send header “Privacy:none” to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2001p2
TN 108 0 0 0
SCH3928
TN 108 0 0 1
ECHG yes
ITEM cls nama ddga
...
```

5.7.3. Enable Call Forward for Phone

In this section, it shows how to configure Call Forward feature at the system level and phone level.

- a) Select **Customer > 00 > Call Redirection**. The Call Redirection page is shown as **Figure 45**.
- **Total redirection count limit: 0** (unlimited)
 - **Call Forward: Originating**
 - **Number of normal ring cycle of CFNA: 4**

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Days for day option 2:

Days for day option 3:

Redirection Holidays

Do not disturb hunting:

Total redirection count limit:

Options:

Call forward reminder tone for 500/2500 sets

CFNA treatment for call waiting calls on a DN

DID call to second degree busy treatment

Message center

Prevention of reciprocal call forward

Call forward:

Originating

Forwarding

Number of normal ringing cycles for CFNA

Option 0:

Option 1:

Option 2:

Number of distinctive ringing cycles for CFNA

Option 0:

Option 1:

Option 2:

Calls routed to message center

No answer DID calls:

No answer non-DID calls:

DID calls to busy telephones:

Save

Cancel

Figure 45 – Call Redirection

b) To enable **Call Forward All Call (CFAC)** for phone over trunk by using LD 11, change its CLS to **CXFA**, **SFA** then program the forward number on the phone set. Following is the configuration of a phone has CFAC enabled with forwarding number is 66139675279.

REQ: prt
TYPE: 2007
TN 108 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2007
TN 108 0 00 02 VIRTUAL
TYPE 2007

...
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD **SFA** MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDA CDMA LLCN MCTD CLBD AUTU
GPUD DPUD DNDD **CFXA** ARHD CLTD ASCD
...
19 **CFW 16 66139675279**
...

c) To enable **Call Forward Busy (CFB)** for phone over trunk by using LD 11, change its CLS to **FBA, HTA, SFA** then program the forward number as **HUNT**. Following is the configuration of a phone has CFB enabled with forward number 66139675279.

REQ: prt
TYPE: 2007
TN 108 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2007
TN 108 0 00 02 VIRTUAL
TYPE 2007

...
CLS UNR **FBA** WTA LPR MTD FNA **HTA** TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD **SFA** MRD DDV CNID CDCA MSID DAPA BFED RCBF
...
HUNT 66139675279
...

d) To enable **Call Forward No Answer (CFNA)** for phone over trunk by using LD 11, change its CLS to **FNA**, **SFA** then program the forward number as **FDN**. Following is the configuration of a phone has CFNA enabled with forward number 66139675279.

```
REQ: prt
TYPE: 2007
TN 108 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2007
TN 108 0 00 02 VIRTUAL
TYPE 2007
...
FDN 66139675279
...
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBF
...
```

5.7.4. Enable Call Waiting for Phone

In this section, it shows how to configure Call Waiting feature at phone level.

- a) Login Call Server CLI (please refer to **Section 5.1.2** for more detail).
- b) Configure Call Waiting feature for phone by using LD 11 to change CLS to **HTD**, **SWA** and adding a **CWT** key.

```
REQ: prt
TYPE: 2004p2

TN 108 0 0 0
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES 2004P2
TN 108 0 00 00 VIRTUAL
TYPE 2004P2
...
CLS UNR FBD WTA LPR MTD FNA HTD TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWA LNA CNDA
...
```

```
KEY 00 SCR 5983 0   MARP
CPND
CPND_LANG ROMAN
NAME Car2 i2004p2
XPLN 13
DISPLAY_FMT FIRST, LAST
01 CWT
...
```

5.8. Administer NRS/SPS

In this section, it shows how to configure a NRS/SPS on the CS1000. Follow the below steps to setup the NRS/SPS server. It is assumed that the NRS/SPS has been deployed on the CS1000 UCM environment with all latest Service Pack installed.

5.8.1. Create a New Domain Name on NRS/SPS

This section shows how to create a new domain name for this test configuration.

- Under the **Element Names**, select the **NRSM on car1-sps** (not shown).
- The **Network Routing Service Manager** page will appear (**Figure 46**), click on the radio button of the **Standby database**. Then select **Numbering Plans -> Domains**. Click on the **Service Domain (1)** tab to add a domain.

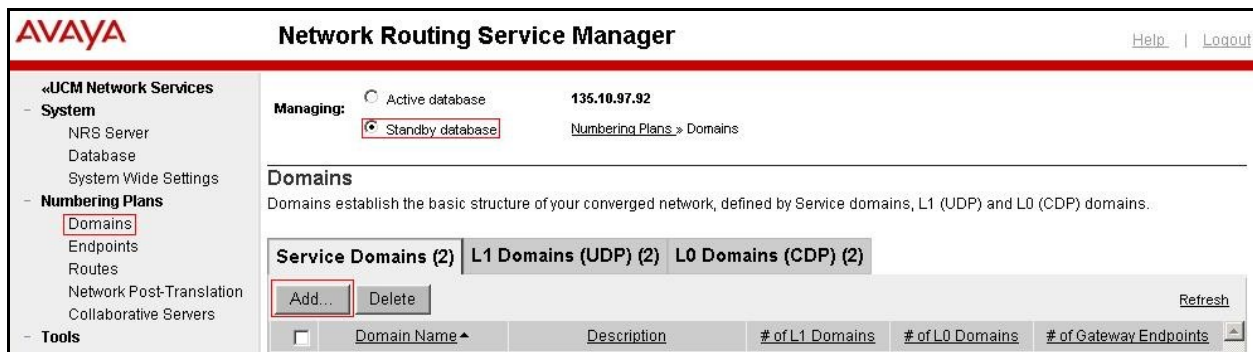


Figure 46 – Domains Page

- Click on the **Add** button and enter the domain name to be added, in this case it is **bvwddev.com** (not shown). Click **Save**.
- Select the **L1 Domains (UDP) (1)**, from the **Filter by Domain**, select the newly created domain **bvwddev.com**. Click on the **Add** button and enter **udp** as L1 Domain name. Click **Save** (not shown).
- Select the **L0 Domains (CDP) (1)**, from the **Filter by Domain**, select the newly created domain **bvwddev.com**. Click on the **All L1 Domain** pull down menu to choose the **udp**. Click on the **Add** button and enter **cdp** as L0 Domain name. Click **Save** (not shown).
- From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data of the domain name to save in to the **Active Database**. Click on the **Commit** button (not shown).

5.8.2. Configure NRS/SPS to create a Dynamic Gateway Endpoint for SIP Signaling Gateway

This section shows how to add a gateway endpoint to the NRS/SPS.

a) Click on the radio button of the **Standby database**, select the **Numbering Plan** -> **Endpoints**. Enter the endpoint name and choose the values which are highlighted in red boxes as shown in **Figure 47**. Click on the **Add** button.

The screenshot shows the Avaya Network Routing Service Manager (NRS) web interface. The left sidebar contains a navigation menu with categories: «UCM Network Services», System, Numbering Plans, and Tools. Under «UCM Network Services», the 'Endpoints' link is highlighted. The main content area is titled 'Managing' and shows the 'Standby database' selected. Below this, there is a 'Search for Endpoints' section with a search bar and filters. The 'Limit results to Domain' filter is set to 'bwwdev.com', and the transport protocol is set to 'udp'. The 'Add...' button is highlighted in red.

Figure 47 – Adding New Gateway Endpoints

b) The detail gateway endpoint configuration page will appear (**Figure 48** and **Figure 49**).

- **End point name:** car2-ssg-carrier
- **Trust Node:** checked
- **Endpoint authentication enabled:** Authentication off
- **SIP support:** Dynamic SIP endpoint
- **SIP mode:** Proxy Mode
- **SIP UDP transport enabled:** checked
- **SIP UDP port:** 5060

AVAYA Network Routing Service Manager Help | Logout

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint bvwdev.com / udp / cdp)

End point name: *

Description:

Trust Node: ☒

Tandem gateway endpoint name:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

* Required value Save Cancel

Figure 48 – Gateway Endpoint Details Configuration for CS1000 SSG Pages 1

AVAYA Network Routing Service Manager Help | Logout

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint bvwdev.com / udp / cdp)

SIP support:

SIP mode: ☒ Proxy Mode ☐ Redirect Mode

SIP TCP transport enabled: ☐

SIP TCP port:

SIP UDP transport enabled: ☒

SIP UDP port:

SIP TLS transport enabled: ☐

SIP TLS port:

Persistent TCP support enabled: ☒

End to end security support: ☐

Network Connection Server enabled: ☐

Redundancy enabled:

Main endpoint name:

Redundant endpoint name:

* Required value Save Cancel

Figure 49 – Gateway Endpoint Details Configuration for CS1000 SSG Pages 2

c) Click **Save**

d) From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data to save to the **Active Database**. Click on the **Commit** button (not shown).

5.8.3. Configure NRS/SPS to create a Static Gateway Endpoint for the Acme Packet Session Border Controller

This section shows how to add the Acme Packet Session Border Controller as a static gateway endpoint on the NRS/SPS as shown in **Figure 50** and **Figure 51**.

a) Click on the radio button of the **Standby database**, select the **Numbering Plan -> Endpoints**. Enter the endpoint name and choose the values which are highlighted in red boxes as shown in **Figure 47**. Click on the **Add** button.

- **End point name:** acme
- **Trust Node:** checked
- **Endpoint authentication enabled:** Authentication off
- **Static endpoint address:** 135.10.97.184
- **SIP support:** Static SIP endpoint
- **SIP mode:** Proxy Mode
- **SIP UDP transport enabled:** checked
- **SIP UDP port:** 5060

AVAYA Network Routing Service Manager [Help](#) | [Logout](#)

Managing: ☐ Active database 135.10.97.92
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint bywdev.com / udp / cdp)

End point name: acme *

Description:

Trust Node: ☒

Tandem gateway endpoint name: Not Applicable

Endpoint authentication enabled: Authentication off

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

* Required value

Save Cancel

Figure 50 – Gateway Endpoint Details Configuration for SBC Pages 1

AVAYA Network Routing Service Manager Help | Logout

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint bvwdev.com / udp / cdp)

Static endpoint address:

H.323 support:

SIP support:

SIP mode: ☒ Proxy Mode ☐ Redirect Mode

SIP TCP transport enabled: ☐

SIP TCP port:

SIP UDP transport enabled: ☒

SIP UDP port:

SIP TLS transport enabled: ☐

SIP TLS port:

Persistent TCP support enabled: ☒

End to end security support: ☐

Network Connection Server enabled: ☐

Redundancy enabled:

★ Required value

Save **Cancel**

Figure 51 – Gateway Endpoint Details Configuration for SBC Pages 2

b) Click **Save**.

c) From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data to save to the **Active Database**. Click on the **Commit** button (not shown).

Note: After adding the SIP Signaling Gateway (SSG) and the Acme Packet Session Border Controller to the NRS/SPS, **Active database** shows IP addresses of the gateways as in **Figure 52** (highlighted red box). It means the registration has been successful for dynamic endpoint and all gateways are in service.

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135.10.97.92

Numbering Plans » Endpoints

Search for Endpoints

Hide

Enter an endpoint ID (use * for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID: *

Limit results to Domain: All service domains / All L1 domains / All LD domains

Results per page: 50 Search

Gateway Endpoints (3) User Endpoints (0)

SIP phone context... Refresh

	ID	Supported Protocols	SIP mode:	Call Signaling IP	Description	# of Routing Entries	Context
1	Node075	RAS H.323 endpoint	Not Applicable	Not registered		1	Frox.com.ads / udp1 / cdp1
2	acme	Static SIP endpoint	Proxy Mode	135.10.97.184		15	bwdev.com / udp / cdp
3	car2-ssg-carrier	Dynamic SIP endpoint	Proxy Mode	135.10.97.168		1	bwdev.com / udp / cdp

1 - 3 of 3 Gateway Endpoint(s) Page 1 of 1 First Previous Next Last

Figure 52 – Gateway Endpoints successfully register to NRS/SPS

5.8.4. Creating Routing Entry for the SSG on the NRS/SPS

In this section, it describes how to create the routing entry on the NRS/SPS to route the call from PSTN via Acme Packet Session Border Controller to the SSG. In the test configuration, we have used the routing entry 913324 which is sending from CenturyLink as the first 6 digits of the DID's range, 913 - 324 – XXXX.

a) Click radio button **Standby database**, then on the **Network Routing Service Manager** page as shown in **Figure 46**, select **Numbering Plan -> Routes**. On the **Routing Entries** page, choose and change the attributes highlighted in red boxes to the values as shown in **Figure 53**. Select SSG endpoint then click the **Add** button.

AVAYA Network Routing Service Manager

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Routes](#)

Search for Routing Entries [Hide](#)

Enter a DnPrefix and Dn Type (use * for all) and click Search. You may narrow the search by specifying a particular domain.

DN Prefix: DN Type:

Limit results to Domain: / /

Endpoint Name:

Results per page:

Routing Entries (1) Default Routes (0) Emergency Fallback Routes (0)

Figure 53 – Routing Entries for CS1000 SSG

- b) **Add Routing Entry** page will appear as shown in **Figure 54**. Fill in the textboxes with:
- DN Type: **Private level 0 regional (CDP steering code)**
 - DN Prefix: **913324**
 - Route cost: **1**

AVAYA Network Routing Service Manager

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Routes > Routing Entry](#)

Add Routing Entry { bwdev.com / udp / cdp / car2-ssg-carrier }

DN type:

DN prefix: *

Route cost: * (1-255)

* Required value.

Figure 54 – Add Routing Entry for CS1000 SSG

- c) Click **Save**.
- d) From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data to save to the **Active Database**. Click on the **Commit** button (not shown).

5.8.5. Creating Routing Entry for the SBC on the NRS/SPS

This section describes how to create the routing entries on the NRS/SPS to route the call from CS1000 to PSTN via the Acme Packet Session Border Controller. In the testing configuration,

the following routing entries are used: **0, 411, 911, 1866, 318, 1318, 613, 1613, 913738, 913533** and **913856**. In the below example, it shows the configuration only for entry **613**. Others entries can be done the same way.

a) Click radio button **Standby database**, then on the **Network Routing Service Manager** page as shown in **Figure 46**, select **Numbering Plan -> Routes**. On the **Routing Entries** page, choose and change the attributes highlighted in red boxes to the values as shown in **Figure 55**. Select “acme” endpoint then Click the **Add** button.

AVAYA Network Routing Service Manager

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Routes](#)

Search for Routing Entries [Hide](#)

Enter a DnPrefix and Dn Type (use * for all) and click Search. You may narrow the search by specifying a particular domain.

DN Prefix: * DN Type: All DN Types

Limit results to Domain: bwdev.com, udp, cdp

Endpoint Name: acme

Results per page: 50 [Search](#)

Routing Entries (15) **Default Routes (0)** **Emergency Fallback Routes (0)**

[Add...](#) [Copy...](#) [Move...](#) [Import...](#) [Export...](#) [Routing test...](#) [Delete](#) [Refresh](#)

Figure 55 – Route Entries for ACME

b) Add Routing Entry page will appear as shown in **Figure 56**. Fill in the textboxes with:

- DN Type: Private level 0 regional (CDP steering code)
- DN Prefix: 613
- Route cost: 1

AVAYA Network Routing Service Manager

Managing: ☐ Active database **135.10.97.92**
☒ Standby database [Numbering Plans > Routes > Routing Entry](#)

Add Routing Entry { bwdev.com / udp / cdp / acme }

DN type: Private level 0 regional (CDP steering code)

DN prefix: 613 *

Route cost: 1 * (1-255)

* Required value. [Save](#) [Cancel](#)

Figure 56 – Add Routing Entry for SBC

c) Click **Save**.

d) From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data to save to the **Active Database**. Click on the **Commit** button (not shown).

6. Configure Acme Packet Net-Net 3800 Session Border Controller

This section describes the configuration of the Acme Packet Session Border Controllers necessary for interoperability with the CS1000 and CenturyLink systems. The Acme Packet Session Border Controller was configured via the Acme Packet Command Line Interface (ACLI). This section assumes the reader is familiar with accessing and configuring the Acme Packet Session Border Controller.

This section will not attempt to describe each component in its entirety but instead will highlight critical fields in each component which relates to the functionality in these Application Notes and the direct connection to CS1000. The remaining fields are generally the default/standard value used by the Acme Packet Session Border Controller for that field.

In this testing, according to the configuration reference **Figure 1**, the Avaya elements reside on the Private side and the CenturyLink elements reside on the Public side of the network.

6.1. Acme Packet Command Line Interface Summary

The Acme Packet Session Border Controller is configured using the Acme Packet Command Line Interface (ACLI). The following are the generic ACLI steps for configuring various elements.

1. Access the console port of the Acme Packet Session Border Controller using a PC and a terminal emulation program such as HyperTerminal (use the RJ-45 to DB9 adapter as packaged with the Session Border Controller for cable connection). Use the following settings for the serial port on the PC.
 - Bits per second: 115200
 - Data bits: 8
 - Parity: None
 - Stop bits: 1
 - Flow control: None
2. Log in to the Acme Packet Session Border Controller with the proper user password.
3. Enable the Super-user mode by entering the **enable** command and then the super user password. The command prompt will change to include a “#” instead of a “>” while in Super user mode. This level of system access (i.e. at the “acmesystem#” prompt) will be referred to as the **main** level of the ACLI. Specific sub-levels of the ACLI will then be accessed to configure specific **elements** and specific **parameters** of those elements.
4. In Super-user mode, enter the **configure terminal** command. The **configure terminal** command is used to access the system level where all operating and system elements may be configured. This level of system access will be referred to as the **configuration** level.
5. Enter the name of an element to be configured (e.g., **system**).
6. Enter the name of a sub-element, if any (e.g., **phy-interface**).

7. Enter the name of an element parameter followed by its value (e.g., **name INSIDE**).
8. Enter **done** to save changes to the element. Use of the **done** command causes the system to save and display the settings for the current element.
9. Enter **exit** as many times as necessary to return to the configuration level.
10. Repeat **Steps 5 - 9** to configure all other elements.
11. Enter **exit** to return to the main level.
12. Type **save-config** to save the entire configuration.
13. Type **activate-config** to activate the entire configuration.

After accessing different levels of the ACLI to configure elements and parameters, it is necessary to return to the main level in order to run certain tasks such as saving the configuration, activating the configuration, and rebooting the system.

Note – Acme Packet Net-Net 3800 provisioning applicable to the reference configuration is shown in **bold** text. Other parameters and setting are shown for informational purposes.

6.2. Physical and Network Interfaces

As part of the compliance test, the Ethernet interface slot 01/port 0 of the Acme Packet Session Border Controller was connected to the external un-trusted network. The Ethernet slot 0/port 0 was connected to the internal corporate LAN. A network interface was defined for each physical interface to assign it a routable IP address.

The physical interface below defines the ports on the interface connected to the network on which the Avaya elements reside.

```

phy-interface
  name                INSIDE
  operation-type      Media
  port                0
  slot                0
  virtual-mac
  admin-state         enabled
  auto-negotiation    enabled
  duplex-mode         FULL
  speed               100
  overload-protection disabled
  last-modified-by    admin@console
  last-modified-date  2011-01-08 20:06:19

```

The physical interface below defines the ports on the interface connected to the network on which the CenturyLink elements reside.

phy-interface		
name	OUTSIDE	
operation-type	Media	
port	0	
slot	1	
virtual-mac		
admin-state		enabled
auto-negotiation		enabled
duplex-mode		FULL
speed		100
overload-protection		disabled
last-modified-by		admin@console
last-modified-date		2011-01-08 20:06:30

The network interface below defines the IP addresses on the interface connected to the network on which the Avaya elements reside.

network-interface		
name	INSIDE	
sub-port-id		0
description		
hostname		
ip-address	10.10.97.184	
pri-utility-addr		
sec-utility-addr		
netmask	255.255.255.192	
gateway	10.10.97.129	
sec-gateway		
gw-heartbeat		
state		disabled
heartbeat		0
retry-count		0
retry-timeout		1
health-score		0
dns-ip-primary		
dns-ip-backup1		
dns-ip-backup2		
dns-domain		
dns-timeout		11
hip-ip-list	10.10.97.184	
ftp-address		
icmp-address	10.10.97.184	
snmp-address		
telnet-address		
ssh-address		
last-modified-by		admin@console
last-modified-date		2011-01-08 20:07:11

The network interface below defines the IP addresses on the interface connected to the network on which the CenturyLink elements reside.

network-interface	
name	OUTSIDE
sub-port-id	0
description	
hostname	
ip-address	10.10.98.98
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.224
gateway	10.10.98.97
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	
dns-timeout	11
hip-ip-list	10.10.98.98
ftp-address	
icmp-address	10.10.98.98
snmp-address	
telnet-address	
ssh-address	
last-modified-by	admin@console
last-modified-date	2011-01-10 15:26:28

6.3. Realm

A realm represents a group of related Acme Packet Session Border Controller components. Two realms were defined for the compliance test.

The realm configuration “INSIDE” below represents the internal network on which the Avaya elements reside.

realm-config	
identifier	INSIDE
description	
addr-prefix	0.0.0.0
network-interfaces	
	INSIDE:0
mm-in-realm	disabled
<Text removed for brevity>	

The realm configuration “OUTSIDE” below represents the external network on which the CenturyLink system resides.

```

realm-config
  identifier                OUTSIDE
  description
  addr-prefix              0.0.0.0
  network-interfaces
    OUTSIDE:0
  mm-in-realm              disabled

<Text removed for brevity>

```

6.4. Session Agent

A session agent defines the characteristics of a signaling peer to the Acme Packet Session Border Controller such as CS1000 and/or Service Provider SBC.

The **session agent** below represents the CenturyLink border element. The SBC will attempt to send calls to the border element. The **in-manipulationid** and **out-manipulationid** are defined in the SIP header manipulation applying to the OUTSIDE realm.

```

session-agent
  hostname                 20.20.196.157
  ip-address               20.20.196.157
  port                    5060
  state                   enabled
  app-protocol             SIP
  app-type
  transport-method        UDP
  realm-id                OUTSIDE
  egress-realm-id
  description              CenturyLink_CS1K7.5
  carriers
  allow-next-hop-ip        enabled
  constraints              disabled
  <Text removed for brevity>
  ping-method              OPTION;hops=0
  ping-interval            30
  ping-send-mode           keep-alive
  <Text removed for brevity>
  in-manipulationid        CL_TO_CS1K7.5_NAT_IP
  out-manipulationid        CS1K7.5_TO_CL_NAT_IP

```

The **session agent** below represents the configuration for inside interface to connect to NRS mentioned in **Section 5.8**.

```
session-agent
  hostname          10.10.97.172
  ip-address        10.10.97.172
  port              5060
  state             enabled
  app-protocol      SIP
  app-type
  transport-method  UDP
  realm-id          INSIDE
  egress-realm-id
  description       CenturyLink_CS1K7.5
  carriers
  allow-next-hop-ip enabled
  constraints        disabled
```

<Text removed for brevity>

6.5. SIP Configuration

The SIP configuration (*sip-config*) defines the global system-wide SIP parameters.

The key SIP configuration (*sip-config*) field is:

- **home-realm-id**: The name of the realm on the private side of the Acme Packet Session Border Controller.
- **egress-realm-id**: The name of the realm on the private side of the Acme Packet Session Border Controller.

```
sip-config
  state             enabled
  operation-mode     dialog
  dialog-transparency enabled
  home-realm-id      INSIDE
  egress-realm-id    INSIDE
  nat-mode           None
```

<Text removed for brevity>

6.6. SIP Interface

The SIP interface (*sip-interface*) defines the receiving characteristics of the SIP interfaces on the Acme Packet Session Border Controller. Two SIP interfaces were defined, one for each realm.

The SIP interface below is used by the Acme Packet Session Border Controller to communicate with CS1000 system.

```

sip-interface
state                enabled
realm-id            INSIDE
description
sip-port
  address            10.10.97.184
  port                5060
  transport-protocol UDP
  tls-profile
  allow-anonymous    all

<Text removed for brevity>

```

The SIP interface below is used by the Acme Packet Session Border Controller to communicate with the CenturyLink system.

```

sip-interface
state                enabled
realm-id            OUTSIDE
description
sip-port
  address            10.10.98.98
  port                5060
  transport-protocol UDP
  tls-profile
  allow-anonymous    all

<Text removed for brevity>

```

6.7. SIP Manipulation

SIP manipulations are rules used to modify the SIP messages (if necessary) for interoperability.

The following sip-manipulation **CS1K7.5_TO_CL_NAT_IP** is applied to **OUTSIDE** realm *out-manipulationid*. These rules perform the following:

- The header rule **manipRURI** changes Avaya Domain Name/IP address to 20.20.196.157 (CenturyLink border element) in the Request URI headers sent to CenturyLink.
- The header rules **manipTo** perform address translation and topology hiding for SIP messages between the CenturyLink system and the Avaya elements.

sip-manipulation	
name	CS1K7.5_TO_CL_NAT_IP
description	
split-headers	
join-headers	
header-rule	
name	manipRURI
header-name	request-uri
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	INVITE
match-value	
new-value	
element-rule	
name	modRURI
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	20.20.196.157
header-rule	
name	manipTo
header-name	To
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	To
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	20.20.196.157
header-rule	
name	delete_mcdn
header-name	Content-Type
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	delete_nt_mcdn
parameter-name	application/x-nt-mcdn-frag-hex;version=ssLinux-7.50.17;base=x2611

type	mime
action	delete-element
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	
element-rule	
name	delete_nt_epid
parameter-name	application/x-nt-epid-frag-hex;version-ssLinux-7.50.17;base=x2611
type	mime
action	delete-element
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	
header-rule	
name	delete_X_nt_e164_clid
header-name	X-nt-e164-clid
action	delete
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
header-rule	
name	delete_Alert_Info
header-name	Alert-info
action	delete
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
last-modified-by	admin@135.10.98.105
last-modified-date	2011-05-24 15:31:59

The following sip-manipulation **CL_TO_CS1K7.5_NAT_IP**, *in-manipulationid*, is applied to **OUTSIDE** realm and translated the SIP header information for CS1000 to understand. These rules perform the following:

- The header rule **manipRURI** change IP address to the Avaya CS1000 Domain Name in the Request URI headers sent to the CS1000 elements.

sip-manipulation	
name	CL_TO_CS1K7.5_NAT_IP
description	
split-headers	
join-headers	
header-rule	
name	manipRURI
header-name	request-uri
action	manipulate
comparison-type	case-sensitive
msg-type	request
methods	INVITE
match-value	
new-value	
element-rule	
name	modRURI
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	bvwdev.com
last-modified-by	admin@console
last-modified-date	2011-01-14 12:22:39

6.8. Steering Pools

Steering pools define the range of ports to be used for the RTP voice stream. Two steering pools were defined, one for each realm.

The key steering pool (*steering-pool*) fields are:

- **ip-address:** The address of the interface on the Acme Packet Session Border Controller.
- **start-port:** An even number of the port that begins the range.
- **end-port:** An odd number of the port that ends the range.
- **realm-id:** The realm to which this steering pool is assigned.

steering-pool	
ip-address	10.10.98.98
start-port	20000
end-port	40000
realm-id	OUTSIDE
network-interface	
last-modified-by	admin@console
last-modified-date	2011-01-08 20:09:01
steering-pool	
ip-address	10.10.97.184
start-port	20000
end-port	40000
realm-id	INSIDE
network-interface	
last-modified-by	admin@console
last-modified-date	2011-01-08 20:09:08

6.9. Local Policy

The local policies below govern the routing of SIP messages from elements on the network on which the Avaya elements, (e.g. CS1000), reside to the CenturyLink system and vice versa.

local-policy	
from-address	ctlvoice.com
to-address	9133245982
	9133245983
	9133245984
	9133245985
	9133245986
	9133245987
	9133245988
	9133245989
	9133245990
source-realm	OUTSIDE
Description	CL_TO_CS1K7.5
activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
last-modified-by	admin@console
last-modified-date	2011-03-01 11:13:58
policy-attribute	
next-hop	10.10.97.172
realm	INSIDE
action	none
terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
(con't)	

days-of-week	U-S
cost	0
app-protocol	SIP
state	enabled
methods	
media-profiles	
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	

local-policy	
from-address	anonymous.invalid bvwdev.com
to-address	*
source-realm	INSIDE
description	CS1K_TO_CL
activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
last-modified-by	admin@console
last-modified-date	2011-03-10 17:24:09
policy-attribute	
next-hop	20.20.196.157
realm	OUTSIDE
action	none
terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
app-protocol	SIP
state	enabled
methods	
media-profiles	
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	

7. Verification Steps

The following steps may be used to verify the configuration.

7.1. General

Place an **inbound/ outbound** call from/ to a PSTN phone to/ from an internal CS1000 phone, answer the call, and verify that two-way speech path exists. Check call display name and number to ensure the correct info was sent/ received. Perform hold/ retrieve. Verify the call remains stable for several minutes and disconnect properly.

7.2. Verify Call Establishment on CS1000 Call Server

a) Active Call Trace (LD 80)

The following is an example of one of the commands available on the CS1000 to trace the DN when the call is in progress and or idle. The call scenario involved PSTN phone number 6139675258 calling 9133245982 on the CS1000.

- Login Call Server CLI (please refer to Section 5.1.2 for more detail)
- Login to the Overlay command prompt, issue the command **LD 80** and then **trace 0 5982**.
- After call is released, issue command **trac 0 5982** again to see if the DN is released back to idle state.

Below is the actual output of the Call Server Command Line mode when the 5982 is in call state:

```
>ld 80
TRA000
.trac 0 5982

ACTIVE VTN 108 0 00 02

ORIG VTN 100 0 00 00 VTRK IPTI RMBR 100 1 INCOMING VOIP GW CALL
FAR-END SIP SIGNALLING IP: 135.10.97.184
FAR-END MEDIA ENDPOINT IP: 135.10.97.184 PORT: 21826
FAR-END VendorID: Not available
TERM VTN 108 0 00 02 KEY 0 SCR MARP CUST 0 DN 5982 TYPE 2007
SIGNALLING ENCRYPTION: INSEC
MEDIA ENDPOINT IP: 135.10.98.20 PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW PAYLOAD 10 ms VAD OFF
RFC2833: RXPT 101 TXPT N/A DIAL DN 5982
MAIN_PM ESTD
TALKSLOT ORIG 24 TERM 29
EES_DATA:
NONE
QUEU NONE
CALL ID 0 33530

---- ISDN ISL CALL (ORIG) ----
CALL REF # = 385
BEARER CAP = VOICE
HLC =
CALL STATE = 10 ACTIVE
CALLING NO = 16139675258 NUM_PLAN:E164 TON:INTERNATIONAL ESN:UNKNOWN
CALLED NO = 9133245982 NUM_PLAN:UNKNOWN TON:UNKNOWN ESN:UNKNOWN

.
TIM000 15:00 16/3/2011 CPU 0
```

And this is the example after the call on 5982 is completed.

```
.trac 0 5982
IDLE VTN 108 0 00 02 MARP
```

b) SIP Trunk monitoring (LD 32)

Place a call inbound from PSTN (6139675258) to CS1000 (9133245982). Then check the SIP Trunk status by using LD 32

```
>ld 32
NPR000
.stat 100 0
031 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
.
```

And this is the example after the call is completed; the BUSY trunk changes its state to IDLE.

```
.stat 100 0
032 UNIT(S) IDLE
000 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

7.3. Protocol Traces

Below are Wireshark traces of the same call scenario that has been made in section 7.2.

The following SIP headers are inspected:

- RequestURI: verify the request number and either SIP domain
- From: verify the display name and display number.
- To: verify the display name and display number.
- History-Info: verify the call forward information and reason code.
- P-Asserted-Identity: verify the display name and display number.
- Privacy: verify the “user, id” masking.

The following attributes in SIP message body are inspected:

- Connection Information (c): verify IP address of far end endpoint
- Time Description (t): verify session timeout of far end endpoint
- Media Description (m): verify audio port, codec, DTMF event description
- Media Attribute (a): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

a) SIP call is captured at Acme Packet Session Border Controller OUTSIDE interface.

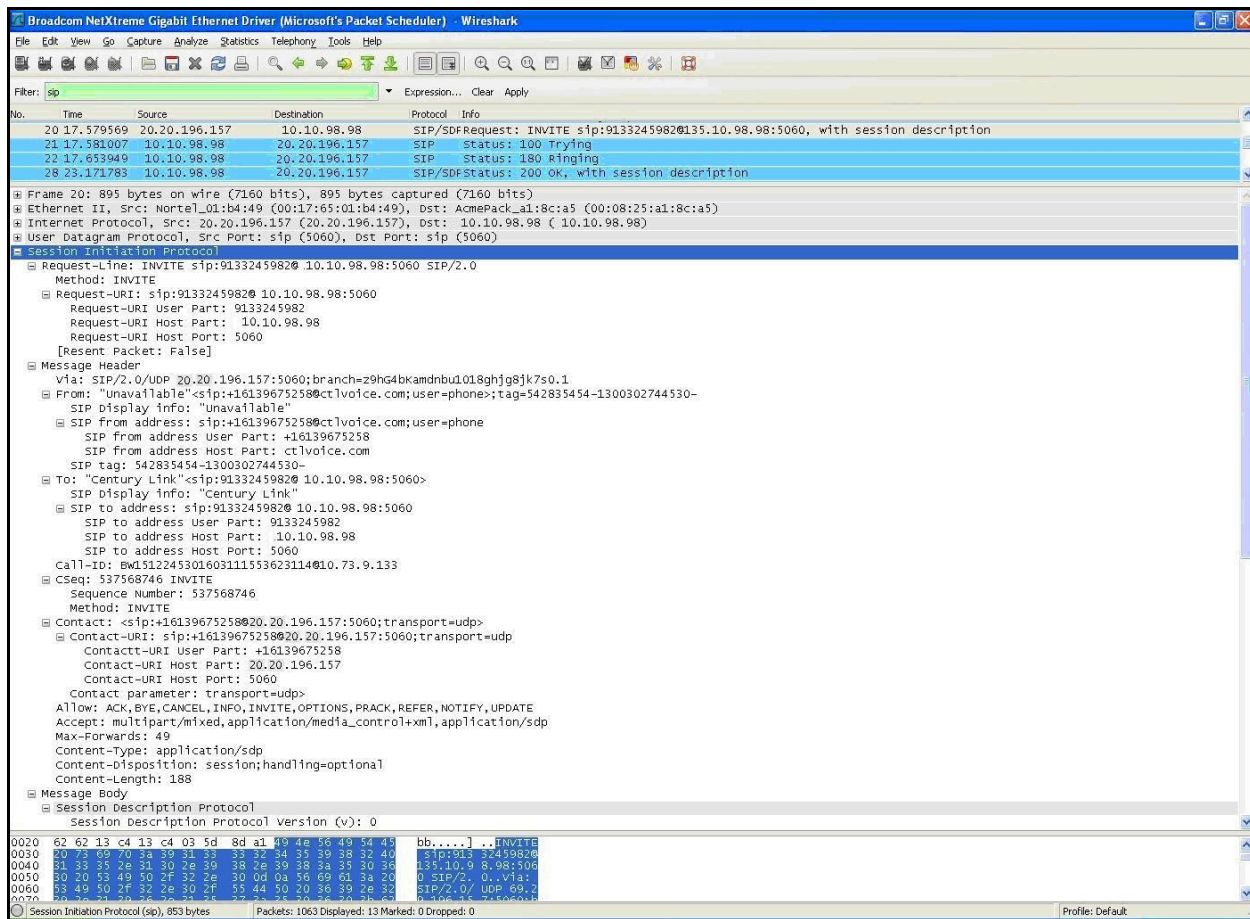


Figure 57: IP packet capture for SIP call at SBC

b) SIP call is captured at CS1000 NRS/SPS TLAN interface.

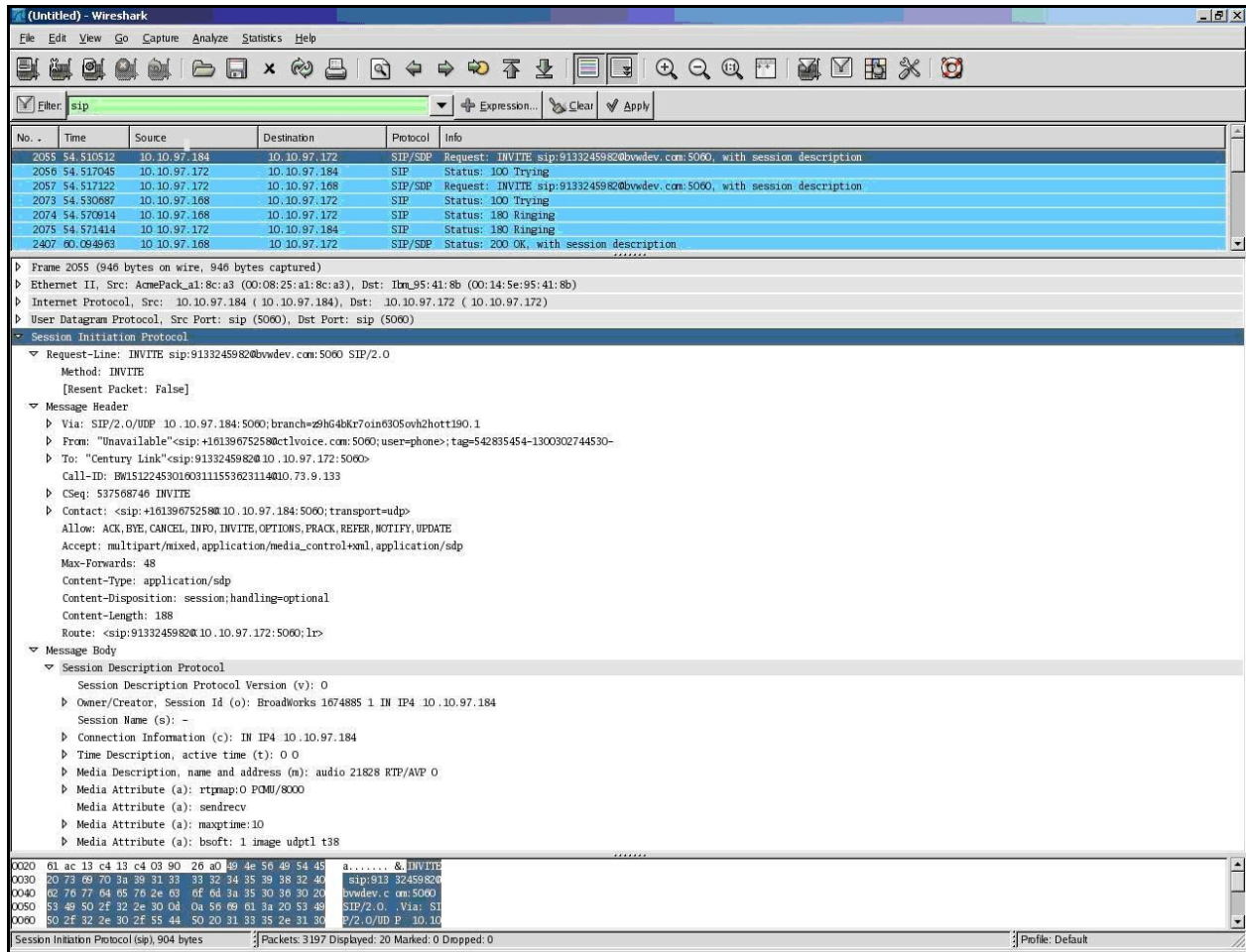


Figure 58: IP packet capture for SIP call at CS1000 NRS/SPS

8. Conclusion

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test result met the objectives outlined in **Section 2.1**. The CenturyLink system is considered **compliant** with the Avaya Communication Server 1000 Release 7.5.

9. Additional References

Product documentation for Avaya products may be found at:

<http://support.avaya.com/css/appmanager/public/support>

[1] *Network Routing Service Fundamentals, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-130, Revision 03.02, November 2010.*

[2] *IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-313, Revision: 05.02, November 2010*

[3] *Communication Server 1000E Overview, Avaya Communication Server 1000, Release 7.5, Document Number NN43041-110, Revision: 05.02, January 2011*

[4] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-116, Revision 05.08, January 2011*

[5] *Communication Server 1000 Dialing Plans Reference, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-283, Revision 05.02, November 2010*

[6] *Product Compatibility Reference, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-256, Revision 05.02, February 2011*

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